MODULE - 1

ELECTRONIC CIRCUITS:

POWER SUPPLIES: Block diagram, Rectifiers, Reservoir, and Smoothing circuits, Full-wave rectifiers, Bi-phase rectifier circuits, voltage regulators, output resistance and voltage regulation, voltage multipliers.

AMPLIFIERS: Types of Amplifiers, gain, input and Output resistance, frequency response, Bandwidth, Phase shift, Negative feedback, Multi-stage amplifier.

OPERATIONAL AMPLIFIER: Operational Amplifier parameter Characteristics, Configurations, Operational amplifier circuits.

OSCILLATORS: positive feedback, Conditions for Oscillation, ladder network oscillator, wein bridge Oscillator, Multivibrator, single-stage astable oscillator, crystal Controlled Oscillator.

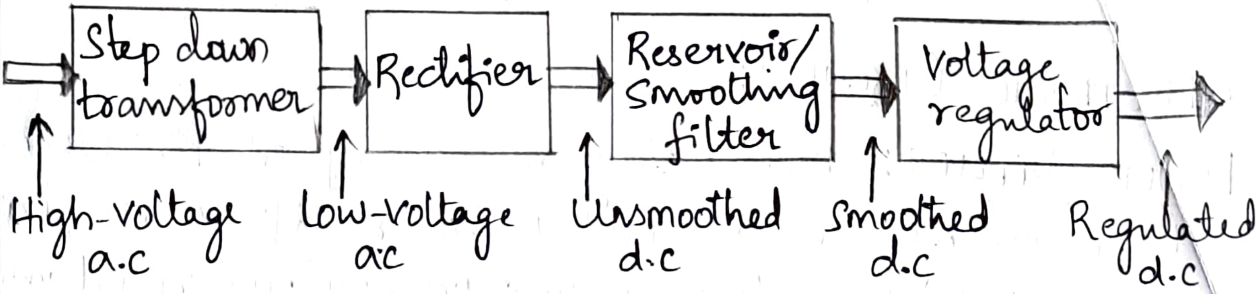
POWER SUPPLIES:

Most of the electronic circuits require a source of well regulated d.c at voltages of typically between 5V and 30V. In some cases, this supply can be derived directly from batteries (Eg: 6V, 9V and 12V).

The basic block diagram of a d.c power supply is shown in figure below.

The main components of a power supply consists of step-down transformer, Rectifier, Reservoir, and a voltage regulator.

fig: Block diagram of d.c



- Since the input mains is at a relatively high voltage, a step-down transformer of appropriate turns ratio is used to convert this to a low voltage.
- The a.c output from the transformer secondary is then rectified using conventional silicon rectifier diodes to produce an unsmoothed output.
- This is then smoothed and filtered before being applied to a circuit which will regulate (or stabilize) the output voltage so that it remains relatively constant in spite of variations in to both load current and incoming mains voltage.

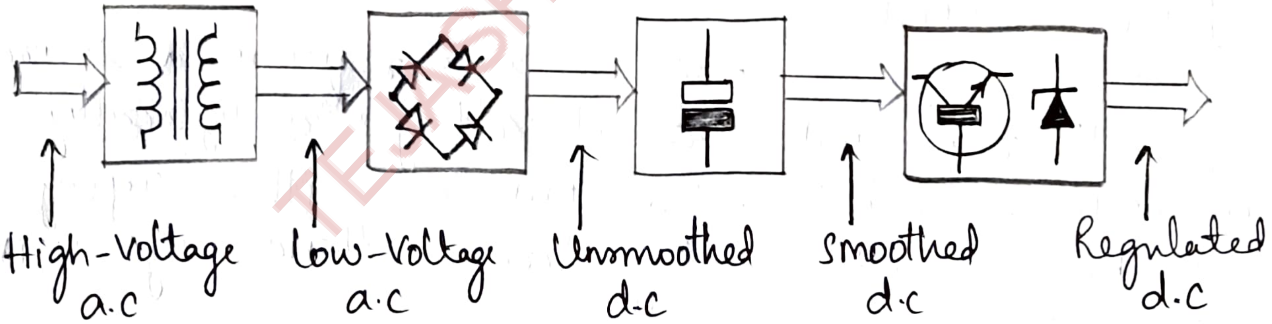


fig: Block diagram of a d.c power supply showing principal components.

- The above figure shows how the electronic components can be used in the realization of the block diagram.

* The iron-cored step-down transformer feeds a rectifier arrangement.

* The output of the rectifier is then applied to a high-value reservoir capacitor. This capacitor stores a considerable amount of charge and is being constantly topped-up by the rectifier arrangement.

* The capacitor also helps to smooth out the voltage pulses produced by the rectifier.

* Finally, a stabilizing circuit provides a constant output voltage.

RECTIFIERS:

* Semiconductor diodes are commonly used to convert alternating current (a.c) to direct current (d.c) are referred to as rectifiers.

* The simplest form of rectifier circuit makes use of a single diode and, since it operates on only either positive or negative half-cycles of the supply, it is known as a half-wave rectifier.

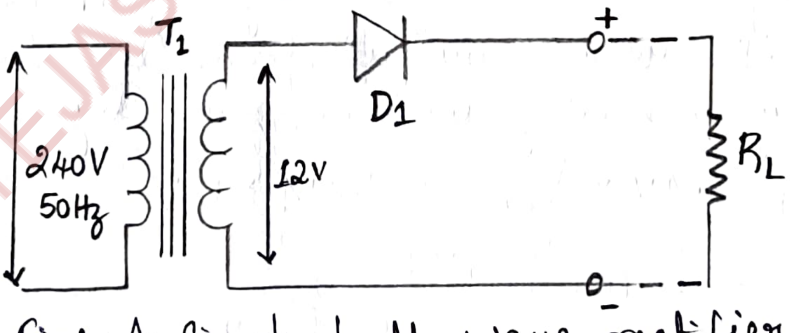


Fig: A simple half-wave rectifier output

* Mains voltage (220V to 240V) is applied to the primary of a step-down transformer (T_1). The secondary of the transformer (T_1) steps down the 240 V_{rms} to 12 V rms.

- During positive half-cycle of the input, the diode D_1 will be forward biased and the current will flow from cathode to anode as shown in figure below and hence the diode D_1 acts as a closed-switch.
- During Negative half-cycle of the input, the diode D_1 will be reverse biased and acts as a open-switch. Hence there is no flow of electric current.
- The Switching action of D_1 results in a pulsating output voltage which is developed across the load resistor R_L.

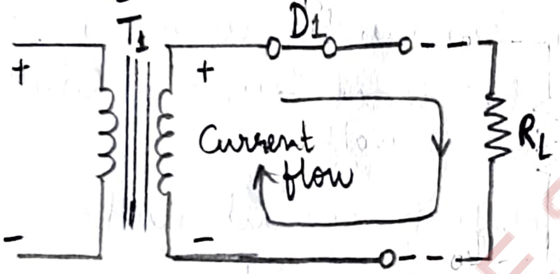


fig (a) D_1 Conducting

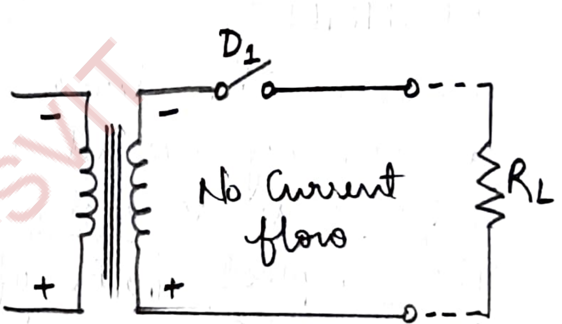


fig (b) D_1 Non-conducting

- When selecting a diode for a particular application. Assuming that the secondary of T_1 provides 12V r.m.s, the peak voltage output from the transformer's secondary winding will be given by:

$$V_{PK} = 1.414 \times V_{r.m.s} = 1.414 \times 12V$$

$$\boxed{V_{PK} = 16.97V}$$

- The peak voltage applied to D_1 will thus be approximately 17V. The negative half cycles are blocked by D_1 and thus only the positive half-cycles appear across R_L.

Peak amplitude $= 17V - 0.7V = \underline{16.3V}$

0.7V - forward threshold voltage

Problem:

* A mains transformer having a turns ratio of 44:1 is connected to a 220V r.m.s mains supply. If the secondary output is applied to a half-wave rectifier, determine the peak voltage that will appear across a load.

Soln: The r.m.s secondary voltage is given by.

$$\frac{V_s}{V_p} = \frac{N_s}{N_p}$$

$$V_s = V_p \times \frac{N_s}{N_p} = 220 \times \frac{1}{44}$$

$$\boxed{V_s = 5V}$$

The peak voltage developed after rectification will be given by:

$$V_{PK} = 1.414 \times 5V = 7.07V.$$

Assuming that the diode is a silicon device with a forward voltage drop of 0.6V, the actual peak voltage dropped across the load will be:

$$V_L = 7.07V - 0.6V$$

$$\boxed{V_L = 6.47V}$$

RESERVOIR AND SMOOTHING CIRCUITS:

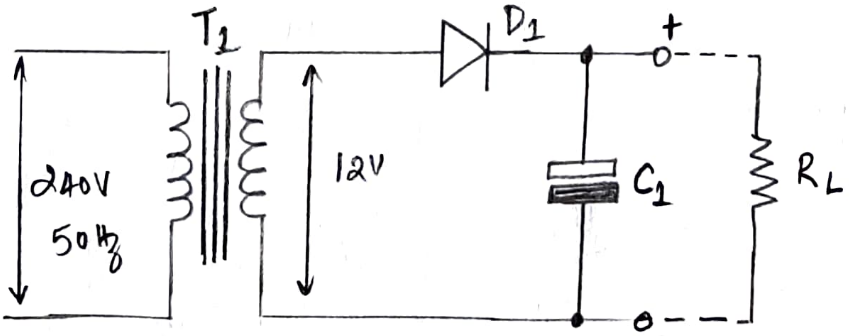


fig: A simple half-wave rectifier circuit with reservoir capacitor.

* The figure above shows a simple half-wave rectifier circuit with reservoir capacitor.
* The Capacitor C_1 has been added to ensure that the output voltage remains at, or near, the peak voltage even when the diode is not conducting.
* When the primary voltage is first applied to T_1, the first positive half-cycle output from the secondary will charge C_1 to the peak value seen across R_L.
* Hence C_1 charges to 16.3V at the peak of the positive half cycle. Because C_1 and R_L are in parallel, the voltage across R_L will be the same as that of C_1
* The time required for C_1 to charge to the maximum (peak) level is determined by the charging circuit time constant.
* The time required for C_1 to discharge is, determined by the Capacitance value and the load resistance R_L.
* Hence C_1 is referred to as a reservoir capacitor. It stores charge during the positive half-cycles of Secondary voltage and releases it during the negative half-cycles.

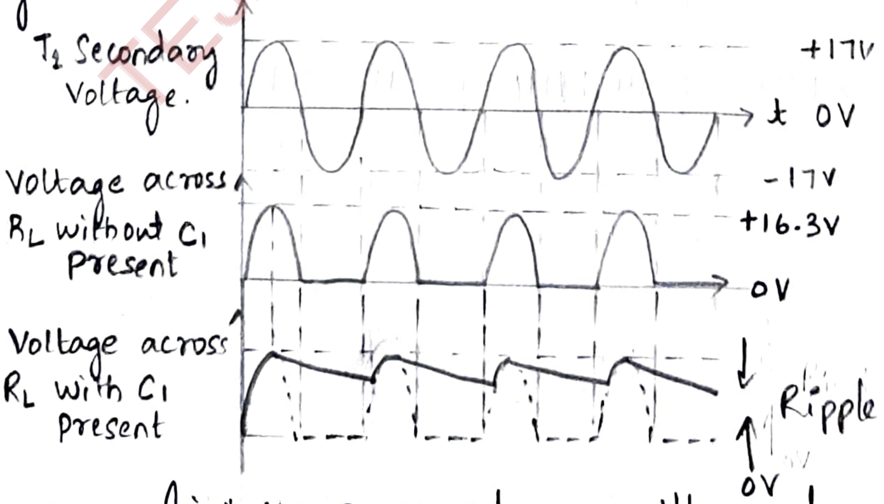


fig: HWR waveforms with and without reservoir Capacitor.

Problem :

* The R-C smoothing filter in a 50Hz mains operated half-wave rectifier circuit consists of $R_1 = 100\Omega$ and $C_2 = 1000\mu F$. If 1V of ripple appears at the input of the circuit, determine the amount of ripple appearing at the output.

Soln: For HWR, f = 50Hz

$$X_C = \frac{1}{2\pi f c} = \frac{1}{2\pi \times 50 \times 1000 \times 10^{-6}}$$

$$\boxed{X_C = 3.18\Omega}$$

$$V_{ripple} = 1 \times \frac{X_C}{\sqrt{R^2 + X_c^2}} = 1 \times \frac{3.18}{\sqrt{100^2 + 3.18^2}}$$

$$V = 0.032V$$

$$\boxed{V = 32mV}$$

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b)

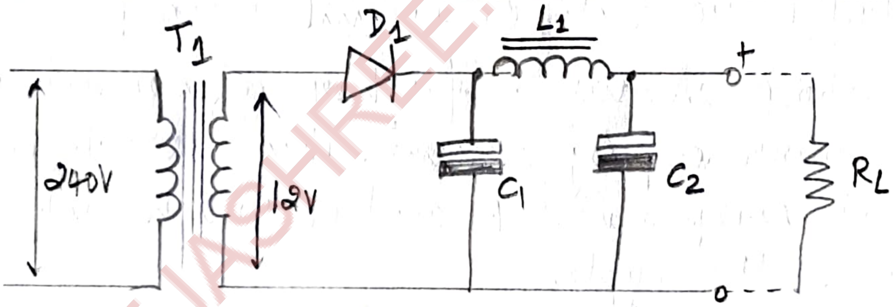


fig: Half wave rectifier circuit with L-C Smoothing filter.

* A further improvement can be achieved by using an inductor L_1 instead of a resistor in the smoothing circuit.

* This circuit also offers the advantage that the minimum d.c voltage is dropped across the inductor.

* The above figure shows the circuit of a half-wave power supply with an L-C smoothing circuit.

- Figure below shows secondary voltage waveforms together with the voltage developed across R_L with and without C_1 present.
- This gives rise to a small variation in d.c output Voltage known as ripple. Since ripple is undesirable we must take additional precautions to reduce it.
- One obvious method is by simply reducing the discharge time constant. This can be achieved by increasing the value of C_1 or by increasing the resistance value of R_L.

SMOOTHING FILTERS:

a)

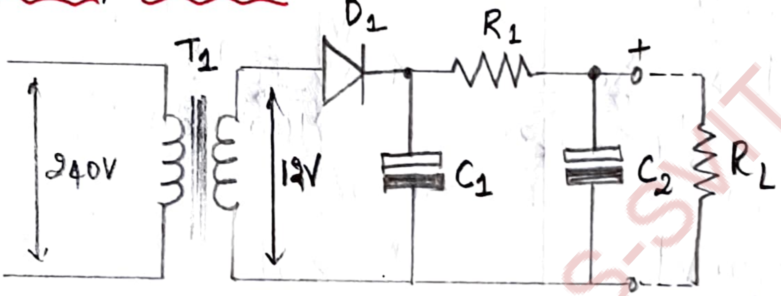


fig: Half wave rectifier circuit with R-C Smoothing filter

- Figure above shows a further refinement of the simple power supply circuit. This circuit employs two additional components R_1 and C_1, which act as a filter to remove the ripple.
- The value of C_1 is chosen so that the component Exhibits a negligible reactance at the ripple frequency.
- In effect, R_1 and C_1 act like a potential divider. The amount of ripple is reduced by an approximate factor equal to:

$$\frac{X_C}{\sqrt{R^2 + X_C^2}}$$

* At the ripple frequency, L_1 exhibits a high value of inductive reactance while C_1 exhibits a low value of capacitive reactance.
* The combined effect is that of an attenuator which greatly reduces the amplitude of the ripple while having a negligible effect on direct voltage.

Problem.

* The L-C Smoothing filter in a 50 Hz mains operated half-wave rectifier circuit consists of $L_1 = 10H$ and $C_2 = 1000\mu F$. If 1V of ripple appears at input of the circuit, determine the amount of ripple appearing at the output.

Soln:

$$f = 50\,Hz \text{ (HWR)}$$

$$X_C = \frac{1}{2\pi f C} = \frac{1}{2\pi \times 50 \times 1000 \times 10^{-6}}$$

$$\boxed{X_C = 3.18\,\Omega}$$

$$X_L = 2\pi f L$$

$$= 2\pi \times 50 \times 10$$

$$\boxed{X_L = 3140\,\Omega}$$

The amount of ripple at the output of the circuit will be approximately.

$$V = 1 \times \frac{X_C}{X_C + X_L} = \frac{3.18}{3140 + 3.18}$$

$$\boxed{V \simeq 0.001V}$$

FULL WAVE RECTIFIERS:

* The half-wave rectifier circuit is relatively inefficient as conduction takes place only on alternate half-cycles
* A better rectifier arrangement would make use of both

positive and negative half-cycles. These full-wave rectifier circuits offer a considerable improvement over their half-wave counterparts.

* There are two basic forms of full-wave rectifier.
1) The bi-phase rectifier
2) The Bridge rectifier.

BI-PHASE RECTIFIER CIRCUITS:

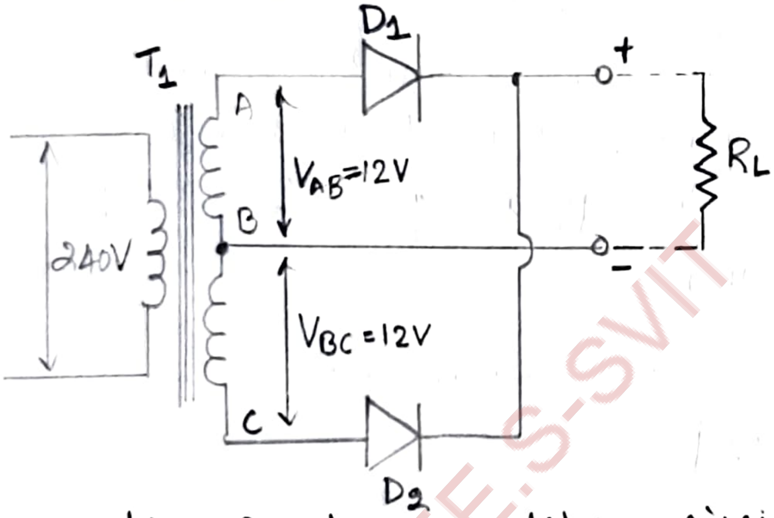


fig: Bi-phase rectifier circuit.

* The figure above shows a simple bi-phase rectifier circuit.

* Mains voltage (240V) is applied to the primary of step-down transformer (T1) which has two identical secondary windings, each providing 12V r.m.s.

* On positive half-cycles, point A will be positive with respect to point B. Similarly point B will be Positive with respect to point C.

* In this condition D_1 will allow conduction while D_2 will not allow conduction. Thus D_1 alone conducts on positive half-cycles.

* On negative half-cycles, point C will be positive w.r.t B. Similarly, point B will be positive w.r.t point A.

* In this condition, D_2 will allow conduction while D_1 will not allow conduction. Thus D_2 alone conducts on negative half-cycles.

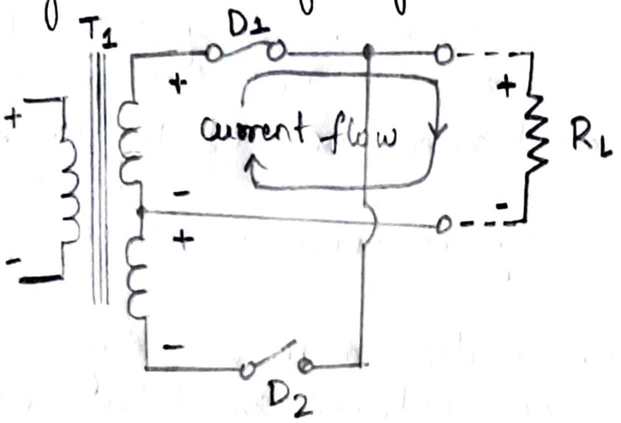


fig (a)

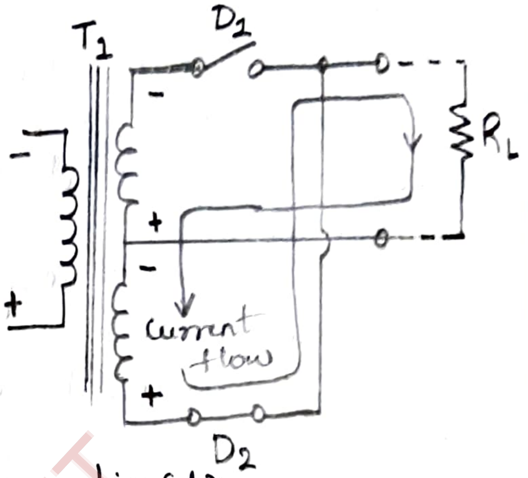
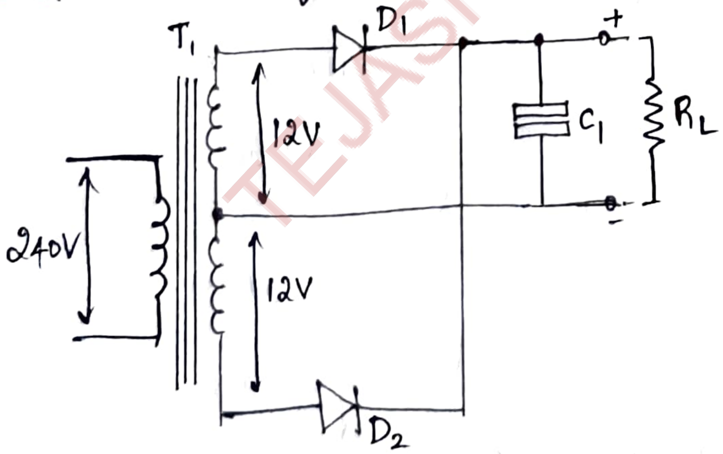


fig (b)

* The figure shows the bi-phase rectifier circuit with diodes replaced by switches. fig(a) shows D_1 conducting on a positive half-cycle while fig(b) shows D_2 conducting on a Negative half cycle.

Bi-phase rectifier with reservoir Capacitor.



* fig above shows how a reservoir Capacitor C_1 can be added to ensure that the output voltage remains at, or near, the peak voltage even when the diodes are not conducting.

- This component operates in exactly the same way as for the half-wave circuit. i.e it charges to approximately 16.3V at the peak of the positive half-cycle.
- The time required for C_1 to charge to the maximum (peak) level is determined by the charging circuit time constant.
- In this circuit, the series resistance comprises the secondary winding resistance together with the forward resistance of the diode and resistance of the wiring and connections.
- Hence C_1 charges very rapidly as soon as either D_1 or D_2 starts to conduct.
- The time required for C_1 to discharge is very much greater and discharge time contrast is determined by the capacitance value and the load resistance R_L.
- In practice, R_L is very much larger than the resistance of the secondary circuit and hence C_1 takes an appreciable time to discharge.
- During this time, D_1 and D_2 will be reverse biased and held in a non-conducting state. As a consequence, the only discharge path for C_1 is through R_L.

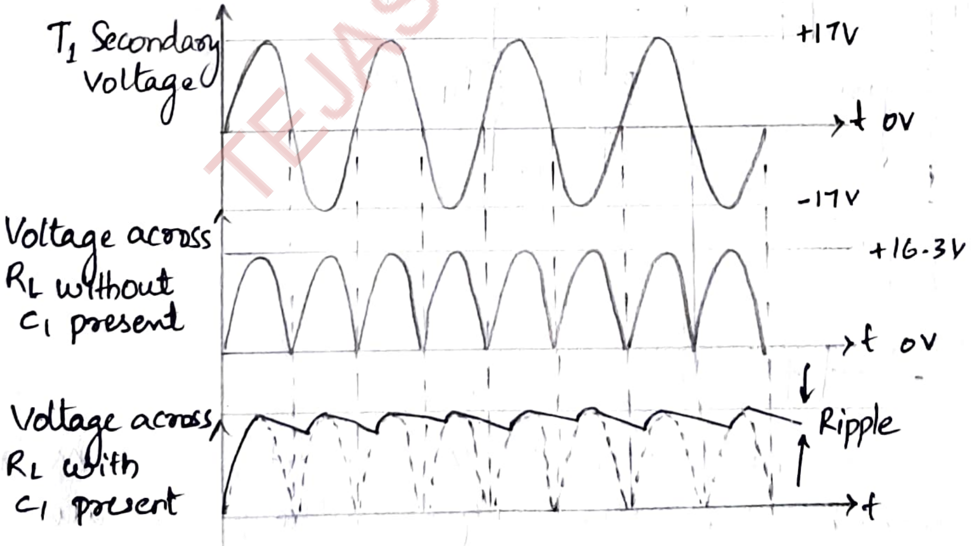


fig: Waveforms for the Bi-phase rectifier.

Bridge Rectifier circuits :

- * An alternative to the use of the bi-phase circuit is that of using a four-diode bridge rectifier. in which opposite pairs of diode conduct on alternate half-cycles.
- * This arrangement avoids the need to have two separate secondary windings.

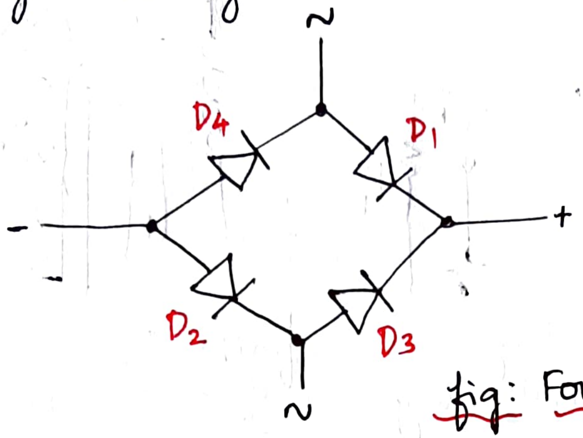


fig: Four diodes connected as a bridge.

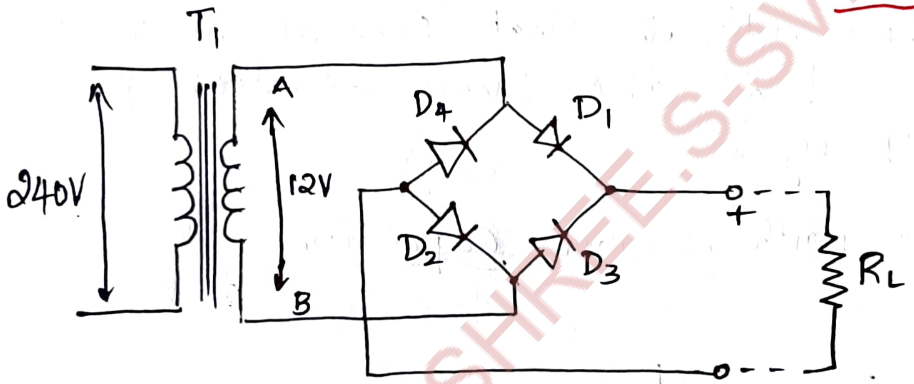


fig: Full wave bridge rectifier circuit.

- * A Full-wave bridge rectifier arrangement is shown in figure above.
- * Mains voltage (240V) is applied to the primary of a step-down transformer (T_1)
- * The Secondary winding provides 12V r.m.s and has a turns ratio of 20:1.
- * On positive half-cycles, point A will be positive with respect to point B. In this condition D_1 and D_2 will allow conduction while D_3 and D_4 will not allow conduction

- Conversely, on negative half-cycles, point B will be positive with respect to point A. In this condition, D_3 and D_4 will allow conduction while D_1 and D_2 will not allow conduction.

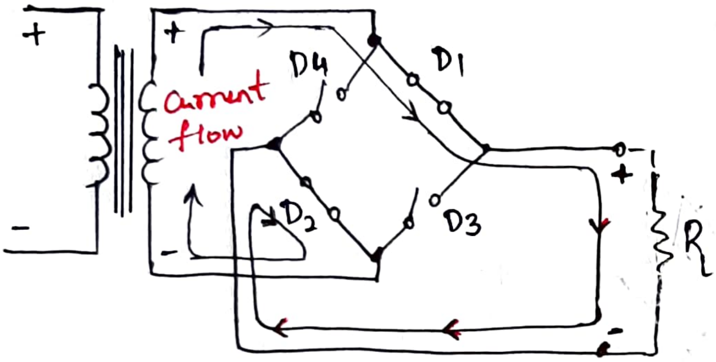
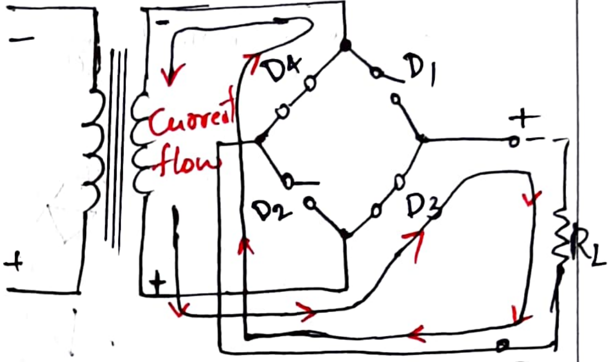


fig (a) D_1 & D_2 Conducting
D_3 & D_4 Non-Conducting



fig(b) D_3 & D_4 Conducting
D_1 & D_2 Non-Conducting

- Once again, the result is that current is routed through the load in the same direction on successive half cycles.

Bridge rectifier with reservoir Capacitor.

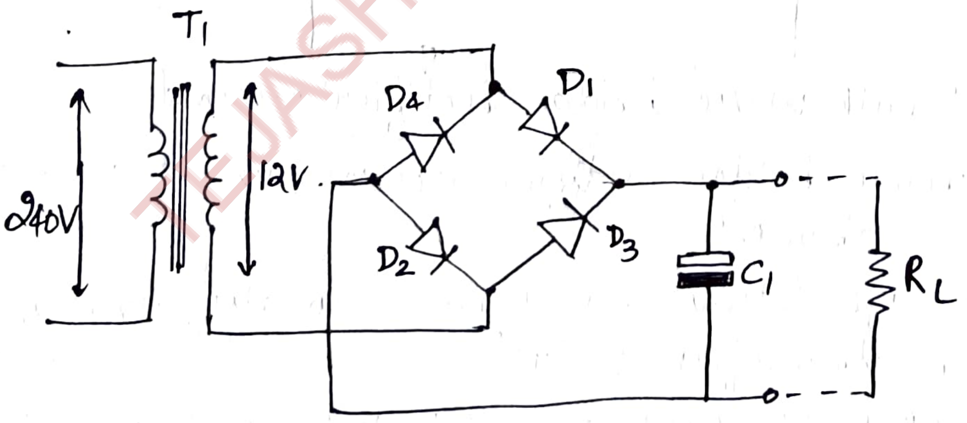


fig: Bridge rectifier with reservoir Capacitor

- The figure above shows how a reservoir Capacitor (C_1) can be added to maintain the output voltage when diodes are not conducting.

* This component operates in exactly the same way as for the bi-phase circuit, i.e it charges to approximately 16.3V at the peak of the positive half-cycle and holds the voltage at this level when the diodes are in their non-conducting states.

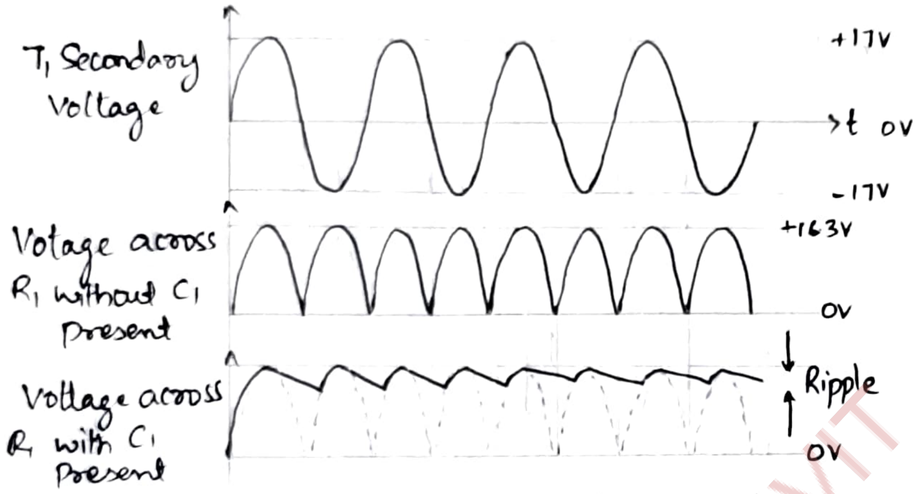


fig: waveforms for the bridge rectifier

VOLTAGE REGULATORS

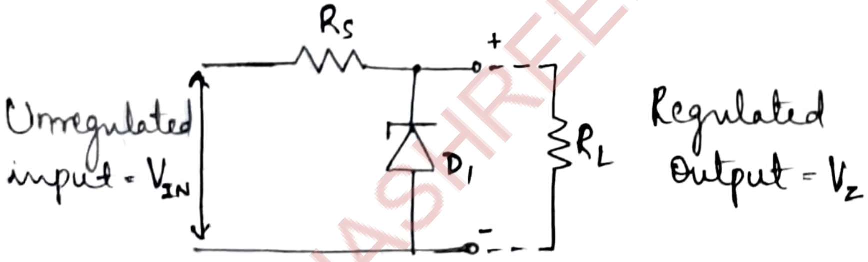


fig: Voltage regulator

* A simple voltage regulator is shown in figure above.
* R_S is included to limit the Zener current to a safe Value when the load is disconnected. when a load (R_L) is connected, the Zener current (I_Z) will fall as current is diverted into the load resistance (allow minimum current 2mA to 5mA in order to ensure that the diode regulates)
* The output Voltage (V_Z) will remain at the Zener voltage

until regulation fails at the point at which the potential divider formed by R_s and R_L produces a lower output voltage that is less than V_Z.

* The ratio of R_s to R_L is thus important. At the point at which the circuit just begins to fail to regulate:

$$\boxed{V_Z = V_I \times \frac{R_L}{R_L + R_S}}$$

Where V_{IN} is the unregulated input voltage. Thus the maximum value of R_s can be calculated from:

$$\boxed{R_{S(max)} = R_L \times \left[\frac{V_{IN}}{V_Z} - 1\right]}$$

* The power dissipated in the zener diode, will be given by $\boxed{P_Z = I_Z \times V_Z}$ Hence the minimum value for R_s can be determined from the off-load condition when:

$$R_{S(min)} = \frac{V_{IN} - V_Z}{I_Z} = \frac{V_{IN} - V_Z}{\left(\frac{P_{Z\,max}}{V_Z}\right)} = \frac{(V_{IN} - V_Z) \times V_Z}{P_{Z\,max}}$$

Thus,

$$\boxed{R_{S(min)} = \frac{V_{IN}V_Z - V_Z^2}{P_{Z\,max}}}$$

where $P_{Z\,max}$ is the maximum power dissipation for the Zener diode.

Problem:

* A 5V Zener diode has a maximum rated power dissipation of 500mW. If the diode is to be used in a simple regulator circuit to supply a regulated 5V to a load having a resistance of 400Ω, determine a

suitable value of series resistor for operation in conjunction with a supply of 9V.

Soln:

Given

$V_Z = 5V$

$V_{IN} = 9V$

$R_L = 400\Omega$

$P_{Z\,max} = 500mW = 0.5W$

$$R_{S\,max} = R_L \times \left(\frac{V_{IN}}{V_Z} - 1\right)$$

$$= 400 \times \left(\frac{9}{5} - 1\right)$$

$$\boxed{R_{S\,max} = 320\,\Omega}$$

$$R_{S\,min} = \frac{V_{IN}V_Z - V_Z^2}{P_{Z\,max}}$$

$$= \frac{(9\times5) - 5^2}{0.5}$$

$$\boxed{R_{S\,min} = 40\,\Omega}$$

Hence a suitable value for R_S would be 150Ω (roughly mid-way between the two extreames)

OUTPUT RESISTANCE AND VOLTAGE REGULATION:

- In a perfect power supply, the output voltage would remain constant regardless of the current taken by the load.
- In practice, however, the output voltage falls as the load current increases. To account for this fact, we say that the power supply has internal resistance (ideally this should be zero). This internal resistance appears

at the output of the supply and is defined as the change in output voltage divided by the corresponding change in output current. Hence,

$$\boxed{R_{out} = \frac{\text{Change in output voltage}}{\text{Change in output current}} = \frac{\Delta V_{out}}{\Delta I_{out}}}$$

Where ΔI_{out} represents a small change in output (load) current and ΔV_{out} represents a corresponding small change in output voltage.

- The regulation of a power supply is given by the relationship:

$$\boxed{\text{Regulation} = \frac{\text{Change in output Voltage}}{\text{Change in line input Voltage}} \times 100\%}$$

- Ideally, the value of regulation should be very small.
- Simple shunt zener diode regulators of the type shown are capable of producing values of regulation of 5% to 10%. More sophisticated circuits based on discrete components produce values between 1% and 5% and integrated circuit regulators often provide values of 1% or less.

Problem.

- The following data was obtained during a test carried out on a d.c power supply.

i) load test

Output voltage (no-load) = 12V

Output voltage (2A load current) = 11.5V

ii) Regulation test

Output voltage (mains input, 220V) = 12V

Output Voltage (mains input, 200V) = 11.9V

Determine (a) the equivalent output resistance of the power supply and (b) the regulation of the Power supply.

Soln: The output resistance can be determined from the load test data:

$$R_{out} = \frac{\text{Change in output Voltage}}{\text{Change in output current}} = \frac{12 - 11.5}{2 - 0}$$

$$\boxed{R_{out} = 0.25\Omega}$$

The regulation can be determined from the regulation test data:

$$\text{Regulation} = \frac{\text{Change in output Voltage}}{\text{Change in line input Voltage}} \times 100\%$$

$$\text{Regulation} = \frac{12 - 1.9}{220 - 200} \times 100\% = \frac{0.1}{20} \times 100\%$$

$$\boxed{\text{Regulation} = 0.5\%}$$

VOLTAGE MULTIPLIERS

- * By adding a second diode and Capacitor, we can increase the output of the simple-half wave rectifier that we seen before.
- * A voltage doubler is shown in figure below

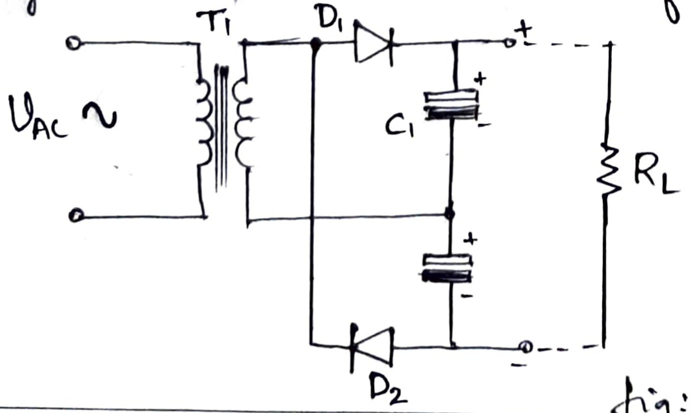


fig: Voltage doubler.

- * In this arrangement C_1 will charge to the positive peak secondary voltage while C_2 will charge to the negative peak secondary voltage.
- * Since the output is taken from C_1 and C_2 connected in series the resulting output voltage is twice that produced by one diode alone.
- * The voltage doubler can be extended to produce higher Voltages using the cascade arrangement shown in figure below.

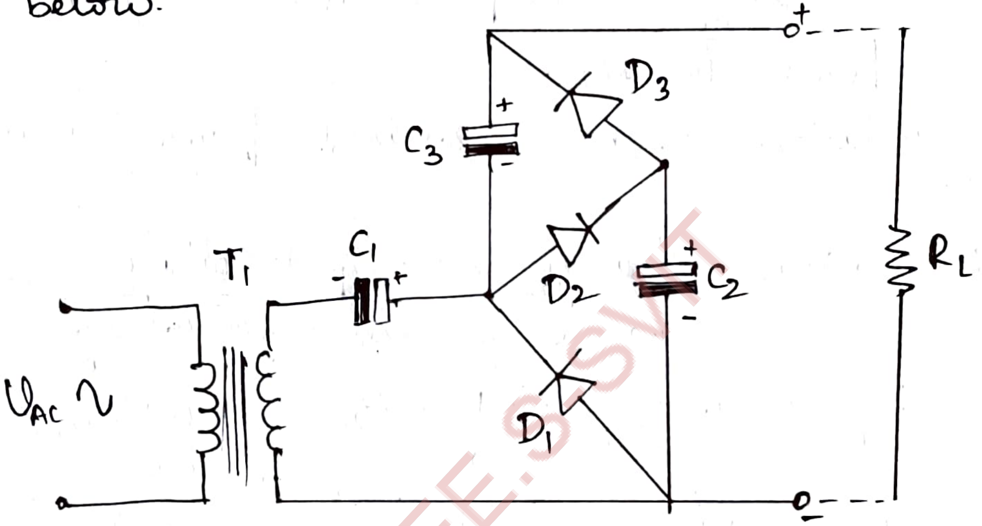


fig: A Voltage tripler.

- * Here C_1 charges to the positive peak secondary voltage, while C_2 and C_3 charge to twice the positive peak Secondary voltage. The result is that the output voltage is the sum of voltage across C_1 and C_3 which is three times the voltage that would be produced by a single diode.

TYPES OF AMPLIFIER

1) **a.c coupled amplifiers:**

In a.c coupled amplifiers, stages are coupled together in such a way that d.c levels are isolated and only the a.c components of a signal are transferred from stage to stage

2) **d.c coupled amplifiers:**

In d.c coupled amplifiers, stages are coupled together in such a way that stages are not isolated to d.c potentials. Both a.c and d.c signal components are transferred from stage to stage.

3) **Large - signal amplifiers:**

Large-signal amplifiers are designed to cater for appreciable voltage and/or current levels (typically from 1V to 100V or more)

4) **Small- Signal amplifiers:**

Small-signal amplifiers are designed to cater for low-level signals (less than 1V). Small- signal amplifiers have to be specially designed to combat the effects of noise.

5) **Audio-frequency amplifiers:**

Audio frequency amplifiers operate in the band of frequencies that is normally associated with audio signals. (E.g. 20Hz to 20kHz)

6) **Wideband amplifiers:**

Wideband amplifiers are capable of amplifying a very wide range of frequencies, typically from a few tens of hertz to several megahertz.

7) **Radio frequency amplifiers:**

Radio frequency amplifiers operate in the band of frequencies that is normally associated with radio signals (e.g from 100 KHz to over 1 GHz)

8) **Low-noise amplifiers:**

Low-noise amplifiers are designed so that they contribute negligible noise (signal disturbance) to the signal being amplified. These amplifiers are usually designed for use with very small signal levels (usually less than 10mV)

GAIN

- One of the most important parameters of an amplifier is the amount of amplification or gain that it provides.
- Gain is simply the ratio of output voltage to input voltage, output current to input current, or output power to input power.
- These three ratios give, respectively the voltage gain, current gain and power gain.

Thus,

$$\boxed{\begin{aligned}\text{Voltage gain } A_V &= \frac{V_{out}}{V_{in}}\\ \text{Current gain } A_I &= \frac{I_{out}}{I_{in}}\\ \text{Power gain } A_P &= \frac{P_{out}}{P_{in}}\end{aligned}}$$

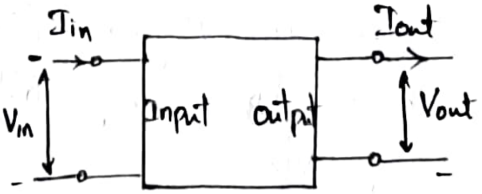


fig: Block diagram of amplifier showing input and output voltages and current.

Since power is the product of current and voltage ($P = V.I$) we can infer that,

$$\boxed{A_P = \frac{P_{out}}{P_{in}} = \frac{I_{out} \times V_{out}}{I_{in} \times V_{in}} = \frac{I_{out}}{I_{in}} \times \frac{V_{out}}{V_{in}} = A_I \times A_V}$$

Problem:

* An amplifier produces an output voltage of 2V for an input of 50mV. If the input and output currents in this condition are respectively 4mA and 200mA, determine a) voltage gain b) current gain c) power gain.

Soln: Given

$V_{out} = 2V$, $V_{in} = 50mV$, $I_{out} = 200mA$, $I_{in} = 4mA$

a) $A_V = \frac{V_{out}}{V_{in}} = \frac{2V}{50mV} = 40$

b) $A_I = \frac{I_{out}}{I_{in}} = \frac{200mA}{4mA} = 50$

c) $A_p = \frac{I_{out} \times V_{out}}{I_{in} \times V_{in}} = \frac{200mA \times 2V}{4mA \times 50mV} = \frac{0.4W}{200\mu W}$

$\boxed{A_p = 2000}$

(or) $A_p = A_V \times A_i$

$= 40 \times 50$

$\boxed{A_p = 2000}$

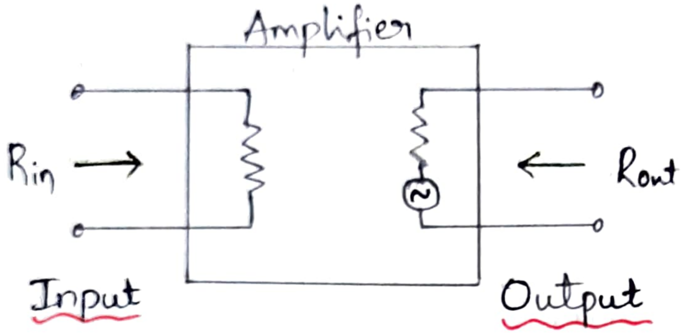
INPUT AND OUTPUT RESISTANCE:

* Input resistance is the ratio of input voltage to input current and it is expressed in ohms. The input of an amplifier is normally purely resistive in the middle of its working frequency range.

* In some cases, the reactance of the input may become appreciable. In such cases we would refer to input impedance rather than input resistance.

* Output resistance is the ratio of open-circuit output voltage to short-circuit output current and is measured in ohms.

* As with input resistance, the output of an amplifier is normally purely resistive and we can safely ignore any reactive component. If this is not the case, we would once again need to refer to output impedance rather than output resistance.



* The figure above shows the input and output resistances are 'seen' looking into the input and output terminals respectively.

Frequency Response:

The frequency response characteristics for Various types of amplifier are shown in figure below

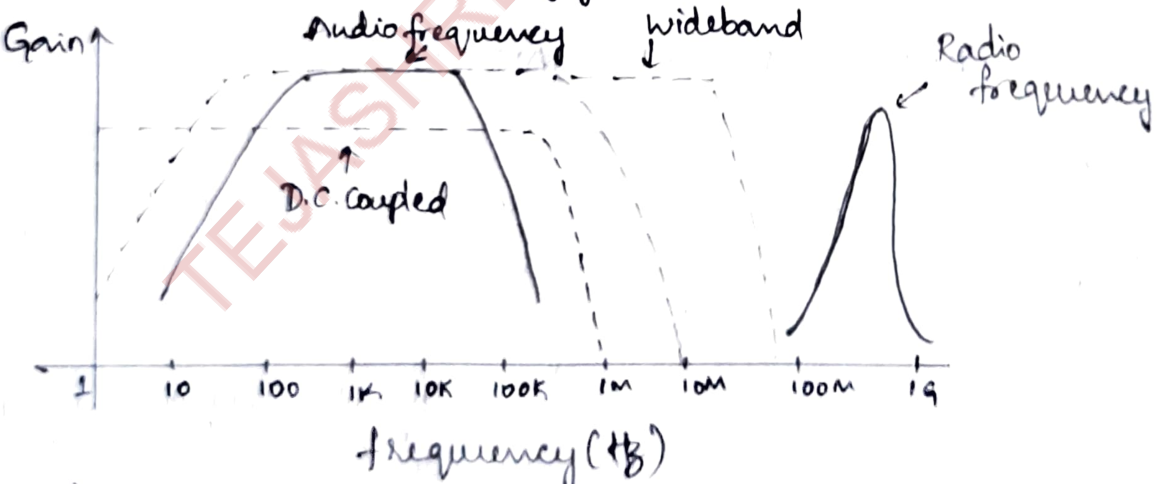
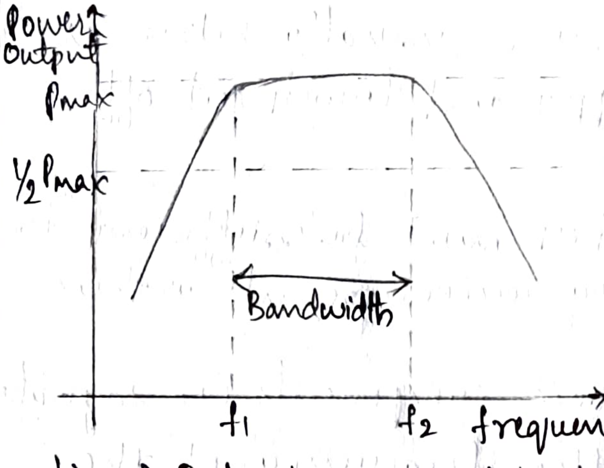


fig: Frequency response and Bandwidth

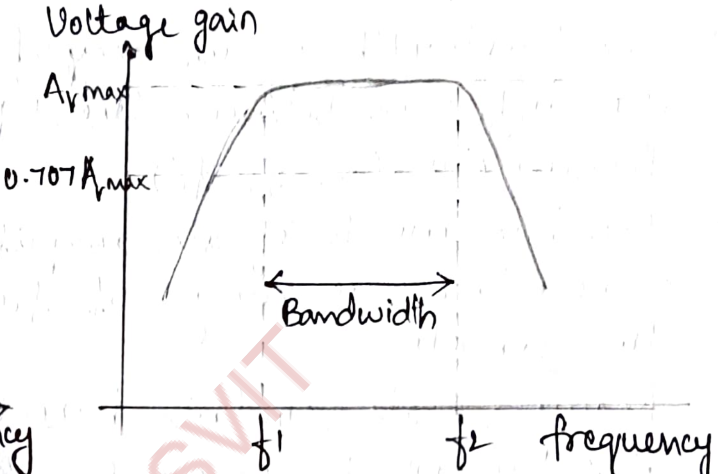
* The frequency response of an amplifier is usually specified in terms of upper and lower cut-off frequency of the amplifier.

* These frequencies are those at which the output power has dropped to 50% (otherwise known as −3dB points) or where the voltage gain has dropped to 70.7% of its mid-band value.

* The figures below show how the bandwidth can be expressed in terms of either power or voltage.



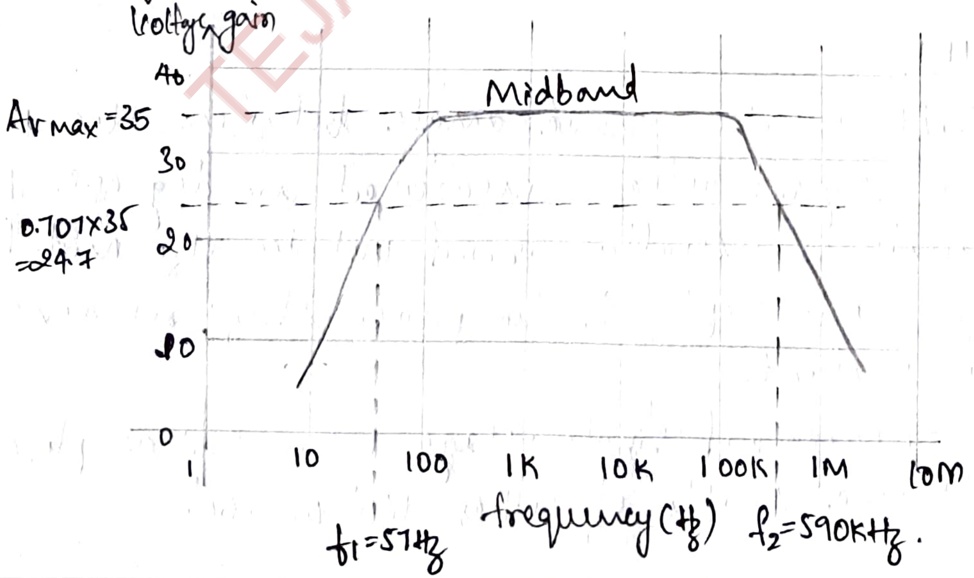
fig(a) Output power plotted again frequency



fig(b) output voltage plotted against frequency.

Problem:

* Determine the mid-band voltage gain and Upper and lower cut-off frequencies for the amplifier whose frequency response is shown below.



Soln: The voltage gain at the two cut-off frequencies can be Calculated from:

$$A_{V\,cut\text{-}off} = 0.707 \times A_{V\max} = 0.707 \times 35$$

$$A_{V\,cut\text{-}off} = 24.7$$

BANDWIDTH:

* The bandwidth of an amplifier is usually taken as the difference between the upper and lower cut-off frequencies (i.e, $f_2 - f_1$).
* The bandwidth of an amplifier must be sufficient to accomodate the range of frequencies present within the signals.
* Many signals contain harmonic components (i.e signals at 2f, 3f, 4f etc, where f is the frequency of the fundamental signal).
* To reproduce a square wave, for example, requires an amplifier with a very wide bandwidth, but it is not Possible to perfectly reproduce such a wave because a square wave comprises of an infinite series of harmonics.

PHASE SHIFT:

* phase shift is the phase angle between the input and output signal voltages measured in degrees. The measurement is usually carried out in the mid-band where, for most amplifiers, the phase shift remains relatively constant.
* Also the conventional single-stage transistor amplifiers provide phase shifts of either 180° or 360°.

NEGATIVE FEEDBACK :

- Many practical amplifiers use negative feedback in order to precisely control the gain, reduce distortion and improve bandwidth.
- The gain can be reduced to a manageable value of feeding back a small proportion of the output.
- The amount of feedback determines the overall (or closed-loop) gain.
- Because this form of feedback has the effect of reducing the overall gain of the circuit, this form of feedback is known as "negative feedback".
- An alternative form of feedback, where the output is fed back in such a way as to reinforce (strengthen) the input signal is known as "positive feedback".

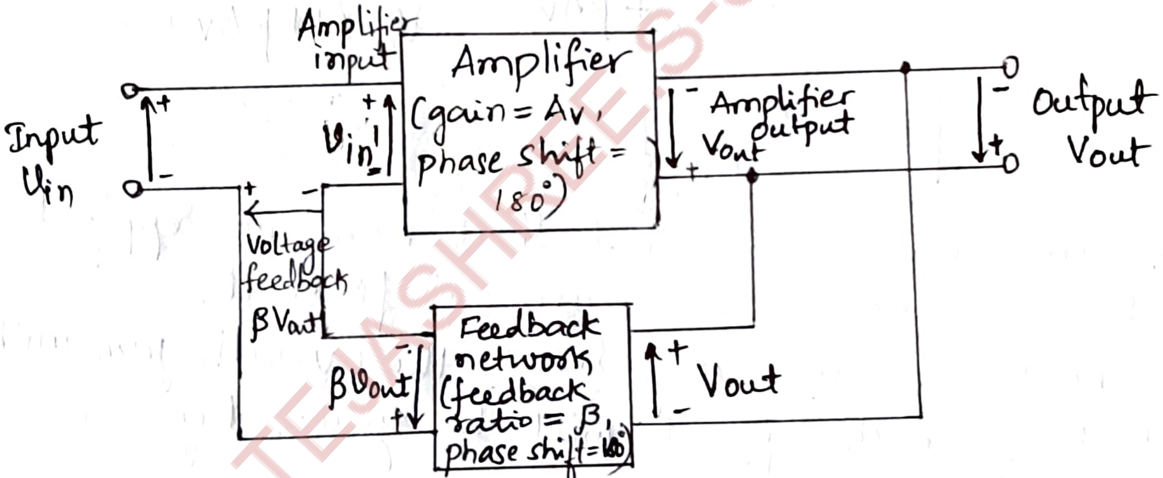


fig: Amplifier with negative feedback applied.

- Figure shows the block diagram of an amplifier stage with negative feedback applied.
- In this circuit, the proportion of the output voltage fed back to the input is given by β and the overall voltage gain will be,

$$\boxed{\text{Overall gain, } G = \frac{V_{out}}{V_{in}}}$$

Now, $V_{in}' = V_{in} - \beta V_{out}$ (Apply KVL)

$$V_{in} = V_{in}' + \beta V_{out} \rightarrow ①$$

and

$$V_{out} = A_V \times V_{in}' \rightarrow ②$$ {A_V - internal gain)

Hence, overall gain is given by

$$G = \frac{V_{out}}{V_{in}} = \frac{A_V \cdot V_{in}'}{V_{in}' + \beta V_{out}}$$

{Sub V_{out}}

$$G = \frac{A_V \cdot V_{in}'}{V_{in}' + \beta(V_{in}' A_V)}$$

$$G = \frac{A_V \cdot V_{in}'}{V_{in}' + \beta V_{in}'} = \frac{A_V \cdot \cancel{V_{in}'}}{\cancel{V_{in}'}(1 + \beta A_V)}$$

$$\frac{A_V \cdot V_{in}'}{V_{in}' + \beta V_{in}' A_V}$$

$$\boxed{G = \frac{A_V}{1 + \beta A_V}}$$

* Hence, the overall gain with feedback negative applied will be less than the gain without feedback.

* Furthermore, if A_V is very large, the overall gain with negative feedback applied will be given by

$$\boxed{G = \frac{1}{\beta}}$$

* Also, loop gain of a feedback amplifier is defined as the product of β and A_V.

Problem:

① An amplifier with negative feedback applied has an open-loop voltage gain of 50 and one-tenth of its output is feedback to the input (i.e $\beta = 0.1$). Determine the overall voltage gain with negative feedback applied.

Soln: Given

$A_V = 50, \ \beta = 0.1$

$$G = \frac{A_V}{1+\beta A_V} = \frac{50}{1+0.1\times 50} = \frac{50}{6}$$

$$\boxed{G = 8.33}$$

② If, in problem ①, the amplifier's open-loop voltage gain increases by 20%. determine the percentage increase in overall voltage gain.

Soln: The new value of voltage gain will be given by

$$A_V = A_V + 0.2A_V$$

$$= 50 + (0.2\times 50)$$

$$\boxed{A_V = 60}$$

Overall gain $G = \frac{A_V}{1+\beta A_V} = \frac{60}{1+(0.1\times 60)} = \frac{60}{7}$

$$\boxed{G = 7.14}$$

The increase in overall voltage gain, expressed as a percentage will thus be,

$$\frac{8.57-8.33}{8.33}\times 100\% = 2.88\%$$

③ An integrated circuit that produces an open-loop gain of 100 is to be used as the basis of an amplifier stage having a precise voltage gain of 20. Determine the amount of feedback required.

Soln: $G = \frac{A_v}{1+\beta A_v}$

Re-arranging the formula for β

$$\beta = \frac{1}{G} - \frac{1}{A_v}$$

$$\beta = \frac{1}{20} - \frac{1}{100}$$

$$\beta = 0.05 - 0.01$$

$$\boxed{\beta = 0.04}$$

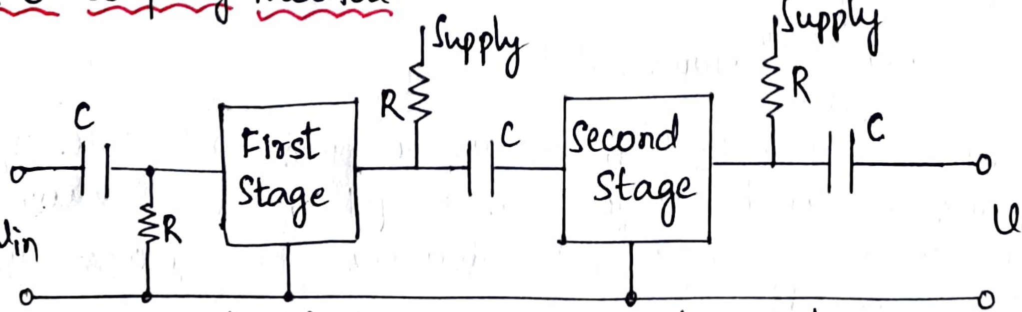
MULTISTAGE AMPLIFIERS:

- * In order to provide sufficiently large values of gain, it is frequently necessary to use a number of interconnected stages within an amplifier.
- * The overall gain of an amplifier with several stages (i.e, a multi-stage amplifier) is simply the product of the individual voltage gains. Hence,

$$\boxed{A_V = A_{V_1} \times A_{V_2} \times A_{V_3} \text{ etc.}}$$

- * However, Bandwidth of multi-stage amplifier will be less than the bandwidth of each individual stage.
- * In otherwords, an increase in gain results in decrease of a bandwidth.
- * Signals can be coupled between the individual stages of a multi-stage amplifier by using different methods.

a) R-C Coupling method.



fig(a) Typical R-C Coupling between stages.

* In this coupling method, the stages are coupled together using capacitors having a low reactance at the signal frequency and resistors.

(b) L-C Coupling Method:

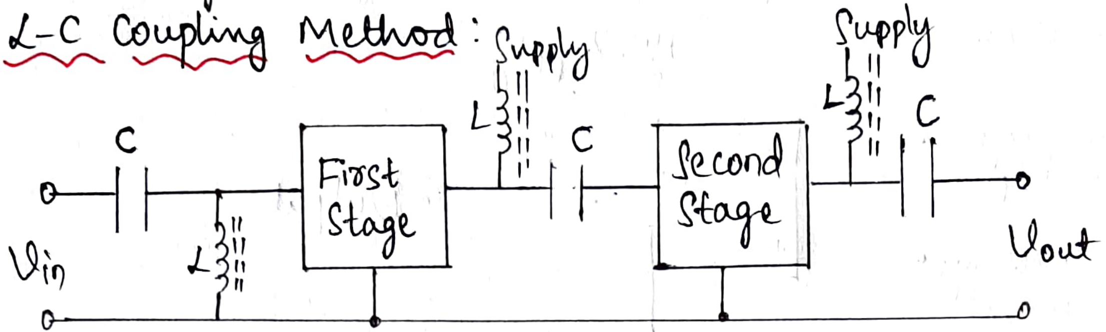


fig: Typical L-C coupling between stages.

* In this method, the inductors have a high reactance at the signal frequency. This type of coupling is generally only used in RF and high-frequency amplifiers.

(c) Transformer Coupling Method.

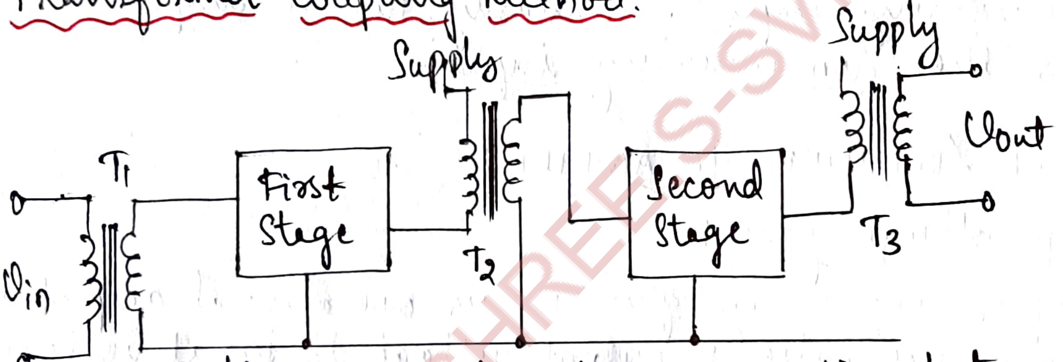


fig: Typical transformer coupling between stages

(4) Direct-Coupling Method:

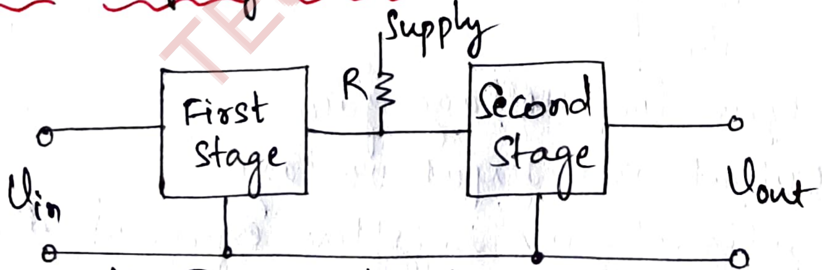
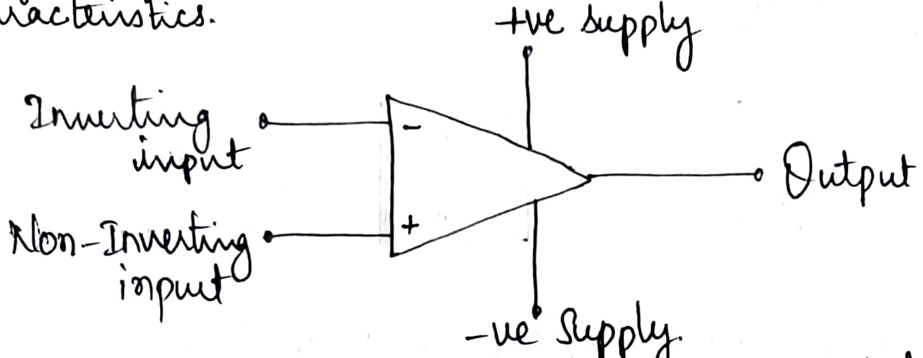


fig: Typical direct-coupling between stages.

OPERATIONAL AMPLIFIERS:

* Operational amplifiers are analogue integrated circuits designed for linear amplification that offer near-ideal characteristics.



* The '+' sign indicates zero phase-shift while '-' sign indicates 180° phase shift. Since 180° phase shift produces an inverted waveform, the '-' input is often referred to as the 'inverting input'. Similarly, the '+' input is known as the 'non-inverting input'.

OPERATIONAL AMPLIFIER PARAMETER:

① Open-loop voltage gain:

The open-loop voltage gain of an operational amplifier is defined as the ratio of output voltage to input voltage measured with no feedback applied.

$$\boxed{A_{V(OL)} = \frac{V_{OUT}}{V_{IN}}}$$

where $A_{V(OL)}$ = open-loop voltage gain

V_{OUT} & V_{IN} = Output and input voltages.

* The open-loop voltage gain is often expressed in decibels (dB) rather than as a ratio.

$$\boxed{A_{V(OL)} = 20 \log_{10} \frac{V_{OUT}}{V_{IN}}}$$

* Most operational amplifiers have open-loop voltage gain of 90 dB.

② CLOSED-loop voltage gain:

The closed-loop voltage gain of an operational amplifier is defined as the ratio of output voltage to input voltage measured with a small proportion of the output fed back to the input.

* closed loop voltage gain is once again the ratio of Output voltage to input voltage but with negative feedback applied.

Hence, $\boxed{A_{V(CL)} = \frac{V_{OUT}}{V_{IN}}}$

where, $A_{V(CL)}$ - open-loop voltage gain

V_{OUT} & V_{IN} – output and input voltages.

Problem:

* An operational amplifier operating with negative feedback produces an output voltage of 2V when supplied with an input of 400μV. Determine the value of closed-loop voltage gain.

Soln

$$A_{V(CL)} = \frac{V_{OUT}}{V_{IN}} = \frac{2}{400 \times 10^{-6}} = \frac{2 \times 10^{6}}{400}$$

$$\boxed{A_{V(CL)} = 5000}$$

$$A_{V(CL)} = 20 \log_{10} (5000) = 20 \times 3.7$$

$$\boxed{A_{V(CL)} = 74 dB}$$

③ Input Resistance:

The input resistance of an operational amplifier is defined as the ratio of input voltage to input current expressed in ohms.

$$\boxed{R_{IN} = \frac{V_{IN}}{I_{IN}}}$$

where R_{IN} - input resistance in ohms.
V_{IN} - input voltage, I_{IN} - input current.

Problem:

* An operational amplifier has an input resistance of 2MΩ. Determine the input current when an input voltage of 5mV is present.

Sol:

$$R_{IN} = \frac{V_{IN}}{I_{IN}}$$

Given: $R_{in} = 2M\Omega$
$V_{IN} = 5mV$

$$I_{IN} = \frac{V_{IN}}{R_{in}} = \frac{5\times10^{-3}}{2\times10^{6}} = 2.5\times10^{-9}A$$

$$\boxed{I_{in} = 2.5nA}$$

④ Output Resistance:

The Output resistance of an operational amplifier is defined as the ratio of open-circuit output voltage to short-circuit output current expressed in ohms.

* Typical values of output resistance range from less than 10Ω to around 100Ω, depending on configuration and amount of feedback employed.

$$\boxed{R_{OUT} = \frac{V_{OUT(OC)}}{I_{OUT(SC)}}}$$

where, R_{OUT} - output resistance in ohms
$V_{OUT(OC)}$ – open circuit output voltage.
$I_{OUT(SC)}$ - Short circuit output current.

⑤ Input-offset voltage:

The voltage that must be applied differentially to the operational amplifier input in order to make the output voltage exactly zero is known as input-offset voltage.

* Input offset voltage may be minimized by applying relatively large amounts of negative feedback.

⑥ Full-power Bandwidth:

The full-power bandwidth for an operational amplifier is equivalent to the frequency at which the maximum undistorted peak output voltage swing falls to 0.707 of its low-frequency value.

* Typical full-power bandwidths range from 10KHz - 1MHz for some high-speed devices.

⑦ Slew Rate:

Slew rate is the rate of change of output voltage with time, when a rectangular step input voltage is applied.

$$\boxed{\text{Slew rate} = \frac{\Delta V_{out}}{\Delta t}}$$

where ΔV_{OUT} - change in output voltage

Δt - corresponding interval of time.

* slew rate is measured in V/s and typical values range from 0.2 V/μs to over 20V/μs.

OPERATIONAL AMPLIFIER CHARACTERISTICS:

The characteristics for an Ideal - operational Amplifiers are

1. The open-loop voltage gain should be very high (ideally infinite)
2. The input resistance should be very high (ideally infinite)
3. The output resistance should be very low (ideally zero)
4. Full-power Bandwidth should be as wide as possible.
5. Slew-rate should be as large as possible.
6. Input offset should be as small as possible.

Problems (Slew-rate)

1. A perfect rectangular pulse is applied to the input of an operational amplifier. If it takes 4μs for the output voltage to change from −5V to +5V. determine the slew rate of the device.

Soln

$$\text{Slew rate} = \frac{\Delta V_{OUT}}{\Delta t} = \frac{10\,V}{4\mu s}$$

$$\text{Slew rate} = 2.5\,V/\mu s$$

2. A wideband operational amplifier has a slew-rate of 15V/μs. If the amplifier is used in a circuit with a voltage gain of 20 and a perfect step input of 100mV is applied to its input, determine the time taken for the output to change level.

Soln

The output voltage change will be 20×100 = 2000mV

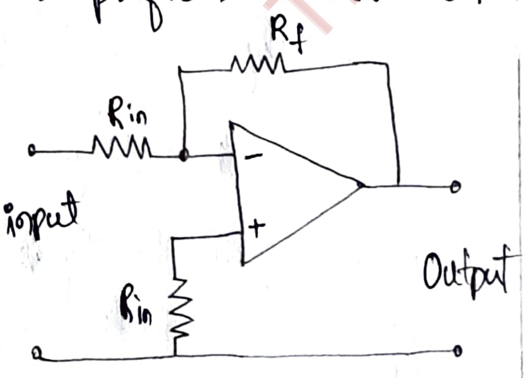
Re-arranging formula for slew-rate,

$$\Delta t = \frac{\Delta V_{OUT}}{\text{slew rate}} = \frac{2V}{15V/\mu s}$$

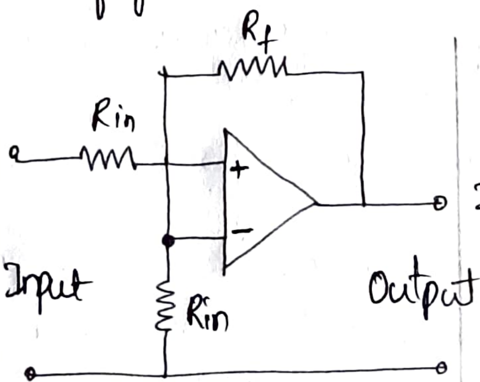
$$\Delta t = 0.133\mu s.$$

OPERATIONAL AMPLIFIER CONFIGURATION:

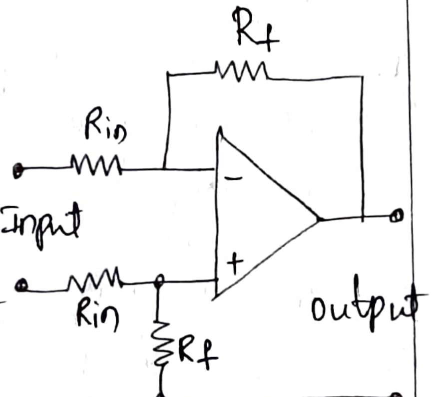
The three basic configurations for operational voltage amplifiers is shown in figure below.



a) Inverting Amplifier



b) Non-Inverting Amplifier



c) Differential Amplifier

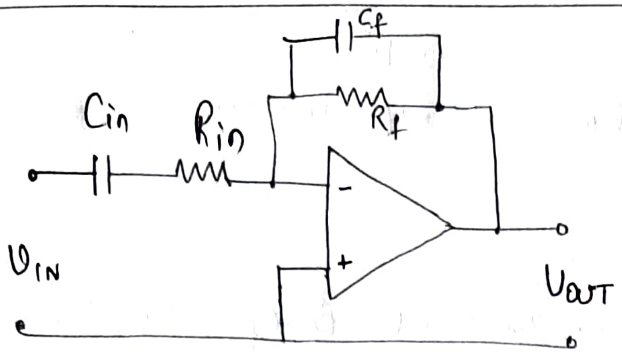


fig: Adding Capacitors to modify the frequency response of an inverting operational Amplifier.

$R_2 = A_V \times R_1$

$$f_1 = \frac{1}{2\pi C_{in} R_{in}} = \frac{0.159}{C_{in} R_{in}}$$

$$f_2 = \frac{1}{2\pi C_f R_f} = \frac{0.159}{C_f R_f}$$

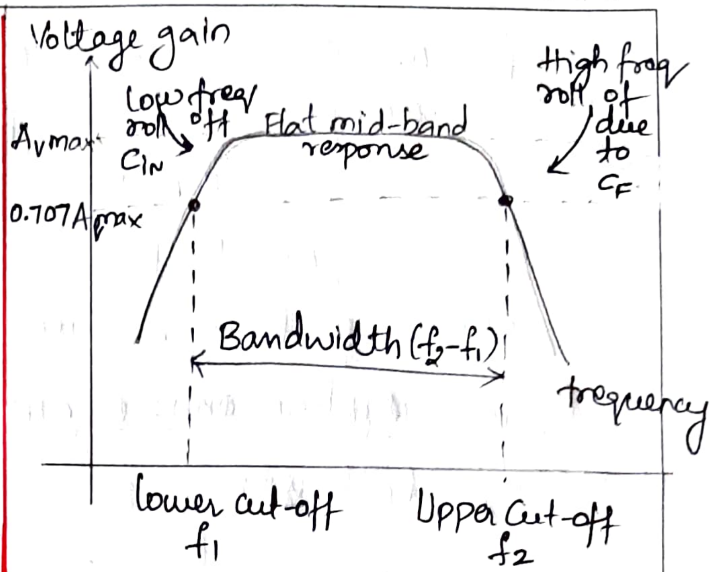


fig: Effect of adding capacitors Cin and Cf to modify the frequency response of an operational amplifier.

Problem:

① An inverting opamp is to operate according to the following specification.
voltage gain - 100
Input resistance (at mid-band -10 KHz)
Lower-cut-off frequency = 250 Hz
Upper-cut-off frequency = 15 KHz
Devise a circuit to satisfy the above specification using an operational amplifier.

Soln $R_{in} = 10\,k\Omega$

The nominal input resistance is the same as the value of R_{in}

$$A_V = \frac{R_2}{R_1}$$

$$f_1 = \frac{0.159}{C_{in} R_{in}}$$

$R_2 = 100 \times 10 k\Omega$, $C_{in} = \frac{0.159}{f_1 R_{in}} = \frac{0.159}{250 \times 10 \times 10^3}$

$R_2 = 1000 k\Omega$

$C_{in} = 63 \times 10^{-9} \Rightarrow \boxed{C_{in} = 63nF}$

$$f_2 = \frac{0.159}{C_f R_f} \Rightarrow C_f = \frac{0.159}{f_2 R_{in}} = \frac{0.159}{15\times10^3 \times 100\times10^3}$$

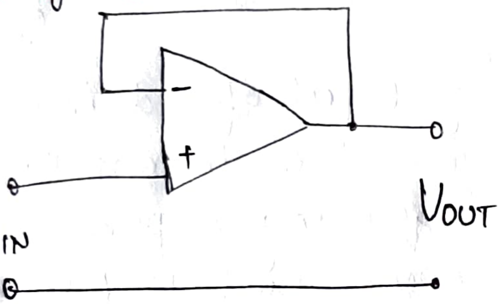
$$C_f = 0.106\times10^{-9}$$

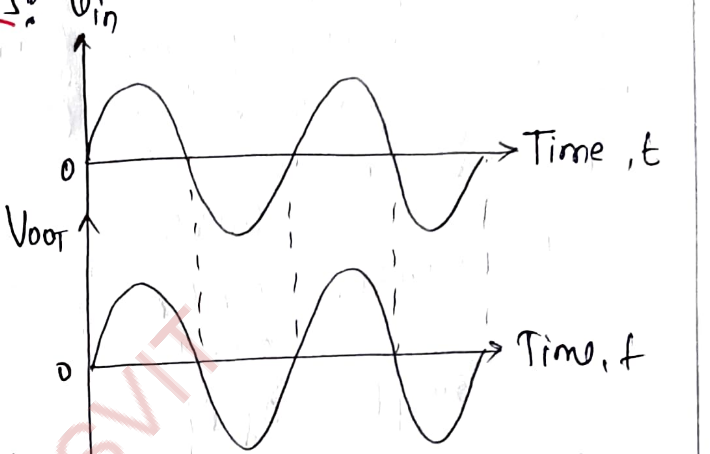
$$\boxed{C_f = 106pF}$$

choose preffered values C_{IN} as 68nF & $C_f = 100pF$.

OPERATIONAL AMPLIFIER CIRCUITS:

① Voltage Follower:





- * This circuit is essentially an inverting amplifier in which 100% of the output is fedback to the input.
- * The amplifier has an unity voltage gain, a very high input and output resistance.

② Differentiators:

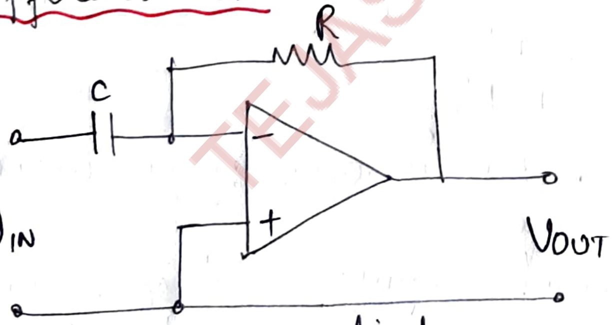
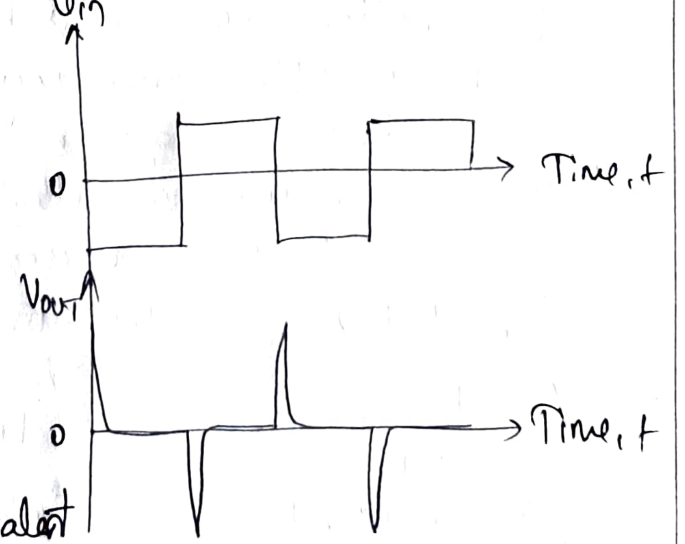
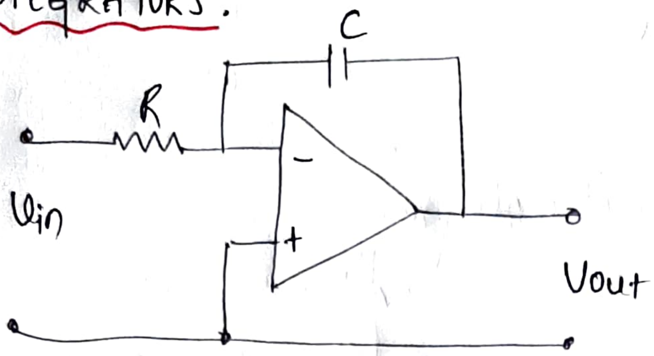


fig: Differentiator



- * A differentiator produces an output voltage that is equivalent to the rate of change of its input.
- * The square wave input is converted to a train of short duration pulses at the output.

③ INTEGRATORS:



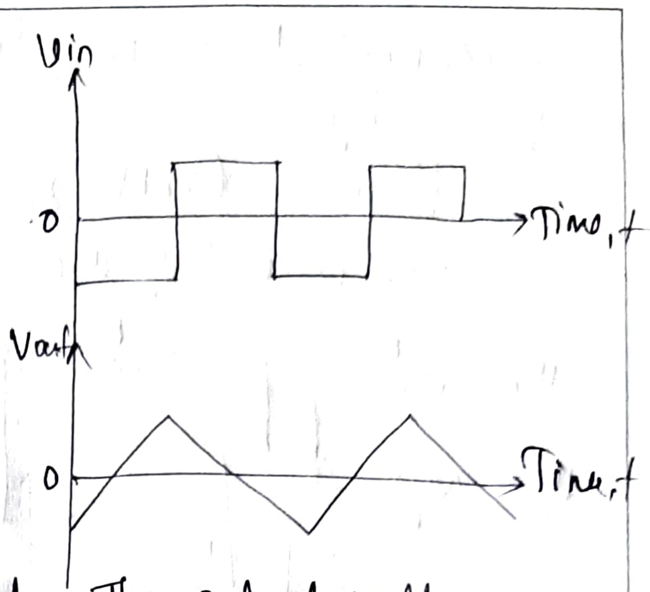
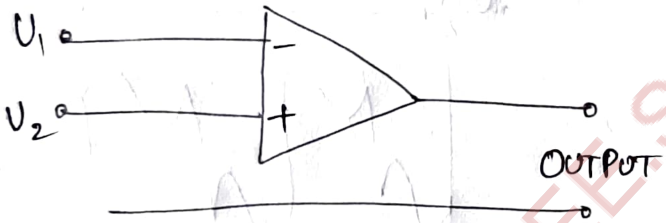


fig: An Integrator

- This circuit provides the opposite function to that of a differentiator. The output voltage ramp up or down according to the polarity of the input.

④ COMPARATORS:



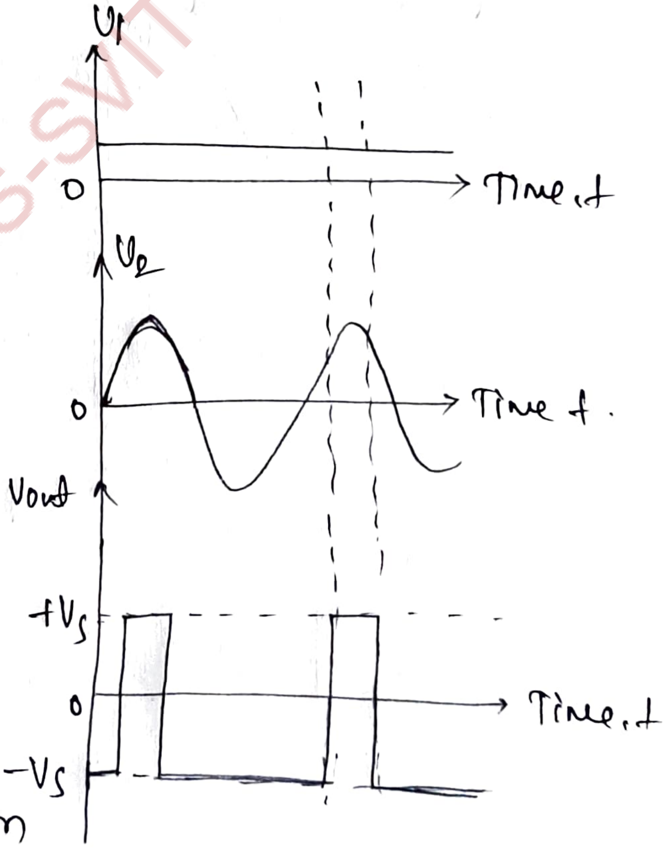


fig: Comparator

- Since no-negative feedback has been applied, this circuit uses the maximum gain of the operational amplifier.
- The output voltage produced by the operational amplifier will thus rise to the maximum possible value.

⑤ SUMMING AMPLIFIERS:

- This circuit produces an output that is the sum of its two input voltages. However, since the operational amplifier is connected in inverting mode, the output voltage is given by,

$$\boxed{V_{OUT} = -(V_1+V_2)}$$

where v_1 and v_2 are input voltages.

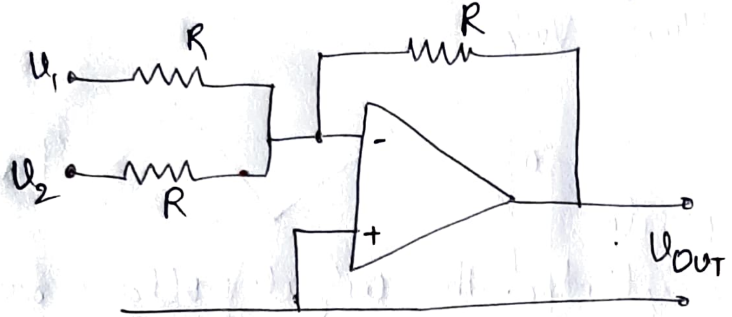
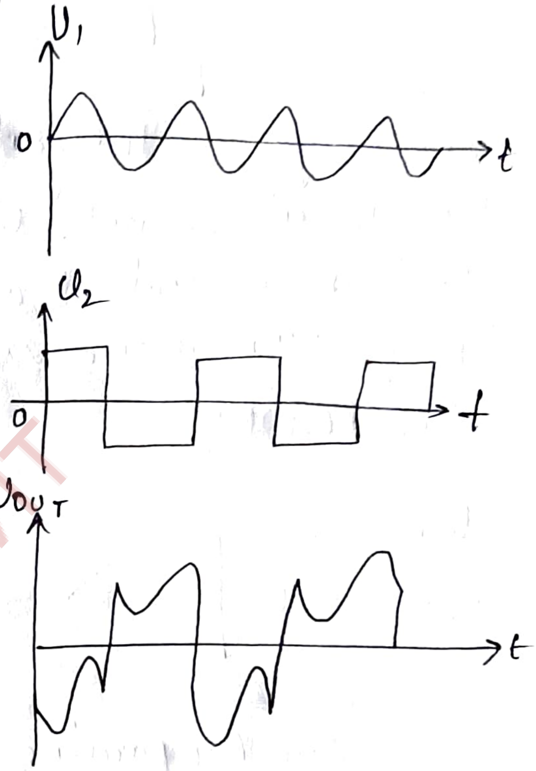


fig: A Summing Amplifier



OSCILLATORS:

POSITIVE FEEDBACK:

An alternative form of feedback, where the output is fed back in such a way as to reinforce the input is known as positive feedback.

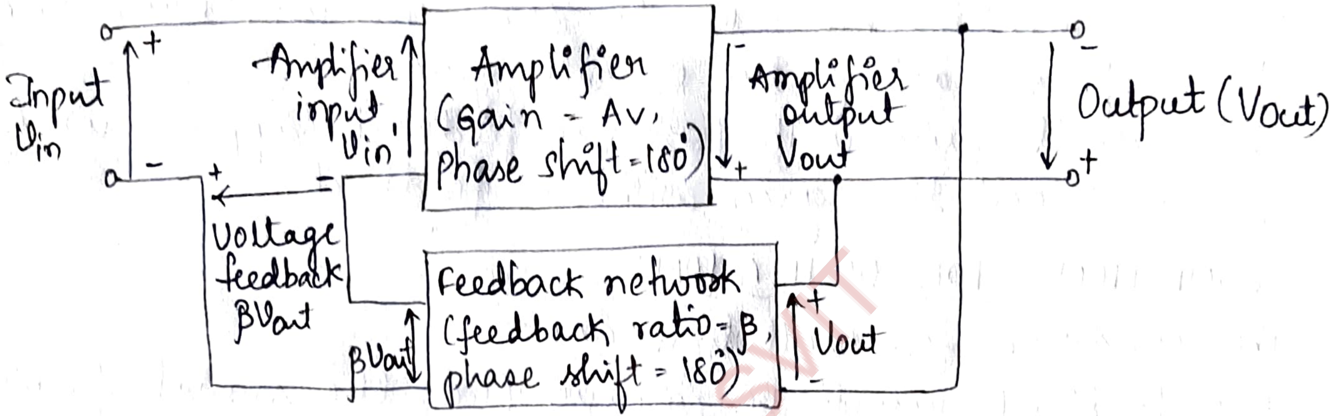


fig: Amplifier with positive feedback applied

- The figure above shows the block diagram of an amplifier stage with positive feedback applied.
- Note that the amplifier provides a phase shift of 180° and the feedback network provides a further 180°. Thus the overall phase shift is 0°. The overall voltage gain G is given by,

$$\boxed{\text{Overall gain, } G = \frac{V_{out}}{V_{in}}}$$

By applying Kirchoff's voltage law

$$\boxed{V_{in}' = V_{in} + \beta V_{out}}$$

Thus, $\boxed{V_{in} = V_{in}' - \beta V_{out}}$

and $\boxed{V_{out} = A_v \cdot V_{in}}$

where Av - internal gain of the amplifier.

Overall gain, $G = \frac{A_v . V_{in}'}{V_{in}' - \beta V_{out}} = \frac{A_v . V_{in}'}{V_{in}' - \beta(A_v \times V_{in}')}$

Thus, $\boxed{G = \frac{A_v}{(1-\beta A_v)}}$

* when loop gain βA_v approaches Unity, The denominator $(1-\beta A_v)$ will become close to zero. This will have the effect of increasing the overall gain.

* The overall gain with positive feedback applied will be greater than the gain without feedback.

CONDITIONS FOR OSCILLATION:

The condition for oscillation are:

1) the feedback must be positive (i.e, the signal fedback must arrive back in-phase with the signal at the input).

2) the overall loop voltage gain must be greater than 1. (i.e, the amplifier's gain must be sufficient to overcome the losses associated with any frequency selective feedback network).

* A number of circuits can be used to provide 180° phase shift, one of the simplest being a three Stage C-R ladder network that we shall meet next.

LADDER NETWORK OSCILLATOR:

* A simple phase-shift oscillator based on a three stage C-R ladder network is shown below.

* TR1 operates as a conventional common-emitter amplifier stage with R_1 and R_2 providing base bias potential and R_3 and C_1 providing emitter stabilization.

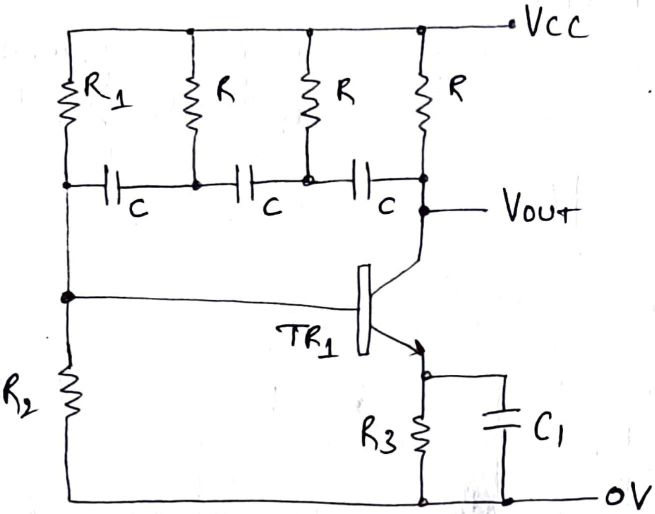


fig: Sine wave Oscillator based on three stage C-R ladder network.

- The total phase shift provided by the C-R ladder network (connected between collector and base) is 180° at the frequency of Oscillation.
- The transistor provides the other 180° phase shift in order to realize an overall phase shift of 360° or 0°.
- The frequency of oscillation of the circuit is

$$\boxed{f = \frac{1}{2\pi\sqrt{6}CR}}$$

- The loss associated with the ladder network is 29, thus the amplifier must provide a gain of at least 29 in order for the circuit to oscillate.

Problem

- Determine the frequency of Oscillation of a three-stage ladder network Oscillator in which C=10nF and R=10KΩ

Soln: Given C=10nF, R=10KΩ

$$f = \frac{1}{2\pi\sqrt{6}CR} = \frac{1}{2\pi\sqrt{6}\times10\times10^{-9}\times10\times10^{3}} = \frac{10^{4}}{15.386}$$

$$\boxed{f = 647Hz}$$

WEIN BRIDGE OSCILLATOR:

- An alternative approach to providing the phase shift required is the use of a wein bridge oscillator network.

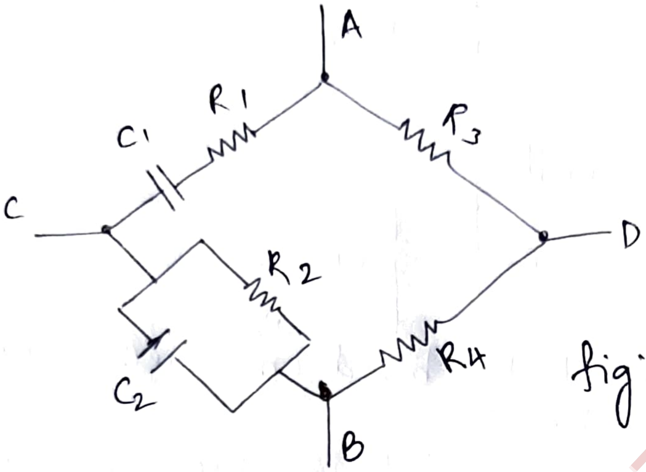


fig: A wein Bridge network.

- Like the C-R ladder, this network provides a phase-shift which varies with frequency.
- The input signal is applied to A and B while the output is taken from C and D.
- At one particular frequency, the phase shift produced by the network will be exactly zero. (input and output signals will be in-phase).
- If we connect the network to an amplifier producing 0° phase shift which has sufficient gain to overcome the losses of the wein bridge, oscillation will result.
- The minimum amplifier gain required to sustain oscillation is given by.

$$\boxed{A_V = 1 + \frac{C_1}{C_2} + \frac{R_2}{R_1}}$$

In most cases, $C_1 = C_2$ and $R_1 = R_2$, Hence the amplifier gain will be $\boxed{A_V = 3}$

- The frequency at which the phase-shift will be zero is given by

$$\boxed{f = \frac{1}{2\pi\sqrt{C_1C_2R_1R_2}}}$$

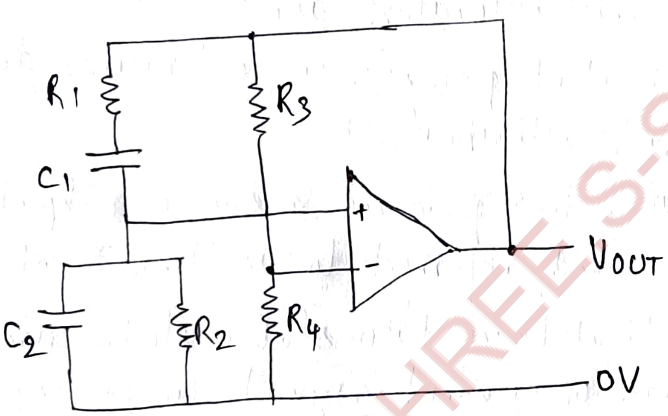
if $R_1 = R_2$ and $C_1 = C_2$

then, $f = \frac{1}{2\pi\sqrt{C^2R^2}}$, $\boxed{f = \frac{1}{2\pi RC}}$

where $R = R_1 = R_2$ and $C = C_1 = C_2$

Problem.

* Figure below shows a wein bridge oscillator based on an operational amplifier. If $C_1 = C_2 = 100nF$. Determine the output frequencies produced by this arrangement (a) when $R_1 = R_2 = 1K\Omega$ and b) when $R_1 = R_2 = 6K\Omega$.



Soln: a) when $R_1 = R_2 = 1K\Omega$

where $R = R_1 = R_2$ and $C = C_1 = C_2$

$$f = \frac{1}{2\pi RC} = \frac{1}{6.28 \times 100 \times 10^{-9} \times 1 \times 10^3}$$

$$\boxed{f = 1.59KHz}$$

b) when $R_1 = R_2 = 6K\Omega$

where $R = R_1 = R_2$ and $C = C_1 = C_2$

$$f = \frac{1}{2\pi RC} = \frac{1}{6.28 \times 100 \times 10^{-9} \times 6 \times 10^3}$$

$$\boxed{f = 265 Hz}$$

MULTIVIBRATORS:

- There are many occasions when we require a square wave output from an oscillator rather than a sine wave output.
- Multivibrators are a family of oscillator circuits that produce output waveforms consisting of one or more rectangular pulses.
- The term 'Multivibrator' simply originates from the fact that this type of waveform is rich in harmonics (i.e 'multiple vibrations').
- Multivibrators use regenerative (i.e, positive) feedback.
- The principal types of multivibrators are

1) Astable multivibrators: that provide a continuous train of pulses. (free-running multivibrators)

2) Monostable multivibrators: that produce a single output pulse (have one stable state and referred to as 'one-shot')

3) Bistable multivibrators: that have two stable states and require a trigger pulse or control signal to change from one state to another.

SINGLE-STAGE ASTABLE OSCILLATOR:

- A simple form of astable oscillator that produces a square wave output can be built using just one operational amplifier as shown below.

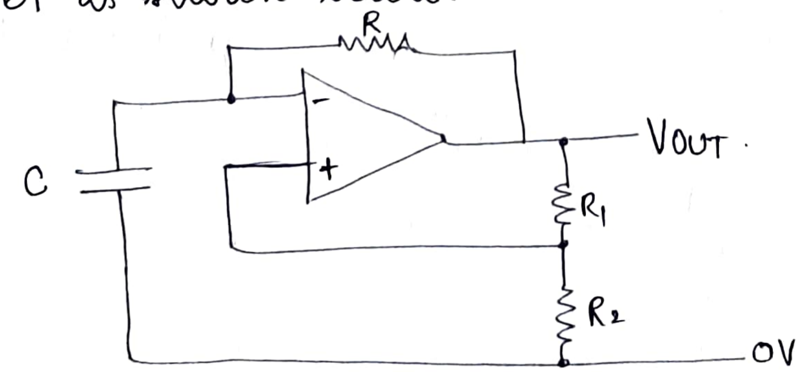


fig: Single-stage astable oscillator using op-Amps.

- The circuit employs positive feedback with the output fedback to the non-inverting input via the potential divider formed by R_1 and R_2.
- This circuit can make a very simple square wave source with a frequency that can be made adjustable by replacing R with a variable or preset resistor.
- Assume that C is initially uncharged and the voltage at the inverting input is slightly less than the voltage at the non-inverting input. The output voltage will rise rapidly to +Vcc and voltage at the inverting input will begin to rise exponentially as capacitor C charges through R.
- Eventually, the voltage at inverting input will have reached a value that causes the voltage at the inverting input to exceed at non-inverting input. At this point, the output voltage will rapidly fall to −Vcc. Capacitor C will then start to charge in the other direction and the voltage at the inverting input will begin to fall exponentially and process continues.
- The Upper threshold voltage is given by

$$\boxed{V_{UT} = V_{CC} \times \frac{R_2}{R_1 + R_2}}$$

- The lower threshold voltage is given by

$$\boxed{V_{LT} = -V_{CC} \times \frac{R_2}{R_1 + R_2}}$$

- Finally, the time for one complete cycle of the output waveform produced by the astable oscillator is given by $\boxed{T = 2CR \ln\left[1 + 2\left(\frac{R_2}{R_1}\right)\right]}$

CRYSTAL CONTROLLED OSCILLATORS:

- A requirement of some oscillators is that they accurately maintain an exact frequency of oscillation.
- In such cases, a quartz crystal can be used as the frequency determining element. The Quartz Crystal vibrates whenever a potential difference is applied across its faces. The frequency of oscillation is determined by the crystal's 'cut' and physical size.
- Most Quartz Crystals can be expected to stabilize the frequency of oscillation of a circuit to within a few parts in a million.
- Crystals can be manufactured for operation in fundamental mode over a frequency range extending from 100KHz to around 20MHz and for overtone operation from 20MHz to well over 100MHz.
- Figure below shows a simple crystal Oscillator circuit in which the crystal provides feedback from the drain to the source of a junction gate FET

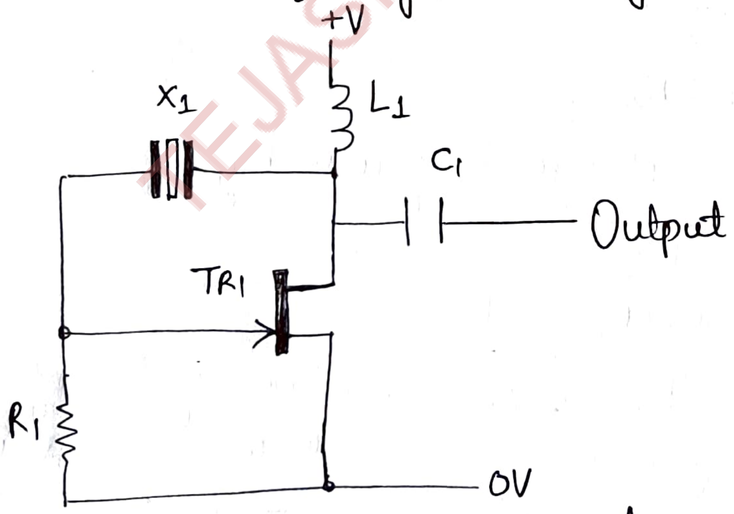


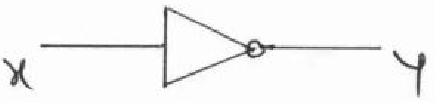
fig: A Simple JFET Oscillator.

Logic Circuits :-

Logic gates are circuits designed to produce the basic logic functions, AND, OR etc. Logic gates having more than one input & having only one output.

Basic Gates:

1) NOT Gate (Inverter):



Truth Table:

Input	Output
X	Y
0	1
1	0

* Inverters are used to complement the logical state i.e a logic 1 input results in a logic 0 output and viceversa.

The Boolean expression for the output is

$$\boxed{Y=\overline{X}}$$

2) AND Gate:-

Truth Table:

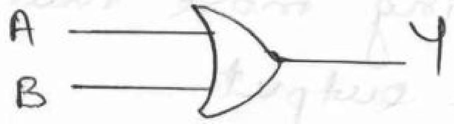
Input		output
A	B	Y
0	0	0
0	1	0
1	0	0
1	1	1

* AND gate produce output logic 1 when all inputs are simultaneously at logic 1.

* If one of the input is logic 0, output is logic 0.

$$\therefore \boxed{Y = A.B.}$$

3) OR Gate:



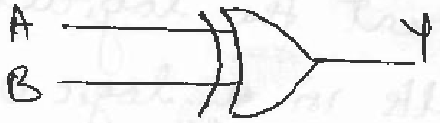
Truth Table:

Input		output
A	B	Y
0	0	0
0	1	1
1	0	1
1	1	1

* OR gates will produce a logic 1 output whenever any one or more inputs are at logic 1.

$$\therefore \boxed{Y = A + B}$$

4) XOR (Exclusive-OR gates):-



Truth Table:

Input		output
A	B	Y
0	0	0
0	1	1
1	0	1
1	1	0

* It produce output logic '1' if odd number of inputs are logic '1' (High).

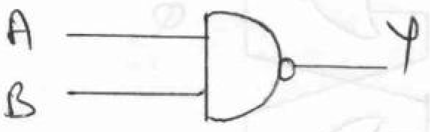
$$\therefore \boxed{Y = A.\bar{B} + \bar{A}.B}$$

Universal Gates:- Basic gates are realized by universal gates.

1) NAND Gate: (NOT-AND)

* It produces output logic 1 if one of the

input logic 0 (complement of AND gate).

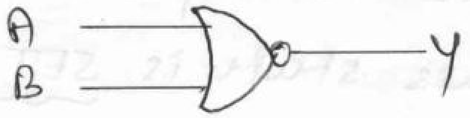


Truth Table:

Input		output
A	B	Y
0	0	1
0	1	1
1	0	1
1	1	0

$$\therefore \boxed{Y = A \cdot B}$$

2) NOR Gate (OR-NOT):



Truth Table:

Input		output
A	B	Y
0	0	1
0	1	0
1	0	0
1	1	0

$$\therefore \boxed{Y = \overline{A+B}}$$

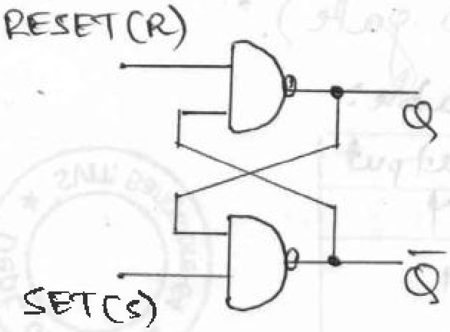
* If one of the input logic 1, the output logic 0, otherwise logic 1.

Bistables:-

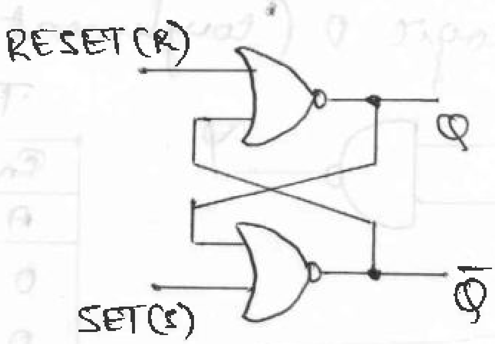
* The output of a bistable has two stable states (logic 0 → Reset, logic 1 → Set)

R-S Bistable:-

* The simplest form of bistable is the R-S bistable. It has two inputs (SET(S), RESET(R))
* Two outputs (Q and $\overline{Q}$, complement each other).
* Cross-Coupled NAND and NOR gates shown below.



fig(a): NAND Gate R-S Bistable



fig(b): NOR Gate R-S Bistable.

case i) S=1, R=0 is applied to inputs and output Q=1, $\bar{Q}$=0. This state is SET state.

case ii) S=0, R=1 is applied to inputs, and output Q=0, $\bar{Q}$=1. This state is RESET state.

Clocked SR Circuit :-

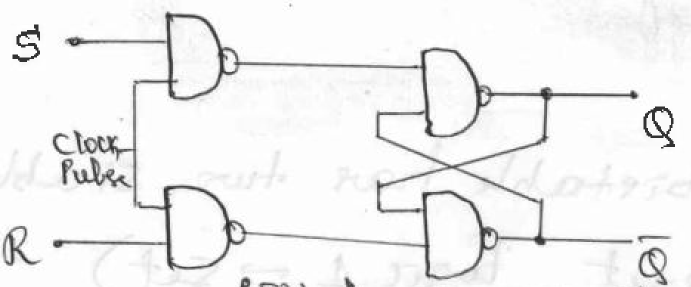


fig: Logic circuit.

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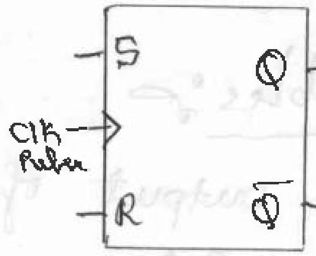


Truth Table:

Input		output	
S	R	Q	$\bar{Q}$
0	0	No change	
0	1	0	1
1	0	1	0
1	1	Invalid	

- If S=0, R=0 No change i.e previous output retaine.
- If S=0, R=1 ⇒ Q=0, R=1 RESET State
- If S=1, R=0 ⇒ Q=1, R=0 SET state
- If S=1, R=1 ⇒ Invalid.

D - Type Bistable :-

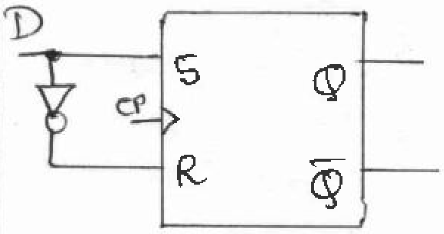


Fig: Symbol.

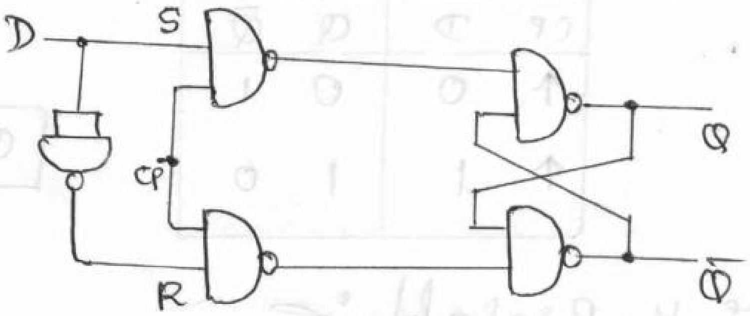


Fig: Diagram

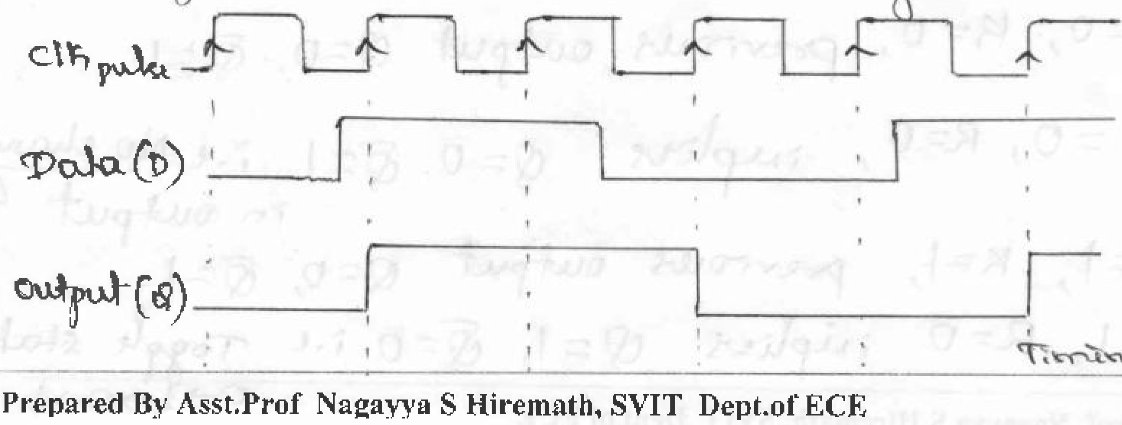
Case i) $D=0, \Rightarrow S=0, R=1$

output $Q=0, \overline{Q}=1$. This state is RESET state

Case ii) $D=1, \Rightarrow S=1, R=0$

output $Q=1, \overline{Q}=0$. This state is SET state.

- Therefore Data/Delay bistable produces output logic 1 if input 'D' logic 1 and output logic 0 if input 'D' logic 0.
- The data input (logic 0 or logic 1) is clocked into the bistable such that the output state only changes when the clock changes state.



Timing diagram.

Truth Table.

Input		output	
CP	D	Q	$\bar{Q}$
↑	0	0	1
↑	1	1	0

$\boxed{Q=D}$

J-K Bistable:-

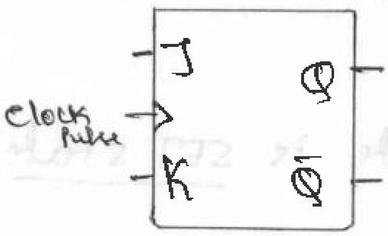


Fig: Symbol.

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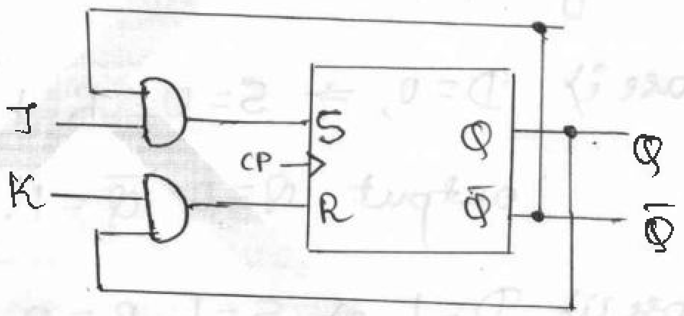


Fig: Logic diagram

Case i) J=1, K=0

Assume $Q=0$, $\bar{Q}=1$, implies $S=1$, $R=0$.

Since $S=1$, $R=0$ the output $Q=1$, $\bar{Q}=0$

This state is SET state.

Case ii) J=0, K=1, previous output $Q=1$, $\bar{Q}=0$

∴ $S=0$, $R=1$. This implies $Q=0$, $\bar{Q}=1$

Case iii) J=0, K=0, previous output $Q=0$, $\bar{Q}=1$

∴ $S=0$, $R=0$, implies $Q=0$, $\bar{Q}=1$ i.e No change in output

Case iv) J=1, K=1, previous output $Q=0$, $\bar{Q}=1$

∴ $S=1$, $R=0$ implies $Q=1$, $\bar{Q}=0$ i.e Toggle state.

0→1→0→1

Truth Table:

Input		Output	
J	K	Q	$\bar{Q}$
0	0	No change	
0	1	0	1
1	0	1	0
1	1	Toggle	

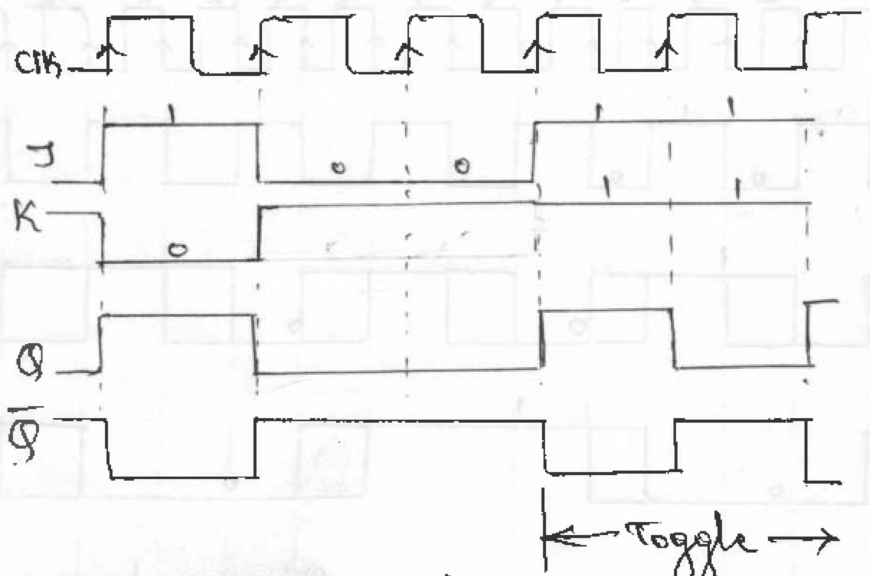


fig: Timing diagram

Four stage Binary Counter using J-K bistables:

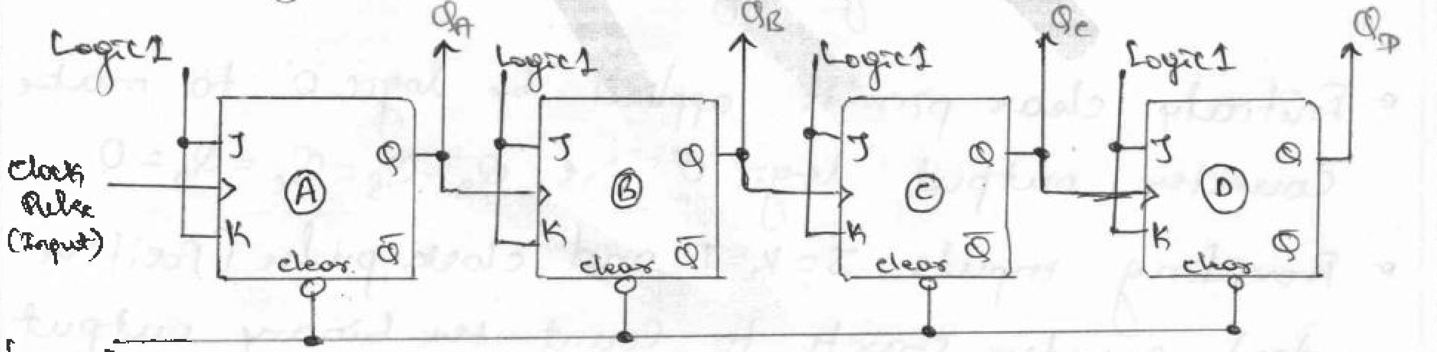
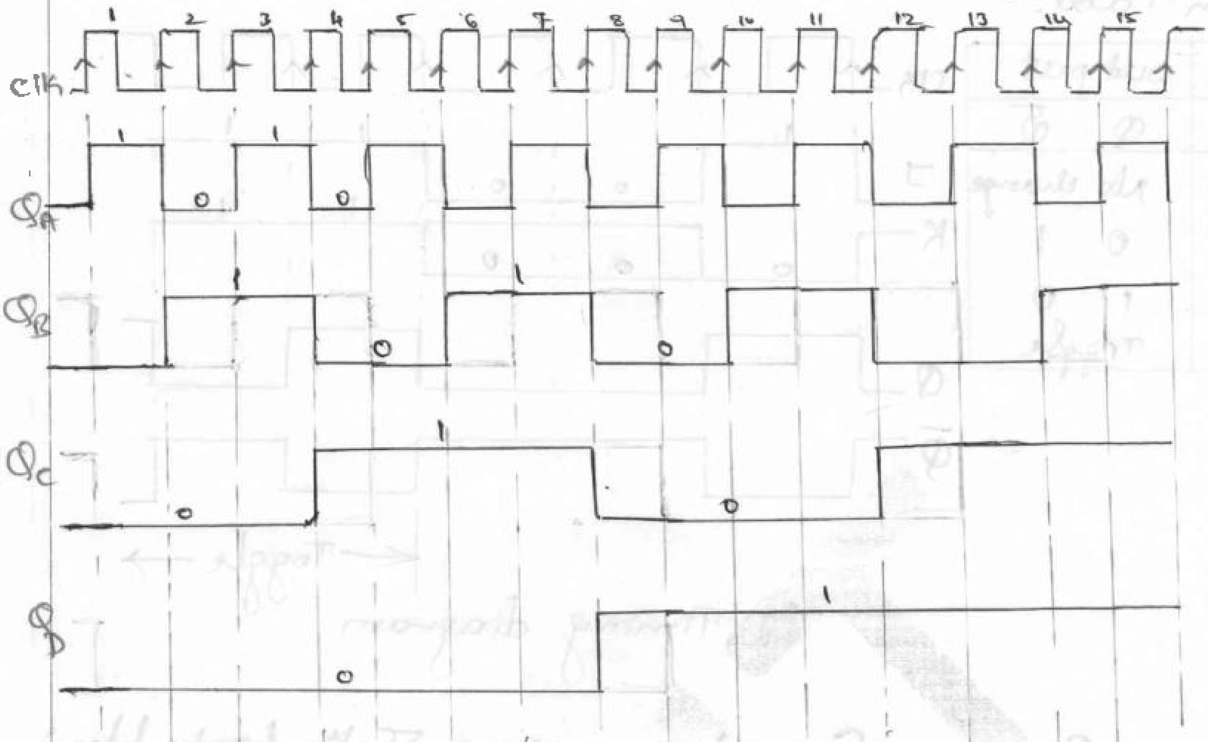


Fig: Logic diagram.

Truth Table:

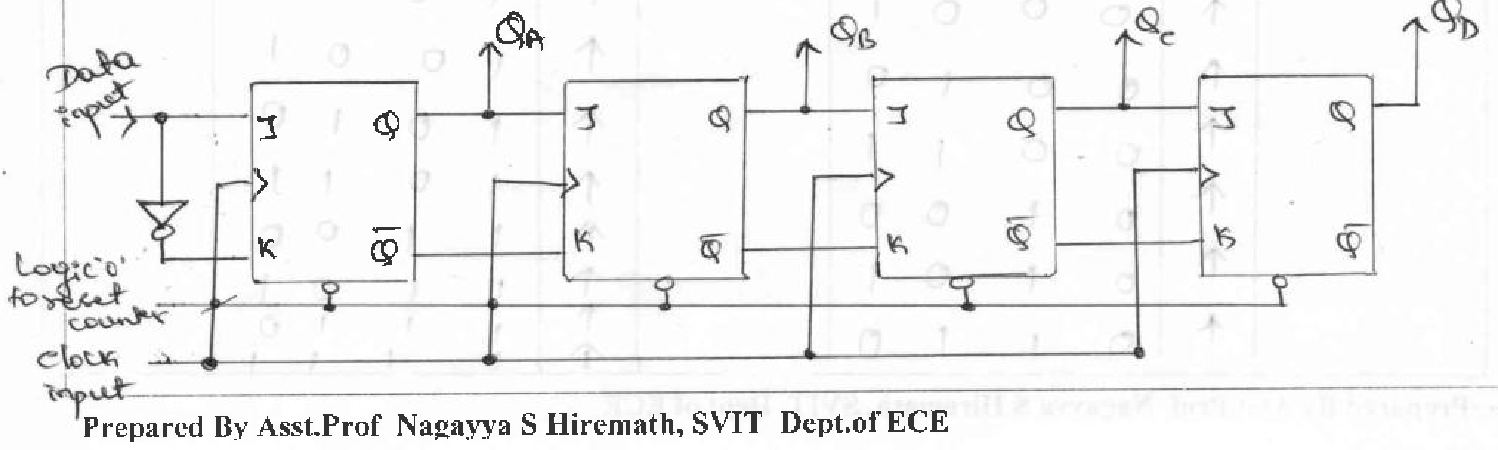
Clock Pulse Input	Q_D	Q_C	Q_B	Q_A
↑	0	0	0	0
↑	0	0	0	1
↑	0	0	1	0
↑	0	0	1	1
↑	0	1	0	0
↑	0	1	0	1
↑	0	1	1	0
↑	0	1	1	1
↑	1	0	0	0
↑	1	0	0	1
↑	1	0	1	0
↑	1	0	1	1
↑	1	1	0	0
↑	1	1	0	1
↑	1	1	1	0
↑	1	1	1	1



Timing diagram.

- Initialy clear pin is applied by logic '0' to make counter output logic '0' i.e $Q_A = Q_B = Q_C = Q_D = 0$.
- Providing inputs J=K=1 and clock pulse (Positive edge), counter starts to count see binary output from 0000 → 1111, as shown in waveform for every clock pulse. output of each bistable is connected as clock input to next bistable.

Four stage shift Register using JK bistable:-



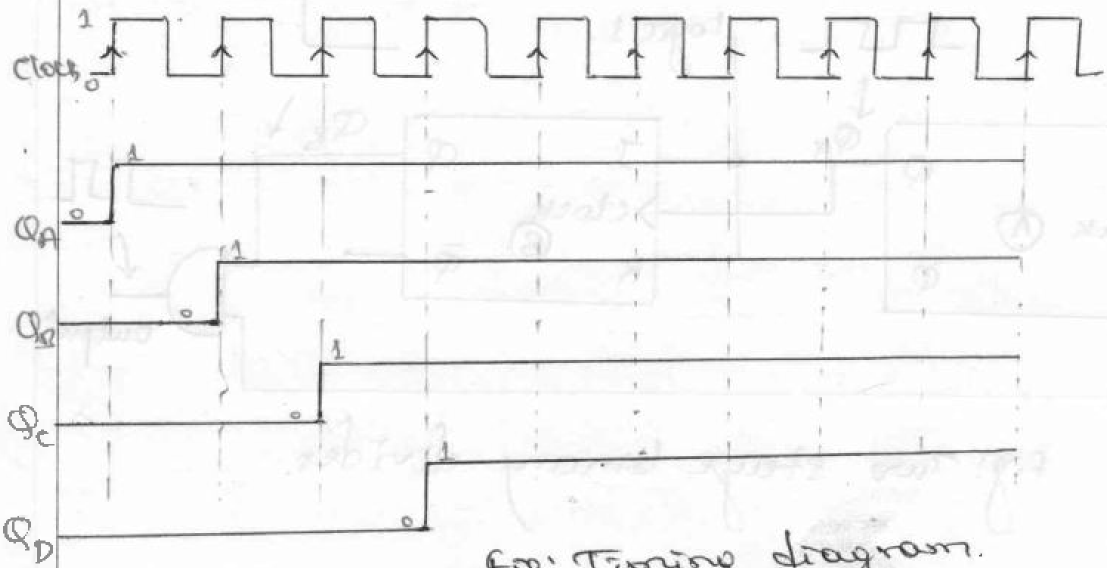


fig: Timing diagram.

Truth Table:

Clock Pulse	Q_A	Q_B	Q_C	Q_D
↑	1	0	0	0
↑	1	1	0	0
↑	1	1	1	0
↑	1	1	1	1

- Initially logic '0' is applied to reset pin to reset the counter $Q_A = Q_B = Q_C = Q_D = 0$.
- clock pulse will be applied simultaneously to all flipflop.
- For first clock pulse $Q_A = 1$, $Q_B = Q_C = Q_D = 0$
- For second clock pulse $Q_A = Q_B = 1$, $Q_C = Q_D = 0$
- For third clock pulse $Q_A = Q_B = Q_C = 1$, $Q_D = 0$.
- For Fourth clock pulse $Q_A = Q_B = Q_C = Q_D = 1$.
- Corresponding waveform as shown in figure.

Two-stage Binary Divider:-

- A two-stage binary divider as shown in fig using JK bistable.
- Initially logic '0' is applied to reset pin to reset the counter $Q_B = Q_A = 0$.

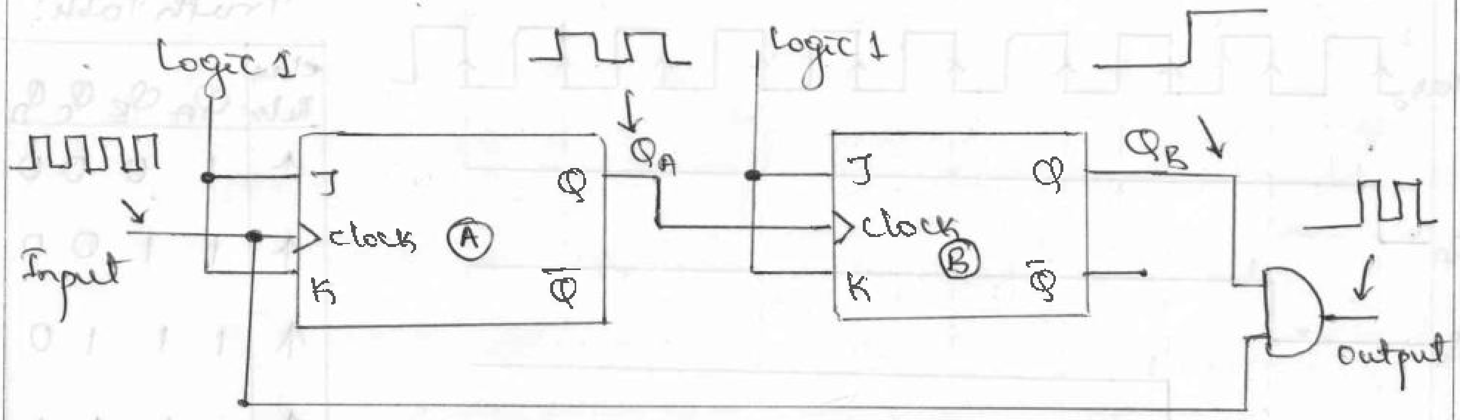


Fig: Two stage binary divider.

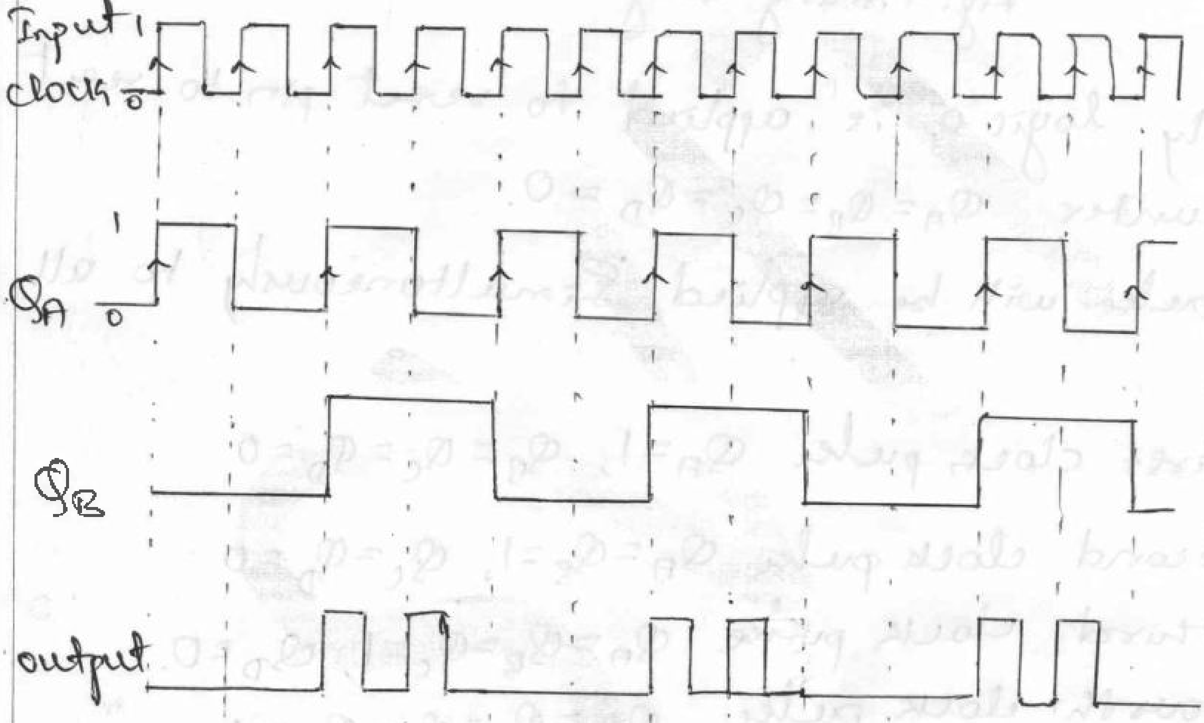


fig: waveform

- After applying first clock pulse $Q_B = 0$, $Q_A = 1$
- For second clock pulse $Q_B = 1$, $Q_A = 0$
- Counting pulse and AND gate o/p as shown in table.

Truth Table:

clock	Q_B	Q_A	Output
↑	0	0	0
↑	0	1	0
↑	1	0	1
↑	1	1	1

Data Representation:-

Number System:

- Decimal Numbers:

The base-10 number system Contains 10 characters (0, 1, 2, 3, 4, 5, 6. 7, 8. 9).

- Binary Numbers:

Only two character (0,1).

- Octal Numbers:

The octal numbers system has eight characters (0, 1, 2, 3, 4, 5, 6. 7).

- Hexadecimal Numbers:

The hexadecimal numbers has (0, 1 2, 3, 4, 5. 6. 7, 8. 9, A, B, C, D, E, F).

Problems: ① Find decimal equivalent of 11001.011_2

soln:

$$N = (1\times2^4) + (1\times2^3) + (0\times2^2) + (0\times2^1) + (1\times2^0) + (0\times2^{-1}) + (1\times2^{-2}) + (1\times2^{-3})$$

$$\therefore N = 16 + 8 + 0 + 0 + 1 + 0 + 0.25 + 0.125$$

$$N = 25.375_{10}$$

② Convert the number 010011110111.110101010_2 to hexadecimal.

soln: Partition the binary number into group of four, starting at the radix point and going left and Right.

$\underline{0100}\ \underline{1111}\ \underline{0111}.\underline{1101}\ \underline{0101}\ \underline{0000}$

4 F 7 D 5 0

$\therefore 4F7.D50_{16}$

Binary (Base2)	Denary (Decimal-base10)	Hexadecimal (Base-16)
0 0 0 0	0	0
0 0 0 1	1	1
0 0 1 0	2	2
0 0 1 1	3	3
0 1 0 0	4	4
0 1 0 1	5	5
0 1 1 0	6	6
0 1 1 1	7	7
1 0 0 0	8	8
1 0 0 1	9	9
1 0 1 0	10	A
1 0 1 1	11	B
1 1 0 0	12	C
1 1 0 1	13	D
1 1 1 0	14	E
1 1 1 1	15	F

③ Convert A3 into Binary.

Sol: A3 → 1010, 3 → 0011

$\therefore A3 = 1010\,0011_2$

Bit: Single binary number (0/1)

Byte: Group of eight bits. (8 bits)

word: Group of 16 bits (2 Bytes)

Doubleword: Group of 32 bits (2 word / 4 Bytes)

Nibble: Group of 4 bits.

Data Types:-

* A byte of data can be stored at each address within the total memory space of a microprocessor system.

Table: Data Types.

Data Type	Bits	Range of Values
Unsigned byte	8	0 to 255
Signed byte	8	-128 to +127
Unsigned word	16	0 to 65,535
Signed word	16	-32,768 to +32,767

1's Complement: find 1's Complement of 00101

Soln: 00101 ⟹ 11010 (∵ Replace 0 by 1, 1 by 0).

2's Complement: Add '1' to 1's Complement number.

Ex: 10010 ⇒ 1's Complement

$$\begin{array}{r} 01101 \\ +1 \\ \hline 01110 \end{array}$$

2's Complement number.

Signed Number:

If MSB bit is '1', the number is negative number or signed number.

Ex: 10000001

MSB (Most significant bit)

LSB (Least significant bit)

Data storage:-

- The semiconductor ROM within a microprocessor system provides storage for the program code as well as any permanent data that requires storage.

All of these data referred to as non-volatile because they remain intact when the power supply is disconnected.

- The semiconductor RAM within a microprocessor system provides storage for the transient data and variables that are used by program.

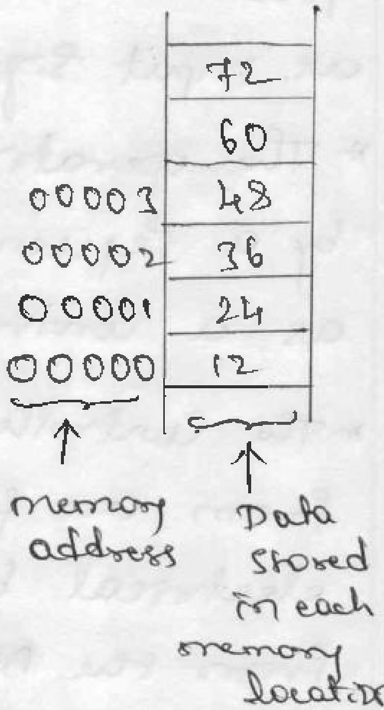
* It is important to note that any program or data stored in RAM will be lost when the power supply is disconnected.

* storage is usually Kbytes.

1 Kbyte = 1024 bytes

i.e 2^{10} = 1024 bytes.

Ex: 2K × 8bits = 2 Kbytes.



	72
	60
00003	48
00002	36
00001	24
00000	12

A Microcontroller System:-

* A microcontroller is a small computer on a single metal oxide-semiconductor (MOS) integrated circuit (IC) chip.

* A microcontroller contains a CPU along with memory and programmable input/output peripherals.

* Microcontrollers are used in ~~Comp~~ Consumer Electronics Products such as Toys, Cameras, Robots washing machine, Microwave ovens etc.

* The arrangement of a typical microcontroller system as shown in fig

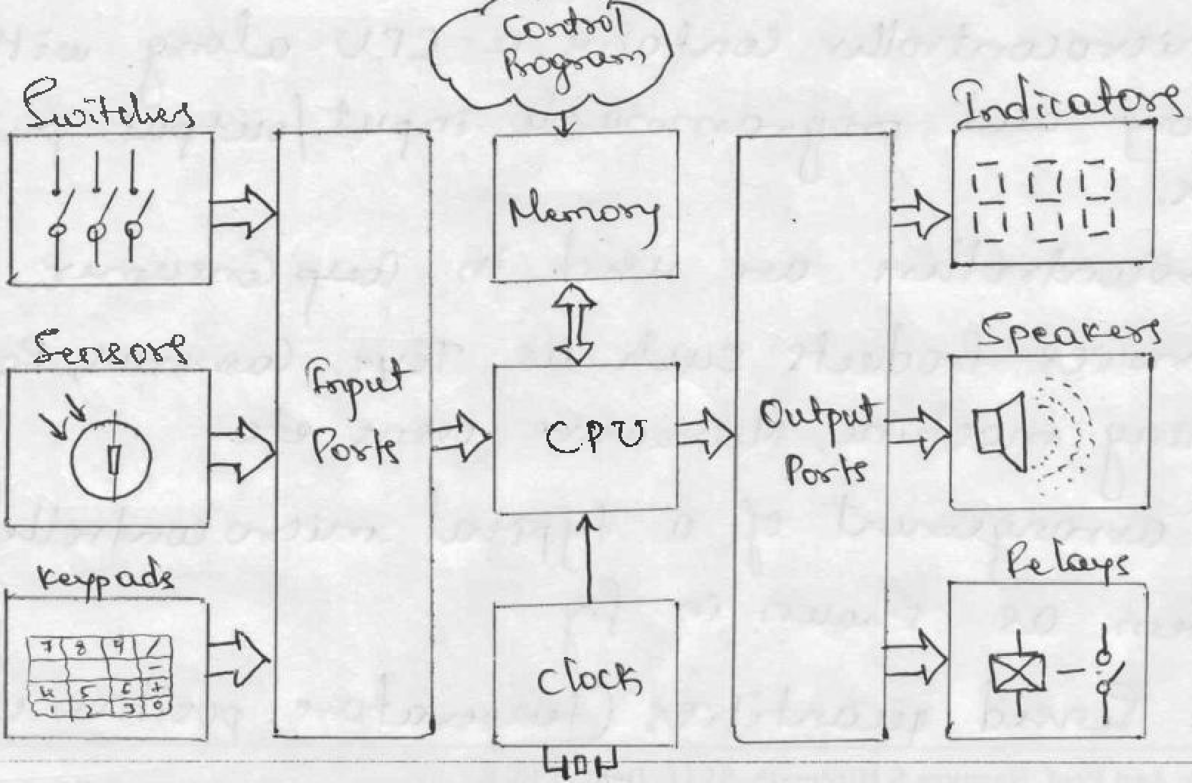
* The sensed quantities (temperature, position etc)

are converted to corresponding electrical signals by means of a number of sensors. The outputs from the sensors (Digital/Analogue form) are passed as input signals to the microcontroller.

* The operation of the microcontroller is controlled by a sequence of software instructions known as a control program.

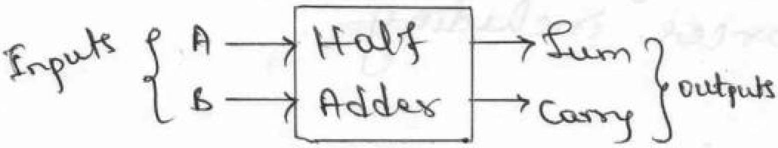
* The controlled device generally converts energy from one form into another form. For example electrical heater that converts electrical energy from the AC mains supply into heat energy.

* Microcontrollers must also have a CPU capable of performing simple arithmatic, logical and timing operations.



* The input port signals can be derived from a number of sources including.
  - Switches
  - Sensors
  - Keypads
* Output port signals can be connected to a number of devices, including.
  - LED indicators.
  - LED seven-segment displays
  - Motor and actuators, relays, transistor devices.
* Input devices supply information to the computer system from the outside world.
* Output devices are used to communicate information from a computer system to the outside world.
* ADC (Analogue to Digital Converter): It is connected at input side to provide digital signal to the input devices (converts analogue signal to Digital)
* DAC (Digital to Analogue Converter): It is connected at output to display analogue quantity.
* Interface circuit: Additional circuitry connected to the microcontroller.

Half Adder:-



Truth Table:

Inputs		outputs	
A	B	Sum	Carry
0	0	0	0
0	1	1	0
1	0	1	0
1	1	0	1

- This circuit adds two binary variables, yields a Sum and carry.

$$\text{Sum} = \bar{A}B + A\bar{B} = A \oplus B.$$

$$\text{Carry} = AB.$$

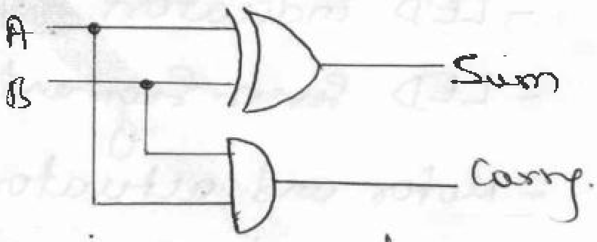
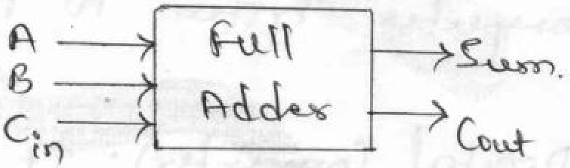


Fig: Logic diagram.

Full Adder:-

- This circuit adds three variables, yields a Sum and carry.



Truth Table:

Inputs			Outputs	
A	B	Cin	Sum	Carry
0	0	0	0	0
0	0	1	1	0
0	1	0	1	0
0	1	1	0	1
1	0	0	1	0
1	0	1	0	1
1	1	0	0	1
1	1	1	1	1

$$\text{Sum} = \bar{A}\bar{B}C_{in} + \bar{A}BC_{in} + A\bar{B}\bar{C}_{in} + ABC_{in}$$

$$\text{Sum} = C_{in}(\bar{A}\bar{B} + AB) + \bar{C}_{in}(\bar{A}B + A\bar{B})$$

$$\text{Sum} = C_{in}(\overline{A \oplus B}) + C_{in}(A \oplus B)$$

$$\boxed{\text{Sum} = A \oplus B \oplus C_{in}}$$

$$C_{out} = \bar{A}BC_{in} + A\bar{B}C_{in} + AB\bar{C}_{in} + ABC_{in}$$

$$C_{out} = C_{in}(\bar{A}B + A\bar{B}) + AB(\bar{C}_{in} + C_{in})$$

$$\boxed{C_{out} = C_{in}(A \oplus B) + AB}. \quad \because \bar{C}_{in} + C_{in} = 1$$

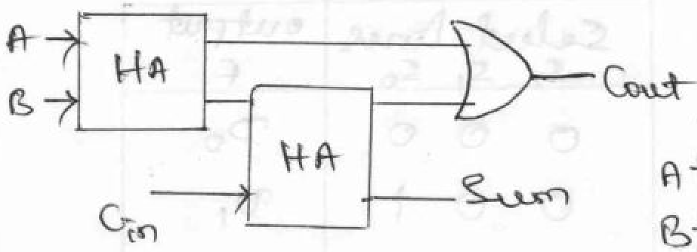


fig: Full adder using two half adder (Block)

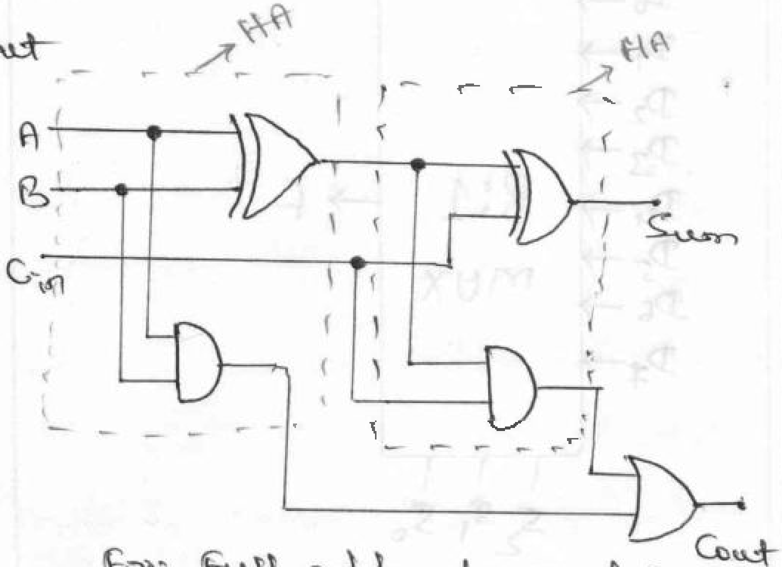
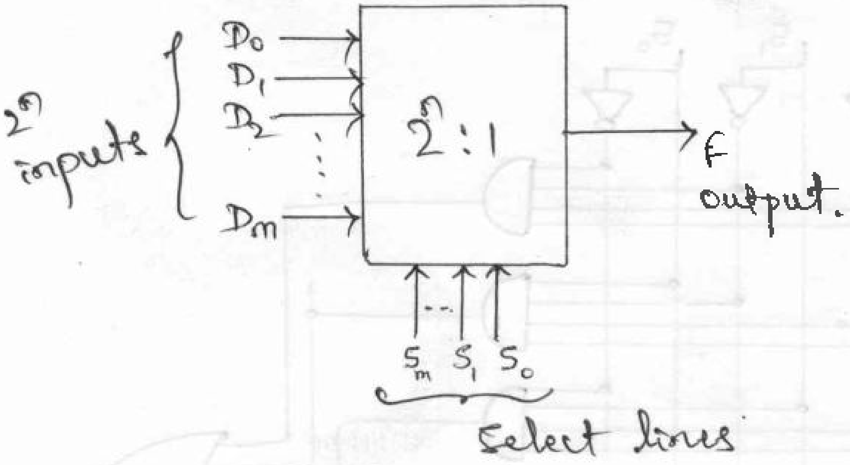


fig: Full adder Logic diagram using two half adder.

Multiplexer (MUX):-

* A digital multiplexer connects one of the inputs to a single output line based on select line values.

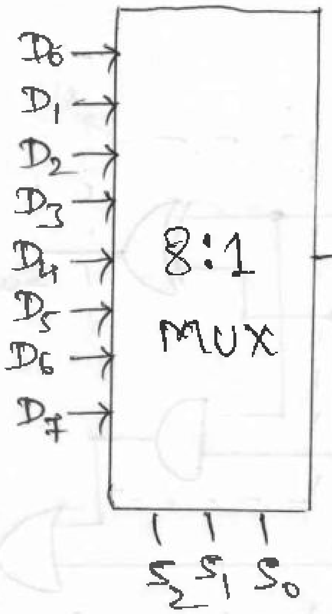


* 'n' select lines, 2^n inputs and only '1' output.

8 to 1 MUX:-

* It consists of 8 inputs, 3 select lines and only one output.

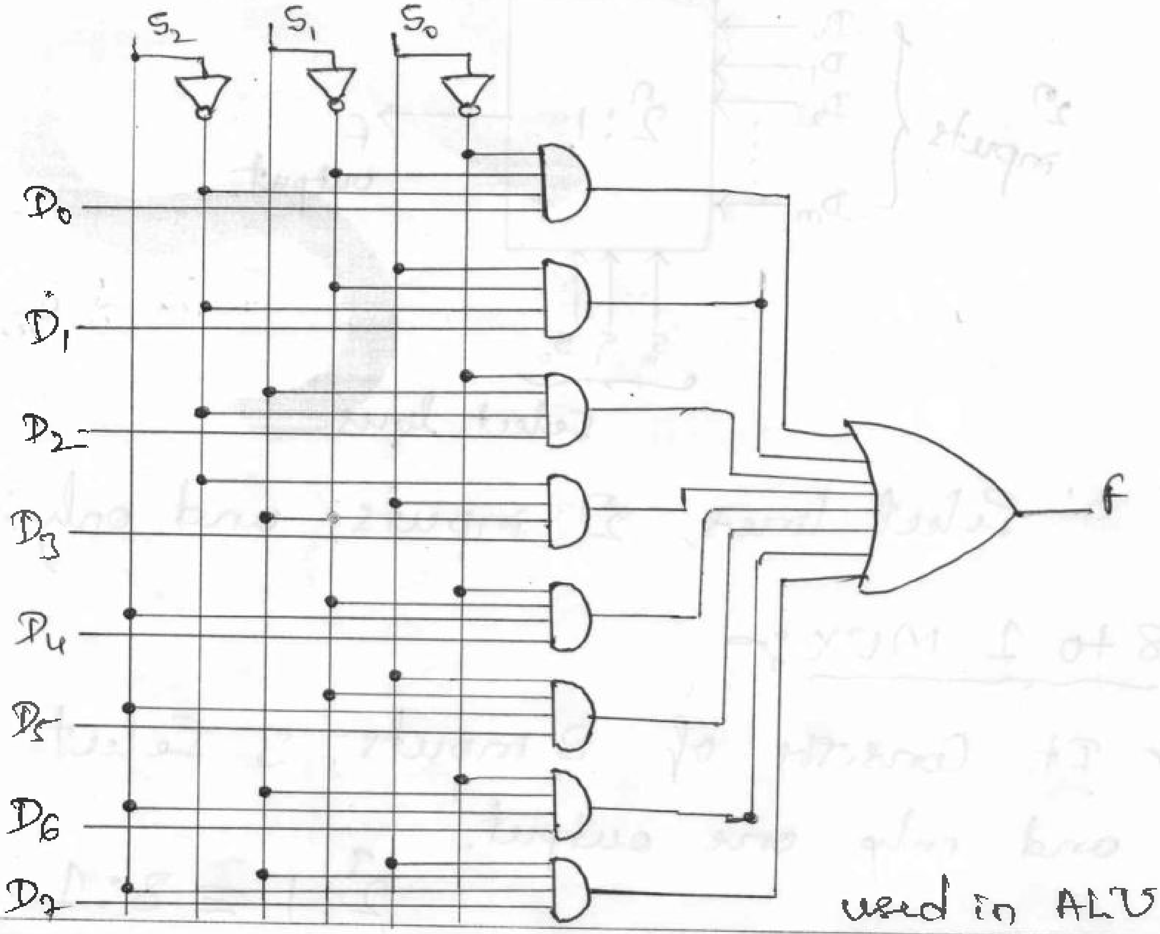
$2^3:1 \equiv 8:1$



Truth Table:

Select lines			output
S_2	S_1	S_0	F
0	0	0	D_0
0	0	1	D_1
0	1	0	D_2
0	1	1	D_3
1	0	0	D_4
1	0	1	D_5
1	1	0	D_6
1	1	1	D_7

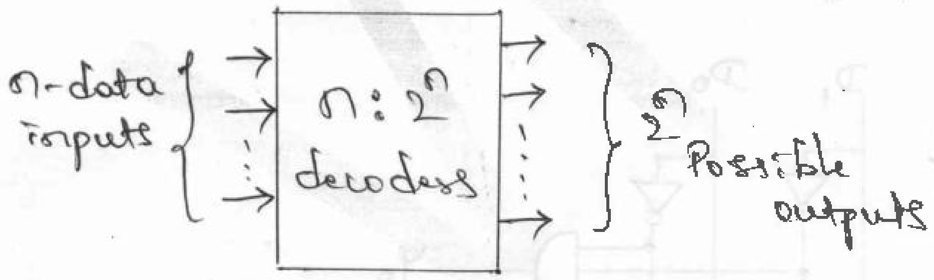
$$F = \bar{S}_2\bar{S}_1\bar{S}_0 D_0 + \bar{S}_2\bar{S}_1 S_0 D_1 + \bar{S}_2 S_1\bar{S}_0 D_2 + \bar{S}_2 S_1 S_0 D_3 + S_2\bar{S}_1\bar{S}_0 D_4$$
$$+ S_2\bar{S}_1 S_0 D_5 + S_2 S_1\bar{S}_0 D_6 + S_2 S_1 S_0 D_7$$



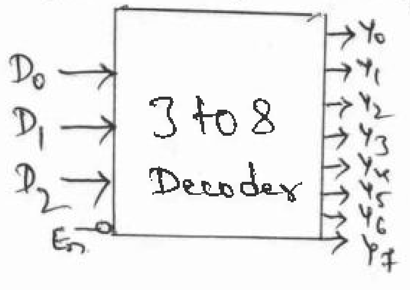
used in ALU.

Decoders :

- Decoders are a class of a combinational logic circuit that converts a set of input variables representing a code into a set of output variables representing a different code.
- A decoder is a multiple input multiple output logic circuit which converts coded inputs into coded output where input and output codes are different.
- 'n' number of input lines and 2^n output lines.



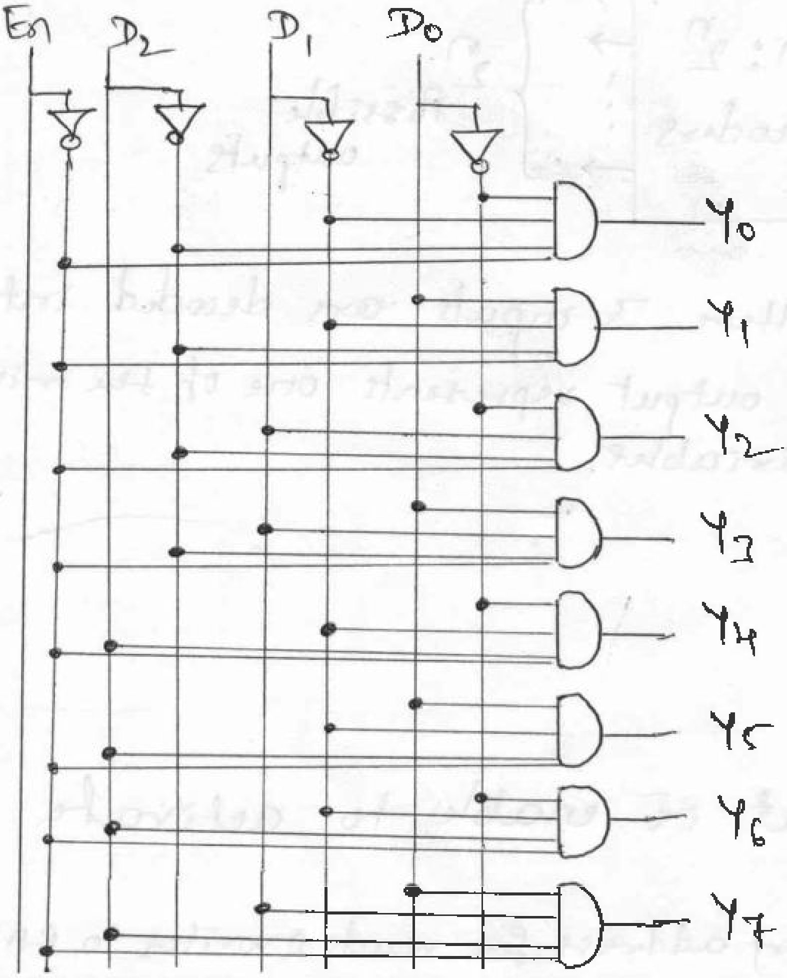
3 to 8 Decoder :- Here 3 inputs are decoded into 8 outputs and each output represents one of the minterms of 3 input variables.



- Additional input is enable, to activate the decoder.
- Used to decode memory address for reads & writes to RAM.

Truth Table:

Inputs				Outputs							
Enable	Select Inputs										
En	D2	D1	D0	Y7	Y6	Y5	Y4	Y3	Y2	Y1	Y0
1	x	x	x	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	1
0	0	0	1	0	0	0	0	0	0	1	0
0	0	1	0	0	0	0	0	0	1	0	0
0	0	1	1	0	0	0	0	1	0	0	0
0	1	0	0	0	0	0	1	0	0	0	0
0	1	0	1	0	0	1	0	0	0	0	0
0	1	1	0	0	1	0	0	0	0	0	0
0	1	1	1	1	0	0	0	0	0	0	0



3 to 8.
fig: Logic diagram

Shift Registers :-

- A register is simply a collection of flipflops taken as entity.
- The basic function of a register is to hold information within a digital system so as to make it available to the logic elements during computing process.
- Registers that are capable of moving information position wise upon the occurance of a clock signal are called shift registers.

4 Bit Shift Register :

- A 4 bit shift register is as shown in fig.
- Initially logic '0' is applied to clear pin to make all flipflop output zero. $Q_3 = Q_2 = Q_1 = Q_0 = 0$.
- All the flipflops are triggered by common clock pulse, data (1010) is now fed in Least significant digit (LSD) first and MSD last as shown in Table.

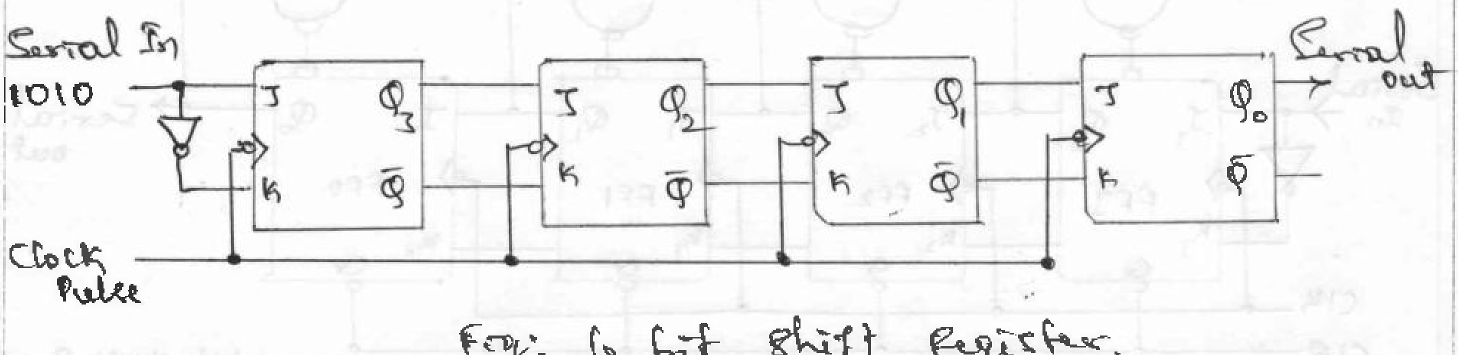


Fig: 4 bit shift Register.

Input		output			
clock Pulse	Serial In	Q_3	Q_2	Q_1	Q_0 (Serial out)
0	0	0	0	0	0
1	1	1	0	0	0
2	0	0	1	0	0
3	1	1	0	1	0 →Data entered
4	0	0	1	0	1
5	0	0	0	1	0
6	0	0	0	0	1
7	0	0	0	0	0

Register Type :-

① Serial in Serial out (SISO)

② Serial in Parallel out (SIPO)

③ Parallel in Serial out (PISO)

④ Parallel in Parallel out (PIPO).

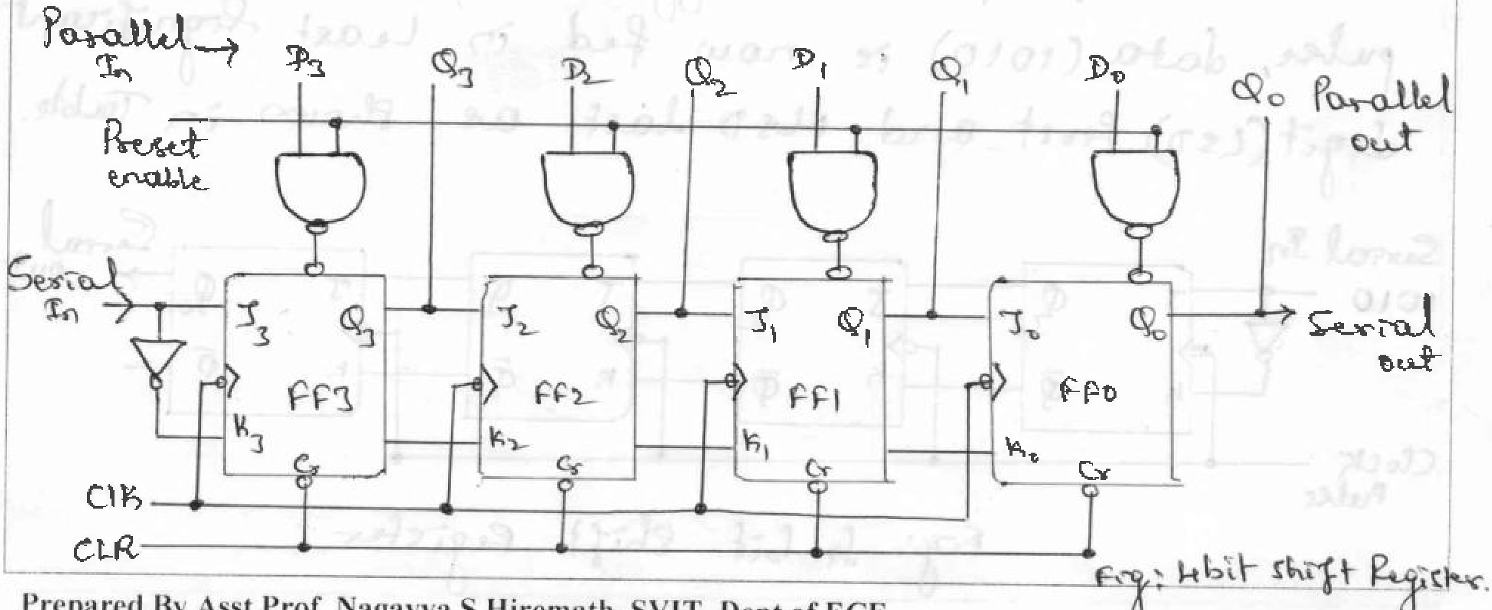


Fig: 4bit shift Register.

* Let the input 1101 be entered serially (SISO).

SISO:

Clock	Serial Data	Q_3	Q_2	Q_1	Q_0
0	x	0	0	0	0
1	1	1	0	0	0
2	0	0	1	0	0
3	1	1	0	1	0
4	1	1	1	0	1
5		0	1	1	0
6		0	0	1	1
7		0	0	0	1

* During the first positive edge of clock pulse, $Q_3 = 1$ and all other output remains '0'.
* The output of flipflop FF-3 is fed to flipflop FF-2

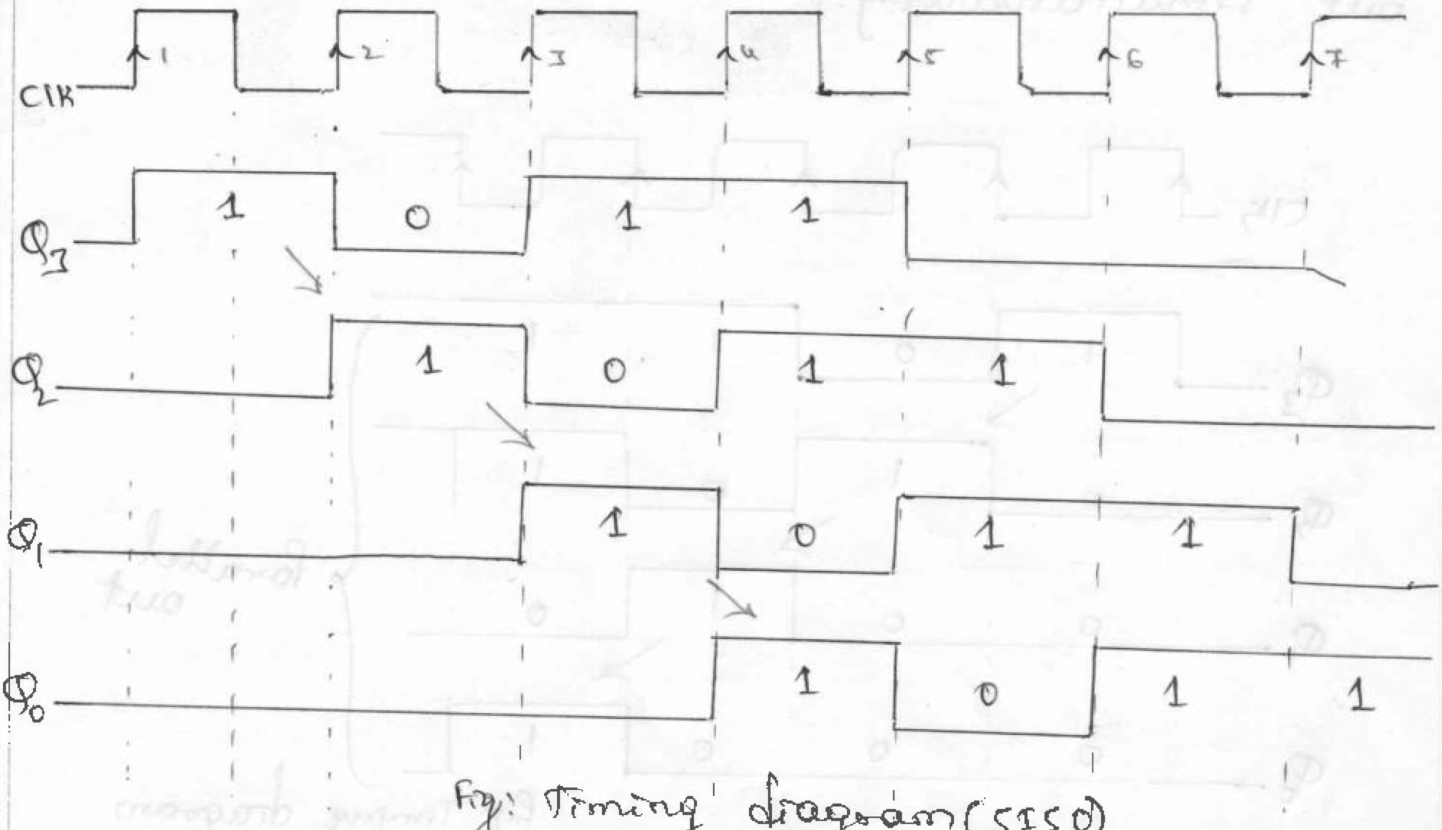


Fig: Timing diagram (SISO)

* Let the input 1101 be entered Serially (SIPO).

PISO: Truth Table:

Clock	Serial data	Q_3	Q_2	Q_1	Q_0
0	x	0	0	0	0
1	1	1	0	0	0
2	0	0	1	0	0
3	1	1	0	1	0
4	1	1	1	0	1

→ Parallel out

* Initially, all flipflops are cleared i.e $Q_3=Q_2=Q_1=Q_0=0$

* The data 1101 is loaded into flipflop by starting from LSB bit first.

* All bits are entered into flipflop in similar manner as SISO but all bits are at the output are taken out simultaneously.

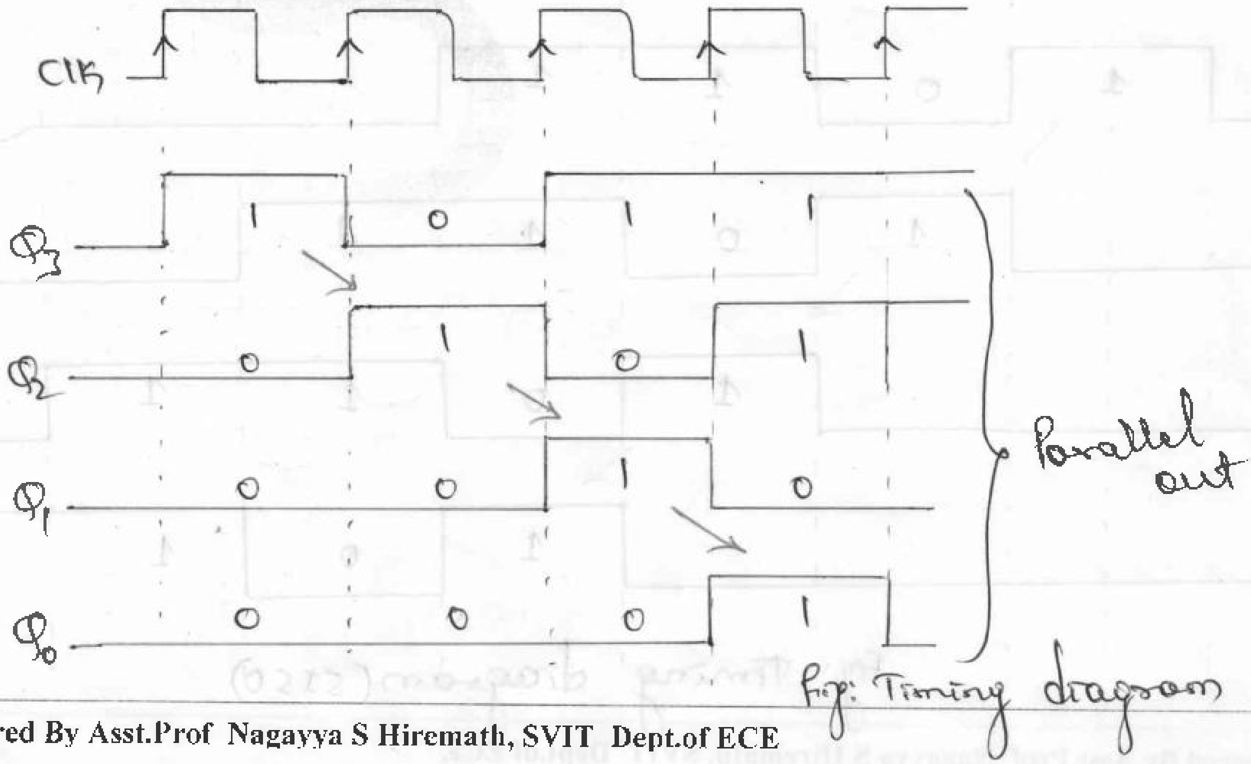


Fig: Timing diagram

* Let the input 1101 be entered Parally (PIPO).

PIPO:

Clock	Parallel Data	Q_3	Q_2	Q_1	Q_0
0	X	0	0	0	0
1	1 1 0 1	1	1	0	1

* In single clock pulse Parallel data entered input will get output side parallely.

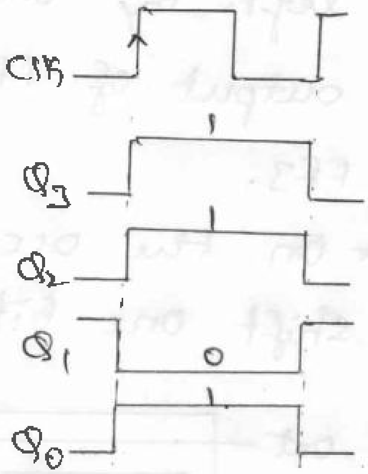


Fig: Timing diagram

PISO:

* Let the input 1101 be entered Parally (PISO).

Clock	Parallel data	Q_3	Q_2	Q_1	Q_0
0	X X X X	0	0	0	0
1	1 1 0 1	1	1	0	1
2		0	1	1	0
3		0	0	1	1
4		0	0	0	1

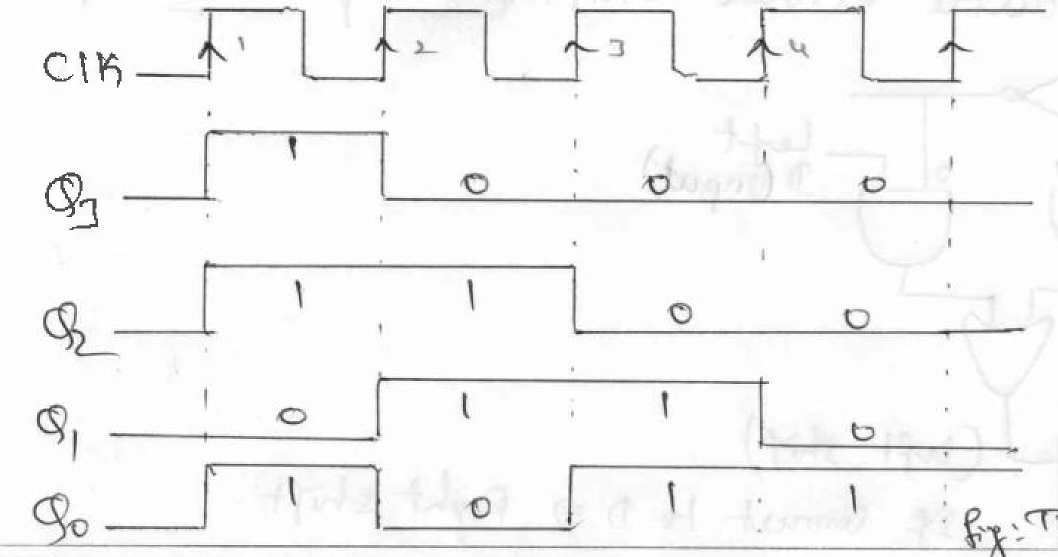


fig: Timing diagram

Shifting of Data to left :-

- In this register, the data is shifted from left by one bit at clock pulse by feeding output of FF0 to input of FF1 and FF1 to FF2 to FF3.
- On the occurance of a pulse, the data word will shift one bit to left. This logic as shown in fig.

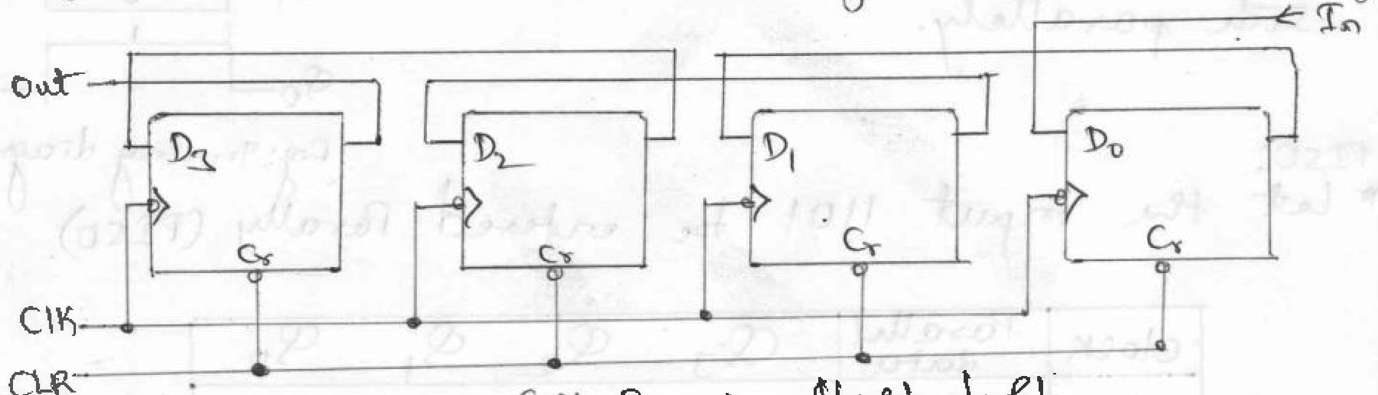
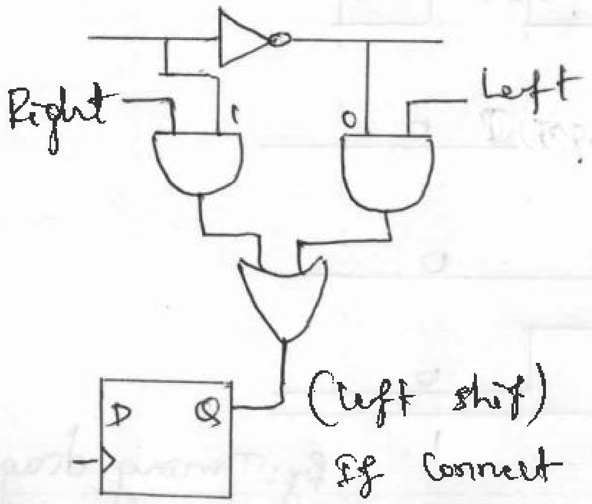


fig: Register Shift left.

Bi-Directional Register:-

- Connecting following circuit (AND-OR logic), to D flipflop, data either shift in right or left shift.



(left shift)

If connect to D ⇒ Right shift

Counters:-

* A Counter is a circuit that counts the number of occurances of an input.

* Each count, a binary number is called a state of the Counter.

Types of Counters:

1) Asynchronous Counter. (Ripple Counter)

2) Synchronous Counter.

* In case of asynchronous counter, not all the flip-flops are clocked simultaneously, whereas in a synchronous counters, all the flip-flops are clocked simultaneously.

* If the counter counts in increasing order with successive clock pulses, it is called an UP-Counter

* If it decreases, it is called a DOWN-Counter.

Asynchronous-Counter:-

* A 3 bit (modulo-8) asynchronous counter as shown in fig below.

* As J and K are tied together to 1 input, the JK flipflop act as T-flipflop.

* The output of one flipflop feeds the clock of the next flipflop to be triggered.

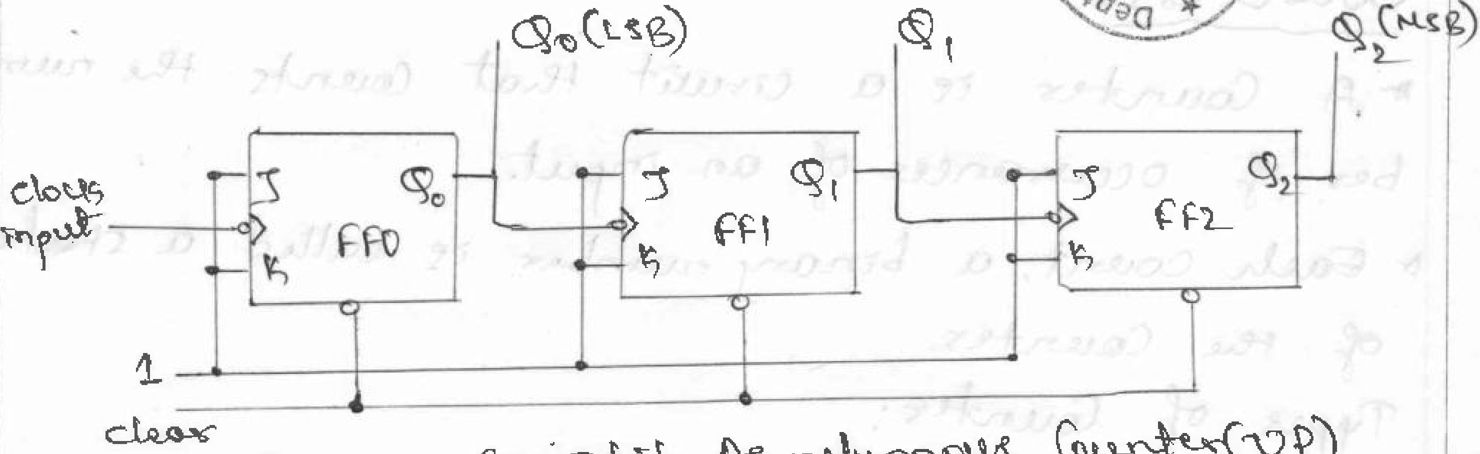


Fig: 3 bit Asynchronous Counter (UP)

- All the flipflops are initially cleared.
- The clock pulses are then applied with each clock pulse, Q toggles.

Truth Table:

Input clock	Q_3	Q_2	Q_1
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

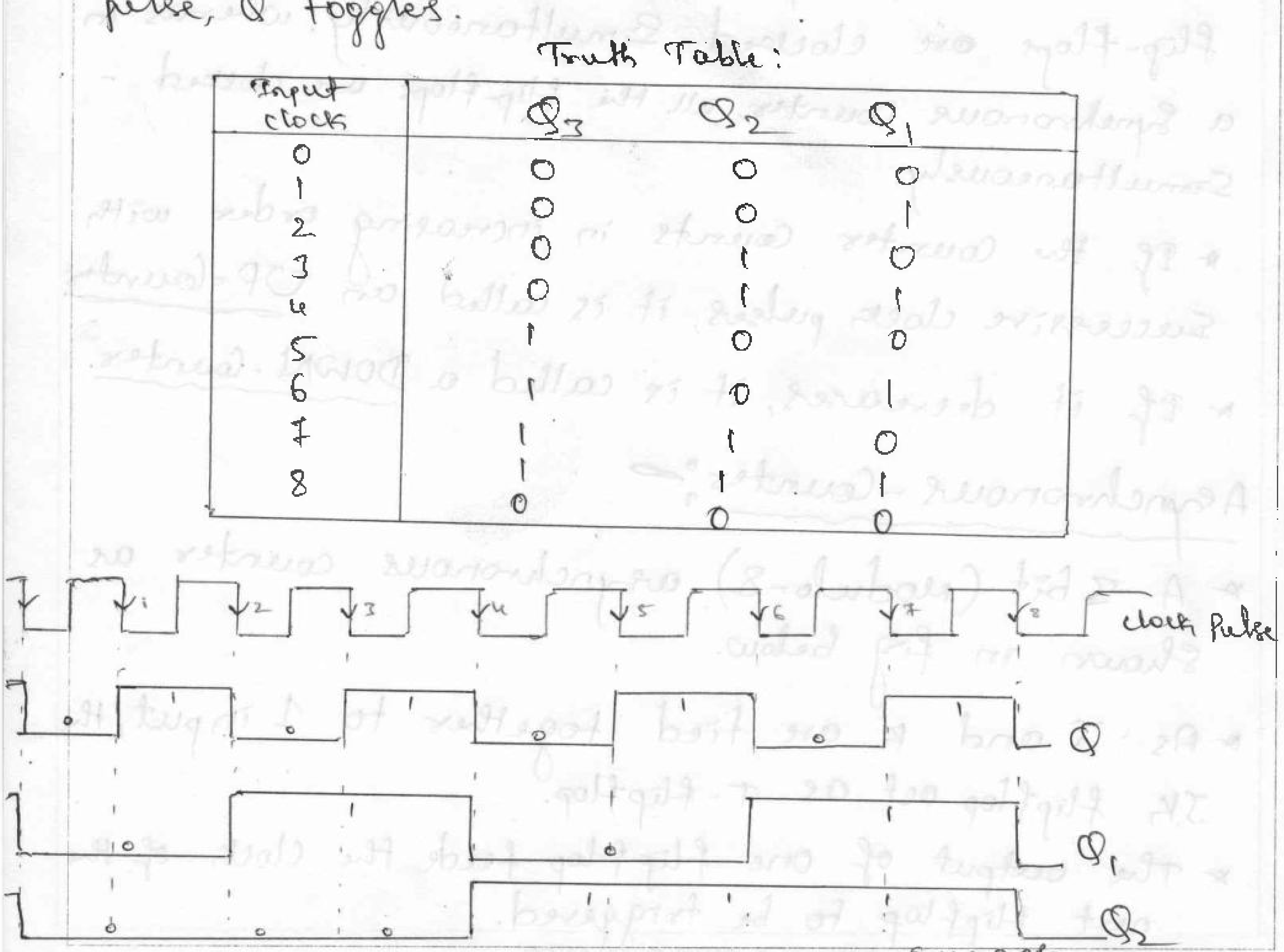


Fig: waveform.

Embedded System:

- An embedded system is an electronic/electro-mechanical system designed to perform a specific function and a combination of both hardware and firmware (software).
- Every embedded system is unique and the hardware as well as the firmware is highly specialized to the application domain.
- Embedded systems are becoming an inevitable part of any product or equipment in all fields including household appliances, telecommunications, medical equipment, industrial control, consumer products, etc.
- Embedded system is a combination of 3 things
  1. Hardware
  2. Software
  3. Mechanical component & it is supposed to do only one specific task only

Examples:

Example1: Washing Machine

A washing machine from an embedded systems point of view has:

a.Hardware: Buttons, displays & buzzer, electronic circuitry.

b.Software: It has a chip on the circuit that holds the software which drives controls & monitors various operations possible.

c.Mechanical components: the internals of a washing machine which actually wash the clothes control the input and output of water.

Example-2: Air Conditioner

An Air Conditioner from an embedded systems point of view has:

a.Hardware: Remote, display & buzzer, infrared Sensors, electronic circuitry

b.Software: It has a chip on the circuit that holds the software which drives control & monitors the various operations possible. The software monitors the external temperature through the sensors and then releases the coolant or suppresses it.

c. Mechanical components: The internals of an air conditioner the motor, the outlet, etc.-

Differences between General Purpose computing system and Embedded system:

Criteria	General Purpose computing system	Embedded system
Contents	A system which is a combination of a generic hardware and general-purpose operating system for executing a variety of applications	A system which is a combination of a special-purpose hardware and embedded OS for executing a variety of applications
OS	It contains a general-purpose operating system (GPOS)	It may or may not contain an operating system for functioning
Alterations	Applications are alterable (programmable) by the user. (It is possible for end user to re-install the OS and also add or remove user applications)	The firmware of the Embedded system is pre-programmed and it is non-alterable by the end user.
Key Factor	Performance is the key deciding factor in the selection of the system. Faster is better.	Application specific requirements (like performance, power requirements, memory usage etc) are the key deciding factor.
Power consumption	More	Less
Response Time	Not Critical	Critical for some applications
Execution	Need not be deterministic	Deterministic for certain types of ES like 'Hard real time systems'.

Classification of Embedded Systems:

The classification of embedded system is based on following criteria's:

- On generation
- On complexity & performance
- On deterministic behaviour
- On triggering

On generation:

1. First generation (1G):
   - Built around 8bit microprocessor & microcontroller.
   - Simple in hardware circuit & firmware developed.
   - Examples: Digital telephone keypads.
2. Second generation (2G):
   - Built around 16-bit µp & 8-bit µc.
   - They are more complex & powerful than 1G µp & µc.
   - Examples: SCADA systems
3. Third generation (3G):
   - Built around 32-bit µp& 16-bit µc.
   - Concepts like Digital Signal Processors (DSPs), Application Specific Integrated Circuits (ASICs) evolved.

   Examples: Robotics, Media, etc.
4. Fourth generation:
   - Built around 64-bit µp & 32-bit µc.
   - The concept of System on Chips (SoC), Multicore Processors evolved.
   - Highly complex & very powerful.

   Examples: Smart Phones.

On complexity & performance:

1. Small-scale:

   Simple in application need

   Performance not time-critical.

   Built around low performance& low cost 8 or 16 bit µp/µc.

   Example: an electronic toy
2. Medium-scale:

   Slightly complex in hardware & firmware requirement.

   Built around medium performance & low cost 16 or 32 bit µp/µc.

   Usually contain operating system.

   Examples: Industrial machines.
3. Large-scale:

   Highly complex hardware & firmware

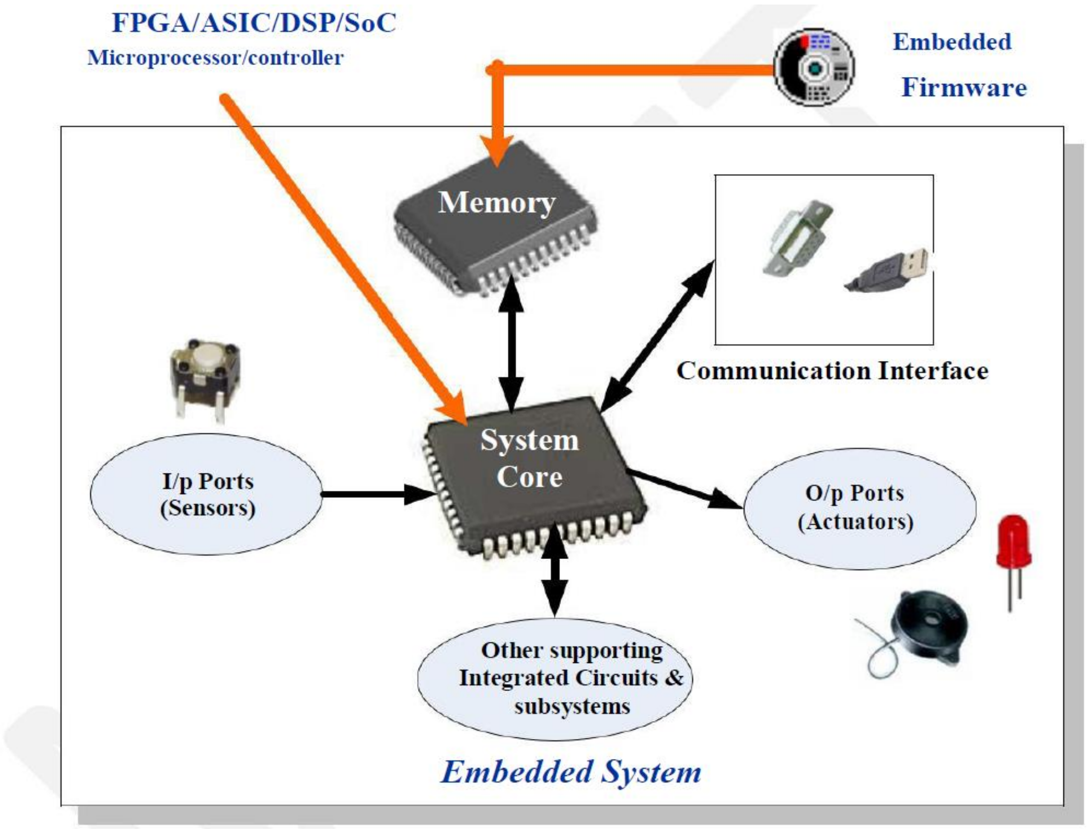
Major Application Areas of ES:

The application areas and the products in the embedded domain are countless. A few of the important domains and products are listed below:

- Consumer electronics: Camcorders, cameras, etc.
- Household appliances: Television, DVD players, washing machine, fridge, microwave oven, etc.
- Home automation and security systems: Air conditioners, sprinklers, intruder detection alarms, closed circuit television cameras, fire alarms, etc.
- Automotive industry: Anti-lock braking systems (ABS), engine control, ignition systems, automatic navigation systems, etc.
- Telecom: Cellular telephones, telephone switches, handset multimedia applications, etc.
- Computer peripherals: Printers, scanners, fax machines, etc.
- Computer networking systems: Network routers, switches, hubs, firewalls, etc.
- Healthcare: Different kinds of scanners, EEG, ECG machines etc.
- Measurement & Instrumentation: Digital multi meters, digital CROs, logic analyzers PLC systems, etc.
- Banking & Retail: Automatic teller machines (ATM) and currency counters, point of sales (POS).
- Card Readers: Barcode, smart card readers, hand held devices, etc.

Elements of an embedded system:

- An embedded system is a combination of 3 things, Hardware Software Mechanical Components and it is supposed to do one specific task only.
- A typical embedded system contains a single chip controller which acts as the master brain of the system. Diagrammatically an embedded system can be represented as follows:
- Embedded systems are basically designed to regulate a physical variable (such Microwave Oven) or to manipulate the state of some devices by sending some signals to the actuators or devices connected to the output port system (such as temperature in Air Conditioner), in response to the input signal provided by the end users or sensors which are connected to the input ports. Hence the embedded systems can be viewed as a reactive system.



- The control is achieved by processing the information coming from the sensors and user interfaces and controlling some actuators that regulate the physical variable.
- Keyboards, push button, switches, etc. are Examples of common user interface input devices and LEDs, LCDs, Piezoelectric buzzers, etc examples for common user interface output devices for a typical embedded system. The requirement of type of user interface changes from application to application based on domain.
- Some embedded systems do not require any manual intervention for their operation. They automatically sense the input parameters from real world through sensors which are connected at input port. The sensor information is passed to the processor after signal conditioning and digitization. The core of the system performs some predefined operations on input data with the help of embedded firmware in the system and sends some actuating signals to the actuator connect connected to the output port of the system.
- The memory of the system is responsible for holding the code (control algorithm and other important configuration details). There are two types of memories are used in any embedded system. Fixed memory (ROM) is used for storing code or program.

- The user cannot change the firmware in this type of memory. The most common types of memories used in embedded systems for control algorithm storage are OTP, PROM, UVEPROM, EEPROM and FLASH.
- An embedded system without code (i.e. the control algorithm) implemented memory has all the peripherals but is not capable of making decisions depending on the situational as well as real world changes.
- Memory for implementing the code may be present on the processor or may be implemented as a separate chip interfacing the processor
- In a controller based embedded system, the controller may contain internal memory for storing code such controllers are called Micro-controllers with on-chip ROM. eg. Atmel AT89C51.

The Core of the Embedded Systems:

The core of the embedded system falls into any one of the following categories.

- General Purpose and Domain Specific Processors
  - Microprocessors
  - Microcontrollers
  - Digital Signal Processors
- Programmable Logic Devices (PLDs)
- Application Specific Integrated Circuits (ASICs)
- Commercial off the shelf Components (COTS)

General Purpose and Domain Specific Processor:

- Almost 80% of the embedded systems are processor/ controller based.
- The processor may be microprocessor or a microcontroller or digital signal processor, depending on the domain and application.

Microprocessor:

- A silicon chip representing a Central Processing Unit (CPU), which is capable of performing arithmetic as well as logical operations according to a pre-defined set of Instructions, which is specific to the manufacturer
- In general the CPU contains the Arithmetic and Logic Unit (ALU), Control Unit and Working registers
- Microprocessor is a dependant unit and it requires the combination of other hardware like Memory, Timer Unit, and Interrupt Controller etc for proper functioning.

- Intel claims the credit for developing the first Microprocessor unit Intel 4004, a 4 bit processor which was released in Nov 1971. Developers of microprocessors:
- Intel – Intel 4004 – November 1971(4-bit)
- Intel – Intel 4040.
- Intel – Intel 8008 – April 1972.
- Intel – Intel 8080 – April 1974(8-bit).
- Intel – Intel 8085 – 1976.

Microcontroller:

- A highly integrated silicon chip containing a CPU, scratch pad RAM, Special and General purpose Register Arrays, On Chip ROM/FLASH memory for program storage, Timer and Interrupt control units and dedicated I/O ports
- Microcontrollers can be considered as a super set of Microprocessors
- Microcontroller can be general purpose (like Intel 8051, designed for generic applications and domains) or application specific (Like Automotive AVR from Atmel Corporation. Designed specifically for automotive applications)
- Since a microcontroller contains all the necessary functional blocks for independent working, they found greater place in the embedded domain in place of microprocessors
- Microcontrollers are cheap, cost effective and are readily available in the market
- Texas Instruments TMS 1000 is considered as the world's first microcontroller.

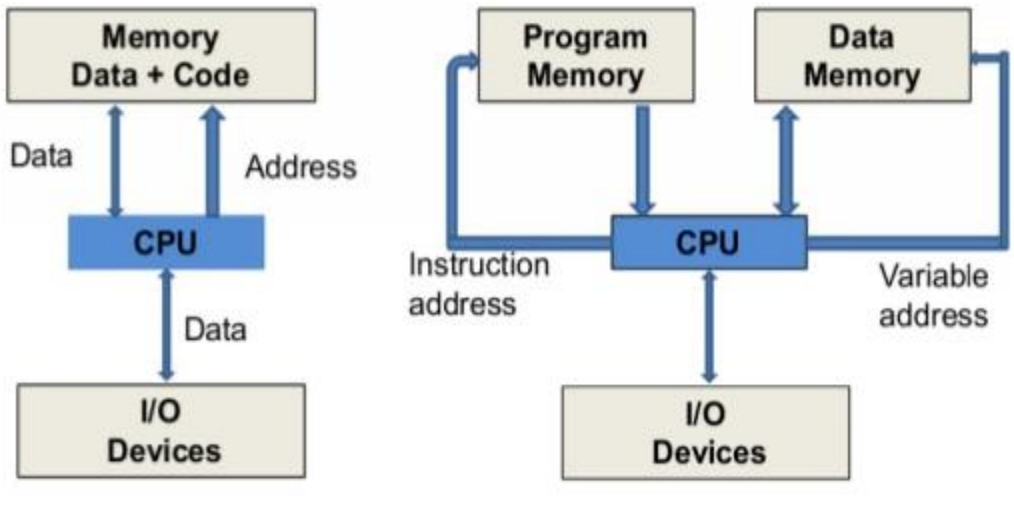
Differences between Microprocessor and Microcontroller:

Microprocessor	Microcontroller
1. Microprocessor is widely used in computer systems.	1. Microcontroller is widely used in embedded systems.
2. It has only a CPU embedded into it	2. It has a CPU, a fixed amount of RAM, ROM and other peripherals all embedded on it.
3. In case of microprocessors we have to connect all the components externally so the circuit becomes large and complex.	3. As all the components are internally connected in microcontroller so the circuit size is less.
4. It consumes more power.	4. It consumes less power than a microprocessor.
5. It has very less internal register storage so it has to rely on external storage. So, all memory based external commands which results in high processing time.	5. It has many registers so processing time is less.

Difference between CISC and RISC Processor:

CISC	RISC
1. Complex instructions.	1. Simple instructions.
2. Main focus is hardware.	2. Main focus is software.
3. Complexity lies in Processor.	3. Complexity lies in Compiler.
4. Multiple clock cycle.	4. Single clock cycle.
5. Transistors are used to store complex instructions.	5. Transistors are used for storing memory.
6. CISC has 100-300 minimum Instructions.	6. RISC uses few instructions (30-40).
7. 8-10 Addressing modes.	7. Few Addressing modes.
8. Variable size/length instructions.	8. Fixed size/length instructions.

Von Neuman Architecture and Harvard Architecture:



Von Neumann Machine

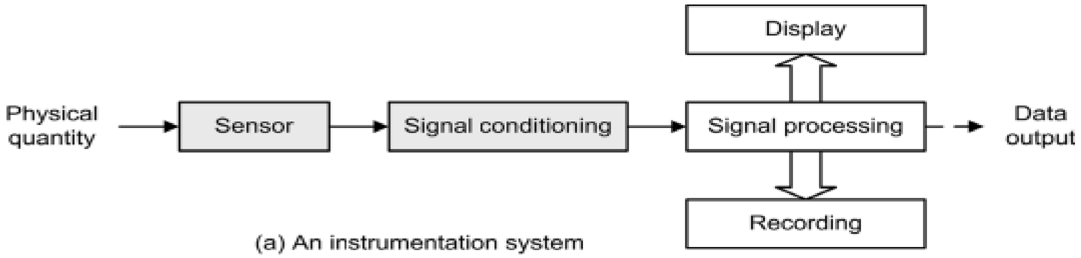
Harvard Machine

Von-Neuman	Harvard
1. Here the RAM and ROM is not separated, and a single memory connection is given to CPU.	1. Here the CPU is connected separately with RAM and ROM.
2. Less space is required.	2. More space is required.

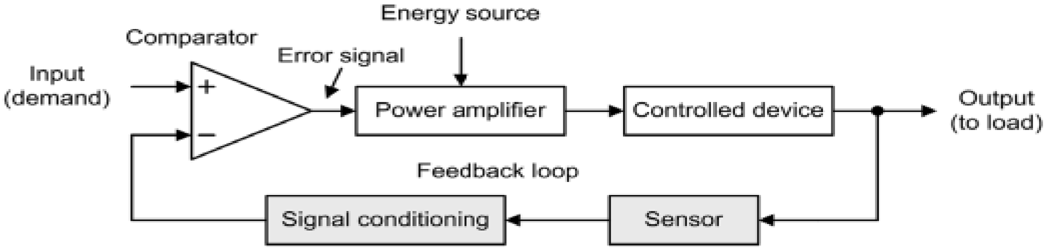
3. Speed is low because fetching data and instructions at the same time is unable. At a time, it can fetch either a data or instruction.	3. Speed is more because it takes less time to fetch data and instruction at the same time from 2 memories called instruction memory and data memory.
4. There is a common bus for transferring data and instruction.	4. There are separate buses to transfer data and instruction.
5. Used in personal computers and small sized computers.	5. This architecture is used in signal processing and the microcontrollers.
6. Same memory address is used for both instructions and data.	6. Separate physical address is used for both instruction and data.
7. Here the two clock cycles are needed to execute a single instruction.	7. A Single clock is needed to execute a instruction.
8. Instructions and read/write operations cannot be accessed by the CPU at the same time.	8. CPU can access both instructions and write/read operations at the same time.
9. These architecture-based computers are somewhat cheaper than Harvard architecture.	9. This is expensive than Von Neumann architecture.

Instrumentation and control systems:

- Instrumentation and control systems Fig.(a) Shows the arrangement of an instrumentation system. The physical quantity to be measured (e.g. temperature) acts upon a sensor that produces an electrical output signal. This signal is an electrical analogue of the physical input but note that there may not be a linear relationship between the physical quantity and its electrical equivalent.



(a) An instrumentation system

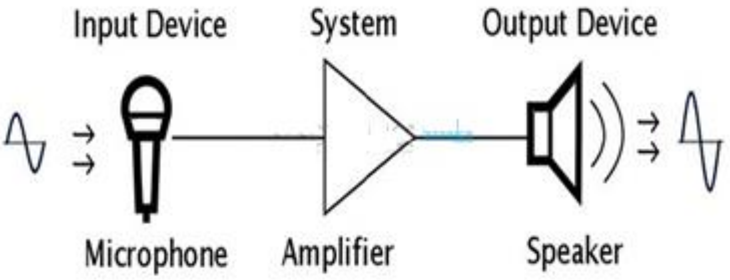


(b) A control system

- Fig.(b) shows the arrangement of a control system. This uses negative feedback in order to regulate and stabilize the output. It thus becomes possible to set the input or demand (i.e. what we desire the output to be) and leave the system to regulate itself by comparing it with a signal derived from the output (via a sensor and appropriate signal conditioning).
- A comparator is used to sense the difference in these two signals and where any discrepancy is detected the input to the power amplifier is adjusted accordingly. This signal is referred to as an error signal (it should be zero when the output exactly matches the demand). The input (demand) is often derived from a simple potentiometer connected across a stable d.c. voltage source while the controlled device can take many forms (e.g. a d.c. motor, linear actuator, heater, etc.).

Transducers:

- Transducers are devices that convert energy in the form of sound, light, heat, etc., into an equivalent electrical signal, or vice versa. Before we go further, let's consider a couple of examples that you will already be familiar with. A loudspeaker is a transducer that converts low frequency electric current into audible sounds. A microphone, on the other hand, is a transducer that performs the reverse function i.e. that of converting sound pressure variations into voltage or current. Loudspeakers and microphones can thus be considered as complementary transducers.



- Transducers may be used both as inputs to electronic circuits and outputs from them. From the two previous examples, it should be obvious that a loudspeaker is an output transducer designed for use in conjunction with an audio system. A microphone is an input transducer designed for use with a recording or sound reinforcing system.

Sensors:

- A sensor is a special kind of transducer that converts energy from one form to another for any measurement or control purpose ex. A Temperature sensor. The signal produced by a sensor is an electrical analogy of a physical quantity, such as distance, velocity, acceleration, temperature, pressure, light level, etc. The signals returned from a sensor, together with control inputs from the user or controller (as appropriate) will subsequently be used to determine the output from the system. The choice of sensor is governed by a number of factors including accuracy, resolution, cost and physical size.

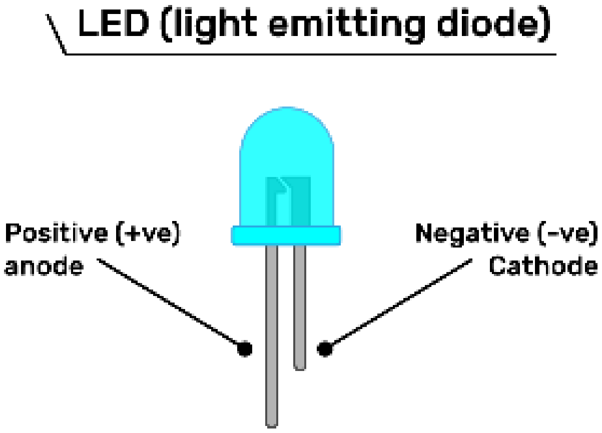
- Sensors can be categorized as either active or passive. An active sensor generates a current or voltage output. A passive transducer requires a source of current or voltage and it modifies this in some way (e.g. by virtue of a change in the sensor's resistance). The result may still be a voltage or current but it is not generated by the sensor on its own.
- Sensors can also be classed as either digital or analogue. The output of a digital sensor can exist in only two discrete states, either 'on' or 'off', 'low' or 'high', 'logic 1' or 'logic 0', etc. The output of an analogue sensor can take any one of an infinite number of voltage or current levels. It is thus said to be continuously variable. Table 15.3 provides details of some common types of sensor.

Actuator:

Actuator is used for output. It is a transducer that may be either mechanical or electrical which converts signals to corresponding physical actions.

LED (Light Emitting Diode):

LED is a p-n junction diode and contains a CATHODE and ANODE for functioning the anode is connected to +ve end of power supply and cathode is connected to –ve end of power supply. The maximum current flowing through the LED is limited by connecting a resistor in series between the power supply and LED as shown in the figure below

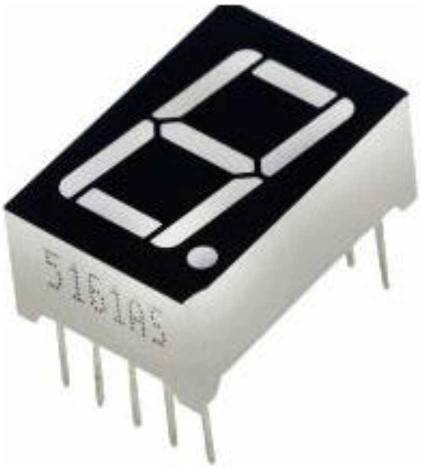


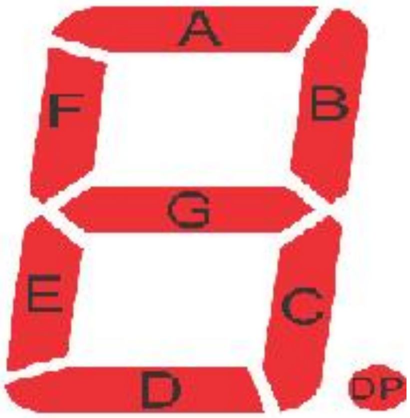
There are two ways to interface an LED to a microprocessor/microcontroller:

- The Anode of LED is connected to the port pin and cathode to Ground: In this approach the port pin sources the current to the LED when it is at logic high (ie. 1).
- The Cathode of LED is connected to the port pin and Anode to Vcc : In this approach the port pin sources the current to the LED when it is at logic high (ie. 1). Here the port pin sinks the current and the LED is turned ON when the port pin is at Logic low (ie. 0)

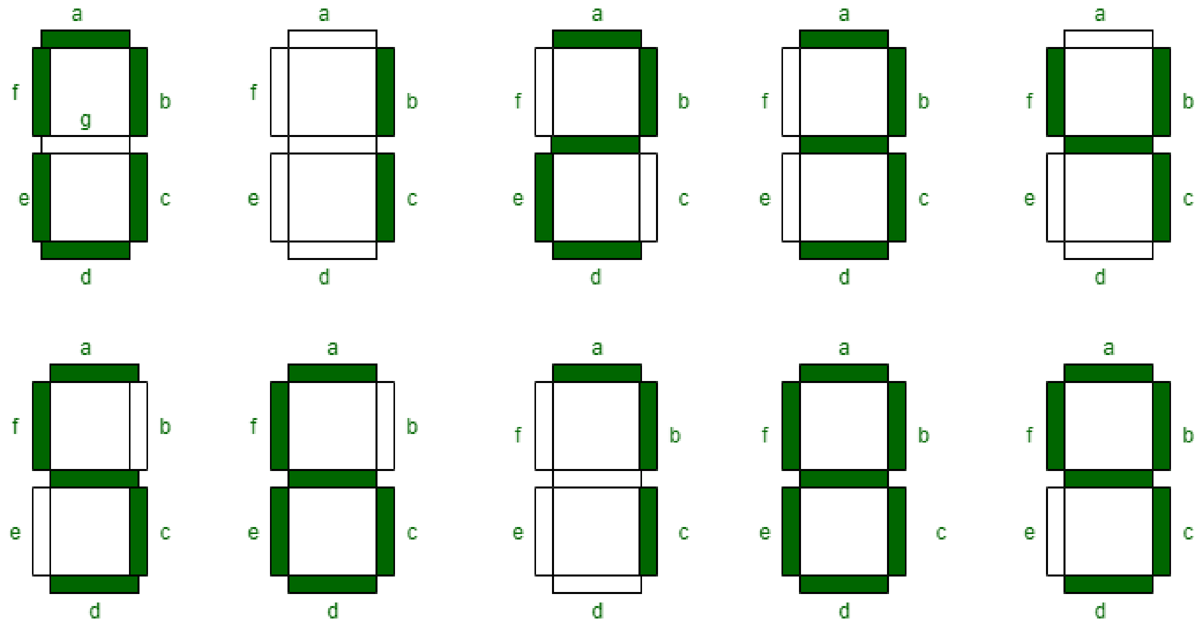
7-segment display:

A seven-segment display (SSD), or seven-segment indicator, is a form of electronic display device for displaying decimal numerals that is an alternative to the more complex dot matrix displays. Seven-segment displays are widely used in digital clocks, electronic meters, basic calculators, and other electronic devices that display numerical information.



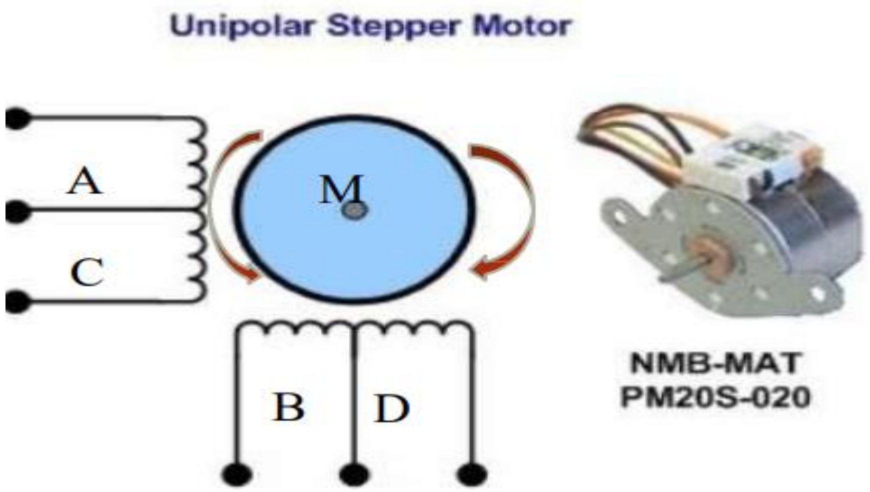


- The seven elements of the display can be lit in different combinations to represent the Arabic numerals. Often the seven segments are arranged in an oblique (slanted) arrangement, which aids readability.
- In most applications, the seven segments are of nearly uniform shape and size (usually elongated hexagons, though trapezoids and rectangles can also be used), though in the case of adding machines, the vertical segments are longer and more oddly shaped at the ends in an effort to further enhance readability.
- The segments of a 7-segment display are referred to by the letters A to G, where the optional decimal point (an "eighth segment", referred to as DP) is used for the display of non-integer numbers.



Stepper motor:

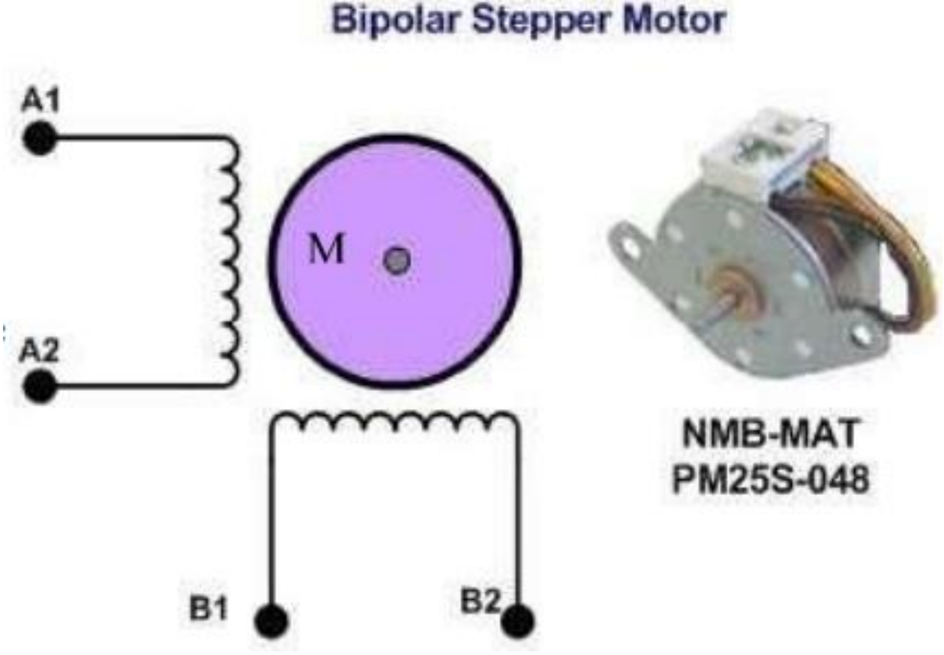
- ➢ A stepper motor is an electro-mechanical device which generates discrete displacement (motion) in response to dc electrical signals.
- ➢ It differs from the normal dc motor in its operation.
- ➢ The dc motor produces continuous rotation on applying dc voltage whereas a stepper motor produces discrete rotation in response to the dc voltage applied to it.
- ➢ Stepper motors are widely used in industrial embedded applications, consumer electronic products and robotics control systems.
- ➢ The paper feed mechanism of a printer/fax makes use of stepper motors for its functioning.
- ➢ Based on the coil winding arrangements, a two-phase stepper motors classified into two. They are:
  - ▪ Unipolar
  - ▪ Bipolar
- ➢ A unipolar stepper motor contains two windings per phase.
  The direction of rotation (clockwise or anticlockwise) of a stepper motor is controlled by changing the direction of current flow.



- ➢ Current in one direction flows through one coil and in the opposite direction flows through the other coil.
- ➢ It is easy to shift the direction of rotation by just switching the terminals to which the coils are connected.
- ➢ Figure illustrates the working of a two-phase unipolar stepper motor

Bipolar:

- ➢ A bipolar stepper motor contains single winding per phase.
- ➢ For reversing the motor rotation the current flow through the windings is reversed dynamically.
- ➢ It requires complex circuitry for current flow reversal.



- There is one disadvantage of unipolar motors. The torque generated by them is quite less. This is because the current is flowing only through the half the winding. Hence they are used in low torque applications.
- On the other hand, bipolar stepper motors are a little complex to wire as we have to use a current reversing H bridge driver IC like an L293D. But the advantage is that the current will flow through the full coil. The resulting torque generated by the motor is larger as compared to a unipolar motor.
- The stepping of stepper motor can be implemented in different ways by changing the sequence of activation of the stator windings. The different stepping modes supported by stepper motor are explained below.
- **Full Step:** In the step mode both the phases are energized simultaneously. The coils A, B, C and D are energized in the following order:
- It should be noted that out of the two windings, only one winding of a phase is energized at a time

Step	Coil A	Coil B	Coil C	Coil D
1	H	H	L	L
2	L	H	H	L
3	L	L	H	H
4	H	L	L	H

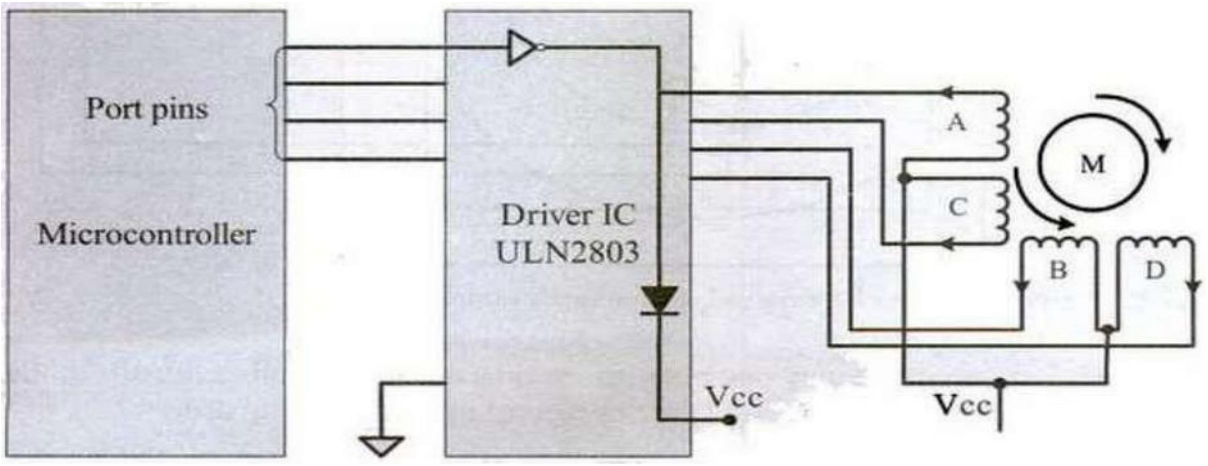
- **Wave Step:** In the wave step mode only one phase is energized at a time and each coils of the phase is energies alternatively. The coils A, B, C and D are energized in the following order:

Step	Coil A	Coil B	Coil C	Coil D
1	H	L	L	L
2	L	H	L	L
3	L	L	H	L
4	L	L	L	H

- **Half Step:** It uses the combination of wave and full step. It has the highest torque and stability. The coil energizing sequence for half step is given below.

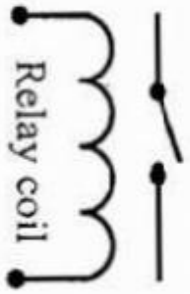
Step	Coil A	Coil B	Coil C	Coil D
1	H	L	L	L
2	H	H	L	L
3	L	H	L	L
4	L	H	H	L
5	L	L	H	L
6	L	L	H	H
7	L	L	L	H
8	H	L	L	H

- The rotation of the stepper motor can be reversed by reversing the order in which the coil is energized. Two-phase unipolar stepper motors are the popular choice for embedded applications.
- The current requirement for stepper motor is little high and hence the port pins of a microcontroller/processor may not be able to drive them directly.
- Also the supply voltage required to operate stepper motor varies normally in the range 5V to 24 V.
- Depending on the current and voltage requirements, special driving circuits are required to interface the stepper motor with microcontroller/processors.
- The following circuit diagram illustrates the interfacing of a stepper motor through a driver circuit connected to the port pins of a microcontroller/processor.

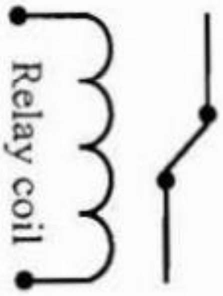


Relay:

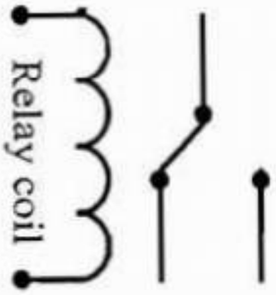
- Relay is an electro-mechanical device. In embedded application, the 'Relay' unit acts as dynamic path selectors for signals and power.
- The 'Relay' unit contains a relay coil made up of insulated wire on a metal core and a metal armature with one or more contacts.
- 'Relay' works on electromagnetic principle. When a voltage is applied to the relay coil, current flows through the coil, which in turn generates a magnetic field.
- The magnetic field attracts the armature core and moves the contact point. The movement of the contact point changes the power/signal flow path.
- 'Relays' are available in different configurations. Figure given below illustrates the widely used relay configurations for embedded applications.



Single pole single throw normally open



Single pole single throw normally closed



Single pole double throw

- The Single Pole Single Throw configuration has only one path for information flow. The path is either open or closed in normal condition.
- For normally Open Single Pole Single Throw relay, the circuit is normally open and it becomes closed when the relay is energized.
- For normally closed Single Pole Single Throw configuration, the circuit is normally closed and it becomes open when the relay is energized. For Single Pole Double Throw Relay, there are two paths for information flow and they are selected by energizing or de-energizing the relay.

Piezo Buzzer:

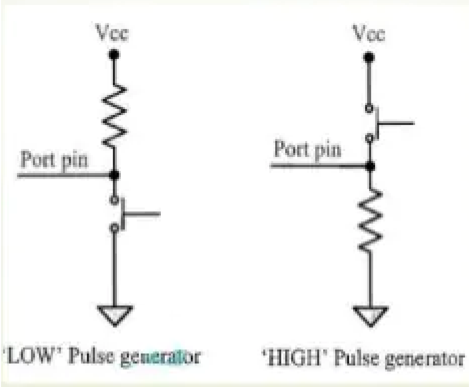
- Piezo buzzer is a piezoelectric device for generating audio indications in embedded application.

- A piezoelectric buzzer contains a piezoelectric diaphragm which produces audible sound in response to the voltage applied to it.
- Piezoelectric buzzers are available in two types: ‘Self-driving’ and ‘External driving’.
- The ‘Self-driving’ circuit contains all the necessary components to generate sound at a predefined tone. It will generate a tone on applying the voltage.

- External driving piezo buzzers supports the generation of different tones. The tone can be varied by applying a variable pulse train to the piezoelectric buzzer.

Push Button Switch:

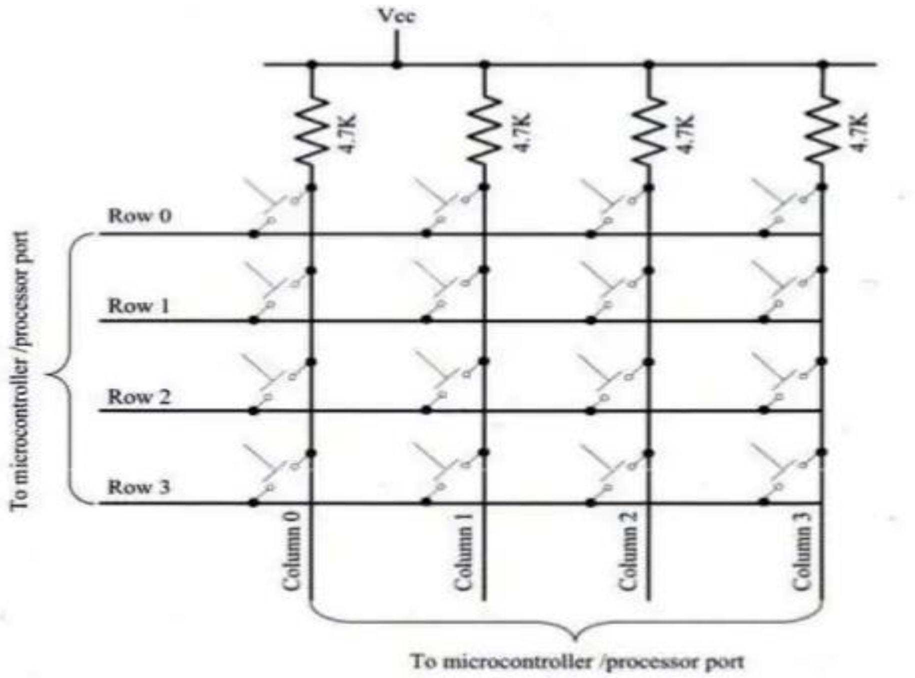
- It is an input device. Push button switch comes in two configurations, namely ‘Push to Make’ and ‘Push to Break’.



- In the ‘Push to Make’ configuration, the switch is normally in the open state and it makes a circuit contact when it is pushed or pressed.
- In the ‘Push to Break’ configuration, the switch is normally in the closed state and it breaks the circuit contact when it is pushed or pressed.
- In the embedded application push button is generally used as reset and start switch.

Keyboard:

- Keyboard is an input device for user interfacing. If the number of keys required is very limited, push button switches can be used and they can be directly interfaced to the port pins for reading.



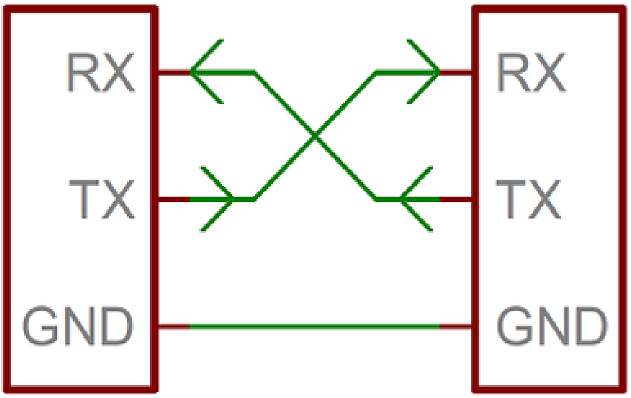
- Matrix keyboard is an optimum solution for handling large key requirements. It greatly reduces the number of interface connections.
- For example, for interfacing 16 keys, in the direct interfacing technique 16 port pins are required, whereas the matrix keyboard only 8 lines are required. The 16 keys are arranged in a 4 column*4 row matrix.
- In a matrix keyboard the keys arranged in matrix fashion. For detecting a key press, the keyboard uses the scanning technique, where each row of the matrix is pulled low and the columns are read.
- After reading the status of each column corresponding to a row. The row is pulled high and the next row is pulled low and the status of the columns read.
- This process is repeated until the scanning for all rows are completed. When row is pulled low and ifa key connected to the row is pressed, reading the column to which the key is connected will give logic 0.

Communication Interface:

UART:

- In UART communication, two UARTs communicate directly with each other. The transmitting UART converts parallel data from a controlling device like a CPU into serial form, transmits it in serial to the receiving UART, which then converts the serial data back into parallel data for the receiving device.

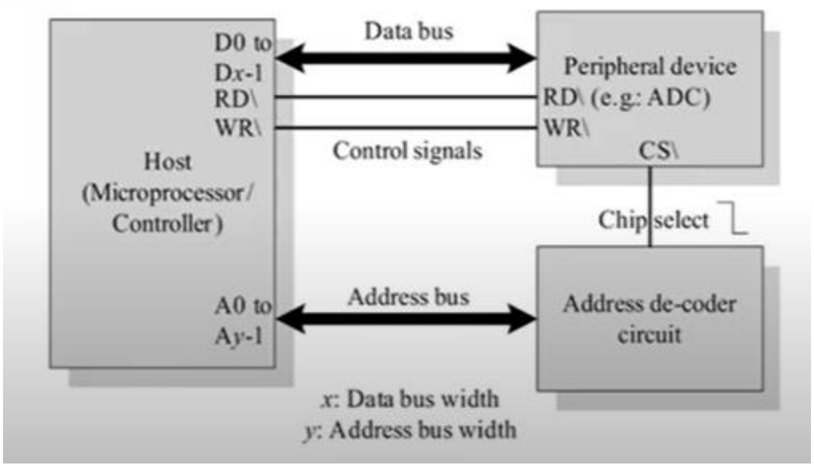
- Only two wires are needed to transmit data between two UARTs. Data flows from the Tx pin of the transmitting UART to the Rx pin of the receiving UART:



- UARTs transmit data asynchronously, which means there is no clock signal to synchronize the output of bits from the transmitting UART to the sampling of bits by the receiving UART.
- Instead of a clock signal, the transmitting UART adds start and stop bits to the data packet being transferred. These bits define the beginning and end of the data packet so the receiving UART knows when to start reading the bits.
- When the receiving UART detects a start bit, it starts to read the incoming bits at a specific frequency known as the baud rate. Baud rate is a measure of the speed of data transfer, expressed in bits per second (bps). Both UARTs must operate at about the same baud rate. The baud rate between the transmitting and receiving UARTs can only differ by about 10% before the timing of bits gets too far off.

Parallel Interface:

- The on board parallel interface is normally used for or communicating with parallel devices which are memory mapped to the host of the system.
- The host processor /controller of the embedded system contain a parallel bus and the device which supports parallel bus can directly connect to this bus system.
- The communication through the parallel bus is controlled by the control signal interface between the device and the host.
- The control signals for communication includes read /write signal and device select signal. The device normally contains device select line and the device becomes active only when this line is asserted by the host processor.



- The direction of data transfer can be controlled through the control signal lines for Read and Write. Only the host processor has control over the Read and Write control signal. The device is normally memory mapped to the host processor and range of address is assigned to it.
- An address Decoder circuit is used for generating the chip select signal for the device. When the address selected by the processor is within the range assigned for the device, the Decoder circuits activates the chip select line and thereby the device becomes active. The processor then can read or write from or to the device by asserting the corresponding control line

USB:

- USB, short for Universal Serial Bus, is a standard type of connection for many different kinds of devices. Generally, USB refers to the types of cables and connectors used to connect these many types of external devices to computers.

- The Universal Serial Bus standard has been extremely successful. USB ports and cables are used to connect hardware such as printers, scanners, keyboards, mice, flash drives, external hard drives, joysticks, cameras, and more to computers of all kinds, including desktops, tablets, laptops, net books, etc.

- In fact, USB has become so common that you'll find the connection available on nearly any computer-like device such as video game consoles, home audio/visual equipment, and even in many automobiles.
- Many portable devices, like smart phones, eBook readers, and small tablets, use USB primarily for charging. USB charging has become so common that it's now easy to find replacement electrical outlets at home improvement stores with USB ports built it, negating the need for a USB power adapter.

Wi-Fi:

- Wi-Fi is a technology for wireless local area networking with devices based on the IEEE 802.11standards. Wi-Fi is a trademark of the Wi-Fi Alliance, which restricts the use of the term Wi-Fi Certified to products that successfully complete interoperability certification testing.
- Devices that can use Wi-Fi technology include personal computers, video-game consoles, phones and tablets, digital cameras, smart TVs, digital audio players and modern printers.
- Wi-Fi compatible devices can connect to the Internet via a WLAN and a wireless access point. Such an access point (or hotspot) has a range of about 20 meters (66 feet) indoors and a greater range outdoors. Hotspot coverage can be as small as a single room with walls that block radio waves, or as large as many square kilometres achieved by using multiple overlapping access points.

General Packet Radio Service:

- GPRS is a packet oriented mobile data service on the 2G and 3G cellular communication system's global system for mobile communications (GSM). GPRS was originally standardized by European Telecommunications Standards Institute (ETSI) in response to the earlier CDPD and i-mode packet-switched cellular technologies. It is now maintained by the 3rd Generation Partnership Project (3GPP).
- GPRS usage is typically charged based on volume of data transferred, contrasting with circuit switched data, which is usually billed per minute of connection time.

Sometimes billing time is broken down to every third of a minute. Usage above the bundle cap is charged per megabyte, speed limited, or disallowed.

- GPRS is a best-effort service, implying variable throughput and latency that depend on the number of other users sharing the service concurrently, as opposed to circuit switching, where a certain quality of service (QoS) is guaranteed during the connection. In 2G systems, GPRS provides data rates of 56–114 kbit/second. 2G cellular technology combined with GPRS is sometimes described as 2.5G, that is, a technology between the second (2G) and third (3G) generations of mobile telephony. It provides moderate-speed data transfer, by using unused time division multiple access (TDMA) channels in, for example, the GSM system. GPRS is integrated into GSM Release 97 and newer releases.

SAI VIDYA INSTITUTE OF TECHNOLOGY

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DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGINEERING

SUBJECT: BASIC ELECTRONICS & COMMUNICATION

SUBJECT CODE--[21ELN14/24]

MODULE - 4

Syllabus:

Analog and Digital Communication – Modern communication system scheme, Information source, and input transducer, Transmitter, Channel or Medium – Hardwired and Soft wired, Noise, Receiver, Multiplexing, Types of communication systems.

Types of modulation (only concepts) – AM, FM, Phase Modulation, Pulse Modulation, PAM, PWM, PPM, PCM.

Concept of Radio wave propagation (Ground, space, sky**)** Concepts of Sampling theorem, Nyquist rate, Digital Modulation Schemes– ASK, FSK, PSK

Radio signal transmission Multiple access techniques, Multipath and fading, Error Management

Antenna, Types of antennas (only definition and antenna model, **exclude radiation patterns).**

MODERN COMMUNICATION SYSTEM SCHEME:

- Communication engineering deals with the techniques of transmitting information.
- Communication engineering means electrical communication, in which information is transmitted through electrical signals.
- Electrical communication is a process by which the information message is transmitted from one point to another, from one person to another, or from one place to another in the form of electrical signals, through some communication link.
- Basic communication system provides a link between the information source and its destination. The process of electrical communication involves sending, receiving, and processing information in electrical form.
- The information to be transmitted passes through a number of stages of the communication system prior it reaches its destination.
- Figure 4.1 shows a block schematic diagram of the most general form of basic communication system.

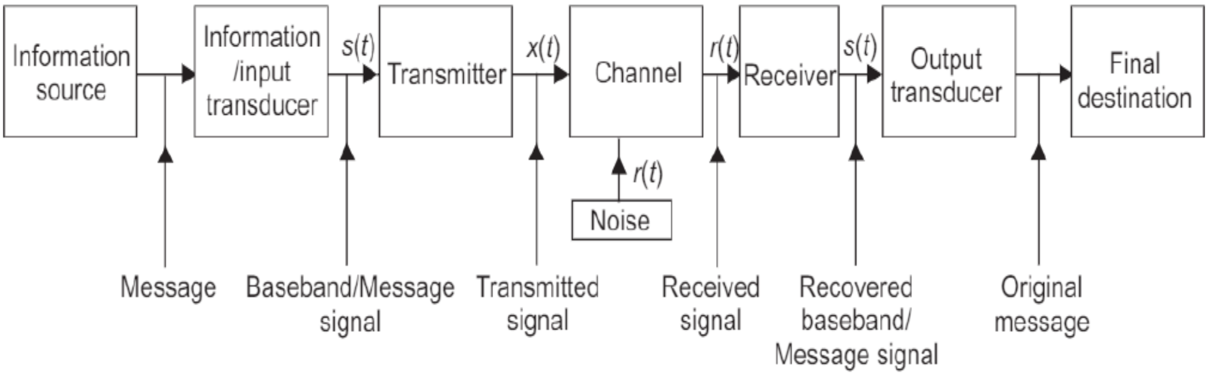


Fig 4.1 Schematic diagram of the most general form of basic communication system.

The main constituents of basic communication system are:

(i) Information source and input transducer

(ii) Transmitter

(iii) Channel or medium

(iv) Noise

(v) Receiver

(vi) Output transducer and final destination.

(i) Information source:

- A communication system transmits information from an information source to a destination and hence the first stage of a communication system is the information source.
- Ex: A sentence or paragraph spoken by a person is a message that contains some information. The

person, in this case, acts as information source. Few other familiar examples of messages are voice, live scenes, music, written text, and e-mail.

- A communication system transmits information in the form of electrical signal or signals.

(i) Input transducer:

- A transducer is a device that converts a non-electrical energy into its corresponding electrical energy called signal and vice versa, e.g., during a telephone conversation, the words spoken by a person are in the form of sound energy.
- An example of a transducer is a microphone. Microphone converts sound signals into the corresponding electrical signals.
- Similarly, a television (TV) picture tube converts electrical signals into its corresponding pictures. Some other examples of transducers are movie cameras, Video Cassette, Recorder (VCR) heads, tape recorder heads, and loudspeakers.
- The information produced by the information source is applied to the next stage, termed the information or input /transducer. This in turn, produces an electrical signal corresponding to the information as output. This electrical signal is called the baseband signal. It is also called a **message signal** s(t).
- There are two types of signals. (a) analog signal, and (b) digital signal.

(a) Analog Signal

- An analog signal is a function of time, and has a continuous range of values. However, there is a definite function value of the analog signal at each point of time.
- A familiar example of analog signal or analog wave form is a pure sine wave form. A practical example of an analog signal is a voice signal. When a voice signal is converted to electrical for by a microphone, one gets a corresponding electrical analog signal.

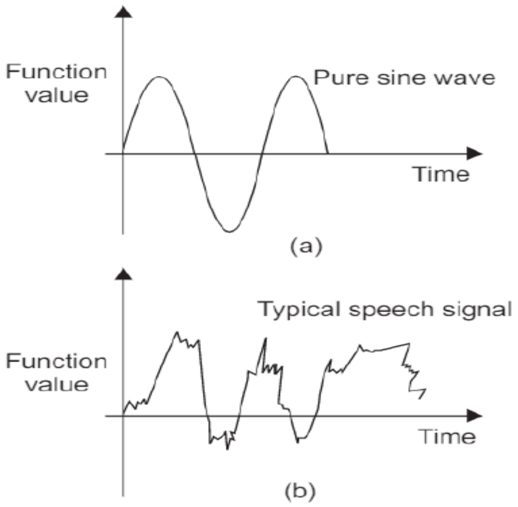


Fig 4.2: Analog signals (a) Pure sine wave (b) Typical speech signal

(b) Digital signal

- A digital signal does not have continuous function values on a time scale. It is discrete in nature, i.e., it has some values at discrete timings.
- A familiar example of a digital signal is the sound signal produced by drumbeats.

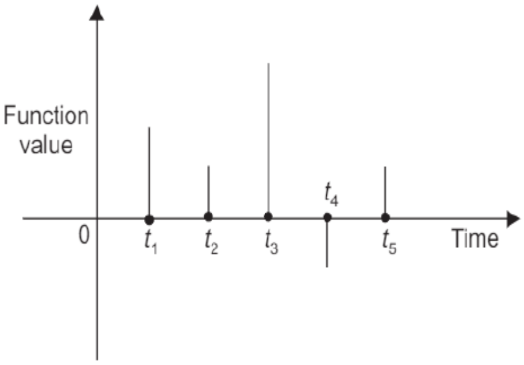


Fig 4.3: Digital signal

- Digital signals in their true sense correspond to a binary digital signal, where the discrete amplitude of the signal is coded into binary digits represented by '0' and '1'.
- The analog signal, which is continuous in time, is converted to discrete time, using a procedure calling sampling. The continuous amplitude of the analog signal is converted to discrete amplitude using a process called quantization. Sampling and quantization are together termed as analog-to-digital conversion (ADC) and the circuitry that performs this operation is called an analog to-digital converter.

(ii) Transmitter:

The transmitter section processes the signal prior transmission. There are two following options for processing signals prior transmission:

(i) The baseband signal, which lies in the low frequency spectrum, is translated to a higher frequency spectrum.

(ii) The baseband signal is transmitted without translating it to a higher frequency spectrum.

- The baseband signal is converted into a corresponding series of sine waves of two different frequencies prior to transmission. Figure 4.4 illustrates this processing.
- The carrier communication system is based on the principle of translating a low frequency baseband signal to higher frequency spectrum. This process is termed as ***modulation.***
- If the baseband signal is a digital signal, the carrier communication system is called a ***digital communication system***. The digital modulation methods are employed for this.
- If the baseband signal is an analog signal, the carrier communication system is called as an ***analog communication system*** and for processing the analog modulation techniques are used.

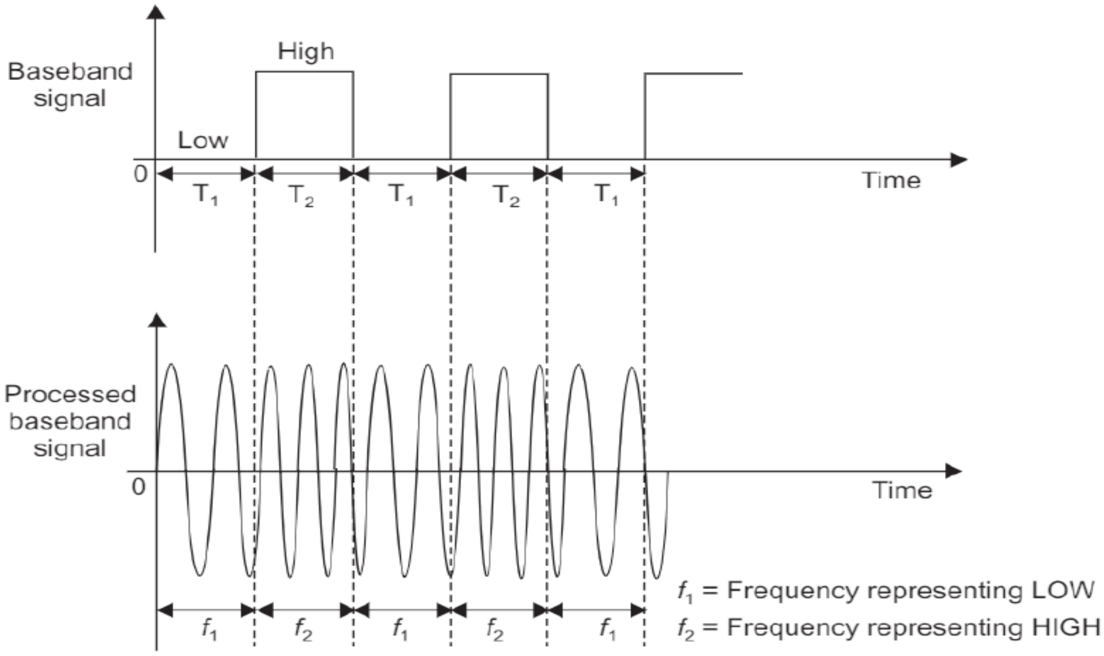


Fig: 4.4 The processing of a baseband signal

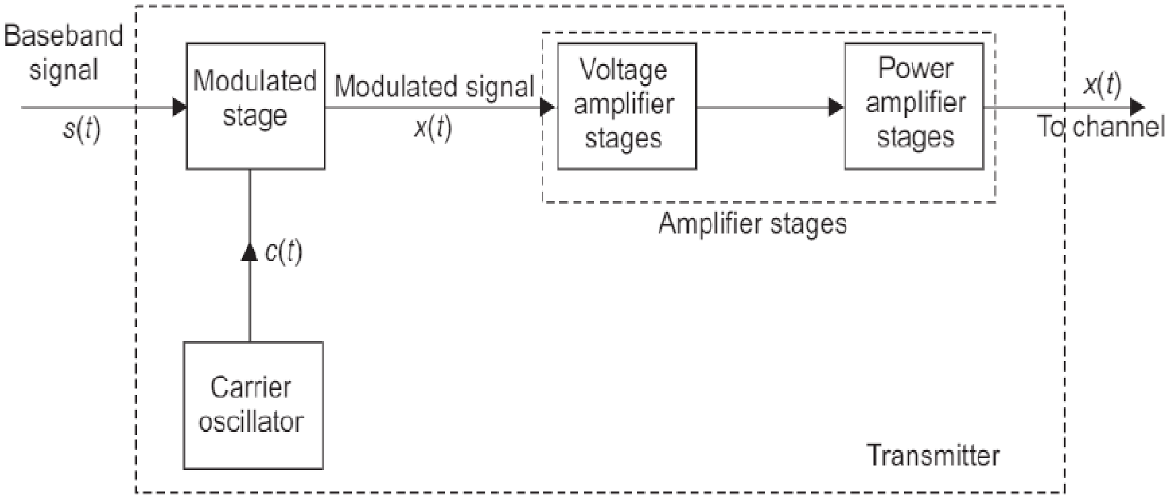


Fig. 4.5 Block diagram representing schematic of an analog transmitter section

- Figure 4.5 shows the baseband signal, s(t) applied to the modulated stage. This stage translates the baseband signal from its low frequency spectrum to high frequency spectrum. This stage also receives another input called the *carrier signal*, c(t), which is generated by a high frequency carrier oscillator.
- Modulation takes place at this stage with the baseband and the carrier signals as two inputs after modulation, the baseband signal is translated to a high frequency spectrum and the carrier signal is said to be modulated by the baseband signal.
- The output of the modulated stage is called the *modulated signal*, and is designated as x(t). The voltage of the modulated signal is then amplified to drive the last stage of the transmitter, called the *power amplifier stage* (Fig. 4.5).

- This stage amplifies the power of the modulated signal and thus it carries enough power to reach the receiver stage of the communication system. Finally, the signal is passed to the transmission medium or channel.
- *Radio signals* are transmitted through *electromagnetic (em) waves*, also referred as radio waves, in a radio communication system.
- The radio waves have a wide frequency range starting from a few ten kilo Hertz (Hz) to several thousand Mega Hertz (MHz). This wide range of frequencies is referred as the *radio frequency* (RF) spectrum.

(iii) Channel or Medium:

- After the required processing, the transmitter section passes the signal to the transmission medium. The signal propagates through the transmission medium and is received at the other side by the receiver section. The transmission medium between the transmitter and the receiver is called a channel.
- ***Channel*** is a very important part of a communication system as its characteristics add many constraints to the design of the communication system, e.g., most of the noise is added to the signal during its transmission through the channel.
- Depending on the physical implementations, one can classify the channels in the following two groups:

Hardware Channels:

These channels are manmade structure which can be used as transmission medium. There are following three possible implementations of the hardware channels.

1. Transmission lines
2. Waveguides
3. Optical Fiber Cables (OFC)

- The examples of transmission lines are *Twisted-pair cables* used in landline telephony and coaxial cables used for cable TV transmission. However, transmission lines are not suitable for ultra-high frequency (UHF) transmission.
- To transmit signals at UHF range, *Waveguides* are employed as medium. Waveguides are hollow, circular, or rectangular metallic structures. The signals enter the waveguide, are reflected at the metallic walls, and propagate towards the other end of the waveguide.
- *Optical fiber cables* are highly sophisticated transmission media, in the form of extremely thin circular pipes. e.g., landline telephony and cable TV network.

Software Channels:

There are certain natural resources which can be used as the transmission medium for signals.

Such transmission media are called *software channels.*

- The possible natural resources that can be used as software channels are: *air or open space* and *sea water*.
- The most widely used software channel is air or open space. The signals are transmitted in the form of electromagnetic (em) waves, also called *radio waves*.
- Systems that use radio waves to transmit signals through open space are called radio communication systems, e.g., radio broad cast, television transmission, satellite communication, and cellular mobile communication.

(iv) Noise:

- In electronics and communication engineering, *noise* is defined as unwanted electrical energy of random and unpredictable nature present in the system due to any cause.
- Obviously, noise is an electrical disturbance, which does not contain any useful information. Thus, noise is a highly undesirable part of a communication system, and have to be minimized.

SNR and Noise Figure (F):

- One can define the SNR as the ratio of the signal power to the noise power at a point in the circuit. Now, if P_s, is signal power and P_n, is noise power, then SNR expressed as S/N, is given as

$$\left(\frac{S}{N}\right) = \left(\frac{P_s}{P_n}\right)$$

If $P_s = V_s^2R$ and $P_n = V_n^2R$, then

$$\frac{S}{R} = \frac{P_s}{P_n} = \frac{V_s^2R}{V_n^2R}$$

where V_s, is signal voltage and V_n, is noise voltage.

- In addition, it is assumed that both the signal and noise powers are dissipated in the same resistor R. Therefore, SNR can be expressed in terms of decibels (dB) as

$$\left(\frac{S}{N}\right) dB = 10 \log_{10}\left(\frac{V_s^2}{V_n^2}\right)$$

$$\left(\frac{S}{N}\right) dB = 20 \log_{10}\left(\frac{V_s}{V_n}\right)$$

- For example, if, at a particular point in a circuit, the signal and noise voltages are given as 3.5 mV and 0.75 mV, respectively, SNR in dB is calculated as:

$$\left(\frac{S}{N}\right) dB = 20 \log_{10}\left(\frac{3.5}{0.75}\right)$$

$$\left(\frac{S}{N}\right) dB = 20 \log_{10}(4.66)$$

$$\left(\frac{S}{N}\right) dB = 13.38\, dB$$

Clearly, the SNR of the circuit at the point is 13.38 *dB.*

- The ***Noise figure*** **(F)** is the measure of the noise introduced by the circuit. It is defined as the ratio of the signal-to-noise power at the input of the circuit and the signal-to-noise power at the output of the circuit. Noise figure (r) can be expressed as

$$F = \frac{\frac{S}{N}\ Power\ at\ the\ input\ terminals\ of the\ circuit}{\frac{S}{N}\ Power\ at\ the\ output\ terminals\ of the\ circuit}$$

Receiver:

- The task of the receiver is to provide the original information to the user. This information is altered due to the processing at the transmitter side.
- The signal received by the receiver, thus does not contain information in its original form. The receiver system receives the transmitted signal and performs some processing on it to the original baseband signal.
- The function of the receiver section is to separate the noise from the received signal, and then recover the original baseband signal by performing some processing on it.
- The receiver performs an operation known as *demodulation,* which brings the baseband signal from the higher frequency spectrum to its original low-frequency spectrum. The demodulation process removes the high frequency carrier from the received signal and retrieves the original baseband.

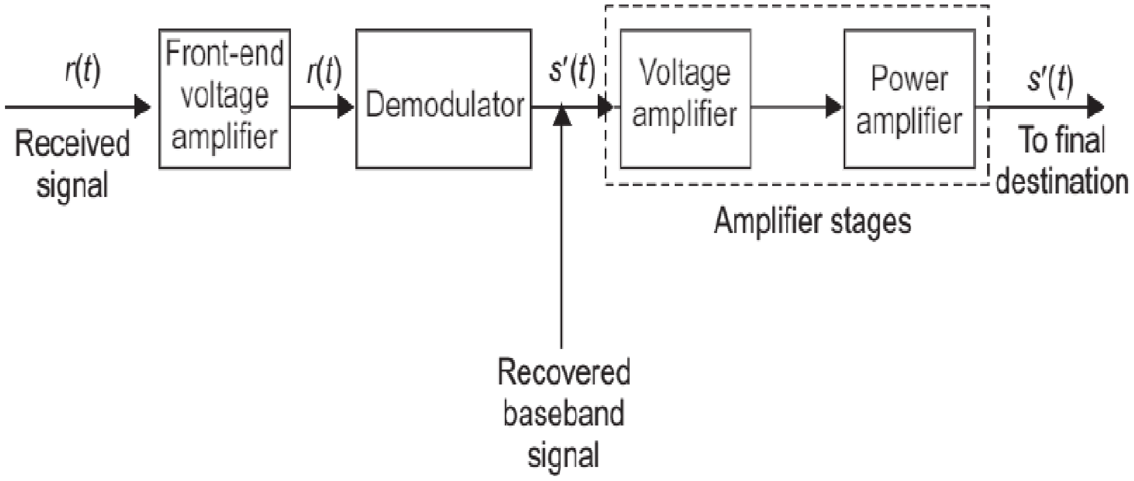


Fig: 4.6 Detailed block diagram of a typical receive section

- From Fig. 4.6 it is evident that the received signal, r(t), is first amplified by the front-end voltage amplifier. This is done to strengthen the received signal, which is weak and to facilitate easy processing. Next, this signal is given to the demodulator, which in turn, demodulates the received signal to recover the original baseband signal. After recovering the original baseband signal, its voltage and power is amplified prior it to final destination block.

MULTIPLEXING

- This is a technique that is most widely used in nearly all types of communication systems, radio and line communication systems.
- Basically, multiplexing is a process which allows more than one signal to transmit through a single channel.
- The use of multiplexing also makes the communication system economical because more than one signal can be transmitted through a single channel.
- Multiplexing is possible in communication system only through modulation.
- To consider multiplexing, let us consider the following example. If many people speak loudly and simultaneously, then it becomes nearly impossible to understand their conversion because the overall result is noise. This noise is the result of mixing of all the speeches. The human ear is not capable of separating these intermingled speeches and therefore no intelligent words are communicated to brain. The same situation is now applied to the transmission of audio signals. These audio signals may come from, say ten different persons. While the speech frequency of different persons will be different, all the ten signals will lie in the same audio range of 20 Hz to 20 kHz.

TYPES OF COMMUNICATION SYSTEMS

One may categorize communication systems based on:

- The *physical infrastructure* pertains to the type of the channel used and the hardware design of the transmitting and receiving equipment.
- The signal *specifications* signify the nature and type of the transmitted signal

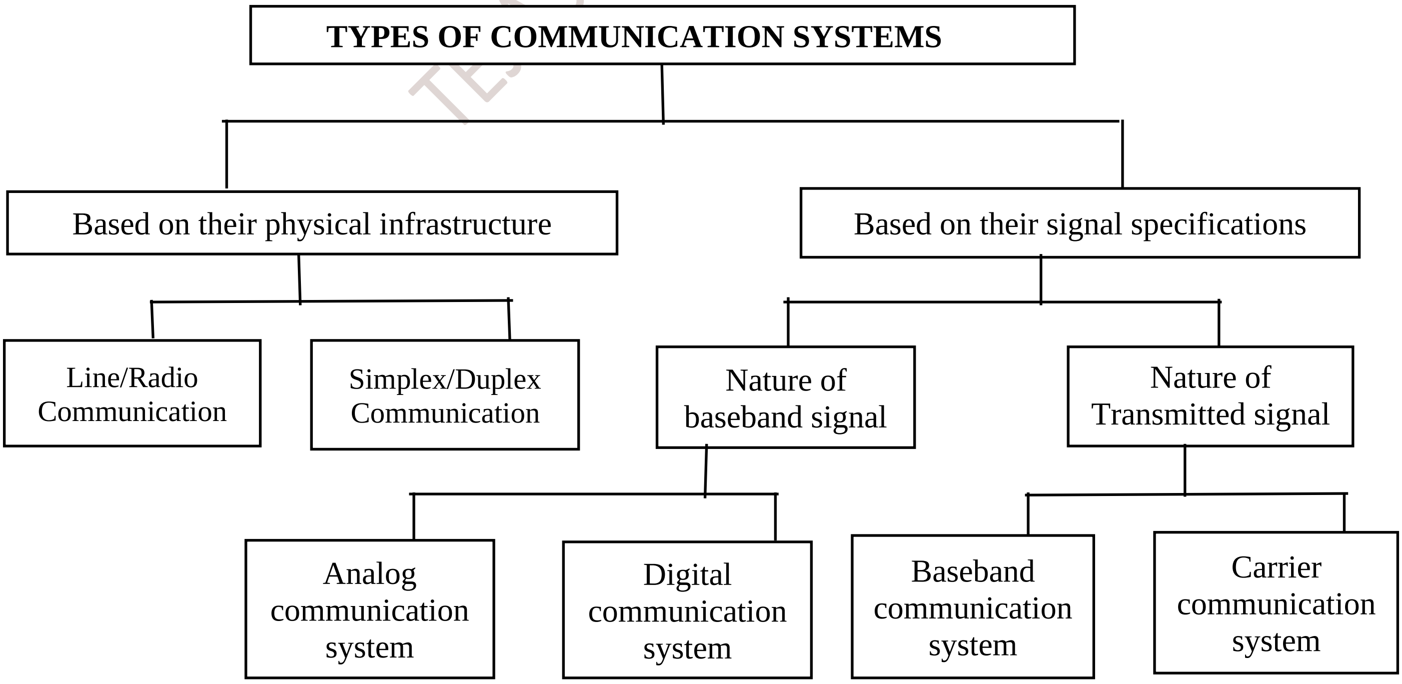


Fig 4.7: Types of Communication Systems

MODULATION

Modulation is the process of changing the parameters of the carrier signal, in accordance with the instantaneous values of the modulating signal.

TYPES OF MODULATION:

- Continuous-wave Modulation
    - Amplitude Modulation
    - Frequency Modulation
    - Phase Modulation
- Pulse modulation
    - PAM
    - PWM
    - PPM
    - PCM

1. *Amplitude modulation (AM)*

- AM is defined as the modulation technique in which the instantaneous amplitude of the carrier signal is varied in accordance with the instantaneous amplitude of the analog modulating signal to be transmitted while the frequency and the phase of the carrier signal remain unchanged.
- Figure 4.8 shows the high frequency carrier signal, modulating signal and the modulated signal.

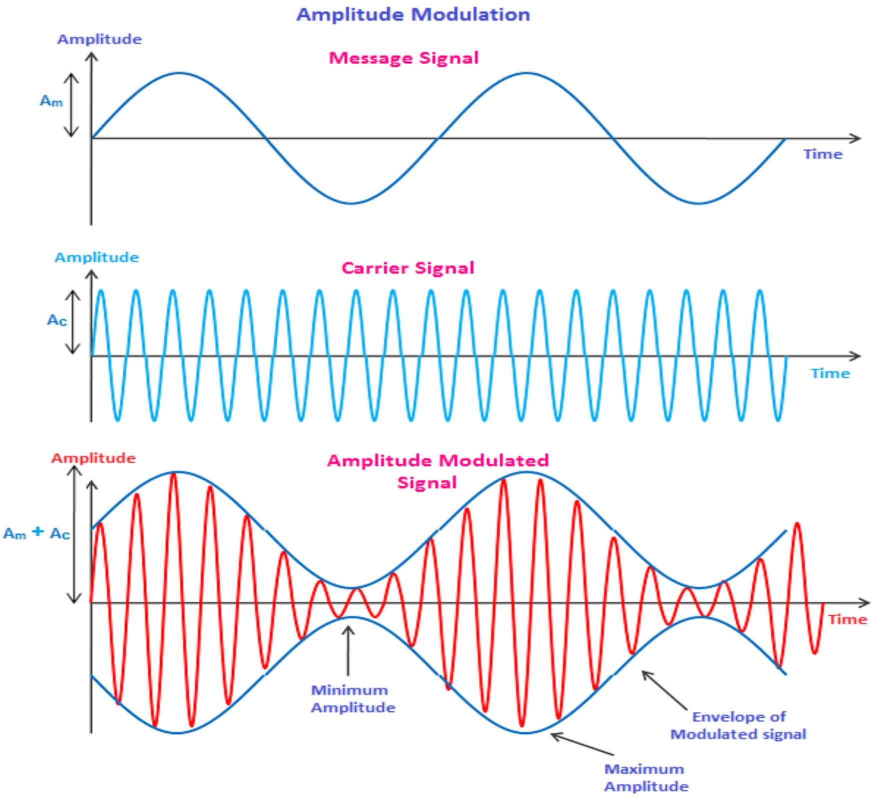


Fig: 4.8 Waveforms of Amplitude modulation

- It can be clearly seen from the figure 4.8 that the modulating signal seems to be superimposed on the carrier signal. The amplitude variations in the peak values of the carrier signal exactly replicate the modulating signal at different points in time which is known as an envelope.

2. ***Frequency Modulation:***

- A modulating signal may vary the frequency of the carrier keeping the amplitude and phase constant. This type of modulation is called Frequency modulation. Broadly speaking, the frequency modulation is the process of changing the frequency of the carrier voltage in accordance with the instantaneous value of the modulating voltage.
- The original frequency of the carrier signal is called Centre or resting frequency and denoted by f_c. The amount by which the frequency of the carrier wave changes or shifts above or below the resting frequency is termed as frequency deviation (Δf). This means $\Delta f \propto m(t)$.
- The total variation is frequency of F.M. wave from the lowest to the highest is termed as carrier saving (CS), i.e., US = 2 x frequency deviation in Centre frequency or CS=2 Δf.
- Modulation index in F.M. is the ratio of frequency deviation to the modulating frequency,

$$\text{i.e. } \mu_f = \frac{Frequency\ Deviation}{Modulating\ Frequency} = \frac{\Delta f}{f_m}$$

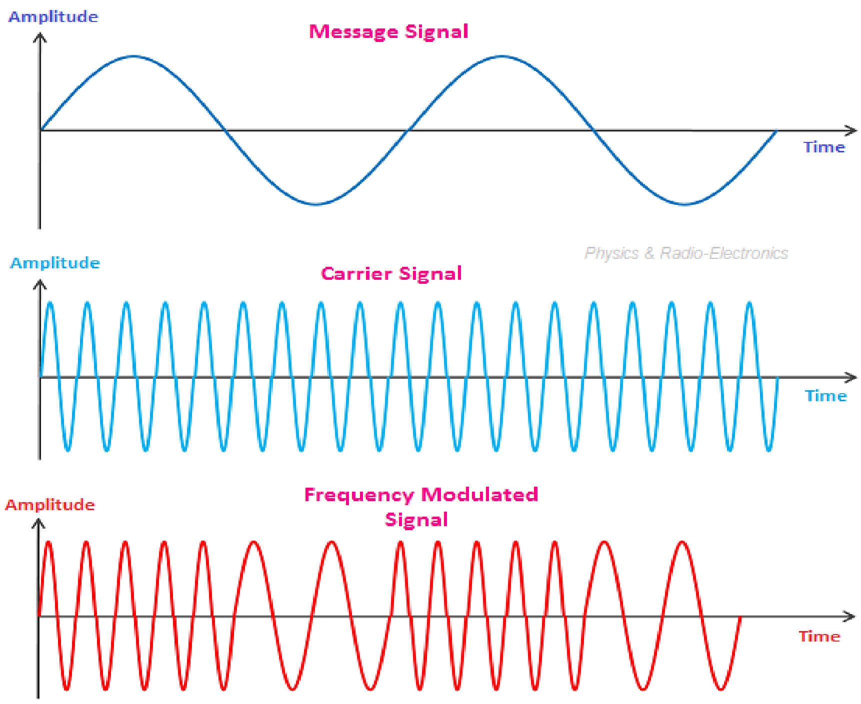


Fig: 4.9 Waveforms of Frequency modulation

3. ***Phase modulation:***

- PM is another form of angle modulation. Phase modulation is the process in which the instantaneous phase of the carrier signal is varied in accordance with the instantaneous amplitude of the modulating signal.

- In this type of modulation, the amplitude and frequency of the carrier signal remains unaltered after pulse modulation.

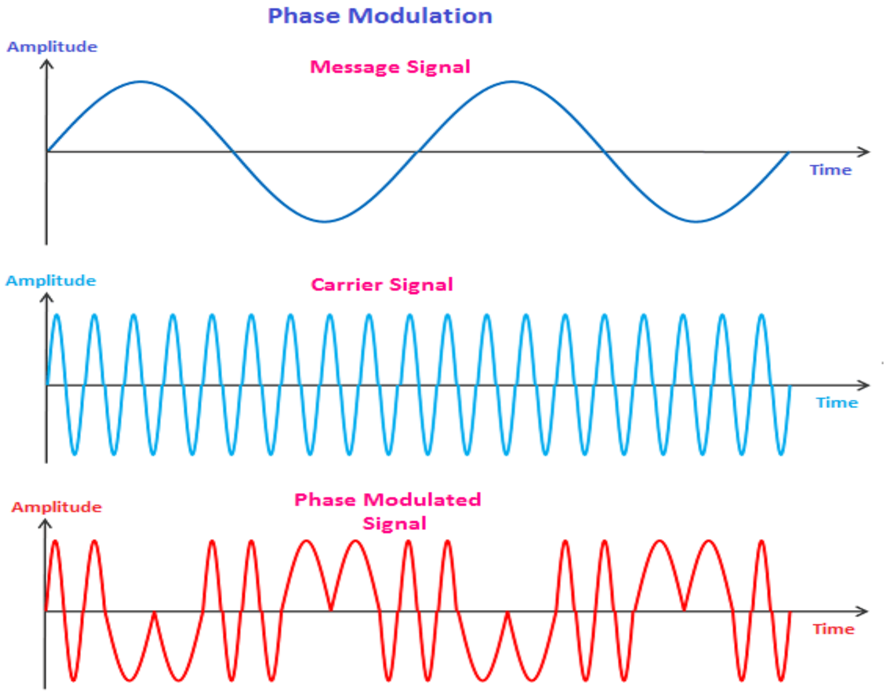


Fig: 4.10 Waveforms of Phase modulation

Pulse modulation:

- It may be used to transmit analog information, such as continuous speech or data. It is system in which continuous waveforms are sampled at regular intervals. Pulse modulation may be subdivided into two categories, analog and digital.
- Pulse-amplitude and pulse-time modulation are both analog, while the pulse code and delta modulation system are both digital.

4.1 *Phase-amplitude modulation (PAM):*

- PAM is the simplest form of pulse modulation. PAM is a pulse modulation system in which the signal is sampled at regular intervals, and each sample is made proportional to the amplitude of the signal at the instant of sampling. The pulses are then sent by either wire or cable, or else are used to modulate a carrier.
- The ability to use constant-amplitude pulses is a major advantage of pulse modulation, and since PAM does not use constant- amplitude pulses, it is infrequently used. When it is used, the pulses frequency- modulate the carrier.

4.2 *Pulse width or pulse-duration modulation (PWM or PDM):*

- In this system, the starting time and amplitude of each pulse are constant but the width or duration of each pulse is made proportional to the instantaneous value of analog signal.

- PDM has the disadvantage, when compared with pulse-position modulation (PPM), that its pulses are of varying width and therefore of varying power content. This means that the transmitter must be powerful enough to handle the maximum- width pulses, although the average power transmitted is perhaps only half of the peak power. PWM still works if synchronization between transmitter and receiver fails, whereas PPM does not.

4.3 *Pulse position-modulation (PPM):*

In this system, the amplitude and width of the pulses is kept constant, while the position of each pulse, in relation to the position of a recurrent reference pulse is varied by instantaneous sampled value of the modulating wave. As compared to PWM, PPM has the advantage of requiring constant transmitter power output, but the disadvantage of depending on transmitter receiver synchronization.

Pulse-code modulation (PCM):

PCM is a digital process in which the message signal is sampled is rounded off to the nearest value of a finite set of allowable values and rounded values are coded. PCM generator produces a series of numbers or digits. Each one of these digits, almost always in binary code, represents the approximate amplitude of the signal sample at that instant.

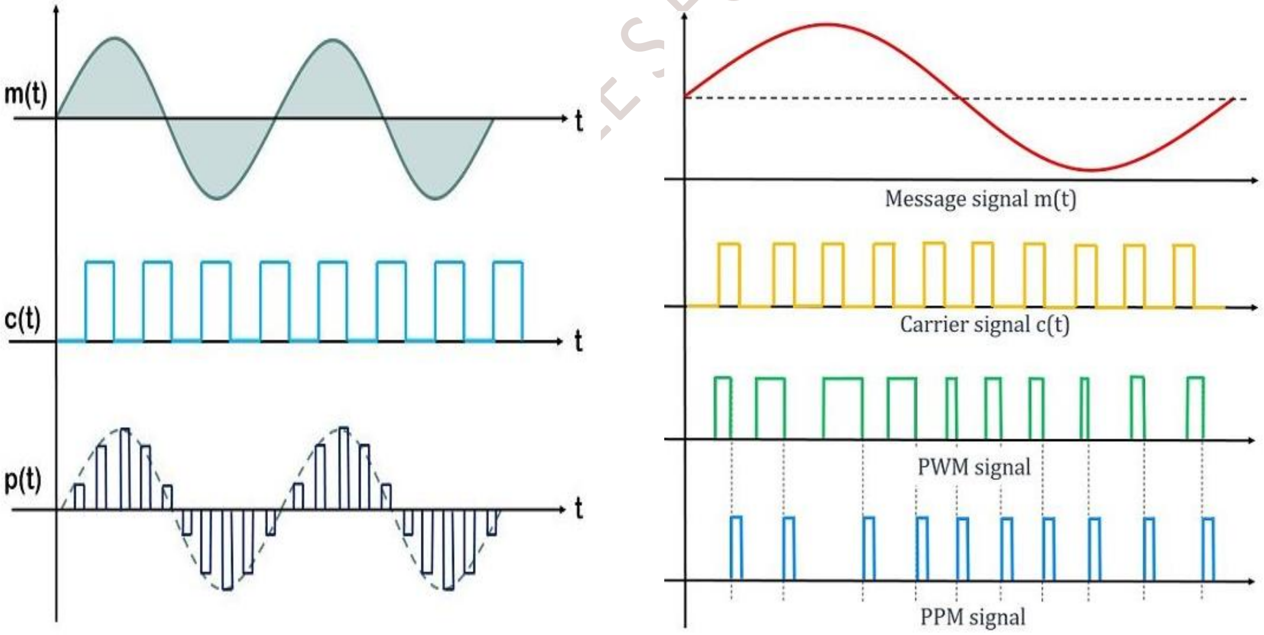


Fig 4.11: Waveforms of PAM, PWM & PPM

Radio wave propagation (Ground, space, sky):

- In *space communication* electromagnetic waves of different frequencies are used to carry information through the physical space acting as the transmission medium. Electromagnetic waves with frequencies extending from about 10 kHz to 300 GHz are classed as ***radio waves.***

- Depending primarily on the frequency a radio wave travels from the transmitting to the receiving antenna in several ways. On the basis of the mode of propagation, radio waves can be broadly classified as:
  (a) ground or surface wave. (b) space or tropospheric wave. (c) sky way.

(a). Ground wave propagation:

- In ground wave propagation, radio waves are guided by the earth and move along its curved surface from the transmitter to the receiver.
- As the waves moves over the ground, they are strongly influenced by the electrical properties of the ground. As high frequency waves are strongly absorbed by ground; ground wave propagation is useful only at low frequencies.
- Below 500 kHz, ground waves can be used for communication within distances of about 1500 km from the transmitter.
- AM radio broadcast in the medium frequency band cover local areas and take place primarily by the ground wave. Ground wave transmission is very reliable whatever the atmospheric conditions be.

(b). Space or tropospheric wave propagation:

- When a radio wave transmitted from an antenna, travelling in a straight line directly reaches the receiving antenna, it is termed as space or tropospheric wave.
- In space wave or line of sight propagation, radio waves move in the earth's troposphere within about 15 km over the surface of the earth.
- The space wave is made up of two components:
  (a) a direct or line-of- sight MGVE form the transmitting to the receiving antenna.
  (b) the ground-reflected urine traversing forms the transmitting antenna to ground and reflected to the receiving antenna.
- Television frequencies in the range 100-220 MHz are transmitted through this mode.

(c). Sky wave propagation:

- In this mode of propagation, radio waves transmitted from the transmitting antenna reach the receiving antenna after reflection form the ionosphere, i.e., the ionized layers lying in the earth's upper atmosphere.
- Short wave transmission around the globe is possible through sky wave via successive reflections at the ionosphere and the earth's surface.

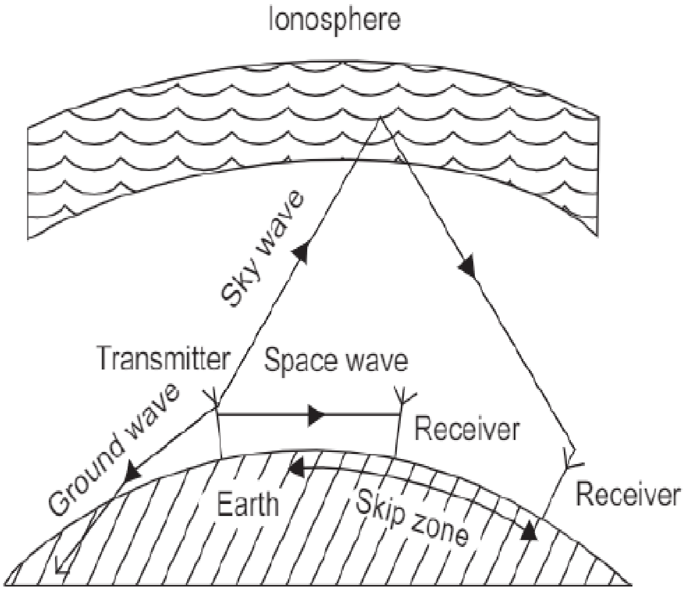


Fig 4.12: Mode of Propagation

Sampling theorem:

- There are two types of signals, continuous-time and discrete-time signals. The processing of discrete-time signals is more flexible and is also preferable than the continuous-time signals.
- The sampling theorem governs the conversion of continuous-time signal into discrete- time signal.
- The concept of sampling provides a widely used method for using discrete-time system technology to implement continuous-time systems and process the continuous-time signals.
- A continuous-time signal may be completely represented in its samples and recovered back if the sampling frequency is $f_s \geq 2f_m$. Here, f_s is the sampling frequency and f_m is the maximum frequency present in the signal.

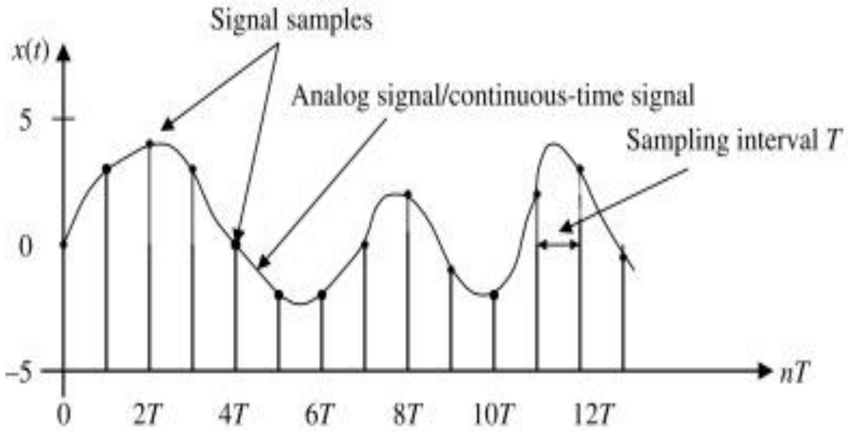


Fig 4.13: Sampling Waveform

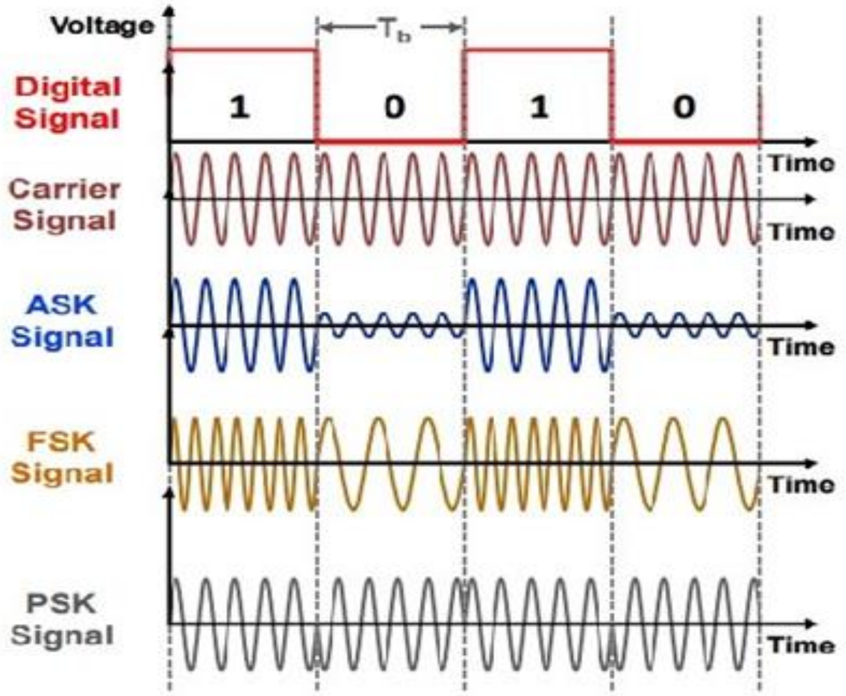
Nyquist Rate:

- When the sampling rate becomes exactly equal to $2f_m$ samples per second, then it is called Nyquist rate. Nyquist rate is the minimum sampling rate.

- A low pass filter is used to recover the original signal from its samples.
- The process of reconstructing the continuous-time signal from its samples is known as interpolation.
- When the sampling frequency is less than the Nyquist rate, aliasing problem is said to occur.
- Aliasing is the phenomenon in which a high frequency component in the frequency spectrum of the signal takes the identity of a lower frequency component in the spectrum of the sampled signal.
- To avoid aliasing:
  - Pre-alias filter must be used to limit the band of frequencies of the signal to f_m Hz.
  - Sampling frequency must be selected such that $f_s \geq 2f_m$.

Digital Modulation Schemes:

- In digital communications, the modulating signal consists of binary data. When it is required to transmit digital signals on a bandpass channel, the amplitude, frequency or phase of the sinusoidal carrier is varied in accordance with the incoming digital data.
- Since, the digital data is in discrete steps, the modulation of the bandpass sinusoidal carrier is also done in discrete steps. Due to this reason, this type of modulation is known as ***digital modulation***.
- Digital modulation schemes as classified as under:
  - Amplitude Shift Keying (ASK)
  - Frequency Shift Keying (FSK)
  - Phase Shift Keying (PSK)
- Because of constant amplitude of FSK or PSK, the effect of non-linearities, noise interference is minimum on signal detection. However, these effects are more pronounced on ASK. Therefore, FSK and PSK are preferred over ASK.
- Coherent digital modulation techniques are those techniques which employ coherent detection. In coherent detection, the local carrier generated at the receiver is phase locked with the carrier at the transmitter. Thus, the detection is done by correlating the received noisy signal and locally generated carrier. The coherent detection is also called synchronous detection.
- ASK signal may be generated by simply applying the incoming binary data and the sinusoidal carrier to the two inputs of a product modulator.
- The demodulation of binary ASK waveform can be achieved with the help of coherent detector.



Radio Signal Transmission

- Fig 4.14 shows the architecture of a wireless communication transmitter. In the figure, the transmitter usually processes the information in two stages. In the first stage, a modulator accepts the incoming bits, and computes symbols that represent the amplitude and phase of the outgoing wave. It then passes these to the analogue transmitter, which generates the radio wave itself.
- The modulation scheme used in Fig. 4.14 is known as quadrature phase shift keying (QPSK).
- A QPSK modulator takes the incoming bits two at a time and transmits them using a radio wave that can have four different states. These have phases of 45°, 135°, 225° and 315°.

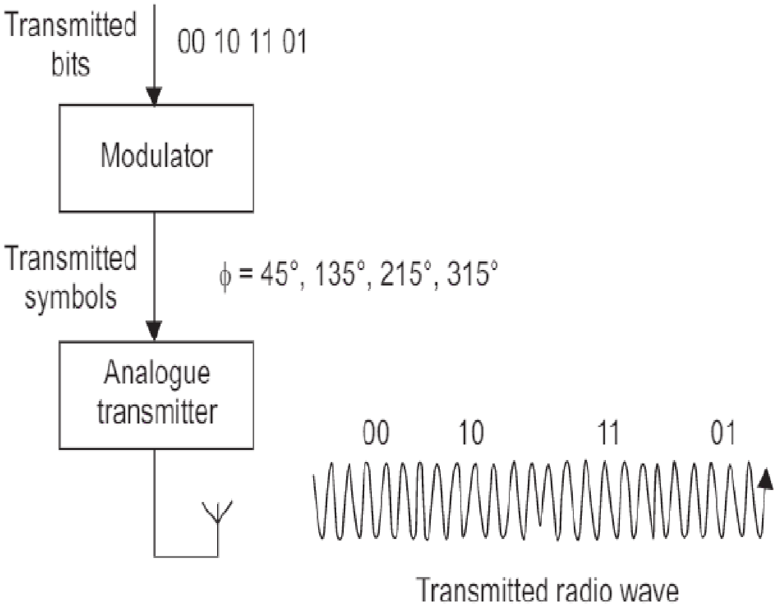
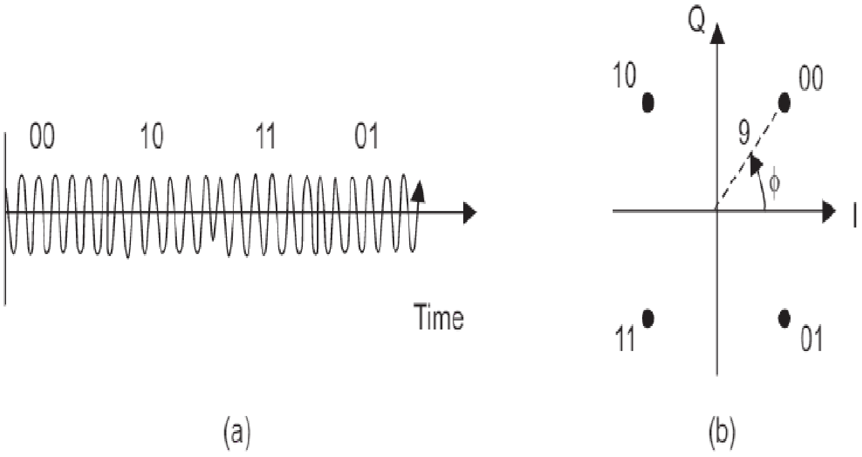


Fig 4.14: Architecture of a wireless communication transmitter



4.15: Quadrature phase shift keying (a) Example QPSK waveform (b) QPSK constellation diagram

- Fig. 4.15 (a), which correspond to bit combinations of 00, 10, 11 and 01 respectively. We can represent the four states of QPSK using the constellation diagram shown in Fig. 4.15 (b).
- In this diagram, the distance of each state from the origin represents the amplitude of the transmitted wave, while the angle (measured anti-clockwise from the x-axis) represents its phase. Usually, it is more convenient to represent each symbol using two other numbers, which are known as the in-phase (I) and quadrature (Q) components. These are computed as follows:

$$I = a \cos \phi$$

$$Q = a \sin \phi$$

where a is the amplitude of the transmitted wave and ϕ is its phase.

- Mathematicians will recognize the in-phase and quadrature components as the real and imaginary parts of a complex number.
- As shown in Fig. 4.16, LTE uses four modulation schemes altogether. Binary phase shift keying (BPSK) sends bits one at a time, using two states that can be interpreted as starting phases of 0° and 180°, or as signal amplitudes of +1 and -1.
- LTE uses this scheme for a limited number of control streams, but does not use it for normal data transmissions.
- 16 quadrature amplitude modulation (16-QAM) sends bits four at a time, using 16 states that have different amplitudes and phases.
- Similarly, 64-QAM sends bits six at a time using 64 different states, so it has a data rate six times greater than that of BPSK.

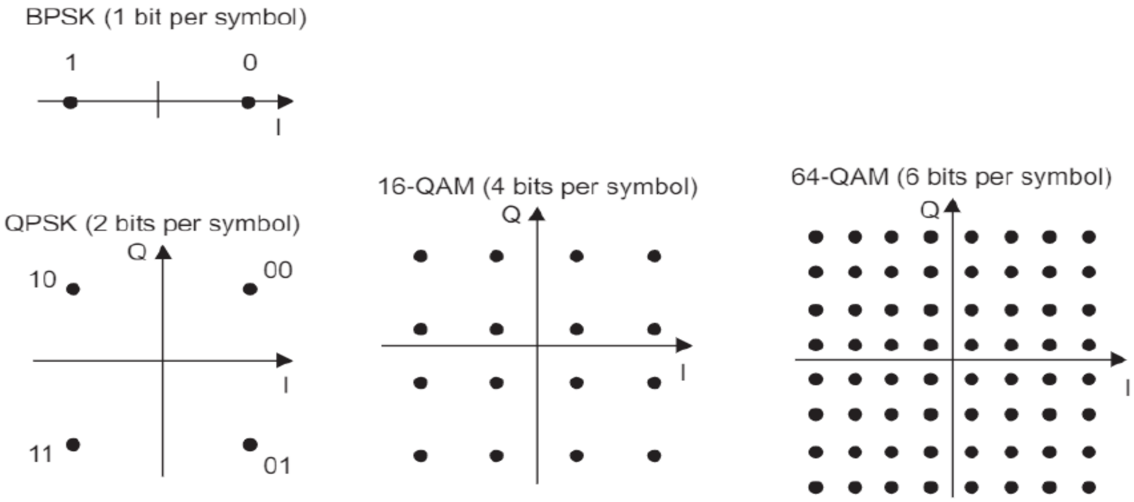


Fig 4.16: Modulation schemes used by LTE

Multiple Access Techniques:

- The techniques described so far work well for one-to-one communications. In a cellular network, however, a base station has to transmit to many different mobiles at once. It does this by sharing the resources of the air interface, in a technique known as multiple access.
- Mobile communication systems use a new different multiple access techniques, two of which are shown in Fig. 4.17 **frequency division multiple access (FDMA)** was used by the first-generation analogue systems. In this technique, each mobile receives on its own carrier frequency, which it distinguishes from the others by the use of analogue filters.
- In **time division multiple access (TDMA)**, mobiles receive information on the same carrier frequency but at different times.
- **GSM** uses a mix of frequency and time division multiple access, in which every cell has several carrier frequencies that are each shared amongst eight different mobiles.
- LTE uses another mixed technique known as orthogonal frequency division multiple access **(OFDMA).**

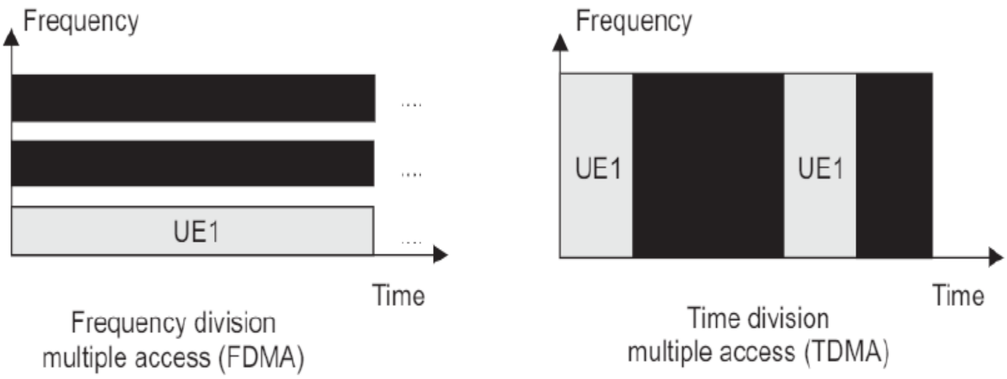


Fig 4.17: Example multiple access techniques

- Third generation communication systems used a different technique altogether, known as **code division multiple access (CDMA).**
- In this technique, mobiles receive on the same carrier frequency and at the same time, but the signals are labelled by the use of codes, which allow a mobile to separate its own signal from those of the others.
- LTE uses a few of the concepts from CDMA for some of its control signals, but does not implement the technique otherwise. Multiple access is actually a generalization of a simpler technique known as multiplexing.

FDD and TDD Modes

- A mobile communication system can operate in the transmission modes as shown in Fig. 4.18.
- When using frequency division duplex (FDD), the base station and mobile transmit and receive at the same time, but using different carrier frequencies.
- Using time division duplex (TDD), they transmit and receive on the same carrier frequency but at different times. FDD and TDD modes have different advantages and disadvantages.
- In FDD mode, the bandwidths of the uplink and downlink are fixed and are usually the same. This makes it suitable for voice communications, in which the uplink and downlink data rates are very similar.
- In TDD mode, the system can adjust how much time is allocated to the uplink and downlink. This makes it suitable for applications such as web browsing.
- TDD mode can be badly affected by interference if, for example, one base station is transmitting while a nearby base station is receiving.
- To avoid this, nearby base stations must be carefully time synchronized and must use the same allocations for the uplink and downlink, so that they all transmit and receive at the same time.
- This makes TDD suitable for networks that are made from isolated hotspots, because each hotspot can have a different timing and resource allocation. In contrast, FDD is often preferred for wide-area networks that have no isolated regions.

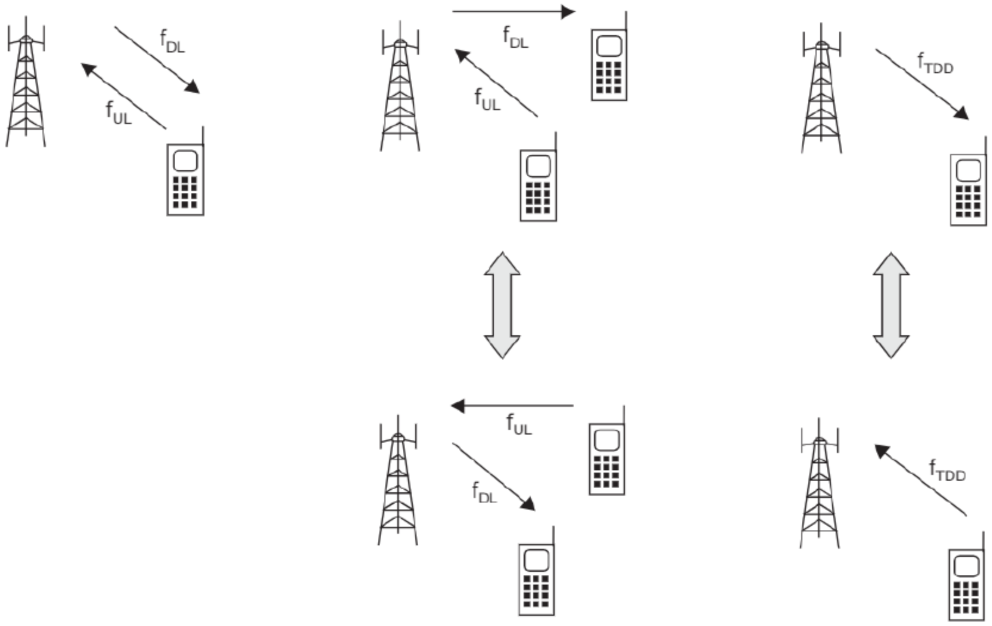


Fig 4.18: Operation of FDD and TDD modes

- When operating in FDD mode, the mobile usually has to contain a high attenuation duplex filter that isolates the uplink transmitter from the downlink receiver.
- LTE supports each of the modes described above. A cell can use either FDD or TDD mode.

MULTIPATH AND FADING

- Propagation loss and noise are not the only problem. As a result of reflections, rays can take several different paths from the transmitter to the receiver. This phenomenon is known as **multipath.**
- At the receiver, the incoming rays can add together in different ways, which are shown in Fig. 4.19. If the peaks of the incoming rays coincide then they reinforce each other, a situation known as ***constructive interference.***
- If, however, the peaks of one ray coincide with the troughs of another, then the result is destructive interference, in which the rays cancel.
- ***Destructive interference*** can make the received signal power drop to a very low level, a situation known as ***fading***. The resulting increase in the error rate makes fading a serious problem for any mobile communication system.

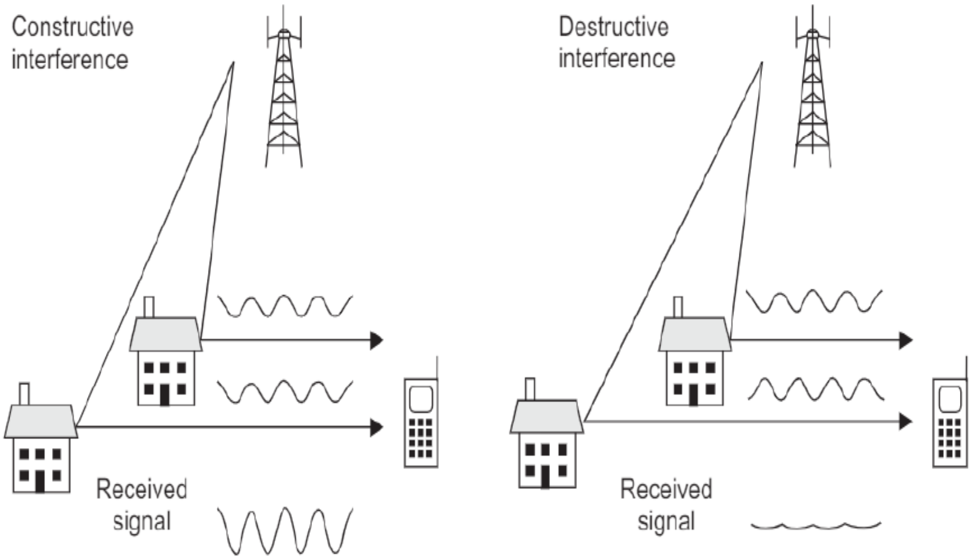


Fig 4.19: Generation of constructive interference, destructive interference and fading in a multipath environment

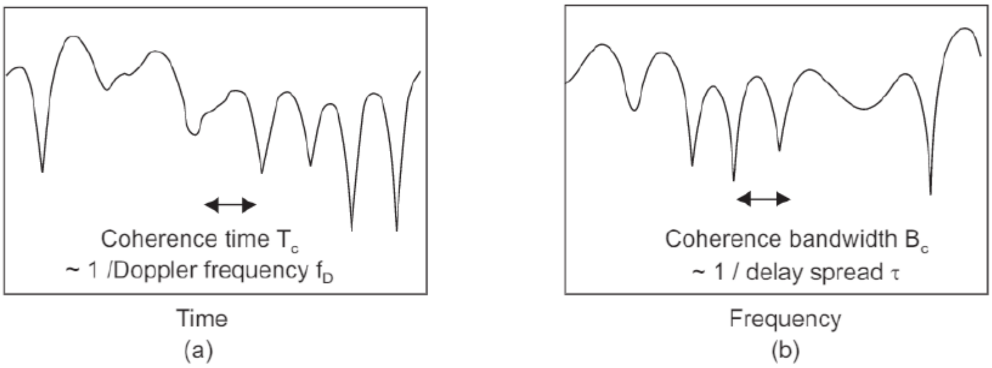


Fig 4.20: Fading as a function of (a) time and (b) frequency

- If the mobile moves from one place to another, then the ray geometry changes, so the interference pattern changes between constructive and destructive.
- Fading is therefore a function of time, as shown in Fig. 4.20 (a). The amplitude and phase of the received signal vary over a timescale called the *coherence time*, T_c which can be estimated as follows:

$$T_c \approx \frac{1}{f_D}$$

Here f_D is the mobile's Doppler frequency:

$$f_D = \frac{v}{c} f_c$$

where f_c is the carrier frequency, v is the speed of the mobile and c is the speed of light (3 x 10^8 m/s).

- For example, a pedestrian might walk with a speed of l ms^{-1} (3.6 km hr^{-1}). At a carrier frequency of 1500 MHz, the resulting Doppler shift is 5 Hz, giving a coherence time of about 200 milliseconds. Faster mobiles move through the interference pattern more quickly, so their coherence time is correspondingly less.

- If the carrier frequency changes, then the wavelength of the radio signal changes. This also makes the interference pattern change between constructive and destructive, so fading is a function of frequency as well 4.20 (b).

Error Management

Forward Error Correction:

- We know that noise and interference lead to errors in a wireless communication receiver. These are bad enough during voice calls, but are even more damaging to important information such as web pages and emails. Fortunately, there are several ways to solve the problem. The most important technique is **forward error correction.**
- In this technique, the transmitted information is represented using a *codeword* that is typically two or three times as long. The extra bits supply additional, redundant data that allow the receiver to recover the original information sequence.
- For example, a transmitter might represent the information sequence 101 using the codeword 110010111. After an error in the second bit, the receiver might recover the codeword 100010111.
- If the coding scheme has been well designed, then the receiver can conclude that this is not a valid codeword, and that the most likely transmitted codeword was 110010111.
- The receiver has therefore corrected the bit error and can recover the original information. The effect is very like written English, which contains redundant letters that allow the reader to understand the underlying information, even in the presence of spelling mistakes.
- The coding rate is the number of information bits divided by the number of transmitted bits (1/3 in the example above). Usually, forward error correction algorithms operate with a fixed coding rate.
- Despite this, a wireless transmitter can still adjust the coding rate using the two- stage process shown in Fig. 4.21.
- In the first stage, the information bits are passed through at fixed-rate coder. The main algorithm used by LTE is known as Turbo coding and has a fixed coding rate of 1/3.
- In the second stage, called rule matching, some of the coded bits are selected for transmission, while the others are discarded in a process known as puncturing.
- The receiver has a copy of the puncturing algorithm, so it can insert dummy bits at the points where information was discarded. It can then pass the result through a turbo decoder for error correction.

- Changes in the coding rate have a similar effect to changes in the modulation scheme. If the coding rate is low, then the transmitted data contain many redundant bits.
- This allows the receiver to correct a large number of errors and to operate successfully at a low SINR, but at the expense of a low information rate.
- If the coding rate is close to 1, then the information rate is higher but the system is more vulnerable to errors.
- LTE exploits this with a similar trade-off to the one we saw earlier, by transmitting with a high coding rate if the received SINR is high and vice versa.

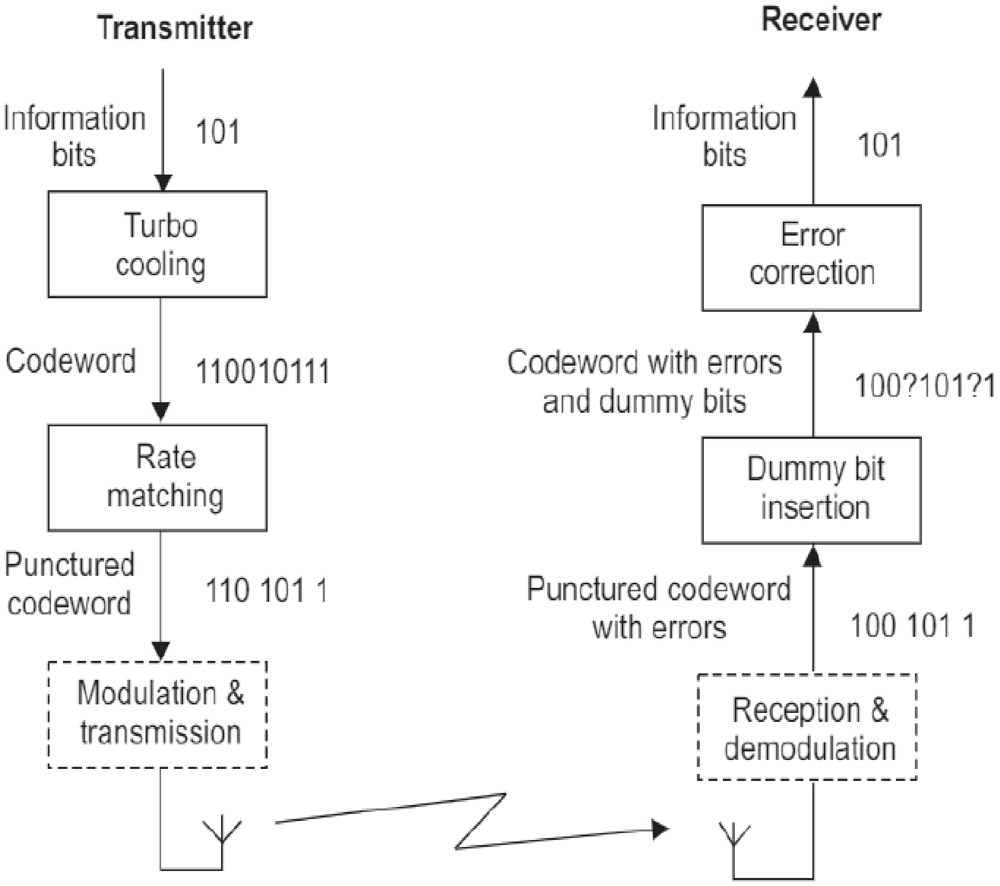


Fig 4.21: Block diagram of a transmitter and receiver using forward error correction and rate Matching

Automatic Repeat Request:

- Automatic repeat request (ARQ) is another error management technique, which is illustrated in Fig 4.22. Here, the transmitter takes a block of information bits and uses them to compute some extra bits that are known as a cyclic redundancy check (CRC).
- It appends these io the information block and then transmits the two sets of data in the usual way.

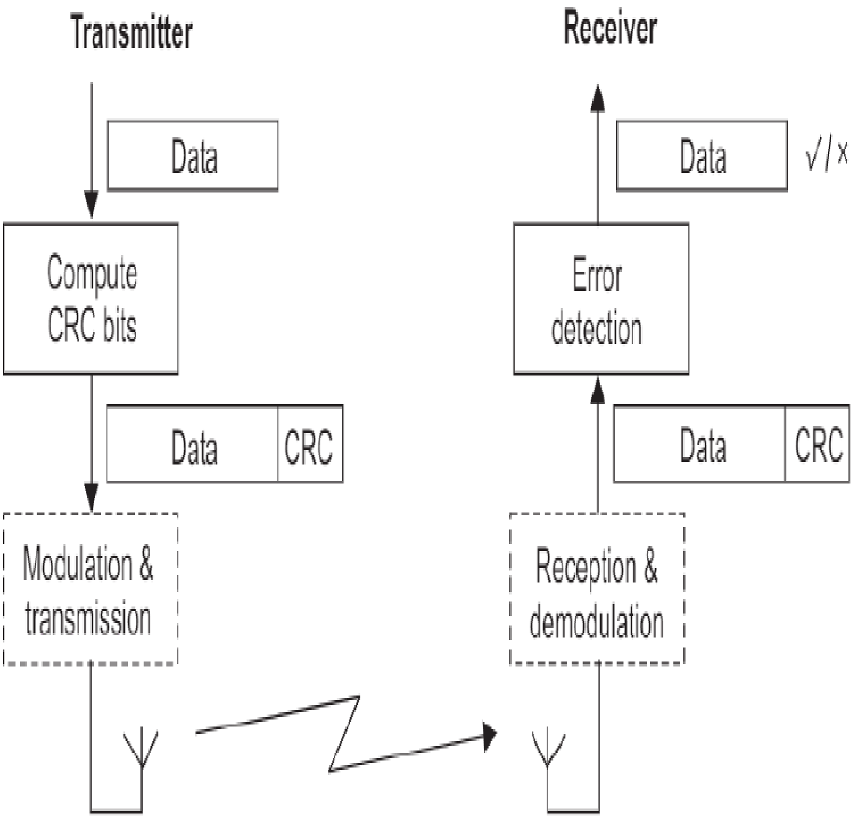


Fig 4.22: Block diagram of a transmitter and receiver using automatic repeat request

- The receiver separates the two fields and uses the information bits to compute the expected CRC bits. If the observed and expected CRC bits are the same, then it concludes that the information has been received correctly and sends a positive acknowledgement back to the transmitter.
- If the CRC bits are different, it concludes that an error has occurred and sends a negative acknowledgement to request a re-transmission.
- Positive and negative acknowledgements are often abbreviated to ACK and NACK respectively.
- A wireless communication system often combines the two error management techniques that we have been describing. Such a system corrects most of the bit errors by the use of forward error correction and then uses automatic repeat requests to handle the remaining errors that leak through.
- Normally, ARQ uses a technique called selective re-transmission Fig 4.23 in which the receiver waits for several blocks of data to arrive before acknowledging them all.
- This allows the transmitter to continue sending data without waiting for an acknowledgement, but it means that any re-transmitted data can take a long time to arrive.
- Consequently, this technique is only suitable for non-real-time streams such as web-pages and emails.

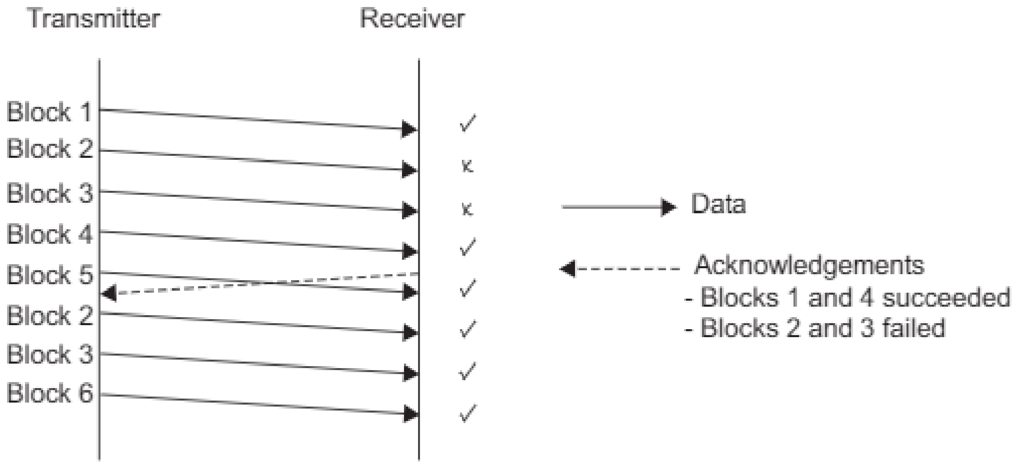


Fig 4.23: Operation of a selective re-transmission ARQ scheme

Antenna:

- An antenna is a device for converting electromagnetic radiation in space into electrical currents in conductors or vice-versa, depending on whether it is being used for receiving or for transmitting, respectively.
- Antennas transform wire propagated waves into space propagated waves. They receive electromagnetic waves and pass them onto a receiver or they transmit electromagnetic waves which have been produced by a transmitter.
- The requirements on the antennas needed for the ever-expanding networks are becoming continually

higher. An antenna must have the following features:

- Strictly defined radiation patterns for a most accurate network planning.
- Growing concern for the level of intermodulation due to the radiation of many HF- carriers via one antenna.
- Dua1 polarization.
- Electrical down-tilting of the vertical diagram.
- Unobtrusive design.

Types of antennas:

1. **Omnidirectional Antenna**

   An omnidirectional antenna is an antenna that has a non-directional pattern (circular pattern) in a given plane with a directional pattern in any orthogonal plane. Examples of omnidirectional antennas are *dipoles* and *collinear antennas*.

2. **Dipole Antennas**

   A dipole antenna most commonly refers to a half-wavelength (L/2) dipole. The physical antenna is constructed of conductive elements whose combined length is about half of a wavelength at its intended frequency of operation. This is a simple antenna that radiates its energy out toward the horizon.

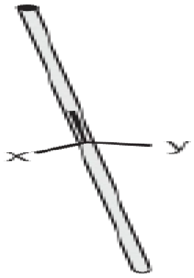


Fig 4.24: Dipole antenna

3. Collinear Omni Antennas:

In order to create an omnidirectional antenna with higher gain, multiple omni directional structures (either wires or elements on a circuit board) can be arranged in a vertical, linear fashion to retain the same omnidirectional pattern in the azimuth plane but a more focused elevation plane beam which then has higher gain. This is frequently referred to as a collinear array.

4. Directional Antennas

- A directional antenna is one that radiates its energy more effectively in one (or some) direction than others. Typically, these antennas have one main lobe and several minor lobes. Examples of directional antennas are *patches* and *dishes*.
- Directional antennas are used for coverage as well as point-to-point links. They can be patch antennas, dishes, horns or a whole host of other varieties. They all accomplish the same goal: radiating their energy out in a particular direction.

5. Patch Antennas

- A patch antenna, in its simplest form, is just a single rectangular (or circular) conductive plate that is spaced above a ground plane.
- Patch antennas are attractive due to their low profile and ease of fabrication. The radiation pattern of a single patch is characterized by a single main lobe of moderate beamwidth.
- Frequently, the beamwidths in the azimuth and elevation planes are similar, resulting in a fairly circular beam, although this is by no means universal.
- The beamwidths can be manipulated to produce an antenna with higher or lower gain, depending on the requirements. An antenna built with a single patch will have a maximum gain of about 9 dBi or a bit less.
- The patch antenna in Fig. 4.25 shows how simple these antennas can be. This is a simple rectangular patch built over a rectangular ground plane.

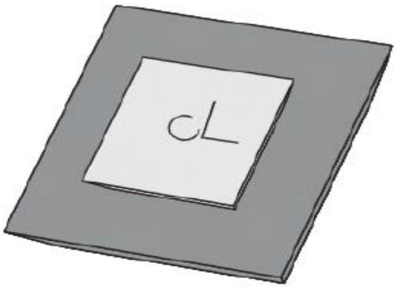


Fig 4.25: Patch antenna

6. Patch Array Antennas:

- A patch array antenna is, in general, some arrangement of multiple patch antennas that are all driven by the same source. Frequently, this arrangement consists of patches arranged in orderly rows and columns (a rectangular array) as shown in Fig. 4.26.
- The reason for these types of arrangements is higher gain. Higher gain commonly implies a narrower beamwidth and that is, indeed, the case with patch arrays. The array shown here has a gain of about 18 dBi with an azimuth and elevation plane beamwidth of about 20 degrees.

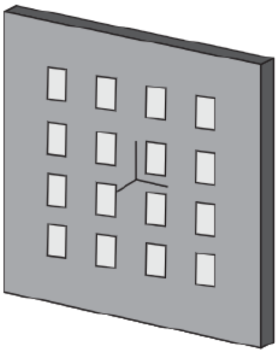


Fig 4.26: 4x4 patch array antenna

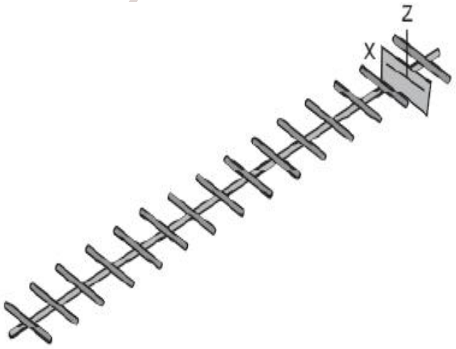


Fig 4.27: Yagi-Antenna model

7. Yagi Antennas

- A Yagi antenna is formed by driving a simple antenna, typically a dipole or dipole-like antenna, and shaping the beam using a well-chosen series of non-driven elements whose length and spacing are tightly controlled.
- The Yagi shown here in Fig. 4.27 is built with one reflector (the bar behind the driven antenna) and 14 directors (the bars in front of the driven antenna).
- Many times, these antennas are designed so that they can be rotated for either horizontal or vertical polarization, so having the same 3-dB beamwidth in each plane is a nice feature in those instances. Again, the Yagi antenna is a directional antenna that radiates its energy out in one main direction. Their directional nature seems to be somewhat intuitive due to their common, tubular form factor. It is easy to visualize aiming these antennas much like a rifle.

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DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGINEERING

SUBJECT: BASIC ELECTRONICS & COMMUNICATION

SUBJECT CODE--[21ELN14/24]

MODULE - 5

Syllabus:

Cellular Wireless Networks - Introduction, cellular telephone system, cellular concept and frequency reuse.

Wireless Network Topologies - First Generation (1G) Technology, Second Generation (2G) Technology,

GSM Communications, GSM System architecture, Third Generation (3G) Technology, CDMA Technology, High-level architecture of LTE, Fourth Generation (4G) Technology, Wireless LAN, Bluetooth, Bluetooth Architecture.

Satellite Communication – Elements of Satellite Communication, Types of satellites – GEO, LEO, MEO.

Optical Fiber Communication - A fiber optic Communication system.

Microwave Communication – Introduction, Frequency modulated microwave communication system.

5.1 Introduction:

To provide wireless communication within a particular geographic region, an integrated network of base stations must be installed to provide sufficient radio coverage to all the mobile users. Also, base station must be connected to a central hub called the mobile switching center (MSC). The key principles of cellular telephone were provided in 1947 by the researchers at bell telephone laboratories and other telecom companies throughout the world. For cellular communications, it was determined that the large geographic area must be subdivided into small sections, called cells which use the concept of frequency reuse to increase the capacity of a wireless and mobile telephone channel.

5.2 Cellular Telephone System:

As shown in Fig 5.1 a cellular system comprises the following basic components.

Mobile Station (MS): This is the mobile handset, which is used by a user to communicate with another user.

Cell: Each cellular service area is divided into small regions called cell (5 to 20 Km).

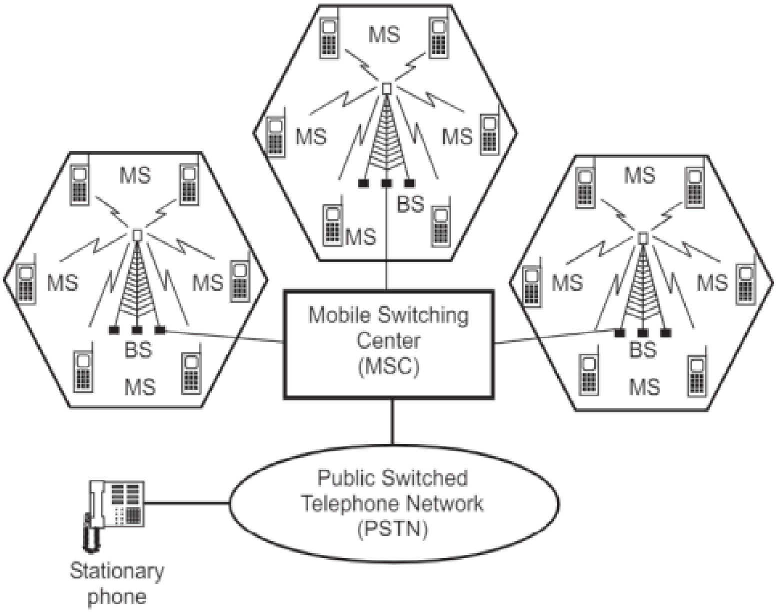

Fig 5.1 Schematic diagram of telephone system

Base Station (BS): Each cell contains an antenna, which is controlled by a small office.

Mobile Switching Center (MSC): Each base station is controlled by a switching office, called mobile switching center.

5.3 Cellular Concept and Frequency Reuse:

5.3.1 Cellular Concept:

- In the early phase, mobile radio system normally used a high-power transmitter with an antenna mounted on a tall tower. This approach gave very good coverage, but it was very difficult to reuse these same frequencies, therefore that the network capacity is low.

- As the demand for mobile service increased, achieving high network capacity by the same radio spectrum was more important than covering large areas:
- In order to solve the capacity problem, the cellular concept was proposed in 1970. The fundamental principle of the cellular concept is to divide the coverage area into a number of smaller areas which are served by their own radio base station.
- Radio channels are allocated to these smaller areas in an intelligent way so as to minimize the interference and improve the performance, and cater to the traffic loads in these areas called *cells.*
- The groups of cells in smaller areas are known as *clusters.* As the population grows, cells can be added to accommodate that growth Frequencies used in one cell cluster can be reused in other cells.
- The cellular concept employs variable low power transmitters, which allow cells to be sized according to the subscriber density and demand of a given area.
- Frequencies used in one cell cluster can be reused in other cells. Small cells will increase the network capacity, but on the other hand will increase the co-channel interference (CCI), therefore affect the quality of service (QoS).
- In order to achieve high capacity while satisfying quality of service expectations, a cellular architecture must be defined so as to be flexible to accommodate system growth.

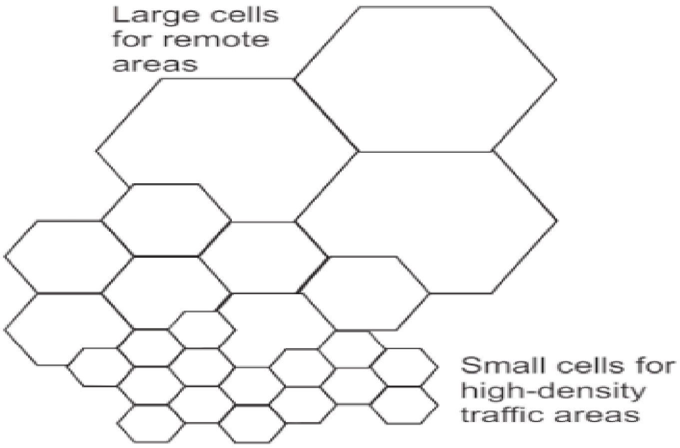

Fig 5.2 Cellular concept in wireless and mobile networks.

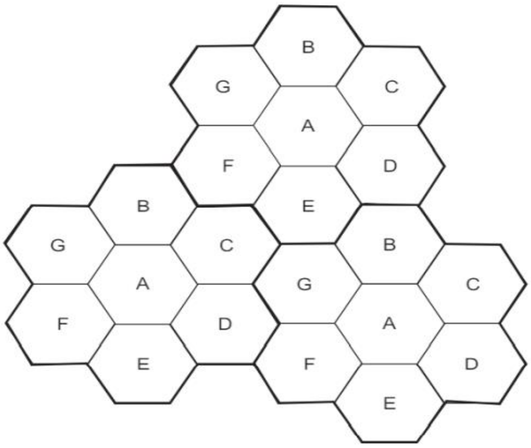

Fig 5.3 Concept of frequency reuse

5.3.2 Frequency Reuse

- Frequency reuse is the core concept of the cellular mobile radio system. The total available channels are divided into a number of channel sets (Actually, frequency reuse pattern is equally to the number of channel sets). Each channel set is assigned to a cell.
- Cells are assigned a group of channels that is completely different from neighboring cell.
- Figure 5.3 illustrates the concept of frequency reuse. Cells with the same alphabet use the same channel set. The same set of channels can be reused in another cell provided that the reuse distance D is fulfilled. The reuse distance is the minimum separation of identical channels that have the same carrier frequency, at which there is acceptable interference:

$$D = \sqrt{3N}\, R$$

Where N is the number of channel sets (cells in a cluster) (in Fig. N=7), R is the radius of a cell.

Reduction of Interference:

- Reusing an identical frequency channel in different cells is limited by co-channel interference between cells, and the co-channel interference can become a major problem.
- One way to reduce co-channel interference (CCI) is to keep the separation between two co-channel cells by a sufficient distance. Another way for controlling CCI is to use directional antennas at the base station (BS), and we call it cell sectoring.
- The adjacent channel interference, coming from neighboring channels and next channels, is another consideration in channel assignment.
- The adjacent channel interference depends on the separation of two adjacent channels, the characteristic of receiver filters, and the distance of two adjacent channel users.
- The near-end ratio interference can occur among the neighboring channels. Therefore, if one channel is assigned to a cell; its adjacent channels cannot be assigned to the same cell and vice versa.

Transmitting and Receiving:

Basic operations of transmitting and receiving in a cellular telephone network are discussed in this section.

Transmitting involves the following steps:

- A caller enters a 10-digit code (phone number) and presses the send button.
- The MS scans the band to select a free channel and sends a strong signal to send the number entered.
- The BS relays the number to the MSC.
- The MSC in turn dispatches the request to all the base stations in the cellular system.
- The Mobile Identification Number (MIN) is then broadcast over all the forward control channels throughout the cellular system. It is known as paging.
- The MS responds by identifying itself over the reverse control channel.
- The BS relays the acknowledgement sent by the mobile and informs the MSC about the handshake.
- The MSC assigns an unused voice channel to the call and call is established.

Receiving involves the following steps:

- All the idle mobile stations continuously listen to the paging signal to detect messages directed at them.
- When a call is placed to a mobile station, a packet is sent to the callees home MSC to find out where it is
- A packet is sent to the base station in its current cell, which then sends a broadcast on the paging channel.
- The called MS responds on the control channel.
- In response, a voice channel is assigned and ringing starts at the MS.

Mobility Management:

- A MS is assigned a home network, commonly known as location area. When an MS migrates out of its current BS into the footprint of another, a procedure is performed to maintain service continuity, known as Handoff management.
- An agent in the home network, called home agent, keeps track of the current location of the MS. The procedure to keep track of the user’s current location is referred to as Location management.
- Handoff management and location management together are referred to as Mobility management.

Handoff:

- At any instant, each mobile station is logically in a cell and under the control of the cells base station.
- When a mobile station moves out of a cell, the base station notices the MSs signal fading away and requests all the neighboring BSS to report the strength they are receiving.
- The BS then transfers ownership to the cell getting the strongest signal and the MSC changes the channel carrying the call. The process is called handoff.
- There are two types of handoffs, hard handoff and soft handoff.
- In a hard handoff, which was used in the early systems, a MS communicates with one BS As a MS moves from cell A to cell B. the communication between the AS and base station of cell A is first broken before communication is started between the MS and the base station of B. As a consequence, the transition is not smooth. For smooth transition from one cell (say A) to another (say B), an MS continues to talk to both A and B.
- As the MS moves from cell A to cell B, at some point the communication broken with the old base station of cell A. This is known as soft handoff.

Roaming:

- Two fundamental operations are associated with location management; location update and paging.
- When a Mobile Station (MS) enters a new Location Area, it performs a location updating procedure by making an association between the foreign agent and the home agent.
- One of the BSs, in the newly visited Location Area is informed and the home directory of the MS is updated with its current location.
- When the home agent receives a message destined for the MS, it forwards the message to the MS via the foreign agent. An authentication process is performed before forwarding the message.

5.4 Wireless Network Topologies:

Wireless network topology is defined as the configuration in which a mobile terminal (MT) communicates with other mobile terminals. Basically, there are two types of topologies used in wireless networks as follows:

1. Ad-Hoc Network Topology:

- Ad-hoc wireless networks do not need any infrastructure to work Each node can communicate directly with other nodes, so no base station is necessary.
- These networks are primarily used by military and also in a few commercial applications for voice and data transmission: This topology is suitable for rapid deployment of a wireless network

in a mobile or fixed environment.

- Fig 5.4 shows a single-hop, ad-hoc network where every user terminal has the functional capability of communicating directly with any other user terminal Nodes with an ad-hoc network can only communicate if they can reach each other physically, i.e., if they are within each other's radio range or if other nodes can forward the message.

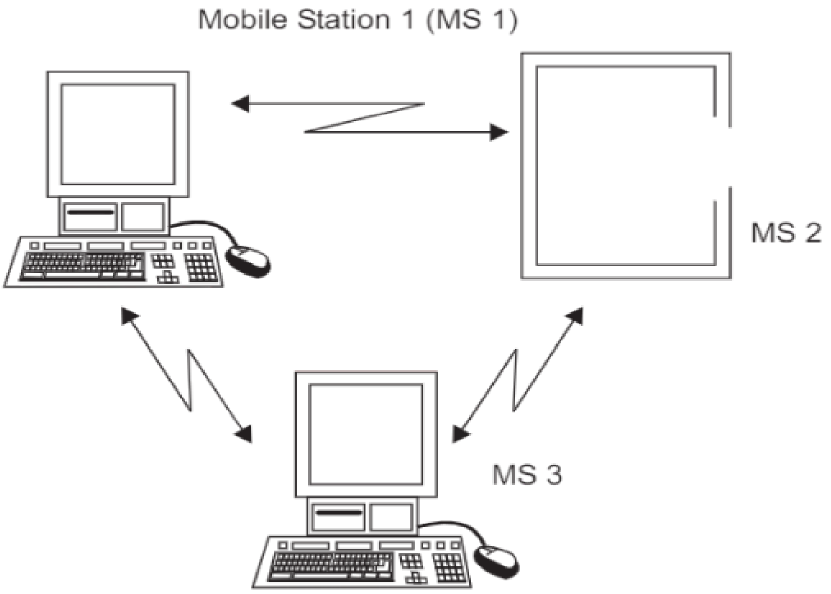

Fig 5.4 Single hop ad-hoc network

- In some ad-hoc networking applications, where users may be distributed over a wide area. a given user terminal may be able to reach only a portion of the other users in the network due to the transmitter signal power limitations.
- In this situation, user terminals will have to cooperate in carrying messages across the network between widely separated stations. These networks are called multi hop ad-hoc networks as shown in Fig 5.5.

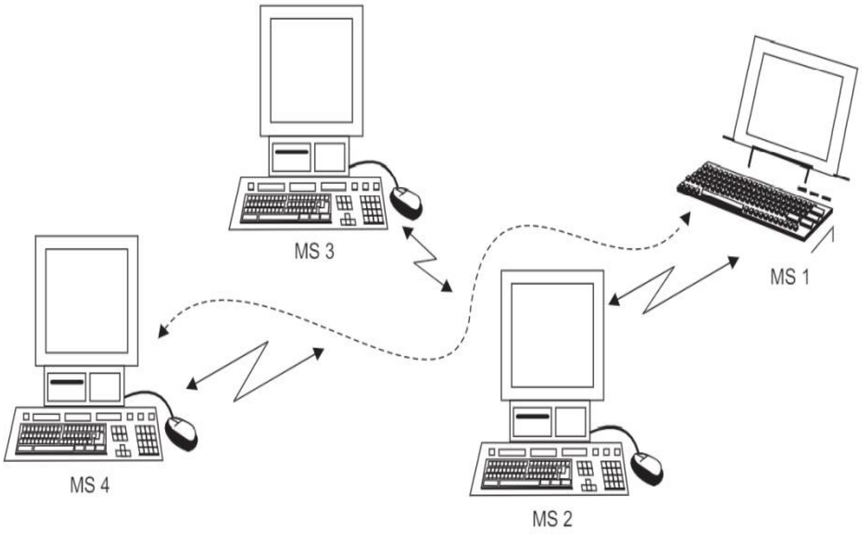

Fig 5.5 Multi hop ad-hoc network.

2. Infrastructure Network Topology:

- In this topology, there is a fixed infrastructure that supports the communication between the mobile terminals and between mobile and fixed terminals.
- This topology is often designed for large coverage areas and multiple base stations (BS) or access point (AP) operations. Figure 5.6 shows the basic operation of an infrastructure network with a single base station.
- Base station (BS) serves as the hub of the network and mobile terminals are located at different positions at the ends of spokes.

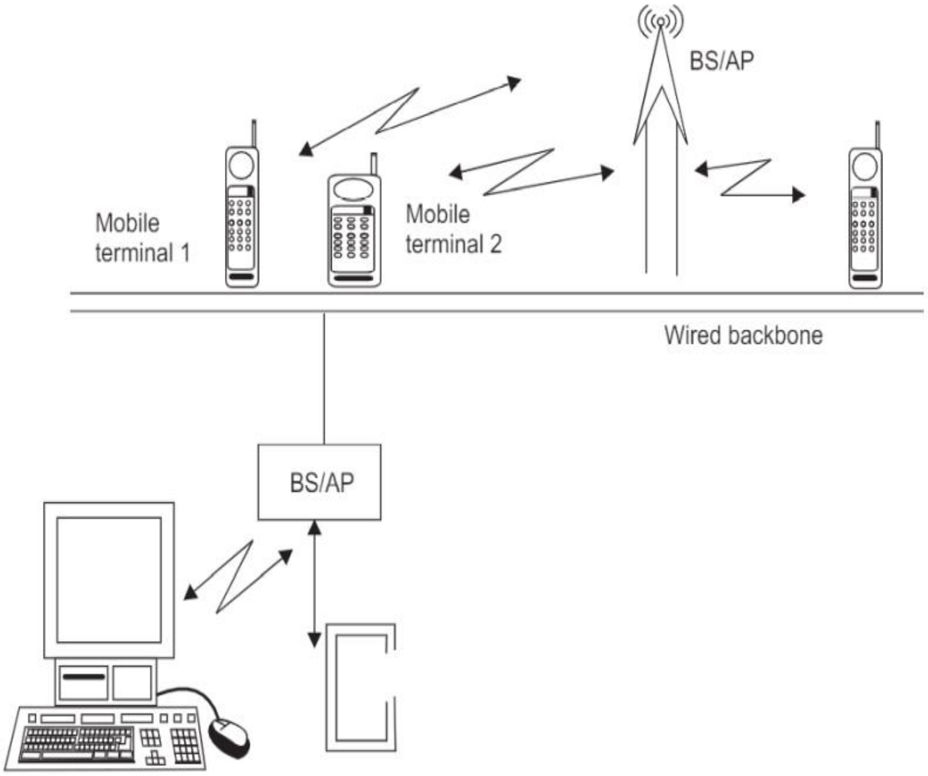

Fig 5.6 Infrastructure network topology.

- Any communication from one wireless station user to another comes through the base station. Thus, we can say that the hub is involved in managing the user access to the network.

FIRST GENERATION (1G) TECHNOLOGY:

- 1G stands for "first generation," refers to the first generation of wireless telecommunication technology, more popularly known as cell phones.
- A set of wireless standards developed in the 1980's, IG technology replaced 0G technology, which featured mobile radio telephones and such technologies as Mobile Telephone System (MTS), Advanced Mobile Telephone System (AMTS), Improved Mobile Telephone Service (IMTS), and Push to Talk (PTT). Its successor, 2G, which made use of digital signals, IG wireless networks used analog radio signals.

- Through 1G, a voice call gets modulated to a higher frequency of about 150 MHz and up as it is transmitted between radio towers. This is done using a technique called Frequency-Division Multiple Access (FDMA).
- In terms of overall connection quality, IG compares unfavorably to its successors. It has low capacity, unreliable handoff, poor voice links, and no security at all since voice calls were played back in radio towers, making these calls susceptible to unwanted eavesdropping by third parties.
- The first generation was designed for voice communication. One example is Advanced Mobile Phone System (AMPS) used in North America.
- AMPS is an analog cellular phone system. It uses 800 MHz ISM band and two separate analog channels; forward and reverse analog channels.

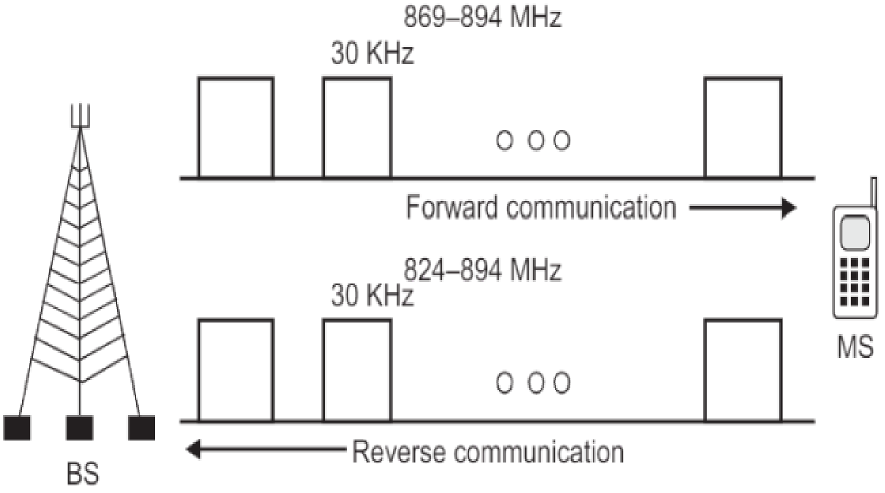

Fig 5.7 frequency bands used in AMPS system.

- The band between 824 to 849 MHz is used for reverse communication from MS to BS. The band between 869 to 894 MHz is used for forward communication from BS to MS. Each band is divided in to 832 30-KHz channels as shown in Fig 5.7.
- As each location area is shared by two service providers, each provider can have 416 channels, out of which 21 are used for control. AMPS use Frequency Division Multiple Access (FDMA) to divide each 25-MHz band into 30-KHz channels as shown in Fig 5.8.

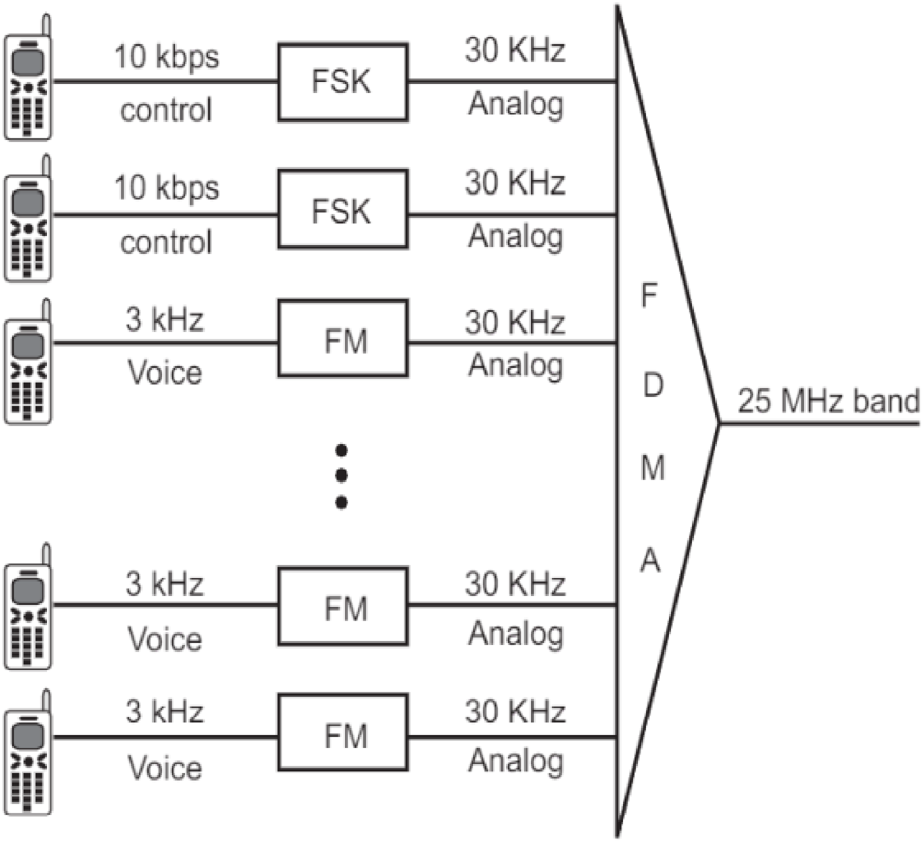

Fig 5.8 FDMA medium access control technique used in AMPS.

SECOND GENERATION (2G) TECHNOLOGY:

- 2G is short for "second-generation" wireless telephone technology. It cannot normally transfer data, such as email or software, other than the digital voice call itself, and other basic ancillary data such as time and date.
- The first-generation cellular network was developed for analog voice communication.
- To provide better voice quality, the second generation was developed for digitized voice communication. Nevertheless, SMS messaging is also available as a form of data transmission
- for some standards.
- Second generation 2G cellular telecom networks were commercially launched on the GSM standard in Finland in 1991. GSM service is used by over 2 billion people across more than 212 countries and territories. The ubiquity of the GSM standard makes international roaming very common between mobile phone operators, enabling subscribers to use their phones in many parts of the world.
- 2G technologies can be divided into Time Division Multiple Access (TDMA) based and Code Division Multiple Access (CDMA) based standards depending on the type of multiplexing used.
- 2G makes use of a CODEC (Compression-Decompression Algorithm) to compress and multiplex digital voice data. Through this technology, a 2G network can pack more calls per

amount of bandwidth as a 1G network. 2G cell phone units were generally smaller than IG units, since they emitted less radio power.

- Some benefits of 2G where Digital signals require consume less battery power, so it helps mobile batteries to last long, Digital coding improves the voice clarity and reduces noise in the line. Digital signals are considered environment friendly. The use of digital data service assists mobile network operators to introduce short message service over the cellular phones.
- Digital encryption has provided secrecy and safety to the data and voice calls. The use of 2G technology requires strong digital signals to help mobile phones work. If there is no network coverage in any specific area, digital signals would be weak.
- Three major systems were evolved as follows:
  - IS-136 (D-AMPS)
  - IS-95 (CDMA)
  - Global System for Mobile (GSM)

D-AMPS:

- D-AMPS is essentially a digital version of AMPS and it is backward compatible with AMPS.
- It uses the same bands and channels and uses the frequency reuse factor of 1/725 frames per second each of 1994 hits, divided in 6 slots shared by three channels.
- Each slot has 324 bits-159 data, 64 control, 101 error correction as shown in Fig 5.9.
- As shown in the figure it uses both IDMA and IDMA medium access control techniques.

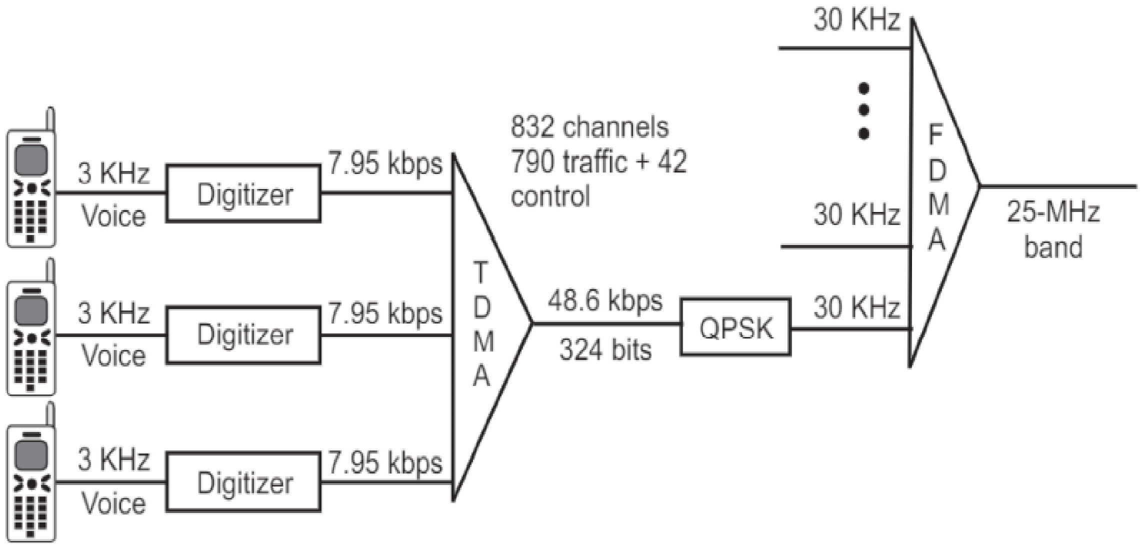

Fig 5.9 D-AMPS

- IS-95 (CDMA): IS-95 CDMA: IS-95 is based on CDMA/DS-SS and FDMA medium access control techniques.
- The forward and backward transmissions are shown in Fig 5.9.

GSM:

- The Global System for Mobile (GSM) communication is a European standard developed to replace the first-generation technology. Uses two bands for duplex communication.
- Each voice channel is digitized and compressed to a 13Kbps digital signal. Each slot carries 156.25 bits, 8 slots are multiplexed together creating a FDM frame, 26 frames are combined to form a multi frame, as shown in Fig 5.10.

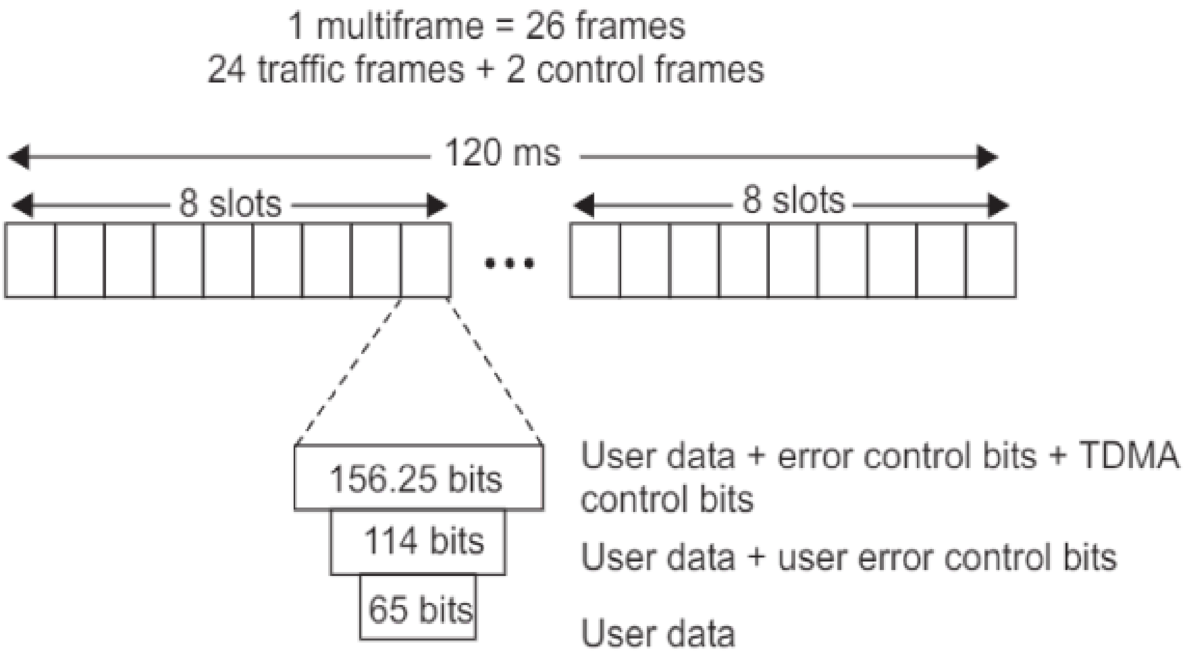

Fig 5.10 Multi frame Components

- For medium access control, GSM combines both TDMA and FDMA. There is large amount of overhead in TDMA; 114 bits are generated by adding extra bits for error correction. Because of complex error correction, it allows a reuse factor as low as 1/3.

Global System for Mobile (GSM) Communications:

- The first GSM system developed was GSM-900 (Phase 1). Now, GSM is known as Global Systems for Mobile Communications Phase I operates in 900 MHz band for voice only.
- Phases 2 introduced in 1993 which included facsimile, video and data communication services.
- The system architecture of GSM is shown in the Fig 5.11. It consists of three major subsystems that interact with each other and with the subscribers through specified network interfaces. The three subsystems are as follows:
  - Mobile station (MS)
  - Base station subsystem (BSS)
  - Network and switching subsystem (NSS)

Mobile Station (MS):

- The MS consists of the physical equipment used by the subscriber to access a mobile network for offered telecommunication services.
- Functionally, the MS includes a Mobile Terminal (MT) and, depending on the services it can support, various Terminal Equipment (TE), and combinations of TE and Terminal Adaptor (TA) functions (the TA acts as a gateway between the TE and the MT).
- Various types of MS, such as the vehicle mounted station, portable station, or handheld station, are used.

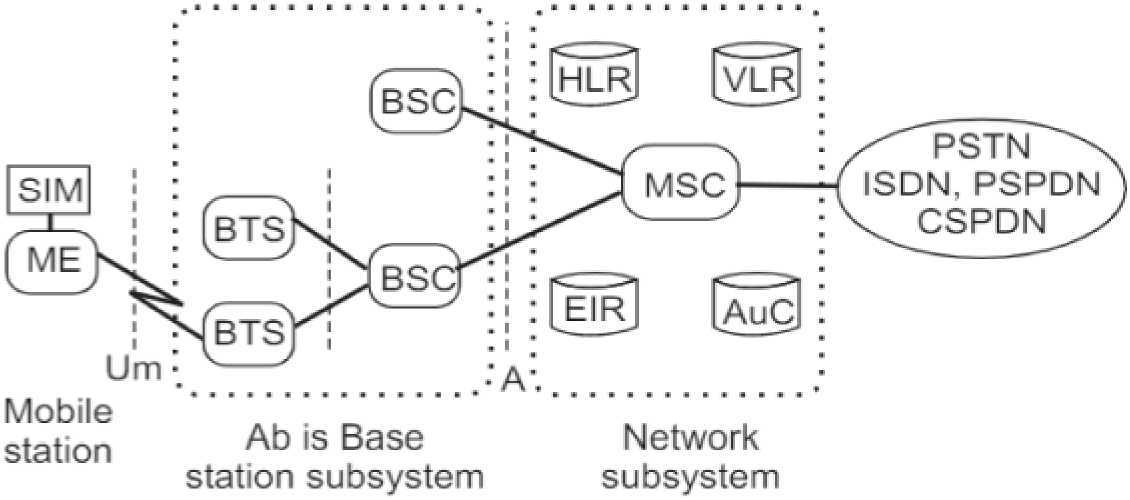

Fig 5.11 GSM architecture.

- Basically, a MS can be divided into two parts. The first part is the Mobile equipment (ME) which contains the hardware and software to support radio and human interface functions.
- The second part contains terminal/user specific data in the form of a smart card known as Subscriber Identity Module (SIM), which can effectively be considered a sort of logical terminal.
- The SIM card plugs into the first part of the MS and remains in for the duration of use. Without the SIM card, the MS is not associated with any user and cannot make or receive calls (except possibly an emergency call if the network allows).
- The SIM card is issued by the mobile service provider after subscription, while the first part of the MS would be available at retail shops to buy or rent. This type of SIM card mobility is analogous to terminal mobility, but provides a personal-mobility-like service within the GSM mobile network.
- An MS has a number of identities including the International Mobile Equipment Identity (IMEI), the International Mobile Subscriber Identity (IMSI), and the ISDN number. The IMSI is stored in the SIM. The SIM card contains all the subscriber-related information stored on the user's side of the radio interface.

Base Station Subsystem (BSS):

- The BSS is the physical equipment that provides radio coverage to prescribed geographical areas, known as the cells. It contains equipment required to communicate with the MS.
- Functionally, a BSS consists of a control function carried out by the base station controller (BSC) and a transmitting function performed by the BTS.
- The BTS is the radio transmission equipment and covers each cell. A BSS can serve several cells because it can have multiple BTSs.
- The BTS contains the Transcoder Rate Adapter Unit (TRAU), In TRAU, the GSM-specific speech encoding and decoding is carried out, as well as the rate adaptation function for data. In certain situations, the TRAU is located at the MSC to gain an advantage of more compressed transmission between the BTS and the MSC

Network and Switching Subsystem (NSS):

- The NSS includes the main switching functions of GSM, databases required for the subscribers, and mobility management.
- Its main role is to manage the communications between GSM and other network users. Within the NSS, the switching functions are performed by the MSC Subscriber information relevant to provisioning of services is kept in the home location register (HLR).
- The other database in the NSS is the visitor location register (VLR). The MSC performs the necessary switching functions required for the MSs located in an associated geographical area, called an MSC area.
- The MSC monitors the mobility of its subscribers and manages necessary resources required to handle and update the location registration procedures and to carry out the handover functions. The MSC is involved in the interworking functions to communicate with other networks such as Public Switched Telephone Network (PSTN) and ISDN.
- The interworking functions of the MSC depend upon the type of the network to which it is connected and the type of service to be performed. The call routing and control and echo control functions are also performed by the MSC.

Home location registers (HLR):

- The HLR is the functional unit used for management of mobile subscribers.
- The number of HLRs in a PLMN varies with the characteristics of the PLMN.
- Two types of information are stored in the HLR: subscriber information and part of the mobile information to allow incoming calls to be routed to the MSC for the particular MS.
- Any administrative action by the service provider on subscriber data is performed in the HLR. The HLR stores IMSI, MS ISDN number, VLR address, and subscriber data.

Visitor Location Register (VLR):

- The VLR is linked to one or more MSCs. The VLR is the functional unit that dynamically stores subscriber information when the subscriber is located in the area covered by the VIR.
- When a roaming MS enters an MSC area, the MSC informs the associated VLR about the MS; the MS goes through a registration procedure.
- The registration procedure for the MS includes these activities:
    - The VLR recognizes that the MS is from another MN.
    - If roaming is allowed, the VLR finds the MS's HLR in its home MN.
    - The VLR constructs a Global Title (GT) from the IMSI to allow signaling from the VLR to the MSS HLR via the PSTN/ISDN networks.
    - The VLR generates a Mobile Subscriber Roaming Number (MSRN) that is used to route incoming calls to the MS.
    - The MSRN is sent to the MS's HLR.
- The information in the VLR includes MSRN, TMSI, the location area in which the MS has been registered, data related to supplementary service. MS ISDN number, IMSI, HER address or GT, and local MS identity, if used.
- The NSS contains more than MSCs, HLRs, and VLRs In order to deliver an incoming call to a GSM user, the call is first routed to a gateway switch, referred to as the Gateway Mobile Service Switching Center (GMSC).
- The GMSC is responsible for collecting the location information and routing the call to the MSC through which the subscriber can obtain service at that instant (i.e., the visited MSC).
- The GMSC first finds the right HER from the directory number of the CSM subscriber and interrogates it.
- The GMSC has an interface with external networks for which it provides gateway fiction, as well as with the 57 signaling network for interworking with other NSS entities.

Operation and Maintenance Subsystem (OMSS):

- The OMSS is responsible for handling system security based on validation of identities of various telecommunications entities. These functions are performed in the Authentication Center (AuC) and Equipment identity register (EIR).
- The AuC is accessed by the HLR to determine whether an MS will be granted service. The EIR provides MS information used by the MSC. The EIR maintains a list of legitimate, fraudulent, or faulty MSS.
- The OMSS is also in charge of remote operation and maintenance functions of the MN. These functions are monitored and controlled in the OMSS.

- The OMSS may have one or more Network Management Centers (NMCs) to centralize MN control.
- The Operational and Maintenance Center (OMC) is the functional entity through which the service provider monitors and controls the system. The OMC provides a single point for the maintenance personnel to maintain the entire system. One OMC can serve multiple MSCs.

THIRD GENERATION (3G) TECHNOLOGY:

- The third-generation systems support high speed packet switched data (up to 2Mbps).
- In fact, GPRS is considered to be a transition step from second generation cellular systems to third generation cellular systems.
- The 3G systems are accepted world-wide and the subscriber is able to get the mobile services from anywhere in the world without replacing his handset or SIM card.
- The subscriber also gets the same environment and services in the visiting network as in his home network also being independent of the terminal. Apart from this, the modern generation cellular systems provide with the framework to build various kind of services (like VPN and conferencing) on the top of core cellular networks.
- Currently the 3G cellular systems are being evolved from the existing cellular networks. Despite the efforts of standardization, UMTS (Universal Mobile Telecommunication System) and CDMA-2000 are the two main 3G networks which are being used. Both these systems use CDMA technology.
- The UMTS system is being promoted by ETSI (European Telecommunication Standards Institute) and is a successor of GSM while CDMA 2000 is successor of IS-95.

CDMA TECHNOLOGY

- CDMA offers several advantages over FDMA and TDMA. Error control coding, spreading of the spectrum, soft handoffs and strict power control are some of those advantages.
- CDMA is primarily an air-interface and access technique that is based on direct sequence-spread spectrum (DS-SS).
- The air interface is significantly different in the case of CDMA compared with TDMA technique.
- The core fixed network infrastructure of CDMA supports the wireless interface is very similar to the structure of the GSM core network.
- After 2000, third generation (3G) systems are being standardized all over the world currently by International Telecommunication Union (ITU) under the banner of International Mobile Telecommunications beyond 2000 (IMT-2000).
- Both IS-136 and IS-95 use CDMA as the air interface and the access method.

- In CDMA, all user data, the control channel and signaling information are transmitted on the same frequency at the same time. Also, CDMA employs powerful error control codes.
- The quality of voice is also improved and the multipath and fading problems are also reduced in CDMA technology.

Capacity of a Mobile Telecommunication System:

- In 1948, Claude Shannon discovered a theoretical limit on the data rate that can be achieved from any communication system. We will write it in its simplest form, as follows:

$$C = B \log_2 (1 + SINR)$$

- Here, SINR is the signal to interference plus noise ratio, in other words the power at the receiver due to the required signal, divided by the power due to noise and interference. B is the bandwidth of the communication system in Hz, and C is the channel capacity in bits.
- It is theoretically possible for a communication system to send data from a transmitter to a receiver without any errors at all, provided that the data rate is less than the channel capacity.
- In a mobile communication system, C is the maximum data rate that one cell can handle and equals the combined data rate of all the mobiles in the cell.

FROM UNIVERSAL MOBILE TELECOMMUNICATION SYSTEM (UMTS) TO LONG-TERM EVOLUTION (LTE):

High Level Architecture of Long-Term Evolution (LTE):

- In 2004, 3GPP began a study into the long-term evolution of UMTS. The aim was to keep 3GPP's mobile communication systems competitive over timescales of 10 years and beyond, by delivering the high data rates and low latencies those future users would require.
- Fig 5.12 shows the resulting architecture and the way in which that architecture developed from that of UMTS.
- In the new architecture, the evolved packet core (EPC) is a direct replacement for the packet switched domain of UMTS and GSM. It distributes all types of information to the user, voice as well as data, using the packet switching technologies that have traditionally been used for data alone.
- The evolved UMTS terrestrial radio access network (E-UTRAN) handles the EPC's radio communications with the mobile, so is a direct replacement for the UTRAN mobile is still known as the user equipment, though its internal operation is very different from before.

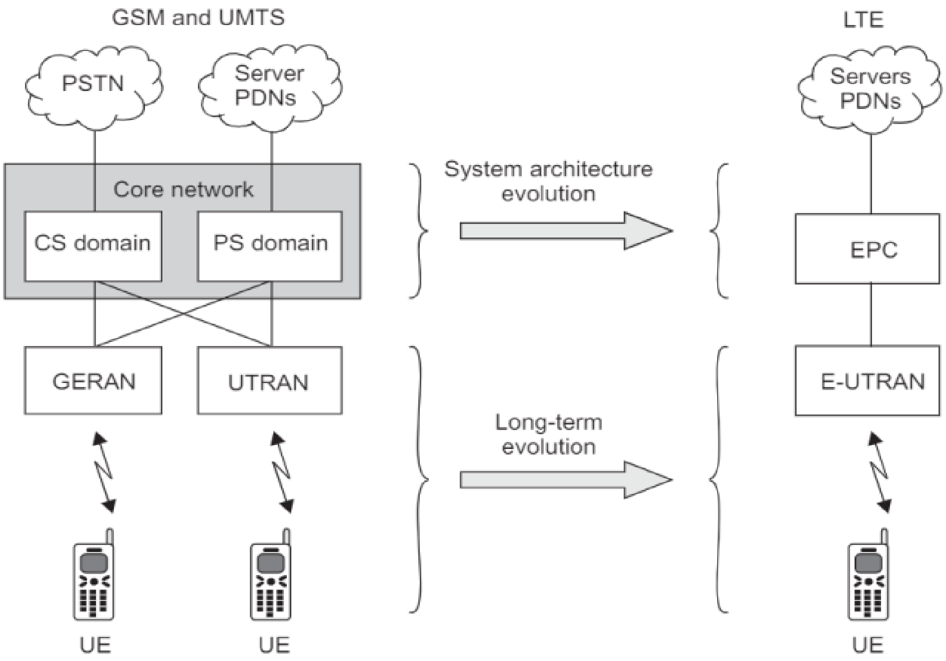

Fig 5.12 Evolution of the system architecture from GSM and UMTS to LTE

- The new architecture was designed as part of two 3GPP work items, namely system architecture evolution (SAE), which covered the core network, and long-term evolution (LTE), which covered the radio access network, air interface and mobile.
- Officially, the whole system is known as the evolved packet system (EPS), while the acronym LTE refers only to the evolution of the air interface. Despite this official usage, LTE has become a colloquial name for the whole system, and is regularly used in this way by 3GPP. We will use LTE in this colloquial way.

FOURTH GENERATION (4G) TECHNOLOGY

- 4G is short for "fourth-generation" wireless telephone technology.
- It is the latest technology which is started to be used in many countries. LTE or Long-Term Evolution is the brand name given to the efforts of 3GPP 4th Generation technology development efforts mostly in Europe and UMB (Ultra-Mobile Broadband) is the brand name for similar efforts by 3GPP2 in North America.
- The High-Level requirements for a 4G technology were identified as:
  - Higher spectral efficiency.
  - Reduced cost per bit.
  - Increased service provisioning by lowering the cost and increasing efficiency.
  - Open interfaces as against closed technologies of the past.
  - Power consumption efficiency.
  - Scalable and flexible usage of frequency bands.

- The technical specifications approved by 3GPP for the LTE project include the use of Orthogonal Frequency Division Multiplexing (OFDM) and advanced antenna technologies such as MIMO (Multiple Input Multiple Output).
- It specifies downlink peak speeds of 326 Mbps and uplink peak speeds of 86Mbps, both in a 20 MHz bandwidth. It also mandates the roundtrip latency between the base station and handsets to 10-milliseconds.
- The LTE-advanced is now de facto 4G mobile communications system, and it is likely to remain so. It does not have a serious competitor in sight 3GPP2 has given up its UMB initiative, and mobile WIMAX (802.16m) has not been able to gain significant market share, though it fulfills the 4G criteria set by the ITU.
- In order to understand LTE-A, it is better to have a look at LTE first. LTE is a very different system from UMTS, as both its architecture and the technologies used are either new or greatly enhanced versions of the old entities.

LTE-A System Architecture:

- Figure 5.13 gives a high-level description of the LTE-A network architecture, Readers who are more familiar with 2G/3G networks may notice the simplicity of the LTE-A architecture.
- In the old GSM there were base transceiver stations (BTS) and base station controllers (BSC), and in UTRA networks, we have NodeBs and radio network controllers (RNC), and several different entities in the core network.

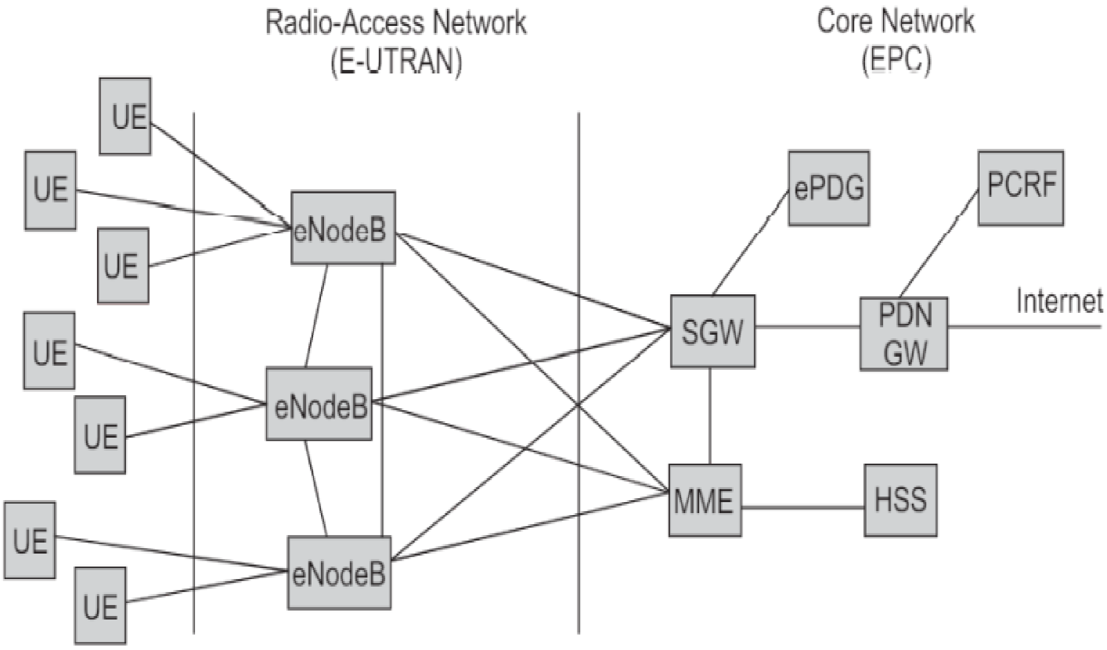

Fig 5.13 LTE system architecture

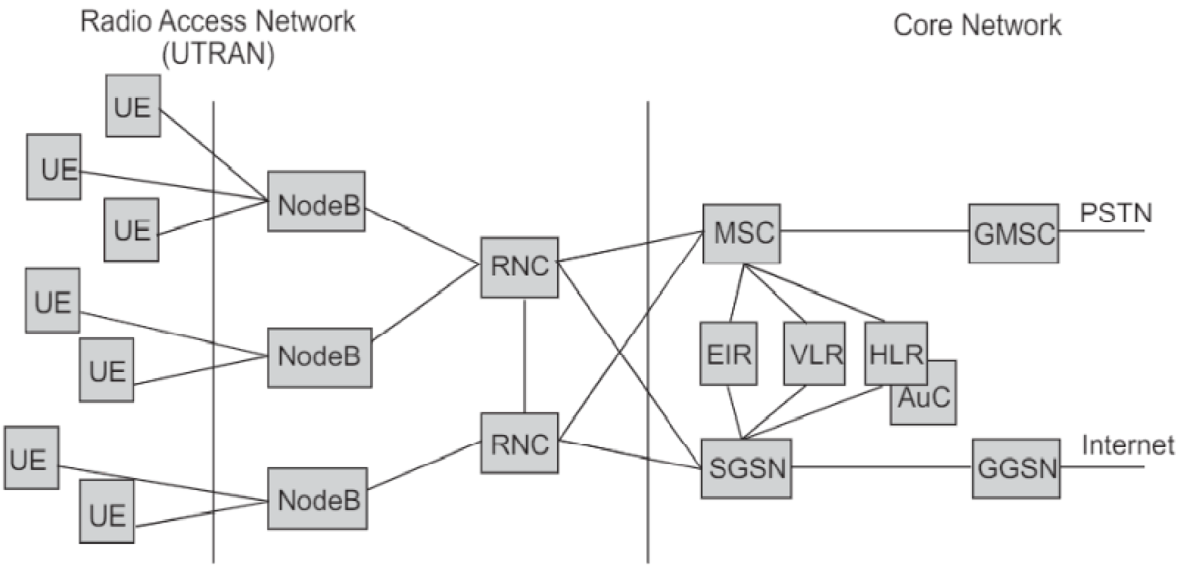

Fig 5.14 ULTRAN system architecture

WIRELESS LAN (WLAN)

- To know about WLAN, first we need to know the definition of LAN, which is simply a way of connecting computers together within a single organization, and usually in a single site.
- Wireless Local Area Network (WLAN) links two or more devices using a wireless communication method. It usually provides a connection through an Access Point (AP) to the wider internet. This gives users the ability to move around within a local coverage area while still be connected to the network. Just as the mobile phone frees people to make a phone call from anywhere in their home, a WLAN permits people to use their computers anywhere in the network area.
- In WLAN, connectivity no longer implies attachment. Local areas are measured not in feet or meters, but miles or kilometers. An infrastructure need not be buried in the ground or hidden behind the walls, so we can move and change it at the speed of the organization.
- The major standards for WLANs are IEEE 802.11 and HIPERLAN.

WLAN Specifications:

The IEEE 802.11 specifications were developed specifically for Wireless Local Area Networks (WLANS) by the IEEE and include four subsets of Ethernet-based protocol standards: 802.11. 802.11a, 802.11b, and 802.11g.

802.11

802.11 operated in the 2.4 GHz range and was the original specification of the 802.11 IEEE standard. This specification delivered 1 to 2 Mbps using a technology known as phase-shift keying (PSK) modulation. This specification is no longer used and has largely been replaced by other forms of the 802.11 standard.

802.11a

- 802.11a operates in the 5-6 GHz range with data rates commonly in the 6 Mbps, 12 Mbps, or 24 Mbps range. Because 802.11a uses the orthogonal frequency division multiplexing (OFDM) standard, data transfer rates can be as high as 54 Mbps.
- OFDM breaks up fast serial information signals into several slower sub-signals that are transferred at the same time via different frequencies, providing more resistance to radio frequency interference. The 802.11a specification is also known as Wi-F15, and though regionally deployed, it is not a global standard like 802.11b.

802.11b

- The 802.11b standard (also known as Wi-Fi) operates in the 2.4 GHz range with up to 11 Mbps data rates and is backward compatible with the 802.11 standard.
- 802.11b uses a technology known as complementary code keying (CCK) modulation, which allows for higher data rates with less chance of multi-path propagation interference.
- The overall benefits include:
    - Up to twice the data rate of conventional 11 Mbps 802.11b standard products.
    - Greater WLAN coverage.
    - Improved security over standard 802.11b.

802.11g

- 802.11g is the most recent IEEE 802.11 draft standard and operates in the 2.4 GHz range with data rates as high as 54 Mbps over a limited distance.

Advantages of WLAN over Wired LAN:

- *Installation:* Wireless LANs are very easy to install. There is no requirement for wiring every workstation and every room. This case of installation makes wireless LANS inherently flexible. If a workstation must be moved, it can be done easily and without additional wiring, cable drops or reconfiguration of the network.
- *Portability:* If a company moves to a new location, the wireless system is much easier to move than ripping up all of the cables that a wired system would have snaked throughout the building. It provides a useful complement to radio-based systems, particularly for systems requiring low cost, light weight, moderate data rates, and only requiring short ranges.

BLUETOOTH

- Bluetooth is a standard used in links of radio of short scope, destined to replace wired connections between electronic devices like cellular telephones, Personal Digital Assistants (PDA), computers, and many other devices.

- Bluetooth technology can be used at home, in the office, in the car, etc. This technology allows to the user's instantaneous connections of voice and information between several devices in real time. The way of transmission used assures protection against interferences and safety in the sending of information.
- The Bluetooth is a small microchip that operates in a band of available frequency throughout the world. Communications can realize point to point and point multipoint.
- The standard Bluetooth operates in the band of 2,4 GHz. This band is available worldwide, however, the width of the band can differ in different countries.

Bluetooth Architecture - Piconets and Scatternets:

- Up to seven slaves can be active and served simultaneously by the master.
- If the master needs to communicate with more than seven devices, it can do so by first instructing active slave devices to switch to low-power park mode and then inviting other parked slaves to become active in the piconet.
- This juggling act can be repeated, which allows a master to serve a large number of slaves.

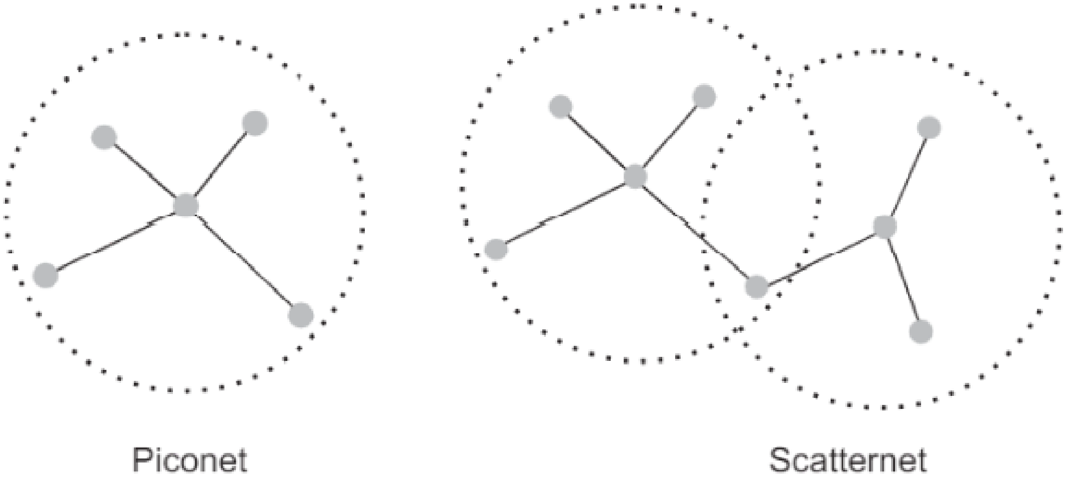

Fig 5.15 Illustration of the concept of piconet and scatternet in Bluetooth

- Most envisioned Bluetooth applications involve local communication among small groups of devices.
- A piconet configuration consisting of two, three, or up to eight devices is ideally suited to meet the communication needs of such applications.
- When many groups of devices need to be active simultaneously, each group can form a separate piconet. The slave nodes in each piconet stay synchronized with the master clock and hop according to a channel-hopping sequence that is a function of the master's node address.
- Since channel-hopping sequences are pseudorandom, the probability of collision among piconets is small. Piconets with overlapping coverage can coexist and operate

independently. Nonetheless, when the degree of overlap is high, the performance of each piconet starts to degrade. In some usage scenarios, however, devices in different piconets may need to communicate with each other.

- Bluetooth defines a structure called scatternet to facilitate interpiconet communication. A scatternet is formed by interconnecting multiple piconets. As shown on the right side of fig 5.15.

SATELLITE COMMUNICATION

ELEMENTS OF SATELLITE COMMUNICATION:

The basic elements of a satellite communication system are shown in the Fig.5.16. Basic elements are:

- **User:** The user generates the baseband signal that proceeds through a terrestrial network and transmitted to a satellite at the earth station.
- **Satellite:** The satellite consists of a large number of repeaters in the space that perform the reception of modulated RF carrier in its uplink frequency spectrum from all the earth stations in the present networks, amplifiers. They retransmit them back to the earth stations in the down link frequency spectrum. To avoid interference, downlink and uplink frequency spectrums should be separate and different.

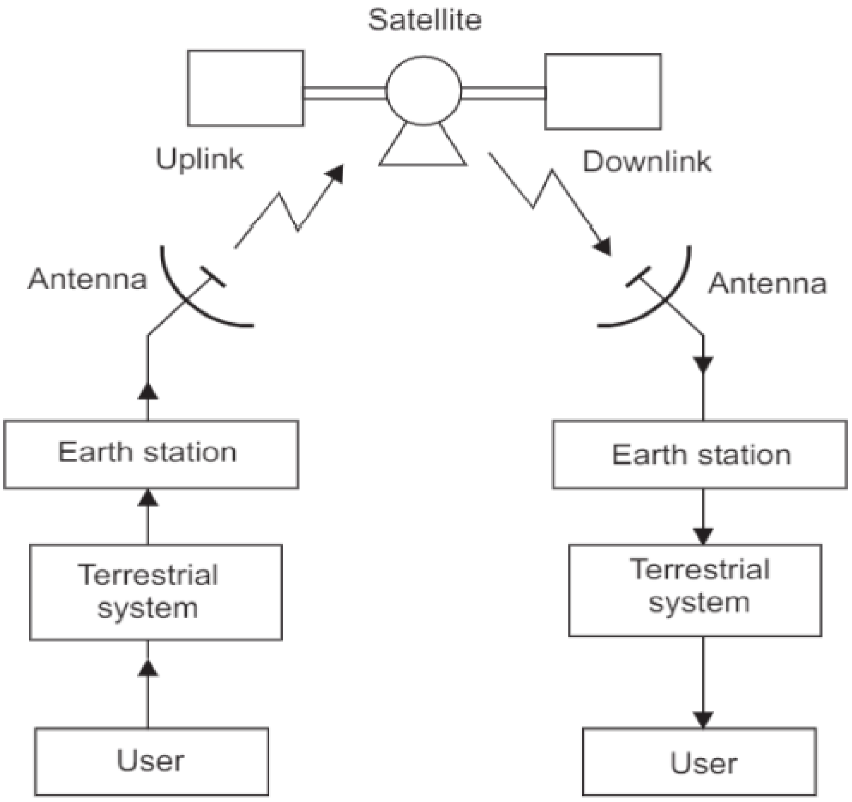

Fig 5.16 Basics elements of a satellite communication system

- **Terrestrial network:** This is a network on the ground which carries the signal from user to earth station. It can be a telephone switch or a dedicated link between the user and the earth station.
- **Earth Station:** It's a radio station located on the earth and used for relaying signals from satellites; It governs all the activities and transmissions happening in the satellite communication.

TYPES OF SATELLITES (BASED ON ORBITS):

Geostationary Earth Orbit (GEO) Satellites:

- GEO satellites are synchronous with respect to earth. Looking from a fixed point from Earth, these satellites appear to be stationary.
- These satellites are placed in the space in such a way that only three satellites are sufficient to provide connection throughout the surface of the Earth. The orbit of these satellites is circular.
- There are three conditions which lead to geostationary satellites. Lifetime expectancy of these satellites is 15 years. The satellite should be placed 37,786 kms (approximated to 36,000 kms) above the surface of the earth.
- These satellites must travel in the rotational speed of earth, and in the direction of motion of earth, that is eastward.
- The inclination of satellite with respect to earth must be 00.
- Geostationary satellite in practical is termed as geosynchronous as there are multiple factors which make these satellites shift from the ideal geostationary condition.
- Gravitational pull of sun and moon makes these satellites deviate from their orbit.
- Over the period of time, they go through a drag. (Earth's gravitational force has no effect on these satellites due to their distance from the surface of the Earth). These satellites experience the centrifugal force due to the rotation of Earth, making them deviate from their orbit.
- The non-circular shape of the earth leads to continuous adjustment of speed of satellite from the earth station.
- These satellites are used for TV and radio broadcast, weather forecast and also, these satellites are opening as backbones for the telephone networks.

Low Earth Orbit (LEO) Satellites:

- These satellites are placed 500-1500 kms above the surface of the earth.
- As LEO satellites circulate on a lower orbit, hence they exhibit a much shorter period that is 95 to 120 minutes.
- LEO system try to ensure a high elevation for every spot-on earth to provide a high quality communication link. Each LEO satellite will only be visible from the earth for around ten

minutes.

- Using advanced compression schemes, transmission rates of about 2,400 bit/s can be enough for voice communication.
- LEOs even provide this bandwidth for mobile terminals with Omni-directional antennas using low transmit power in the range of IW.
- The delay for packets delivered via a LEO is relatively low. Smaller footprints of LEOs allow for better frequency reuse, similar to the concepts used for cellular networks. LEOs can provide a much higher elevation in Polar Regions and better global coverage. These satellites are mainly used in remote sensing and mobile communication services (due to lower latency).

Disadvantages:

- The biggest problem of the LEO concept is the need for many satellites if global coverage is to be reached.
- Several concepts involve 50-200 or even more satellites in orbit. The high number of satellites combined with the fast movements resulting in a high complexity of the whole satellite system.
- The short time of visibility with a high elevation requires additional mechanisms for conviction handover between different satellites.
- One general problem of LFOs in the short lifetime of about five to eight years due to atmospheric drag and radiation.
- Other factors are the need for routing of data packets from satellite to if a user wants to communicate around the world.

Medium Earth Orbit (MEO) Satellites:

- MEO satellites can be positioned somewhere between LEOs and GEOS, both in terms of their orbit and due to their advantages and disadvantages. Using orbits around 10,000 km, the system only requires a dozen satellites which is more than a GEO system, but much less than a LEO system.
- These satellites move more slowly relative to the Earth's rotation allowing a simpler system design (satellite periods are about six hours). Depending on the inclination, a MEO can cover larger populations, so requiring fewer handovers.

Disadvantages:

Due to the larger distance to the earth, delay increases to about 70-80 ms.

- These satellites need higher transmit power and special antennas for smaller footprints.

OPTICAL FIBER COMMUNICATION

A Fiber-Optic Communication System:

A generalized configuration of a fiber-optic communication system is shown in brief description of each block in this figure 5.17 will give us an idea of the prime employed in this system.

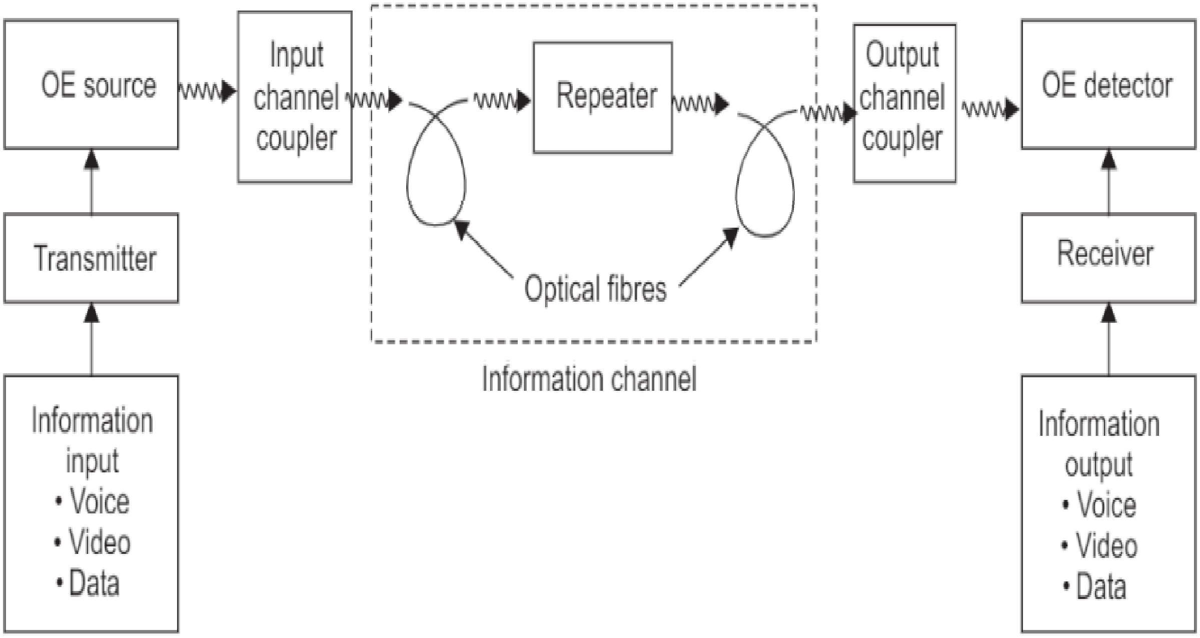

Fig 5.17 Generalized configuration of a fiber optic communication system.

Information Input:

- The information input may be in any of the several physical forms, e.g., voice, video, or data. Therefore, an input transducer is required for converting the non-electrical input into an electrical input.
- For example, a microphone converts a sound signal into an electrical current, a video camera converts an image into an electric current or voltage, and so on. In situations where the fiber-optic link forms a part of a larger system, the information input is normally in electrical form.
- Examples of this type include data transfer between different computers or that between different parts of the same computer. In either case, the information input must be in the electrical form for onward transmission through the fiber-optic link.

Transmitter:

- The transmitter for the modulator, as it is often called) comprises an electronic stage which in converts the electric signal into the proper form and (i) impresses this signal onto the electromagnetic wave (carrier) generated by the optoelectronic source
- The modulation of an optical carrier may be achieved by employing either an analog or a digital signal.

- An analog signal varies continuously and reproduces the form of the original information input, whereas digital modulation involves obtaining information in the discrete forms.
- In the latter the signal is either on or off, with the on state representing a digital 1 and the off state representing a digital 0. These are called binary digits for bits of the digital system. The number of bits per second (bps) transmitted is called the data rate. If the information input is in the analog form, it may be obtained in the digital forms by employing an analog-to-digital converter.
- Analog modulation is much simpler to implement but requires higher signal-to-noise ratio at the receiver end as compared to digital modulation. Further, the linearity needed for analog modulation is not always provided by the optical source, particularly at high modulation frequencies. Therefore, analog fiber-optic systems are limited to shorter distances and lower bandwidths.

Optoelectronic Source:

- An optoelectronic (OE) source generates an electromagnetic wave in the optical range (particularly the near-infrared part of the spectrum), which serves as an information carrier.
- Common sources for fiber-optic communication are the light-emitting diode (LED) and the injection laser diode (ILD). Ideally, an optoelectronic source should generate a stable single frequency electromagnetic wave with enough power for long-haul transmission.
- However, in practice, LEDs and even laser diodes emit a range of frequencies and limited power. The favorable properties of these sources are that they are compact, lightweight, consume moderate amounts of power, and are relatively easy to modulate. Furthermore, LEDs and laser diodes which emit frequencies that are less attenuated while propagating through optical fibers are available.

Channel Couplers:

- In the case of open channel transmission, for example, the radio or television broadcasting system, the channel coupler is an antenna. It collects the signal from the transmitter and directs this to the atmospheric channel. At the receiver end again the antenna collects the signal and routes it to the receiver. In the case of guided channel transmission, e.g., a telephone link, the coupler is simply a connector for attaching the transmitter to the cable:
- In fiber-optic systems, the function of a coupler is to collect the light signal from the optoelectronic source and send it efficiently to the optical fiber cable. Several designs are possible. However, the coupling losses are large owing to Fresnel reflection and limited light gathering capacity of such couplers. At the end of the link again a emipler is required to collect the signal and direct it onto the photodetector.

Fiber-optic Information Channel:

- In communication systems, the term 'information channel' refers to the path between the transmitter and the receiver.
- In fiber-optic systems, the optical signal traverses along the cable consisting of a single fiber or a bundle of optical fibers.
- An optical fiber is an extremely thin strand of ultra-pure glass designed to transmit optical signals from the optoelectronic source to the optoelectronic detector.
- In its simplest form, it consists of two main regions: (1) a solid cylindrical region of diameter 8-100 µm called the core and (ii) a coaxial cylindrical region of diameter normally 125 µm called the cladding.
- The refractive index of the core is kept greater than that of the cladding. This feature makes light travel through this structure by the phenomenon of total internal reflection. In order to give strength to the optical fiber, it is given a primary or buffer coating of plastic, and then a cable is made of several such fibers. This optical fiber cable serves as an information channel.
- For clarity of the transmitted information, it is required that the information channel should have low attenuation for the frequencies being transmitted through it and a large light-gathering capacity. Furthermore, the cable should have low dispersion in both the time and frequency domains, because high dispersion results in the distortion of the propagating signal.

Repeater:

- As the optical signals propagate along the length of the fiber, they get attenuated due to absorption, scattering, etc. and broadened due to dispersion.
- After a certain length, the cumulative effect of attenuation and dispersion causes the signals to become weak and indistinguishable.
- Therefore, before this happens, the strength and shape of the signal must be restored. This can be done by using either a regenerator or an optical amplifier, eg, an erbium-doped fiber amplifier (EDFA), at an appropriate point along the length of the fiber.

Optoelectronic Detector:

- The reconversion of an optical signal into an electrical signal takes place at the OE detector. Semiconductor p-i-n or avalanche photodiodes are employed for this purpose.
- The photocurrent developed by these detectors is normally proportional to the incident optical power and hence to the information input.
- The desirable characteristics of a detector include small size, low power consumption, linearity, flat spectral response, fast response to optical signals, and long operating life.

Receiver:

- For analog transmission, the output photocurrent of the detector is filtered to remove the de bias that is normally applied to the signal in the modulator module, and also to block any other undesired frequencies accompanying the signal.
- After filtering, the photocurrent is amplified if needed. These two functions are performed by the receiver module.
- For digital transmission, in addition to the filter and amplifier, the receiver may include decision circuits. If the original information is in analog form, a digital-to-analog converter may also be required.
- The design of the receiver is aimed at achieving high sensitivity and low distortion. The signal-to-noise ratio (SNR) and bit-error rate (BER) for digital transmission are important factors for quality communication.

Information Output:

- Finally, the information must be presented in a form that can interpreted by human observer. For example, it may be required to transform the electrical output into a sound wave or a visual image.
- Suitable output transducers are required for achieving this transformation In some cases, the electrical output of the receiver is directly usable This situation arises when a fiber optic system forms the link between different computers or other machines.

MICROWAVE COMMUNICATION

Introduction:

- "Microwaves' is a descriptive term finds its origin in the frequencies used for its communication. The term microwaves used to identify e, waves in the frequency spectrum ranging approximately from 1 GHz (10 Hz) to 30 GHz. This corresponds to wavelengths from 30 cm to 1 cm. Sometimes higher frequencies (extending upto 600 GHz) are also called microwaves.
- These waves present several interesting and unusual features not found in other portions of frequency spectrum. These features make microwaves' uniquely suitable for several useful applications.
- A significant advantage associated with the use of microwaves for communications is their large bandwidth.
- Microwave techniques are now being introduced in fast computer operations. Pulses with very small widths are used in high-speed logic circuits.

- Microwaves are extensively used for information relay systems in communication, especially for line-of-sight transmission systems. They also find application in RADAR. The most common use of the microwave systems is to communicate over rough or inaccessible terrain. Besides their most common use in RADAR systems and point-to-point radio communications, microwaves are applied extensively in Research laboratories servicing microwave test equipment and components eg.. microwave oven which typically operates at 2.45 GHz.
- Microwave communication systems cover distances ranging from 15 miles to 4000 miles. Microwave can be categorized as short haul for intrastate communication and long and for interstate communication systems.
- They generally use advanced modulation techniques much as phase shift keying (PSK) or quadrature amplitude modulation (QAM).

MICROWAVE COMMUNICATIONS

- Microwave communications are widely used for telephone networks, in broad cast and television systems and several other communication applications by services, railways, etc.

Frequency Modulated Microwave Communication System

- FM microwave systems, when equipped with suitable multiplexing technology are capable of carrying hundreds of voice and data channels. In addition to point-to-point communications. The FM microwave systems can also be extended to broadcasting television audio signals.

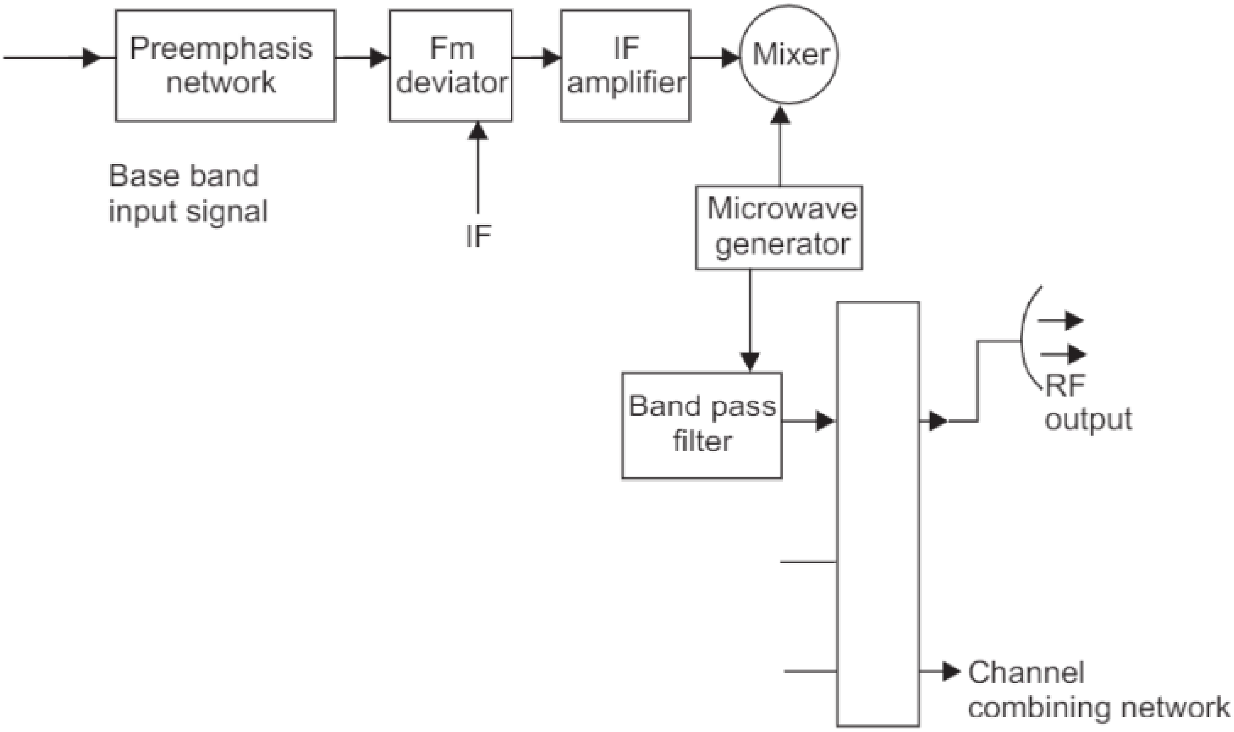

Fig 5.18 Block diagram of FM transmitter

- The baseband input signal can be anything from a FDM voice channel to a TDM channel or from a composite video signal to a wideband data signal.

- The baseband signal is first applied to the preemphasis network that, in general, provides extra amplification to high frequency baseband signals.
- When the signal coming out of the preemphasis circuit is applied to the FM modulator, the low frequencies get frequency modulated by the Intermediate Frequency (IF) carrier and the high frequencies get phase modulated.
- This ensures a more uniform SNR (signal to noise ratio) throughout the frequency range.
- The IF frequencies are generally in the range of 60-80 MHz.
- The modulated output from the FM deviator is passed through the IF amplifier to the mixer. The mixer then converts the signal to microwave frequencies. Using the mixer instead of the multiplier preserves the modulation index and also limits the bandwidth.
- The output of the mixer is passed through the band pass filter to band limit the signal and then to the channel-combining network. Finally, the signal is fed to the transmitter antenna.
- Figure 5.18 shows the simplified transmitter block diagram of a microwave FM transmitter.
- Figure 5.19 shows the need for repeaters and how a virtual line of sight is maintained between the transmitter and the receiver using repeaters.

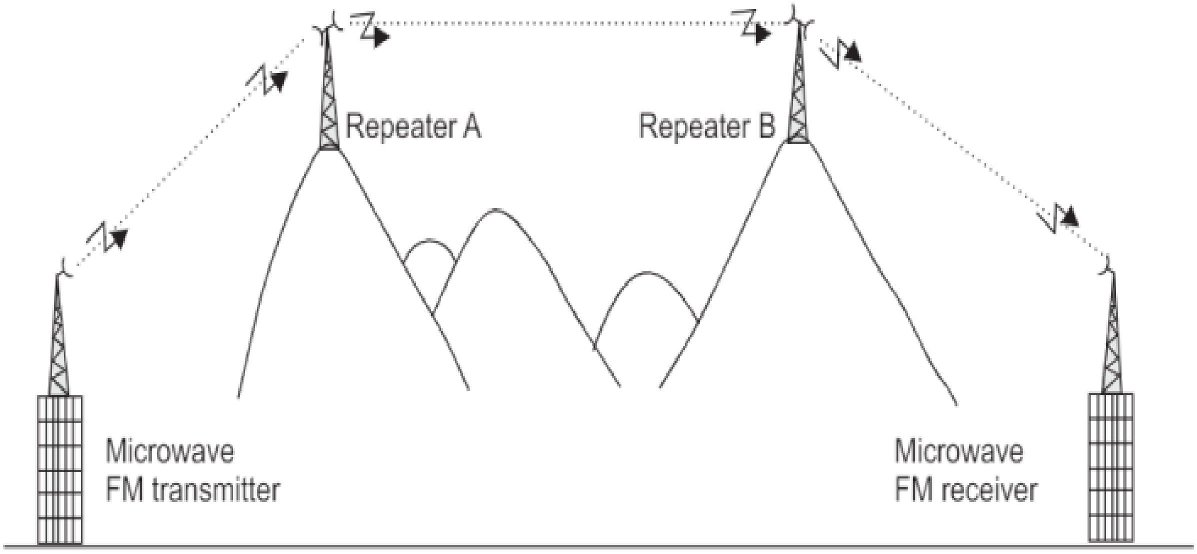

Fig. 5.19 Virtual line of sight for microwave FM transmission using repeaters.

- If the distance between the transmitter and the receiver is less than the maximum distance that ensures the reliable reception of the FM signal, then the communication system does not need any other intermediate station.
- However, there should be a line-of-sight path maintained for reliable transmission and reception. If either if there two conditions are not met, then the system needs intermediate stations that can receive the signal, process it (possibly amplify) and retransmit the signals.

- These intermediate stations are called the repeaters, since their objective is to repeat the signal that they receive (possibly with some amplification) in the direction of the next repeater or the receiver in Fig. the transmitter and the receiver are not in direct line of sight with each other due to the obstructive intermediate terrain (which is caused by the mountains).
- However, by appropriately placing the repeaters, a virtual line of sight is achieved along the path transmitter-repeater A-repeater B-receiver
- In Fig. 5.20, the RF signal picked by the receiving antenna is passed to the channel separation network, which separates the individual channels.
- The bandpass filter then filters out any frequencies that fall outside the bandwidth of the required signal.
- The mixer employs the same RF oscillator frequency as used at the transmitter and converts the RF signal to the IF band.
- The FM detector demodulates the signal which is then passed to the deemphasis network.
- The de-emphasis network applies inverse functionality of the preemphasis network at the transmitting end, to finally restoring the original baseband signal. Figure 5.20 shows the microwave FM receiver block diagram.

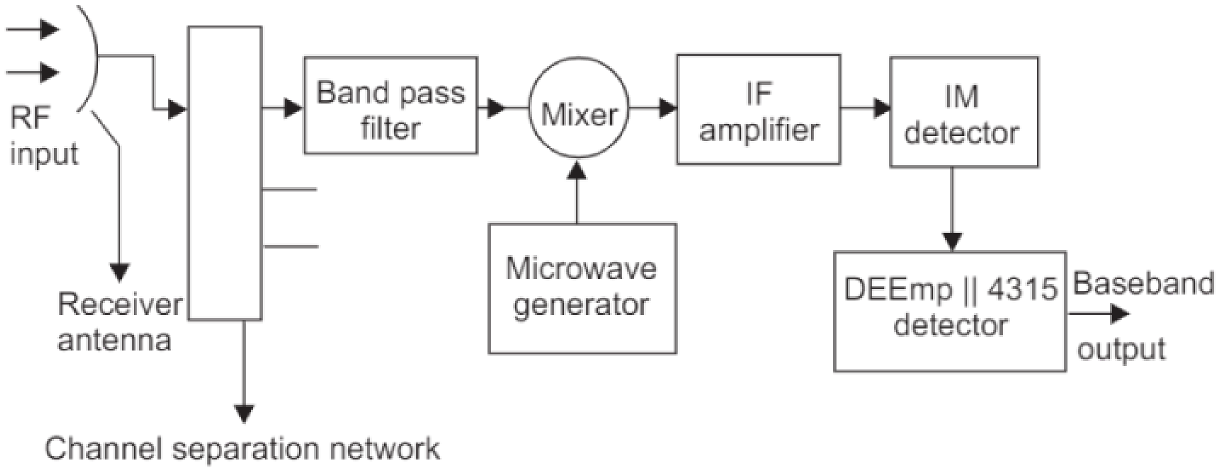

Fig. 5.20 Block diagram of FM Receiver.