Electronic Circuits and Communication Fundamentals

Chapter 1 : Electronic Circuits : Bipolar Junction Transistor

Q. 1 Draw the equivalent circuit of n-p-n transistor and explain the operation of an unbiased transistor.

Ans. :

circuit of n-p-n transistor

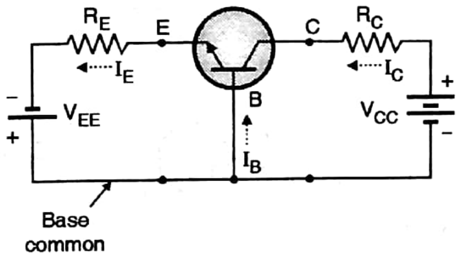


Fig 1.1: n-p-n transistor operation of an unbiased transistor:

For an unbiased transistor no external power supplies are connected to it. As a transistor is formed of two p-n junctions.

Fig. 1.2 shows the depletion regions formed at the B-E and C-B junctions of an n-p-n transistor.

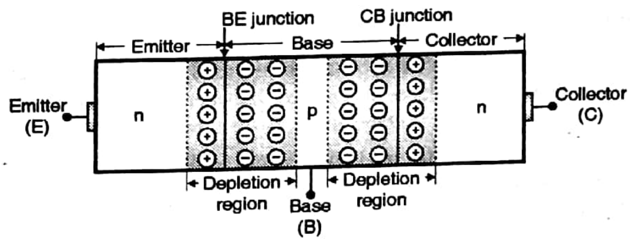


Fig. 1.2 : Depletion regions in an unbiased n-p-n transistor

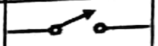
Q. 2 State the regions of operation of a transistor and explain the biasing conditions for the three regions.

Ans. :

Regions of operation of a transistor:

1. Cutoff region (transistor is OFF)
2. Saturation region (transistor is fully ON)
3. Forward active region (in between saturation and cutoff).
4. Inverse active mode

Biasing conditions for the three regions:

Sr. No.	Region of operation	Base emitter junction	Collector base junction	Application
1.	Cutoff region	Reverse biased	Reverse biased	 (B-1619)
2.	Forward active region	Forward biased	Reverse biased	Amplifier
3.	Saturation region	Forward biased	Forward biased	 (B-1620)
4.	Inverse active	Reverse biased	Forward biased	

Q. 3 With the help of neat figures, explain the operation of an n-p-n transistor.

Ans. :

Operation of n-p-n Transistor :

The operation of n-p-n transistor discussed earlier can be simplified as follows :

The forward bias at the B-E junction reduces the barrier potential and causes the electrons to flow from n-type emitter to p-type base.

Holes also will flow from p-type base to n-type emitter. But as the base is more lightly doped than the emitter, almost all the current flowing across the B-E junction consists of electrons entering the base from the emitter. Hence electrons are the majority carriers in an n-p-n transistor.

Some of the electrons entering into the base region do not reach the collector region. Instead they flow out of the base terminal via the base connection as shown in Fig. 1.3, due to recombination. As the base region is very thin and lightly doped, there are very few holes available in the base region for recombination.

Hence about 2% electrons will flow out of base due to recombination.

The remaining 98% electrons cross the reverse biased collector junction to constitute the collector current. They cross the collector region and are collected by the supply V_{CC}.

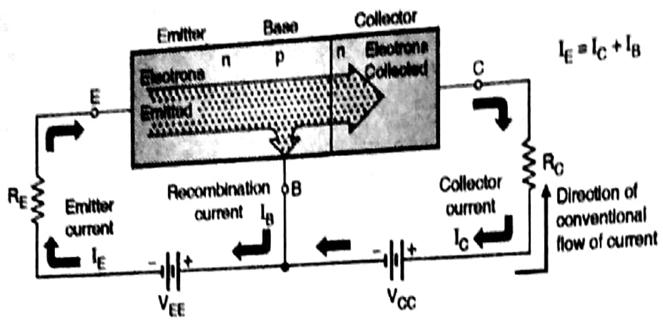


Fig. 1.3 : Operation of an n-p-n transistor

The emitter current is equal to the sum of collector and base currents.

$$\therefore \quad I_E = I_C + I_B$$

Q. 4 With the help of neat diagrams, explain the operation of a p-n-p transistor.

Ans. :

Operation of p-n-p Transistor

The p-n-p transistor behaves exactly in the same way as the n-p-n device. The only difference is, the majority charge carriers are holes instead of electrons. As shown in Fig. 1.4 holes are emitted from the p-type emitter across the forward biased EB junction, into the base.

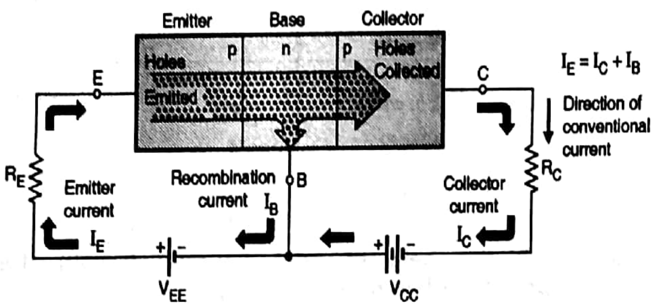


Fig. 1.4 : Operation of a p-n-p transistor

In thelightly doped base there are very few number of electrons available for recombination. Therefore about 2% of total emitted holes will flow out via the base terminal and the remaining are drawn across the collector by the electric field at the reverse biased collector junction.As in case of n-p-n transistor, the forward bias at the EB junction controls the collector and emitter currents.

Q. 5 What is reverse saturation current ?

Ans. :

Reverse saturation current I_{CBO} :

The reverse saturation current of the reverse biased CB junction flows due to the minority carriers hence it is dependent on temperature. We know that,

$$I_C = \beta_{dc} I_B + (1 + \beta_{dc}) I_{CBO}$$

Therefore change in I_{CBO} due to temperature will force the collector current I_C and hence the Q point to change.

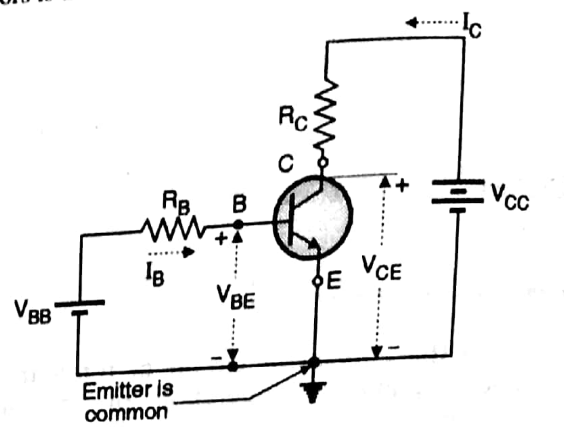
To overcome this problem the biasing circuit must include some kind of "temperature compensation", or "temperature stability" so that the changes in the values of these parameters can be kept under control.

Q. 6 With the help of neat circuit diagram, explain the common emitter configuration of a transistor.

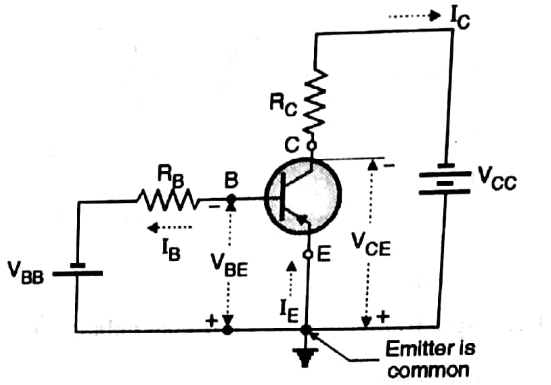
Ans. :

Common Emitter (CE) Configuration :

The common emitter configuration for the p-n-p and n-p-n transistors is as shown in Figs. 1.5(a) and (b).



(a) Common emitter configuration for n-p-n transistor



(b) Common emitter configuration for p-n-p transistor

Fig. 1.5

CE configuration:

The emitter acts as a common terminal between input and output. The input voltage is applied between base and emitter. Hence V_{BE} is the input voltage and I_B is the input current.

The output is taken between the collector and emitter. Therefore V_{CE} is the output voltage and I_C is the output current. In order to operate the transistor in its active region, the base-emitter (BE) junction is forward biased and the collector base junction is reverse biased.

Q. 7 Define the reverse leakage current of a CE configuration.

Ans. :

The reverse leakage current of a transistor operating in the CE configuration is denoted by "I_{CEO}" and is defined as :

Reverse leakage current (CE configuration) :

$$I_{CEO} = (1 + \beta_{dc}) I_{CBO} \qquad ...(1)$$

As the value of β_{dc} is much greater than 1, $I_{CEO} >>> I_{CBO}$. In Equation (1) substitute $I_B = 0$,

$$I_C = (1 + \beta_{dc}) I_{CBO}$$

$$\therefore \quad I_C = I_{CEO} \text{ for } I_B = 0 \qquad ...(2)$$

The reverse leakage current (I_{CEO}) increases with increase in the temperature. This current flows in the same direction as that of I_C. Therefore the collector current I_C will increase with increase in temperature even when I_B is constant.

Refer Fig. 1.6 which shows that if base is open then the collector current is equal to I_{CEO}.

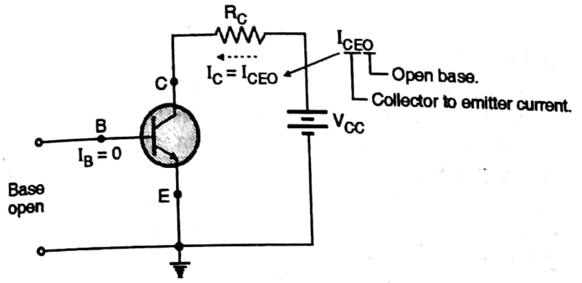


Fig. 1.6 : Reverse leakage current in CE configuration

This is called as thermal instability. So in CE configuration thermal stabilizing circuit must be included.

Q. 8 Why transistor is called a bipolar device ?

Ans. :

The conduction in a bipolar junction transistor takes place due to both, electrons and holes. That is why it is called as a "bipolar" transistor.

If the conduction takes place due to only one type of carriers i.e. majority carriers then the transistor is called as "unipolar" transistor. The example of a unipolar device is the Field Effect Transistor (FET).

Q. 9 Define α and β for a transistor and establish the relationship between them.

Ans. :

ALPHA (α): It is a large signal current gain in common base configuration. It is the ratio of collector current (output current) to the emitter current (input current).

$$\alpha = \frac{\text{Collector current}}{\text{Emitter current}} = \frac{I_C}{I_E}$$

It is a current gain in CB amplifier and it indicates that the amount of emitter current reaching to collector. Its value is unity ideally and practically less than unity.

Beta (β): It is a current gain factor in the common emitter configuration. It is the ration of collector current (output current) to base current (output current).

$$\text{beta} = \frac{I_C}{I_B} \text{ normally Its value is greater than 100}$$

$$\alpha_{dc} = \frac{I_C}{I_E}$$

$$\text{But} \quad I_E = I_C + I_B$$

$$\therefore \quad \alpha_{dc} = \frac{I_C}{I_C + I_B}$$

Dividing numerator and denominator by I_B,

$$\alpha_{dc} = \frac{(I_C / I_B)}{1 + (I_C / I_B)}$$

$$\text{But} \quad (I_C / I_B) = \beta_{dc}$$

$$\therefore \quad \alpha_{dc} = \frac{\beta_{dc}}{1 + \beta_{dc}} \qquad ...(1)$$

This is the relation between α_{dc} and β_{dc}. Similarly we can obtain the expression for β_{dc} in terms of α_{dc} as follows :

$$\beta_{dc} = \frac{I_C}{I_B}$$

$$\text{But} \quad I_B = I_E - I_C$$

$$\therefore \quad \beta_{dc} = \frac{I_C}{(I_E - I_C)}$$

Divide numerator and denominator by I_E,

$$\beta_{dc} = \frac{(I_C / I_E)}{1 - (I_C - I_E)}$$

$$\text{But} \quad (I_C / I_E) = \alpha_{dc}$$

$$\therefore \quad \beta_{dc} = \frac{\alpha_{dc}}{1 - \alpha_{dc}} \qquad ...(2)$$

Thus the relations between α_{dc} and β_{dc} are,

$$\alpha_{dc} = \frac{\beta_{dc}}{1 + \beta_{dc}} \text{ and } \beta_{dc} = \frac{\alpha_{dc}}{1 - \alpha_{dc}}$$

Q. 10 With the help of experimental setup, explain the procedure to plot the input and output characteristics of a transistor in CE configuration.

Ans. : Characteristics of a transistor in common emitter configuration :

Similarly for the common base characteristics, plot the input and output characteristics of a transistor.

Input Characteristics (The Base Curve) : It is a graph of input current I_B versus input voltage V_{BE} at a constant output voltage (V_{CE}). For CE configuration, I_B is the input current and V_{BE} is the input voltage. At constant output voltage V_{CE} the input characteristics of a n-p-n transistor is as shown in Fig. 1.7 The input characteristics also shows the effect of V_{CE}.

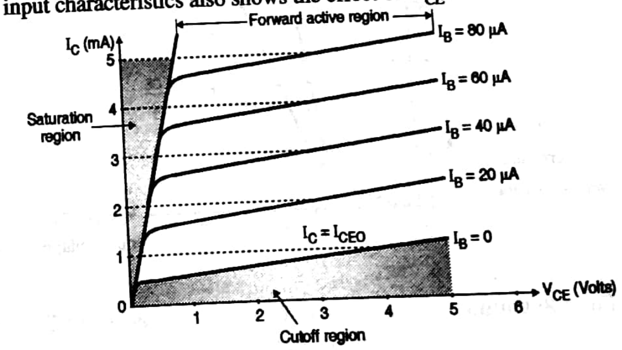


Fig. 1.7: Input characteristics of a transistor in the CE configuration

The input characteristics resembles the forward characteristics of a p-n junction diode. The reason is that B-E junction is a forward biased p-n junction. The base current increases rapidly as the base-emitter voltage crosses the cut in voltage of the BE, p-n junction. The dynamic input resistance is defined as :

$$r_i = \left.\frac{\Delta V_{BE}}{\Delta I_B}\right|_{V_{CE\ constant}}$$

Its value can be obtained from the input characteristics because "r_i" is equal to the reciprocal of slope of the input characteristics. The value of dynamic input resistance "r_i" is low (typically 1 kΩ) for the CE configuration but it is not as low as that of CB configuration.

Effect of change in V_{CE} on the input characteristics :

In CB configuration, the effect of change in V_{CB} on the input characteristics is known as the Early effect. Let consider the effect of change in V_{CE} on the input characteristics. Fig. 1.7 shows that at a constant V_{BE}, if we increase V_{CE} from 5 V to 15 V then the base current decreases from 60 μA to 20 μA. Thus I_B decreases with increase in V_{CE}..

As V_{CE} increase the CB junction is more reverse biased.

↓

The depletion region at CB junction penetrates more into base.

↓

This reduces the electrical width of the base.

↓

The chances of recombination in the base region reduce.

↓

Hence the base current will reduce.

Output Characteristics :

An output characteristics of a CE configuration is the graph of output current (I_C) versus output voltage (V_{CE}) for various fixed values of the input current (I_B).

The typical output characteristics of a n-p-n transistor operating in the CE configuration are as shown in Fig. 1.8

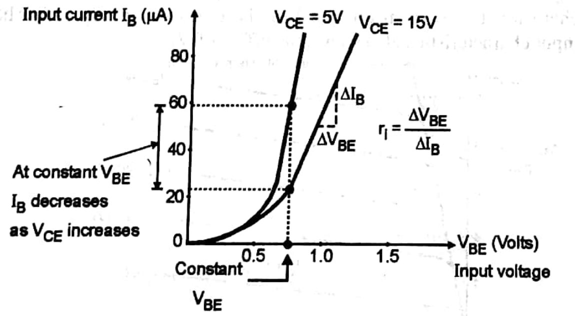


Fig. 1.8: Output characteristics of a n-p-n transistor in CE configuration

As shown in Fig. 1.8, there are three regions of operation namely the cutoff region, active region and saturation region.

1. **Cutoff region :**
   Both the B-E and C-B junctions are reverse biased to operate the transistor in cutoff region. The base current I_B = 0 and the collector current is equal to the reverse leakage current I_{CEO}. The region below the characteristics for I_B = 0 is cutoff region.

2. **Active region :**
   The B-E junction is forward biased, and C-B junction is reverse biased to operate the transistor in the active region. At a constant base current I_B, the collector current I_C increases slightly with increase in the voltage V_{CE}. However the collector current is largely dependent on the base current I_B. At a fixed value of V_{CE}, if I_B is increased, then it will cause I_C to increase substantially. This is because $I_C = \beta_{dc}\, I_B$. This relation is true only for the active region of operation.

3. **Saturation region :**
   The BE junction as well as the collector junction must be forward biased to operate the transistor in its saturation region. The collector base junction can be forward biased if and only if V_{CE} drops down to about 0.2 Volts. Because then V_{BE} = 0.7 V will forward bias the C-B junction. This is as shown in Fig. 1.9 Usually the saturation voltage of a transistor, $V_{CE\ (sat)}$ is between 0.1 to 0.3 Volts. The collector current increases rapidly with increase in V_{CE} as shown in Fig. 1.9.
   I_C is approximately independent of the base current and function of V_{CE}. Therefore in this region the transistor is considered to be a semiconductor resistor of very small value. The transistor is operated as a switch in this region.

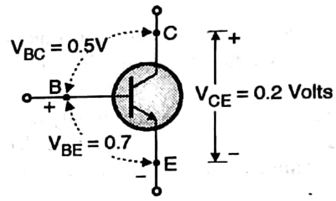


Fig. 1.9 : Forward biasing of CB junction

4. **Dynamic output resistance (r_o) :**
   The dynamic output resistance (r_o) of a transistor in CE configuration is defined as :

$$r_o = \left.\frac{\Delta V_{CE}}{\Delta I_C}\right|_{\text{constant } I_B}$$

   The dynamic output resistance can be obtained as reciprocal of slope of output characteristics. Its value is large in the active region because ΔI_C in this region is very small. However value of r_o will be very small in the saturation region. This is because ΔI_C in that region is large for a small value of ΔV_{CE}. In the active region the typical value of r_o for the CE configuration is 10 kΩ.

5. **Definition of β_{ac} :**

$$\beta_{dc} = \frac{I_C}{I_B}$$

   The value of β_{dc} can be obtained from the output characteristics. At any point on the characteristics we can calculate β_{dc} by taking the ratio of I_C and I_B at that point.
   let us define AC beta of a transistor as :

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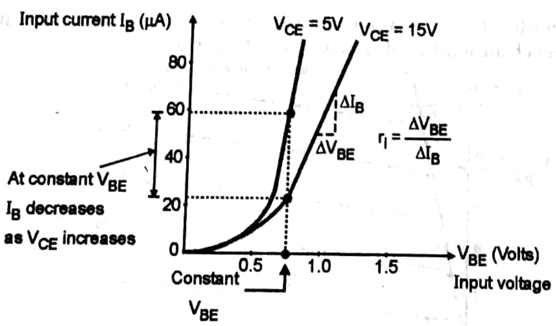

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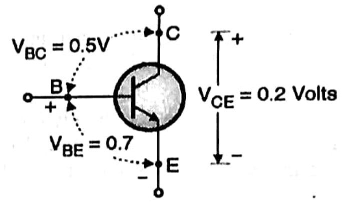

Fig. 1.9 : Forward biasing of CB junction

4. **Dynamic output resistance (r_o) :**
   The dynamic output resistance (r_o) of a transistor in CE configuration is defined as :

$$r_o = \left.\frac{\Delta V_{CE}}{\Delta I_C}\right|_{constant\ I_B}$$

   The dynamic output resistance can be obtained as reciprocal of slope of output characteristics. Its value is large in the active region because Δ I_C in this region is very small. However value of r_o will be very small in the saturation region. This is because Δ I_C in that region is large for a small value of Δ V_{CE}. In the active region the typical value of r_o for the CE configuration is 10 kΩ.

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   let us define AC beta of a transistor as :

$$\beta_{ac} = \left.\frac{\Delta I_C}{\Delta I_B}\right|_{V_{CE}\ constant}$$

Thus the value of ac beta can be obtained at a constant value of V_{CE} from the output characteristics. The values of β_{dc} and β_{ac} are nearly the same.

6. **Maximum V_{CE} and breakdown :**

In the active region the collector junction is reverse biased, so there is a limit on the maximum value of V_{CE}. If V_{CE} exceeds this maximum value, collector junction will breakdown due to the punch through effect. A large current will flow which will generate excessive heat to damage the transistor. Hence for safe operation $V_{CE} < V_{CE\,(max)}$.

Q. 11 List biasing methods for transistor. State necessity for biasing.

Ans. :

Bias stabilization means techniques used to stabilize the Q point. Some of them are as follows :

1. Fixed bias circuit (Single base resistor biasing)
2. Collector to base bias circuit
3. Voltage divider bias circuit
4. Emitter bias.

All electronic devices require proper DC biasing mechanism depending on its application. The main purpose of biasing is to obtain idle or quiescent conditions for the electronic device so that it can perform the required function. For example, when transistor is used in the amplifier circuit, it should work so that transistor remains in the active region i.e. its emitter junction forward biased and collector junction remains in reverse bias always. This can be achieved by transistor biasing with the use of DC batteries and different resistances, which are together, referred as biasing circuit. Proper DC biasing fixes the Q-point on the DC load line such that transistor remains in the active region while the process of amplification. The various parameter of the transistor are considered while designing a biasing circuit.

Q. 12 State the requirements of biasing circuit.

Ans. :

A transistor biasing circuit is expected to satisfy the following requirements :

1. The transistor should be biased in the active region if it is to be used for amplification and in saturation and cut off if it is to be used as a switch.
2. The Q point should be adjusted approximately at the center of the load line for voltage amplifier application.
3. The value of stability factor (S) should be as small as possible. Ideal value of stability factor is 0.
4. Q point should be stabilized by introducing a negative feedback in the biasing circuit.
5. The Q-point should not be affected (its position should not change) due to temperature changes or device to device variation.
6. Bypass capacitor should be included to avoid reduction in voltage gain due to negative feedback.
7. Transistor should be biased in the linear region of the transfer characteristics.

Q. 13 Define biasing of a transistor and state four methods of transistor biasing.

Ans. :

Transistor Biasing is the process of setting a transistors DC operating voltage or current conditions to the correct level so that any AC input signal can be amplified correctly by the transistor.

methods of transistor biasing

Q-point biasing of a transistor

Base Biasing a Common Emitter Amplifier

Fixed Base Biasing a Transistor

Collector Feedback Biasing a Transistor

Dual Feedback Transistor Biasing

Q. 14 Define stability factor of biasing circuit.

Ans. :

The stability of Q point of a transistor amplifier depends on the following three parameters :

1. Leakage current I_{CO}
2. β_{dc}
3. Base to emitter voltage

The effect of these parameters can be expressed mathematically by defining the stability factors for the three parameters individually as follows :

1. Stability factor, $S = \left.\dfrac{\Delta I_C}{\Delta I_{CO}}\right|_{\text{Constant } V_{BE} \text{ and } \beta_{dc}}$

or $\dfrac{\partial I_C}{\partial I_{CO}}$

This represents the change in collector current due to change in reverse saturation current I_{CO}. The other two parameters that means V_{BE} and β_{dc} are assumed to be constant.

2. Stability factor, $S' = \left.\dfrac{\Delta I_C}{\Delta V_{BE}}\right|_{\text{Constant } I_{CO} \text{ and } \beta_{dc}}$

or $\dfrac{\partial I_C}{\partial V_{BE}}$

S′ represents the change in I_C due to change in V_{BE} at constant I_{CO} and β_{dc}.

3. Stability factor, $S'' = \left.\dfrac{\Delta I_C}{\beta_{dc}}\right|_{\text{Constant } I_{CO} \text{ and } V_{BE}}$

or $\dfrac{\partial I_C}{\partial \beta_{dc}}$

These expressions indicate that I_C is collectively dependent on ΔI_{CO}, ΔV_{BE} and $\Delta \beta_{dc}$. The combined effect of these parameters on I_C is mathematically given by,

Total change in collector current,

$$\Delta I_C = S\,\Delta I_{CO} + S'\,\Delta V_{BE} + S''\,\Delta \beta_{dc}$$

Ideally the values of all the stability factors should be zero and practically they should be as small as possible.

Practically the value of S is significantly higher than the other two stability factors. Hence while comparing the biasing circuits, we should focus our attention more on the value of S.

Q. 15 What are the advantages of voltage divider bias circuit ?

Ans.: Advantages of voltage divider bias circuit :

The advantages of voltage divider bias circuit are as follows :

1. It has the smallest value of S among all the biasing circuits. This shows that the bias point stability is highest for the self bias circuit.
2. It is possible to avoid the loss of signal gain by connecting an emitter bypass capacitor across R_E. This does not have any adverse effect on the other advantages of self bias circuit.
3. R_E introduces a negative feedback. This will make the self bias circuit more stable. So all the other advantages of negative feedback get attached to this circuit.
4. Voltage divider bias circuit is therefore the most widely used biasing circuit.

Q. 16 Draw voltage divider bias circuit. How it stabilizes operating point ?

Ans. :

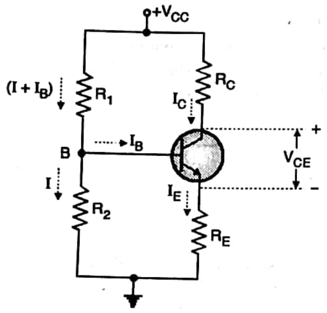


Fig. 1.10: Voltage divider bias

Bias stabilization using voltage divider bias circuit :

- If I_C increases due to change in temperature or β_{dc}
- ⬇ Then I_E increases
- ⬇ Hence drop across R_E increases ($V_E = I_E R_E$)
- ⬇ But V_B is constant. Hence V_{BE} decreases.
- ⬇ Hence I_B decreases.
- ⬇ Hence I_C also decreases. Thus the compensation for increase in I_C is achieved.

Q. 17 Define R_i, R_o, A_I, A_V and A_P for a voltage amplifier.

Ans.:

1. Gain :

The gain of an amplifier is defined as the ratio of output quantity to the input quantity. So the ratio of output voltage to input voltage will be called as "voltage gain A_V" of the amplifier. Similarly the ratio of output current to input current is called as "current gain A_I" of the amplifier.

$$\therefore \text{ Current gain, } A_I = \frac{I_o}{I_i}$$

$$\text{And voltage gain, } A_V = \frac{V_o}{V_i}$$

The voltage and current gains of an amplifier should be as large as possible. Gain is a unitless quantity.

2. Input resistance (R_i) :

It is the resistance seen looking into the input terminals of an amplifier. More generally an input impedance Z_i is defined, however Z_i is resistive for the transistorised amplifiers hence it is replaced by R_i. Ideally R_i should be infinite and practically it should be as large as possible.

3. Output resistance (R_o) :

It is the resistance seen looking into the output terminals of an amplifier when the input signal source V_s is short circuited ($V_i = 0$) and output terminals are open circuited. R_o should be ideally equal to zero and practically it should be as small as possible. Thus current gain, voltage gain, input resistance and output resistance are the four important parameters of an amplifier.

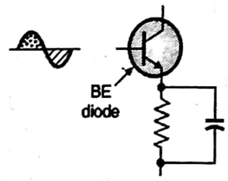
Q. 18 What do you understand by small signal operation ?

Ans. :

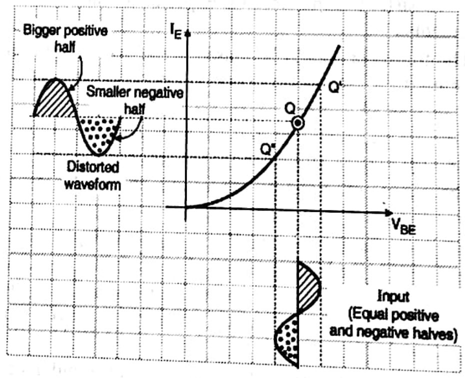
Fig. 1.11(a) shows the VI characteristics of a forward biased BE junction diode of a transistor. The sinusoidal variation in V_{BE} corresponds to the ac signal applied at the base terminal of a transistor.

Q point :

With the variation in input voltage and V_{BE}, the operating point Q moves between the two extremes Q′ and Q″ as shown in Fig. 1.11(b). The size of ac input voltage decides the amount of variation in the position of Q point.



(a) Partially drawn amplifier



(b) Distortion when the input single is too large

Fig. 1.11

Distortion :

The variation in V_{BE} results in corresponding sinusoidal variation in the emitter current as shown in Fig. 1.11(b). The frequency of the ac emitter current is same as that of the ac V_{BE}

voltage. The shape of I_E waveform also is approximately same as that of the V_{BE} waveform.

If the input voltage is too large then the emitter current waveform is **distorted** as shown in Fig. 1.11(b). That means shape of I_E is not exactly same as that of the V_{BE} waveform.

This happens due to the **nonlinear** nature of the characteristics of BE diode. Such a distortion is **undesirable** because it degrades the quality of output.

Reducing the distortion :

One way of reducing the distortion in Fig. 1.11(b) is to keep the ac base voltage **small.** The smaller base voltage would result in a **smaller** swing of Q point.

The smaller swing or variation in the Q point position means less curvature of the graph. If the input signal is small then the graph appears to be **linear** and the distortion will be either **reduced** or **eliminated.**

The 10 % rule :

The total emitter current consists of a dc component and an ac component. Hence the total emitter current is given by,

$$\underbrace{I_E}_{\text{Total emitter current}} = \underbrace{I_{EQ}}_{\text{DC component}} + \underbrace{i_e}_{\text{AC component}}$$

If the ac component i_e is smaller than the dc component I_{EQ} then the distortion can be reduced. This operation is called as small **signal operation.**

The condition for small signal operation is as follows :

Small signal : $i_{e\,(\text{peak to peak})} < 0.1\ I_{EQ}$

That means the peak to peak ac emitter current is less than **10 percent** of the dc emitter current. This is known as the **10 percent** and the amplifiers which satisfy this rule are called as **small signal amplifiers.**

The small signal amplifiers are used at the **front end** (input blocks) of the radio and TV receivers.

Chapter 2 : Power Amplifiers

Q. 1 Explain the concept of power amplifier.

Ans. :

Concept of Power Amplifier :

The small signal amplifiers are designed to amplify signals of very small magnitude.

They operate in the linear portion of the transfer characteristics which is very close to the Q point of the amplifier.

Small signal amplifiers can be analyzed with the help of small signal h-parameters.

Small signal amplifiers are also known as **"voltage amplifiers".** This is because these amplifiers are used primarily for voltage amplification but they are not capable of supplying a large amount of power to the loads such as loud speakers.

Whenever the load demands a large power, use specially designed amplifiers called "power amplifiers".

As shown in Fig. 2.1, these amplifiers convert low power signal at their input to a high power signal.

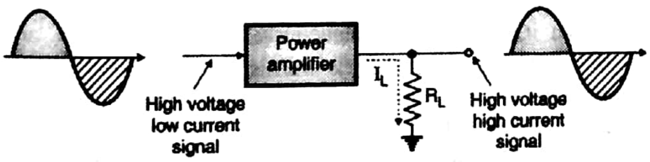


Fig. 2.1: Concept of power amplifiers

The input signal to the large signal amplifier is a high voltage low current signal obtained from a chain of voltage amplifiers. Power is equal to the product of voltage and current. Hence high voltage, low current signal at the input corresponds to a low power signal.

The large signal amplifier (power amplifier) will increase the current sourcing and sinking capability. Therefore at its output we get a high voltage, high current signal that means a high power signal. Thus the power amplifier is basically a **current amplifier.**

Power amplifier is an amplifier which amplifies power or current. It is also called as large signal amplifier. The simplest example of power amplifier is the "emitter follower" circuit which has a unity voltage gain but high current gain.

Power amplifier is usually the last (final) stage in most of the high power circuits. Its output is connected directly to the load. Fig. 2.2 shows the block schematic of an AF amplifier which demonstrates this concept.

The applications of power amplifiers are :

1. Radio receivers
2. Public address (P.A.) systems.
3. CD/casette players
4. TV receivers.

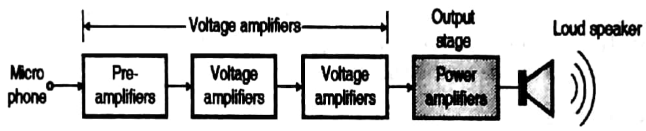


Fig. 2.2 : Simplified block diagram of an AF amplifier

Q. 2 State and explain the important features of power amplifier.

Ans. : Features of a Power Amplifier :

1. Output impedance of the power amplifier should be matched with the load.
2. Power transistors are required to be used.
3. Power amplifiers are bulky.
4. Harmonic distortion is present in their output.
5. They are capable of handling a large power.

Impedance matching :

As the power amplifiers are handling a large amount of power it is important to transfer maximum power to the load. To do so, impedance matching between the output impedance of the power amplifier and load has to be ensured.

As the loads like loud speakers have low impedance, the output impedance of a power amplifier also must be low. Therefore the common collector or emitter follower circuit is normally used as the power amplifiers, because it has a low output impedance.

A transformer may also be used for impedance matching, on the output side.

Use of power transistors :

As the power amplifiers are designed to handle large powers, the transistors used must be capable of withstanding to large voltages and currents.

A large power gets dissipated in these transistors in the form of heat. Therefore we cannot use the ordinary logic level transistors.

Instead specially manufactured power transistors are used.

Power amplifiers are bulky :

The power transistors are bigger in size than the low power transistors. In addition to this, they are mounted on " heat sinks " which are of large surface area.

The " heat sinks " help to reduce the temperature of the power transistors by dissipating the heat to the surroundings. Due to the use of heat sinks and large size power transistors, the power amplifiers become bulky.

Harmonic distortion :

Due to the non-linear characteristics of transistors, harmonic distortion will be present on the output side of the amplifier.

That means, those frequency components which are not present on the input side, will be present in the output.

This will distort the shape of the output waveform. To measure the percentage of distortion, analysis of the output waveform is carried out.

h-parameters cannot be used for the analysis :

The h-parameters are called as small signal h-parameters. They are valid if and only if the input signal is small enough to operate the amplifier close to Q point, on the linear portion of the transfer characteristics of the transistor.

Power amplifiers can not satisfy this condition. Therefore we can not use h-parameters for the analysis of power amplifiers. The analysis of power amplifiers is carried out with the help of a load line drawn on the characteristics of the transistor.

Efficiency :

Efficiency of a power amplifier is defined as the ratio of output power to the total input power.

$$\therefore \quad \% \text{ efficiency } (\eta) = \frac{P_{out}}{P_{in}} \times 100$$

$$\text{But } \quad P_{in} = P_{out} + P_{losses}$$

$$\therefore \quad \% (\eta) = \frac{P_{out}}{P_{out} + P_{losses}} \times 100$$

Where P_{losses} is the power lost in the power transistor.

If the transistor is biased to operate in the active region, then power loss taking place in it will be high and efficiency will be low.

But if the transistor is biased in the cut off region or below cut off and in the saturation region then the power loss taking place in it is low and the efficiency will be increased.

Q. 3 Explain the class A operation of power amplifiers.

Ans. :

Class A Power Amplifier :

A power amplifier is referred to as a class A power amplifier if the transistor used for amplification conducts for the full cycle duration of the input ac signal.

The Q point is adjusted exactly at the center of the load line as shown in Fig. 2.3. Due to this the output signal is obtained for the full cycle of the ac input i.e. for 360° as shown in Fig. 2.3.

The power transistor is biased such that the operating point (Q point) is approximately at the center of the load line as shown in Fig. 2.3

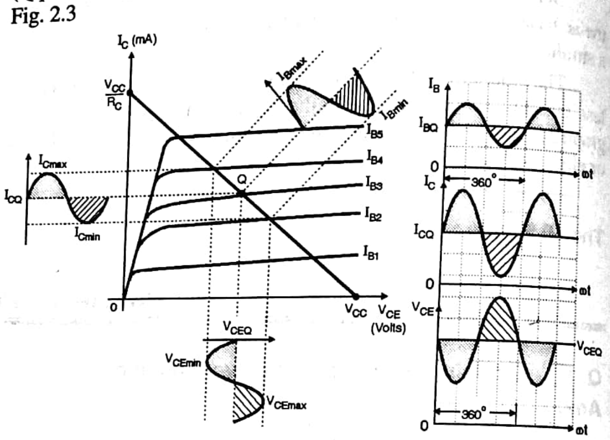


Fig. 2.3 : Graphical representation of class A operation

Now as we apply the ac signal to the base of the transistor, the base current changes sinusoidally above and below the quiescent base current I_{BQ} as shown in Fig. 2.3. In response to the changes in I_B, the collector current changes sinusoidally above and below its quiescent current value I_{CQ}. The collector current and base current are in phase with each other.

Due to changes in I_C, the voltage V_{CE} will also fluctuate sinusoidally as shown in Fig. 2.3. V_{CE} and I_C are 180° out of phase. The transistor remains in the "active region" for all the values of input signal and never enters into the saturation or cutoff regions. As shown in Fig. 2.3, the input signal is amplified faithfully, without introducing any distortions. Thus harmonic contents in the output will be low. As the transistor continuously operates in its active region, the voltage V_{CE} across it and current I_C through it, both are simultaneously high. Therefore a large power will be dissipated in the transistor in the form of heat. Therefore the efficiency of class A power amplifiers is low. In fact it is the lowest of all the power amplifiers. Typically the efficiency (η) of a class A power amplifier lies between 25% to 50%.

Q. 4 State merits, demerits and applications of class A power amplifier.

Ans. :

Merits :

1. Simple construction.
2. Distortionless output voltage.

Demerits:

1. Very low efficiency (25% or 50%)

2. Large power dissipation in the power transistors.

Applications:

1. Linear amplifiers.
2. High gain voltage amplifiers.
3. RF and IF amplifiers in radio and TV.

Q. 5 Explain the class C operation.

Ans. :

Class C Operation :

A power amplifier is referred to as a class C power amplifier if its output is obtained for less than a half cycle period of the input ac signal as shown in Fig. 2.4.

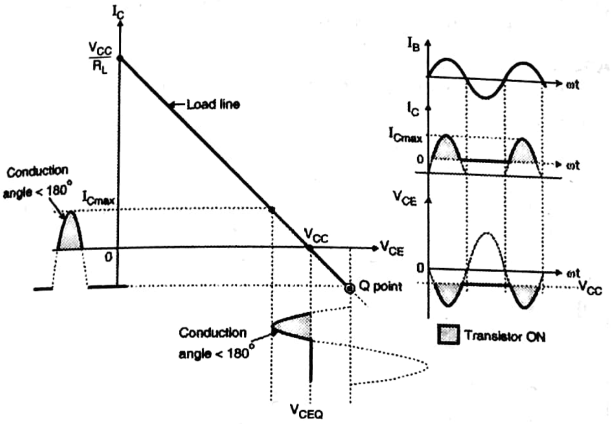


Fig. 2.4 : Class C operation

Thus the power transistor in a class C configuration will conduct for a duration which is less than the period of a half cycle of the ac input signal.

For this the operating point is adjusted to be below the X-axis as shown in Fig. 2.4. Thus the transistor is biased below cut-off.

Due to reduced conduction angle, the output waveform is heavily distorted. The percent distortion is higher than that for a class B amplifier. Therefore class C amplifiers are not used as A.F. power amplifiers.

The efficiency of class C amplifiers is very high. In fact it is the highest of all the power amplifiers. Typically the efficiency is above 95%.

Q. 6 State merits, demerits and applications of class C power amplifiers.

Ans. :

Merits:

1. Very high efficiency (higher than 95%).
2. Low power loss in the power transistors.

Demerits :

1. The output waveform can be distorted.

Applications :

1. The class C amplifiers generally use a tuned circuit as load. Such amplifiers are called as class - C tuned amplifiers.
2. These amplifiers are used as the collector modulator to produce the amplitude modulated signal.

Q. 7 Define oscillator.

Ans. :

Oscillators are basically ac signal generators which you use in laboratories. Oscillators generate sinusoidal voltage of desired magnitude, at desired frequency.

The output voltage and frequency of an oscillator can be variable. The oscillator operates on a dc power supply + V volts and more importantly it produces an alternating output voltage without any alternating signal applied at its input.

Oscillators operate on the principle of positive feedback..

Q. 8 Explain the concept of positive feedback.

Ans. :

There are two types of feedback, namely negative and positive feedback.

The positive feedback is used in oscillators. The concept of positive feedback can be explained with the help of Fig. 2.5.

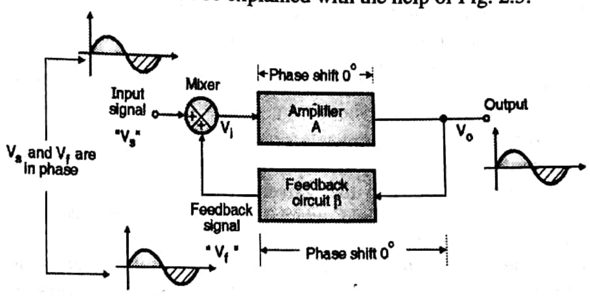


Fig. 2.5. : Principle of positive feedback

An oscillator is an amplifier with positive feedback incorporated in it. A part of the output is fed back through the feedback network and mixer to the amplifier input. The feedback signal is "in phase" with the original input signal as shown in Fig. 2.5. as the phase shift introduced by the feedback network is 0°. The amplifier in Fig. 2.5. is assumed to be a non-inverting amplifier with a voltage gain A which introduces a zero phase shift between its input and output.

Assuming that a sinusoidal input voltage V_s is being applied at the input of the non-inverting amplifier.

As the non-inverting amplifier is being used, there is no phase difference between V_s and V_o. So V_o is an amplified version of V_s with a 0° phase shift. A part of this output voltage is fed back (V_f) to the input side.

The feedback voltage V_f is in phase with the input voltage V_s as the feedback network does not introduce any phase shift. But the feedback voltage amplitude can be adjusted by changing the value of feedback factor β.

$$\therefore \; V_f = \beta V_o$$

if we use an inverting amplifier which introduces a 180° phase shift between V_s and V_o, then the feedback network should introduce a phase shift of 180° in order to bring V_s and V_f in phase with each other, as shown in Fig. 2.6.

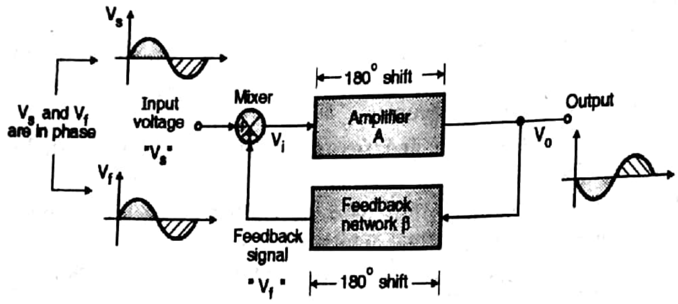


Fig. 2.6 : Principle of positive feedback

Q. 9 State the principle of oscillator.

Ans. :

An oscillator is a circuit which produces a continuous, repeated, alternating waveform without any input. Oscillators basically convert unidirectional current flow from a DC source into an alternating waveform which is of the desired frequency, as decided by its circuit components. The basic principle behind the working of oscillators can be understood by analyzing the behavior of a LC tank circuit shown by Figure 1, which employs an inductor L and a completely pre-charged capacitor C as its components. Here, at first, the capacitor starts to discharge via the inductor, which results in the conversion of its electrical energy into the electromagnetic field, which can be stored in the inductor. Once the capacitor discharges completely, there will be no current flow in the circuit. However, by then, the stored electromagnetic field would have generated a back-emf which results in the flow of current through the circuit in the same direction as that of before.

Q. 10 Explain the Barkhausen criteria.

Ans. :

The Barkhausen criteria should be satisfied by an amplifier with positive feedback to ensure the sustained oscillations.

For an oscillator circuit, there is no input signal "V_s", hence the feedback signal V_f itself should be sufficient to maintain the oscillations.

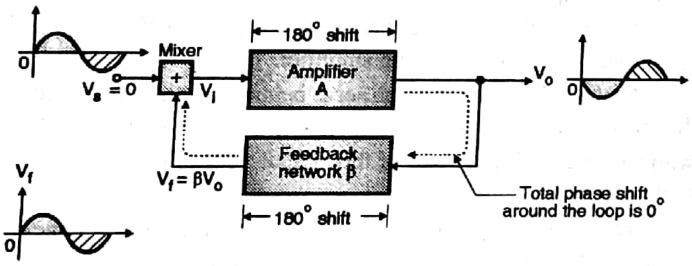


Fig. 2.7 : Block diagram of an oscillator

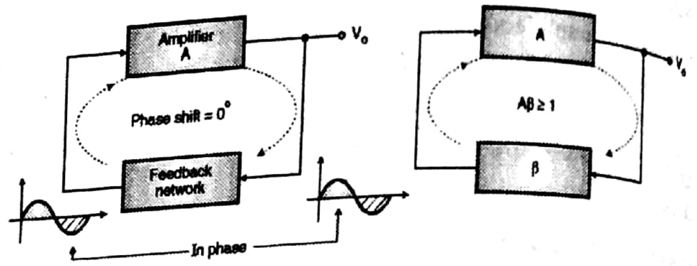
Statement of Barkhausencriterion :

The Barkhausen criterion states that :

1. An oscillator will operate at that frequency for which the total phase shift introduced, as measured from the input terminals, through the amplifier and feedback network and back again to the input is precisely equal to 0° or 360° or integral multiple of 360°.
2. At the oscillator frequency, the magnitude of the product of open loop gain of the amplifier A and the feedback factor β is equal to or greater than unity.

$$\therefore \; |A\beta| \geq 1$$

The product Aβ is called as the "loop gain". These conditions are diagrammatically illustrated in Figs. 2.8(a) and (b).



(a) The phase shift around the loop is 0° (b) Loop gain Aβ≥1

Fig. 2.8. :Barkhausen criterion

Q. 11 Explain the operation of Hartley oscillator and state its applications.

Ans. :

The circuit diagram of a transistorized Hartley oscillator is as shown in Fig. 2.9.

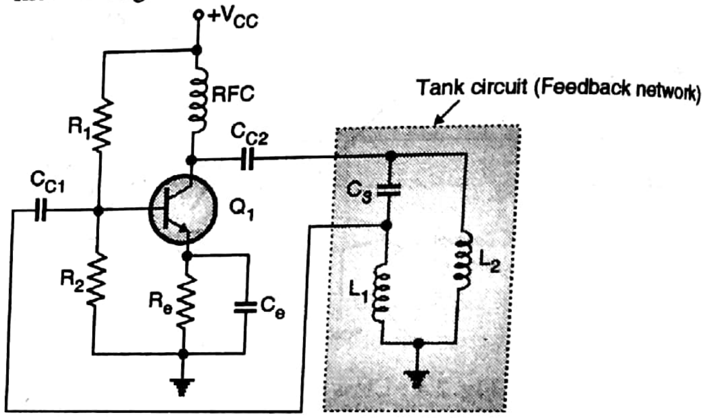


Fig. 2.9: Transistorized Hartley oscillator

Operation of the circuit :

R_1, R_2 and R_e are the resistances for the biasing of the transistor. C_{C1} and C_{C2} are the coupling capacitors, and C_e bypasses R_e. The feedback circuit is formed by the components L_1, L_2 and L_3.

The amplifier is in CE configuration. Hence it provides 180° phase shift between its input and output. The feedback circuit provides an additional 180° phase shift to satisfy the condition for the positive feedback.

The frequency of oscillations is given by :

$$f = \frac{1}{2\pi\sqrt{C_3(L_1+L_2)}}$$

where $(L_1 + L_2)$ is the equivalent inductance.

$$\therefore \quad f = \frac{1}{2\pi\sqrt{C_3 L_{eq.}}}$$

Here not considered the mutual inductance between the two inductances L_1 and L_2. In practice the inductors L_1 and L_2 are wound on the same core so we can't neglect the mutual inductance M present between them. So considering the mutual inductance, the equivalent inductance $L_{eq.}$ is given by,

$$L_{eq.} = L_1 + L_2 + 2M$$

The oscillator frequency can be varied by varying the capacitor C_3. The frequency variation over a wide range can be easily obtained.

Applications :

1. It is used as local oscillator in radio and TV receivers.
2. In the function generators.
3. In RF sources.

Q. 12 Explain the operation of Colpitt's oscillator and state its applications.

Ans. :

The resistors R_1, R_2 and R_e will provide the biasing for the transistor. C_{C1} and C_{C2} are the coupling capacitors and C_e is the bypass capacitor.

The transistor is connected in the CE configuration. Therefore it introduces a phase shift of 180° between its input and output.

The feedback network will provide additional 180° phase shift so as to make the total shift equal to zero. This will satisfy the condition for the positive feedback.

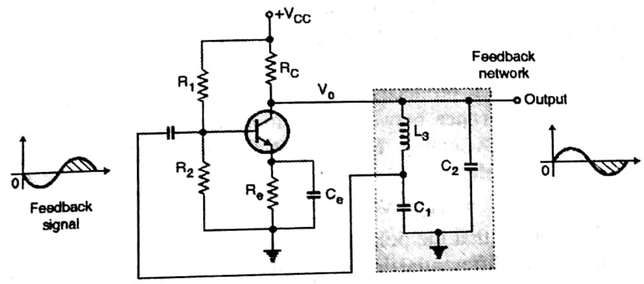


Fig. 2.10: Colpitt's oscillator using a BJT

The frequency of oscillations is given by,

$$f = \frac{1}{2\pi\sqrt{L_3 C}}$$

$$\text{where,} \quad C = \frac{C_1 C_2}{C_1 + C_2}$$

As in the case of transistor phase shift oscillator, the transistor current gain is important. The circuit analysis gives the minimum value of h_{fe} as,

$$h_{fe(min)} = \frac{C_1}{C_2}$$

The behaviour of the Colpitt's oscillator is very similar to that of the Hartley oscillator for the simple reason that both of them use the same basic LC oscillator except for the phase shifting network.

Application :

1. As a high frequency generator

Q. 13 Explain the principle of crystal oscillators.

Ans. :

Principle of crystal oscillators :

Certain materials such as quartz exhibit a unique property called "piezo electric" property.

It states that if mechanical force is applied to a quartz crystal then it generates electric potential.

Also if electric field is applied to a crystal it vibrates mechanically. If we apply mechanical vibrations to a quartz crystal then under proper operating conditions we can obtain electrical oscillations from it.

Q. 14 State advantages of crystal oscillators.

Ans. :

1. Very high frequency stability.
2. Very low frequency drift due to change in temperature and other parameters.
3. It is possible to obtain very high, precise and stable frequency of oscillations.
4. The Q is very high.
5. It is possible to obtain frequencies, higher than the fundamental frequency by operating the crystal in the overtone mode.

Q. 15 Compare LC and crystal oscillators.

Ans. :

Sr. No.	LC oscillators	Crystal oscillators
1.	Frequency of oscillations is dependent on values of L and C.	Frequency of oscillations depends on the dimensions of crystal.
2.	These are preferred at high frequencies.	Preferred at high frequencies.
3.	Hartley, Colpitt's and Clapp oscillators are the examples of LC oscillators.	Miller crystal oscillator and pierce crystal oscillator are the examples.
4.	Poor frequency stability except for the clapp oscillator.	Very high frequency stability.
5.	Used in radio, TV as high frequency sources, frequency synthesizers.	Crystal clock, frequency synthesizer, special type receivers are the applications.

Q. 1 Draw block diagram of typical OP-AMP. Explain function of each block.
May 03, May 04, May 05, Dec. 05, May 09, Dec. 09, May 10

Ans. :

The block diagram of a typical OP-AMP is shown in Fig. 3.1.

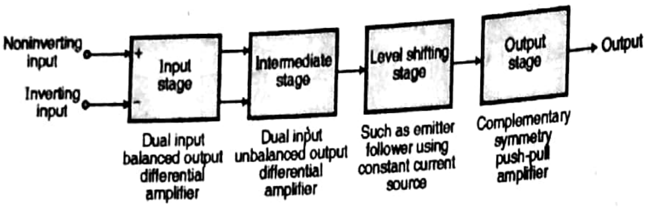


Fig. 3.1 : Block diagram of a typical OP-AMP

The OP-AMP is basically a differential amplifier i.e. it will amplify the voltage which is differentially present between its input terminals.

Input stage :

The input stage is a dual-input, balanced output differential amplifier. The two inputs are inverting and non-inverting input terminals. This stage provides most of the voltage gain of the OP-AMP and decides the value of input resistance R_i.

Intermediate stage :

This is usually another differential amplifier. The input stage drives the stage. This stage is a dual-input unbalanced output (single ended output) differential amplifier.

Level shifting stage :

Due to the direct coupling used between the first two stages, the input of level shifting stage is an amplified signal with some non-zero dc level. Level shifting stage is **used to bring this dc level to zero volts with respect to ground.**

Output stage :

This stage is normally a complementary output stage. It **increases the magnitude of voltage and raises the current supplying capability of the OP-AMP.** It also ensures that the output resistance of OPAMP is low.

Q. 2 Draw and explain equivalent circuit of an ideal op-amp.

Ans. :

Fig. 3.2 shows the equivalent circuit of a practical OP-AMP. It includes important values such as A_V, R_i, R_o etc.

A_V, V_d is the equivalent Thevenin voltage source and R_o is the Thevenin equivalent resistance looking back into the output terminal of an OP-AMP.

The value of input resistance R_i is finite but very high here and that of the output resistance R_o is non-zero because the OP-AMP is non-ideal.

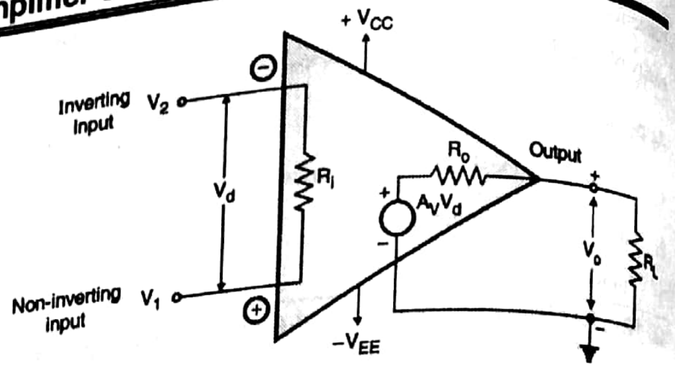


Fig. 3.2 : Equivalent circuit of an OP-AMP

The equivalent circuit of Fig. 3.2 is made of the differential input resistance R_i, the voltage gain A_V and the output resistance R_o. As will be discussed later, the parameters R_i, A_V and R_o are called as the **open-loop parameters.**

$V_d = (V_1 - V_2)$ is called as the differential input voltage, and A_V is called as the open loop gain. Hence the output voltage is given by,

$$V_o = A_V \times V_d = A_V (V_1 - V_2)$$

V_1 and V_2 are the voltages at the non-inverting and inverting input terminals of the OP-AMP, with respect to ground. Since both the input terminals are allowed to be connected to independent potentials, with respect to ground, the input side of the OP-AMP is said to be of the **Double Ended** type. But the output is **single ended** type.

That means it produces output voltage which is proportional only to the **difference** between the input voltages and not to the individual input voltages. **Hence OP-AMPs** are also called as **Difference amplifiers.**

$$V_o = A_V (V_1 - V_2)$$

This shows that the polarity of output voltage depends on the polarity of the differential input signal V_d. The differential input voltage V_d is given by,

$$V_d = \frac{V_o}{A_V}$$

This expression gives use of the value of only the differential input voltage V_d and does not give the individual input voltages V_1 and V_2. The open loop voltage gain A_V is of very large value. Hence the value of V_d even for maximum output voltage is extremely small. For example, to obtain $V_{o\,(max)} = 10$ V a 741 OP-AMP needs

$$V_d = \frac{10}{2 \times 10^5} = 50\ \mu V$$

Thus it need a very small differential input voltage V_d to obtain the maximum possible output voltage

Q. 3 With respect to op-amp explain the ideal characteristics and concept of virtual ground. Explain how op-amp can be used as an averaging amplifier in inverting configuration. Also draw neat circuit diagrams to :

1. **Convert sine wave to square wave using op-amp.**
2. **Detect the crossing of zero's in the generated square wave.** **Dec. 15**

Ans. : The equivalent circuit of an ideal OP-AMP is as shown in Fig. 3.3.to minimise the loading on the input source, a well designed voltage amplifier must draw negligible current from the source and must present a negligible resistance to the output load.

The OP-AMP being a voltage amplifier is no exception, to these requirements.

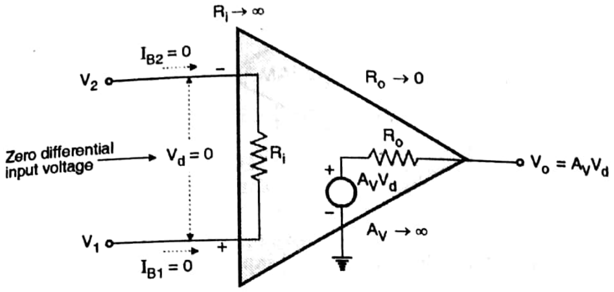


Fig. 3.3. : Equivalent circuit of an ideal OP-AMP

In order to define all the important characteristics of an ideal OP-AMP, refer to Fig. 3.3. in which V_1 and V_2 are the two input signals, V_o is the output voltage, A_V is the open loop voltage gain and R_i is the differential input resistance of the ideal OP-AMP.

V_d is the differential input voltage and I_{B1} and I_{B2} are the currents flowing into the two input terminals. The important characteristics of an ideal OP-AMP are as follows :

1. Infinite voltage gain ($A_V = \infty$) :

The open loop (without any feedback) gain of an ideal OP-AMP is denoted by A_V. It is the differential voltage gain and its value for an ideal OP-AMP is ∞.

$$V_o = A_V \times V_d$$

As $A_V = \infty$, the differential voltage V_d required to obtain the maximum output voltage ($\pm V_{CC}$) will be negligible i.e. $V_d \approx 0V$.

2. Infinite input resistance ($R_i = \infty$) :

The input resistance is also called as differential input resistance and it is defined as the total resistance measured between the two input terminals of an op-amp.

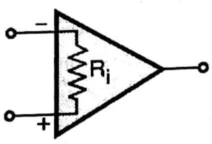


Fig. 3.4. : Differential input resistance

The input resistance R_i of an ideal OP-AMP is infinite. Due to this, the current flowing in each input terminal will be zero i.e. $I_{B1} = I_{B2} = 0$.

Due to infinite input resistance, almost any source can drive it and the source does not get loaded.

3. Zero output resistance ($R_o = 0$) :

The output resistance of an OP-AMP is the resistance viewed from its output terminal as shown in Fig. 3.5.

The output resistance R_o of an ideal OP-AMP is zero. Due to zero output resistance, the ideal OP-AMP can drive infinite number of other devices.

Further there will not be any change in its output voltage due to change in load current. That means its voltage regulation will be good.

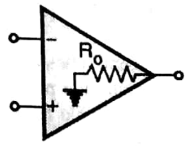


Fig. 3.5. : Output resistance

4. Zero offset voltage :

In practical OP-AMPs a small output voltage is present even though both the inputs V_1 and V_2 are having a zero value. This voltage is called as the offset voltage. For ideal OP-AMPs the offset voltage is zero. That means output voltage will be zero when input voltage is zero.

5. Infinite bandwidth :

Bandwidth of an amplifier is the range of frequencies over which all the signal frequencies are amplified almost equally. The bandwidth of an ideal OP-AMP is infinite. So it can amplify any frequency signal from 0 to ∞ Hz. Thus the gain of an ideal amplifier is constant from 0 frequency (dc signal) to ∞ Hz.

6. Infinite CMRR :

For an OP-AMP, the Common Mode Rejection Ratio (CMRR) is defined as the ratio of differential gain to common mode gain. CMRR is ∞ for an ideal OP-AMP. Thus the output voltage corresponding to the common mode noise is zero.

7. Infinite slew rate ($S = \infty$) :

The slew rate of an ideal OP-AMP is infinite so that the output voltage changes occur simultaneously with the input voltage changes. The slew rate definition and its significance have been explained later.

8. Zero power supply rejection ratio (PSRR = 0) :

PSRR is a parameter which specifies the degree of dependence of the OP-AMP output, on the changes in power supply voltage. For an ideal OP-AMP, PSRR = 0. That means the output voltage does not change due to fluctuations in supply voltage. The characteristics of an ideal OP-AMP are tabulated in Table 3.1.

Table 3.1 : Characteristics of an ideal OP-AMP

Sr. No.	Characteristics	Ideal value
1.	Input resistance	$R_i = \infty \Omega$
2.	Output resistance	$R_o = 0\ \Omega$
3.	Voltage gain	$A_V = \infty$
4.	Bandwidth	$B = \infty$

Sr. No.	Characteristics	Ideal value
5.	Common mode rejection ratio	CMRR = ∞
6.	Slew rate	∞
7.	Offset voltage	0
8.	Power supply rejection ratio (PSRR)	0

Q. 4 Explain two static and two dynamic parameters of OP-AMP. May 12

Ans. : The OP-AMP characteristics are important in practice because it can be use to compare the performance of various OP-AMP ICs and select the best suitable from them for the required application. OP-AMP characteristics are classified into two categories namely DC characteristics and AC characteristics. The DC characteristics include input bias current, input offset current, input offset voltage and thermal drift whereas the AC characteristics include the frequency response, stability of OP-AMP, frequency compensation, slew rate etc.

Open Loop Gain (A_V) :

Open loop gain of a practical OP-AMP is not infinite. It is in the range of a few thousands. The open loop gain of IC 741 is 2×10^5.

Input Resistance (R_i) :

Input resistance of practical OP-AMP is few MΩ. For IC 741 the input resistance is 2 MΩ. For OP-AMPs having FET differential input stage the input resistance can be in the GΩ range ($1\ G\Omega = 1 \times 10^9\ \Omega$).

Output Resistance (R_o) :

Output resistance of a practical OP-AMP is few ohms. For IC 741 the output resistance is 75 Ω.

Bandwidth :

Practical OP-AMPs do not have infinite bandwidth. They have a bandwidth of few hundred kHz. Bandwidth of IC 741 is 1 MHz.

Q. 5 Explain input offset voltage, CMRR and SVRR for operational amplifier. May 14

Ans. : Ideally, for a zero input voltage, the OP-AMP output voltage should be zero. But practically it is not so. This is due to the unavoidable imbalances inside the OP-AMP, specially the imbalances in its differential input stage.

So apply a small differential voltage at the input of the OP-AMP to make the output voltage zero. This voltage is called as input offset voltage.

The input offset voltage is denoted by V_{ios}. The input offset voltage is normally in a few mV range. The value of input offset voltage is temperature dependent. Ideally V_{ios} should be equal to zero.

Q. 6 For OP-AMP explain bias current. May 04

Ans. :

Input bias current I_B is defined as the average of the currents flowing into the two input terminals of the OP-AMP i.e. I_{B1} and I_{B2} as shown in Fig. 3.6.

$$\therefore \text{ Input bias current } I_B = \frac{I_{B1} + I_{B2}}{2}$$

Ideally the currents I_{B1} and I_{B2} must be zero. But for practical OP-AMP they do exist due to the finite value of input resistance R_i. Due to slight difference in the characteristics of the transistors used in the input stage of an OP-AMP, the two currents I_{B1} and I_{B2} are not equal.

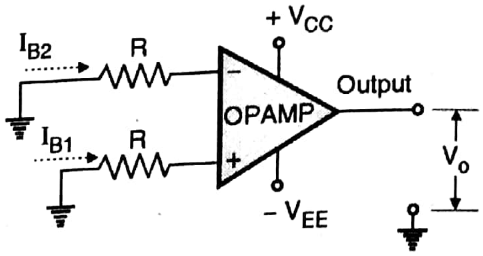


Fig. 3.6 : Bias current and input offset current

The maximum value of I_B is 50 nA for IC 741. It can be reduced to pA level using FET OP-AMPs. The value of input bias current is temperature dependent.

Q. 7 For Op-Amp explain input offset current and give practical value. Dec. 05

Ans. : The input offset current I_{ios} of an OPAMP is defined as the algebraic difference between the currents flowing into its inverting and non-inverting terminals. Mathematically it is expressed as,

$$I_{ios} = |I_{B1} - I_{B2}|$$

Where I_{B1} is the current flowing into the non-inverting input and I_{B2} is the current flowing into the inverting terminal

Ideally, the input offset current must be zero and practically it should be as small as possible. The input offset current exists due to the unequal currents I_{B1} and I_{B2} flowing into the input terminals of the OP-AMP. The values of I_{B1} and I_{B2} are different from each other because the transistors used in the input stage of an OP-AMP are not exactly identical (or matched). The input offset currents for the BJT OP-AMPs is few tens or hundreds of nA. For IC 741C the maximum input offset current is 6 nA. For the FET OP-AMPs it is of the order of few pA.

The input offset current will give rise to a finite output voltage even when the input voltage is zero. That is why it should be reduced to a lowest possible value. The input offset current is temperature dependent.

Q. 8 Explain input offset voltage, CMRR and SVRR for operational amplifier. May 14

Ans. : The change in an OP-AMPs input offset voltage (V_{ios}) due to variation in the supply voltage is called as Power Supply Rejection Ratio (PSRR). It is also called as Supply Voltage Rejection Ratio (SVRR) or Power Supply Sensitivity (PSS). Mathematically PSRR is expressed as,

$$PSRR = \frac{\Delta V_{ios}}{\Delta V}$$

where, ΔV_{ios} = Change in input offset voltage and ΔV = Change in the supply voltage.

PSRR is expressed either in microvolts per volt or in decibels.

For IC 741C, PSRR = 150 μV/V. The value of PSRR should ideally be equal to zero and practically it should be as small as possible.

SVRR is specified assuming that both supply (positive and negative) magnitudes increasing or decreasing simultaneously.

Total Input Offset Voltage :

Various parameters which affect the value of V_{ios} are temperature, CMRR, PSRR etc. The combined effect of these parameters on V_{ios} is expressed in the following equation :

$$V_{ios\,(total)} = V_{ios\,(initial)} + T.C.\,(V_{ios}) \times \Delta T + \frac{\Delta V_1}{CMRR} + \frac{\Delta V}{PSRR} + \frac{\Delta V_o}{A_d}$$

Where $V_{ios\,(initial)}$ = Initial input offset voltage (at ambient temperature)

Q. 9 List down various parameters of Op amp alongwith their typical values for IC741. Also explain what the significance of CMRR and slew rate is ? May 16

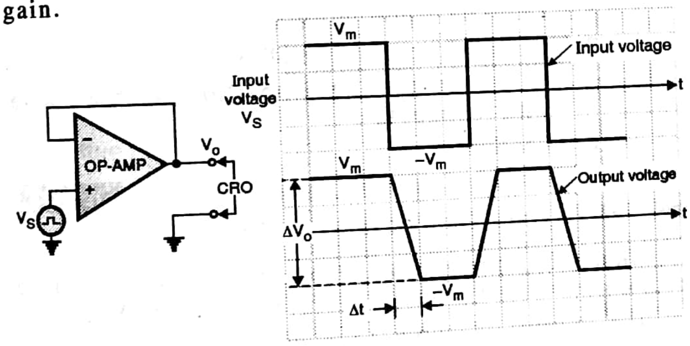
Ans. : Slew rate is defined as the maximum rate of change of output voltage per unit time. Mathematically it is expressed as follows :

$$S = \left.\frac{dV_o}{dt}\right|_{maximum} \quad \text{Volts/}\mu\text{sec}$$

The unit of slew rate are Volts/microseconds.

Importance of slew rate :

Slew rate decides the capability of OP-AMP to change its output rapidly, hence it decides the highest frequency of operation of a given OP-AMP. The value of slew rate depends on the change in voltage gain. Therefore it is generally specified at unity (+ 1) gain.



(a) **Circuit to measure slew rate** (b) **Input and output voltage waveform**

Fig. 3.7

Slew rate should be ideally ∞ and practically as high as possible. Slew rate of 741 OP-AMP is only 0.5 V/μS, which is its biggest drawback. Therefore it cannot be used for high frequency applications. The circuit used for slew rate measurement is shown in Fig. 3.7(a) and the input output voltage waveforms are shown in Fig. 3.7(b). The slew rate is measured from the output voltage waveform as :

$$\text{Slew rate } S = \frac{\Delta V_o}{\Delta t}$$

Ideally the slew rate should be infinite and practically it should be as large as possible.

Use of slew rate to calculate f_m :

Assuming that the input signal V_S is a sinewave, obtain the value of maximum frequency for which the amplifier produces an undistorted output.

Let, $V_S = V_m \sin \omega t$

As we have used a unity gain non-inverting amplifier, the output is exactly equal to the input without any phase shift.

$$\therefore \quad V_o = V_m \sin \omega t$$

Differentiating both sides ,

$$\therefore \quad \frac{dV_o}{dt} = \omega V_m \cos \omega t$$

$\frac{dV_o}{dt}$ will be maximum when $\cos \omega t = 1$ and maximum value of $\frac{dV_o}{dt}$ is nothing but slew rate S.

$$\therefore \quad S = \frac{dV_o}{dt}\,(max) = \omega V_m = 2\pi f_m V_m \text{ V/sec.}$$

$$\therefore \quad f_m = \frac{S}{2\pi V_m}$$

This is the maximum frequency f_m for which the amplifier produces an undistorted output. It is also called as full power bandwidth (FPB).

Q. 10 For Op-Amp explain unity gain bandwidth product and give practical values. Dec. 05

Ans. : The gain bandwidth product (GB) is the bandwidth of the OP-AMP corresponding to the voltage gain of unity (1). The graph of open loop gain versus frequency of IC 741 is shown in Fig. 3.8. It shows that GB = 1 MHz. The other terms used for GB are closed loop bandwidth, unity gain bandwidth and small signal bandwidth.

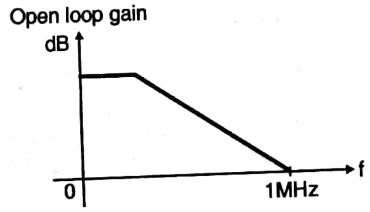


Fig. 3.8 : A graph of open loop gain versus frequency of IC 741

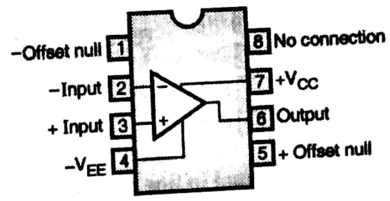
Q. 11 Give technical specifications of OP-AMP 741 with typical values of parameter. May 03. May 05

Ans. : This is one of the oldest and one of the most popular OP-AMP IC. It is a high performance monolithic operational amplifier. It can be used in various analog applications such as integrator, differentiator, summing amplifier etc. Features of this OP-AMP IC are as follows :

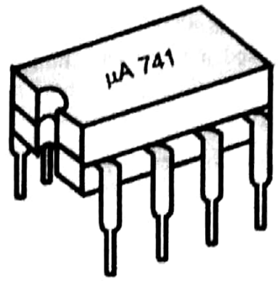
Features of IC 741 : No frequency compensation required. Short circuit protection has been provided. It has the offset voltage null capability. Large common mode and differential voltage ranges. There are no latch ups.

Pin configuration and pin functions :

The pin configuration of OP-AMP IC 741 in the 8 pin mini DIP package is shown in Fig. 3.9(a).



(a) Pin configuration of IC 741 OP-AMP

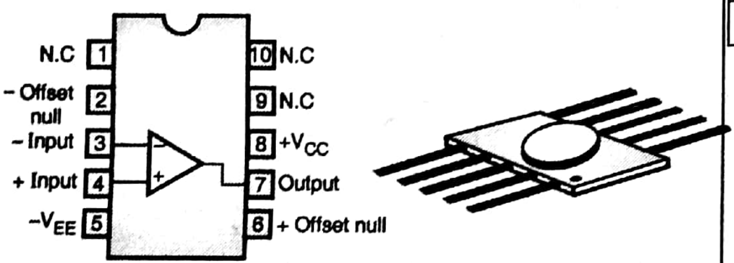


(b) 8 pin mini DIP package for IC 741
Fig. 3.9

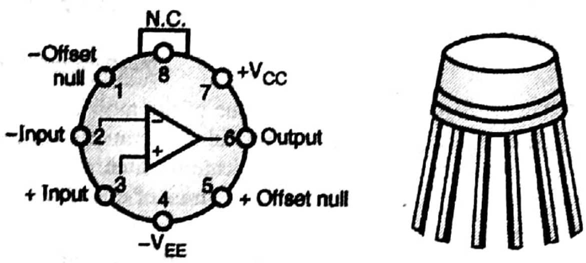
This IC is an 8 pin IC in the dual in line package (DIP). Pin number 7 is for connecting the positive supply voltage $+V_{CC}$ while pin number 4 is to be connected to a negative supply voltage. Thus IC 741 needs a dual polarity power supply. Pin numbers 2 and 3 are inverting and non-inverting inputs respectively and we get the output voltage at pin number 6. Pin numbers 1 and 5 can be used to nullify the offset voltage and pin number 8 is a dummy pin which is not connected anywhere and hence should be left open (unconnected). Even though the mini DIP is the most popular package, the OP-AMP IC 741 is available in some other packages too. They are as follows :

Other packages of OP-AMP :

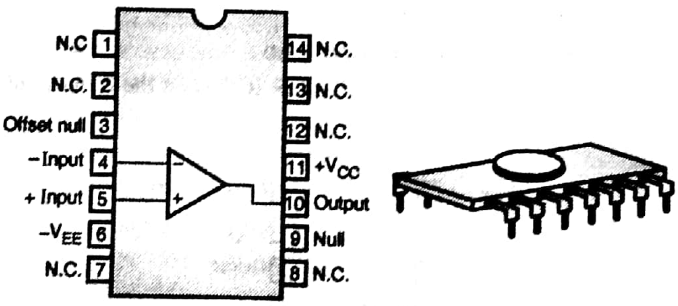
(a) 10 pin flatpack. (b) 8 pin metal can. (c) 14 pin DIP.



(a) 10 pin flatpack



(b) 8 pin-metal can package



(c) 14 lead dual in line package
Fig. 3.10 : Various packages of OP-AMP IC 741

Q. 12 List down various parameters of Op amp alongwith their typical values for IC741. Also explain what the significance of CMRR and slew rate is ? **May 16**

Ans. : The typical values of different important characteristics are listed in Table 3.2, alongwith the ideal values of those characteristics.

Table 3.2 : Important characteristics of IC 741 alongwith those of ideal OP-AMP

Sr. No.	Characteristics	Value for IC 741	Ideal value
1.	Input resistance R_i	2 MΩ	∞
2.	Output resistance R_o	75 Ω	0
3.	Voltage gain A_V	2×10^5	∞
4.	Bandwidth B.W.	1 MHz	∞
5.	CMRR	90 dB	∞
6.	Slew rate S	0.5 V/μS	∞
7.	Input offset voltage V_{ios}	2 mV	0
8.	PSRR	150 μV/V	0
9.	Input bias current I_B	50 nA	0
10.	Input offset current I_{ios}	6 nA	0

Q. 13 Draw :

1. **Open loop configuration**
2. **Closed loop configuration**

With respect to op-amp. Compare the above with respect to :

1. **Feedback**
2. **Ideal and practical gain.** **Dec. 11**

Ans. : Open Loop Configuration of OP-AMP :

All the ideal characteristics of OP-AMP discussed earlier are corresponding to the "open loop operation" of the OP-AMP.

The meaning of "open loop operation" is that there is absolutely no feedback present from the output to input. That means no part of output gets connected to the input.

The configuration of Fig. 3.11 shows the connections of OP-AMP in the "Open Loop" mode. Input signals V_1 and V_2 are applied to the non-inverting and inverting terminals respectively.

The differential signal present between the two inputs is $V_d = (V_1 - V_2)$.

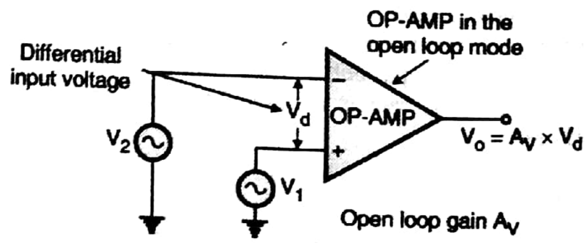


Fig. 3.11 : Open loop configuration

The open loop gain of the OP-AMP is A_V. Its value is very high. The output voltage is given by,

$$V_o = A_V \times V_d = A_V (V_1 - V_2) = \pm V_{sat}$$

Depending on the polarity of V_d, the output voltage will be either equal to $+V_{sat}$ or $-V_{sat}$.

As A_V is very high, a very small value of V_d will drive the OP-AMP into positive or negative saturation. So the output voltage is not proportional to the differential input voltage. It is therefore not possible to use the open loop configuration for linear amplification.

Features of open loop configuration :

1. No feedback.
2. Very high voltage gain.
3. Can not be used as a linear amplifier.
4. Used in comparators.
5. Waveform distortion takes place.
6. Very high input resistance
7. Low output resistance
8. Large bandwidth

In the closed loop configurations, some kind of "feedback" is introduced in the circuit. A part of output is returned back or fed back to the input.

Types of feedback :

The feedback can be one of the following two types :

1. Positive feedback or regenerative feedback OR
2. Negative feedback or degenerative feedback.

Positive Feedback :

If the feedback signal and the original input signal are in phase with each other then it is called as the positive feedback.

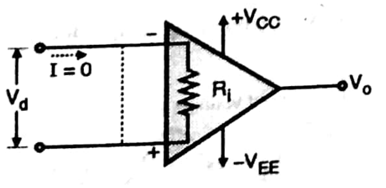
Positive feedback is used in the applications such as "Oscillators" and Schmitt triggers or regenerative comparators.

Negative Feedback :

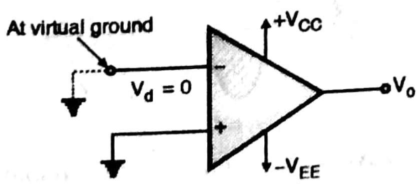
If the feedback signal and the original input signal are 180° out of phase with respect to each other, then it is called as negative feedback. Negative feedback is used in all the amplifiers.

Q. 14 Explain virtual short and virtual ground concept. Dec. 12

Ans. : According to virtual short concept, the potential difference between the two input terminals of an OP-AMP is almost zero. In other words both the input terminals are approximately at the same potential.



(a) Concept of virtual short



(b) Concept of virtual ground

Fig. 3.12

Virtual Short :

The concept of virtual short can be explained as follows :

Assume that the OP-AMP is ideal one. The input impedance (R_i) of an OP-AMP is ideally infinite. Hence current "I" flowing from one input terminal to the other will be zero as shown in Fig. 3.12. Thus the voltage drop across R_i will be zero and both the input terminals will be at the same potential. In other words they are virtually shorted to each other. Other way of explaining this concept is as follows :

The output voltage of an OP-AMP is given by,

$$V_o = A_V \cdot V_d$$

where,

A_V = Open loop gain and V_d = Differential input voltage

$$\therefore \quad V_d = \frac{V_o}{A_V}$$

But $A_V = \infty$ for an ideal OP-AMP and $A_V = 2 \times 10^5$ for IC 741.

$\therefore$ For ideal OP-AMP, $V_d = 0$. Thus the potential difference between the input terminals is zero.

When we short circuit two points, they will have the same potential. Due to this reason, the two OP-AMP terminals which are almost equipotential are said to be virtually (not actually) short circuited.

Q. 15 Explain the concept of virtual ground in operational amplifier. Dec. 16

Ans. : If the non-inverting (+) terminal of OP-AMP is connected to ground as shown in Fig. 3.13(b), then due to the "virtual short" existing between the two input terminals, the inverting (–) terminal will also be at ground potential.

Hence it is said to be at "virtual ground". Similarly if the inverting (–) terminal is connected to ground, then the non-inverting (+) terminal will be at "virtual ground" potential. The concept of virtual ground has been used extensively in analyzing various closed loop configurations, specially use this concept in the inverting amplifier analysis.

Zero Input Current :

As the input resistance of the ideal OP-AMP is infinite, the current flowing into its input terminals is zero. Even for the practical OP-AMPs such as 741C, $R_{in} = 2\ M\Omega$ which is very large. Hence for all the practical purposes we will assume that the input current of an OP-AMP is zero.

Closed Loop Amplifier Configurations :

1. Inverting amplifier 2. Non-inverting amplifier
3. Voltage follower.

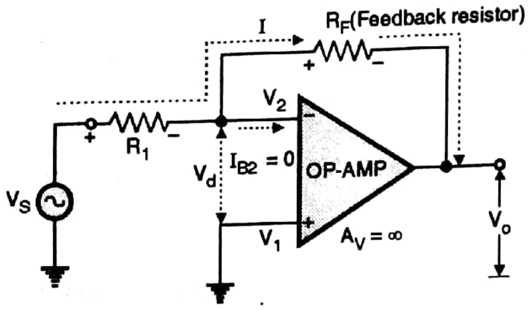
1 The Inverting Amplifier :

The circuit diagram of an inverting amplifier is as shown in Fig. 3.13(a). The signal which is to be amplified is applied at the inverting (–) input terminal of the OP-AMP. The amplified output signal will be 180° out of phase with the input signal. In other words the output signal is "inverted" as shown in Fig. 3.13(b). Therefore this amplifier is known as the inverting amplifier.

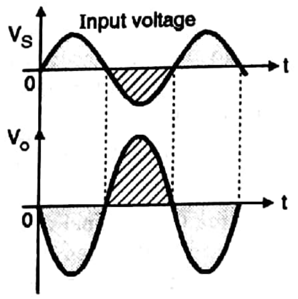
Operation :

The signal to be amplified (V_S) has been connected to the inverting terminal via the resistance R_1.

The other resistor R_F, connected between the output and inverting input terminals is called as the feedback resistance. It introduces a negative feedback. The non-inverting (+) input terminal is connected to ground. As the OP-AMP is an ideal one, its open loop voltage gain $A_V = -\infty$ and input resistance $R_i = \infty$. The negative sign for A_V is due to the inverting configuration. The input and output voltage waveforms are as shown in Fig. 3.13(b). Output is an amplified and inverted version of the input signal V_S.



(a) Inverting amplifier



(b) Waveforms of inverting amplifier

Fig. 3.13

Expression for the closed loop voltage gain :

$$V_o = |A_V| \times V_d \quad ...(1)$$

$$\therefore \quad V_d = \frac{V_o}{|A_V|} \quad ...(2)$$

where, A_V = Open loop gain of OP-AMP.

As A_V of an open loop OP-AMP is ∞.

$$\therefore \quad V_d = \frac{V_o}{\infty} = 0 \quad ...(3)$$

But, $V_d = V_1 - V_2$

$$V_1 - V_2 = 0 \quad ...(4)$$

As the non-inverting (+) input terminal is connected to ground, $V_1 = 0$. Substituting this value in Equation (4),

$$V_2 = 0$$

Thus V_2 is at ground potential.

Since the input resistance $R_i = \infty$, the current going into the OP-AMP will be zero. Therefore the current "I" that passes through R_1 will also pass through R_F as shown in Fig. 3.13(a). As the input voltage V_S is being measured with respect to ground and as V_2 is at ground potential we can say that the input voltage V_S is voltage across R_1 and voltage across R_F is output voltage.

The input voltage V_S is given by,

$$V_S = IR_1 \quad ...(5)$$

And the output voltage is given by,

$$V_o = -IR_F \quad ...(6)$$

write Equations (5) and (6) because V_2 is at approximately ground potential (virtual ground).

Closed loop gain $A_{VF} = \frac{V_o}{V_S}$

Substituting the expressions for V_o and V_S,

$$A_{VF} = \frac{I R_F}{I R_1} = -\frac{R_F}{R_1} \quad ...(7)$$

And $V_o = A_{VF} \times V_S$

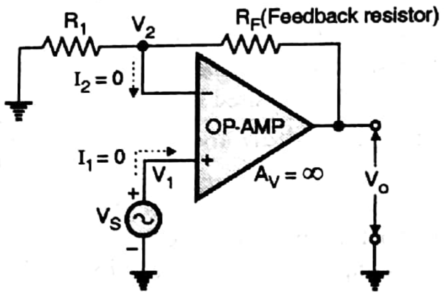
Q. 16 Derive the equation for voltage gain for a non-inverting amplifier. **Dec. 03**

Ans. : The non-inverting amplifier using OP-AMP is as shown in Fig. 3.14(a). Here the signal which is to be amplified is applied to the non-inverting (+) input terminal of the OP-AMP and the inverting (–) input terminal is connected to ground via resistance R_1. As shown in Fig. 3.14(b), the input and output voltages are in phase with each other. The negative feedback is introduced in this circuit via the feedback resistor R_F which is connected between the output and inverting (–) input terminal of OP-AMP.

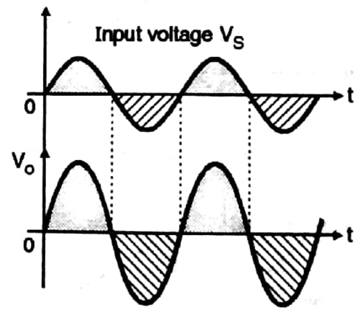
Expression for the closed loop voltage gain (A_{VF}) :

As we are using an ideal OP-AMP, $R_i = \infty$. Therefore the currents entering into both the input terminals of OP-AMP will have a zero value. ($I_1 = I_2 = 0$). Therefore voltage across R_1 is given by,

$$V_2 = \frac{R_1}{(R_F + R_1)} V_o$$



(a) Non-inverting amplifier



(b) Waveforms of non-inverting amplifier

Fig. 3.14

$$V_1 = V_S = V_2$$

Substituting the expression for V_2

$$\therefore \quad V_S = \frac{R_1}{(R_F + R_1)} V_o$$

Therefore the closed loop voltage gain A_{VF} is given as,

$$A_{VF} = \frac{V_o}{V_S} = \frac{R_1 + R_F}{R_1}$$

$$\therefore \quad A_{VF} = 1 + \frac{R_F}{R_1}$$

$$\text{and} \quad V_o = A_{VF} \times V_S$$

Table 3.3 : Comparison of inverting and non-inverting amplifiers

Sr. No.	Parameter	Inverting amplifier	Non-inverting amplifier
1.	Voltage gain.	$A_{VF} = -R_F / R_1$	$A_{VF} = 1 + \frac{R_F}{R_1}$
2.	Phase relation between input and output voltages.	180° out of phase.	In phase.
3.	Value of voltage gain.	Can be greater than, less than or equal to unity.	Always greater than or equal to unity.
4.	Input resistance.	Equal to R_1.	Very large.

Q. 17 With the circuit diagram explain voltage-follower and draw input-output waveforms. May 09, May 11

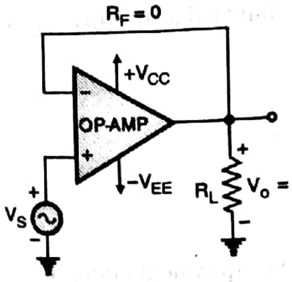
Ans. :

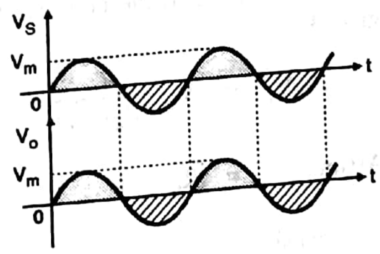
When $R_1 = \infty$ and $R_F = 0$ the non-inverting amplifier gets converted into a voltage follower or unity gain amplifier.

When the non-inverting amplifier is configured so as to obtain a gain of 1, it is called as a voltage follower or unity gain non-inverting buffer. The schematic diagram for a voltage follower is as shown in Fig. 3.15(a).

The voltage follower configuration of Fig. 3.15(a) is obtained by short circuiting R_F and open circuiting R_1 connected in the usual non-inverting amplifier configuration of Fig. 3.14(a).

Thus all the output voltage is fedback to the inverting input of the OP-AMP. Therefore the feedback factor of this circuit i.e. B = 1.





(a) Voltage follower circuit (b) Waveforms of voltage follower

Fig. 3.15

Closed loop gain (A_{VF}) :

Consider the expression for the closed loop gain of a non-inverting amplifier, that is,

$$A_{VF} = 1 + \frac{R_F}{R_1}$$

In this equation, substitute the values of $R_F = 0$ and $R_1 = \infty$ to get the closed loop gain of the voltage follower as :

$$A_{VF} = 1$$

Therefore the output voltage will be equal to and in phase with the input voltage, as shown in Fig. 3.15(b). Thus voltage follower is a non-inverting amplifier with a voltage gain of unity.

The unity gain amplifier does not behave like a conventional voltage amplifier but it acts as a **Resistance Transformer.**

Features of a Voltage Follower Circuit :

The important features of the voltage follower circuit are as follows :

1. Closed loop voltage gain equal to 1 i.e. output is equal to input with no phase shift.
2. Very high input impedance
3. Very low output impedance
4. Large bandwidth.

Q. 18 Explain how op-amp can be used as a summing, scaling and averaging amplifier in the inverting configuration ? Dec. 13

Ans. :

It is possible to apply more than one input signal to an inverting amplifier. This circuit will then add all these input signals to produce their addition at the output.

Such a circuit will then be called as an adder or a summing amplifier. Depending on the polarity or sign of the output voltage the adder circuits can be classified into two categories as :

1. Inverting adder and
2. Non-inverting adder.

Q. 19 Explain how operational amplifier can be used for addition of two AC signals with one DC signal ? Dec. 15

Ans. :

Fig. 3.16 shows the "inverting summing amplifier" configuration with three inputs V_1, V_2 and V_3. Depending on the relation between the feedback resistor R_F and the three input resistances R_1, R_2 and R_3

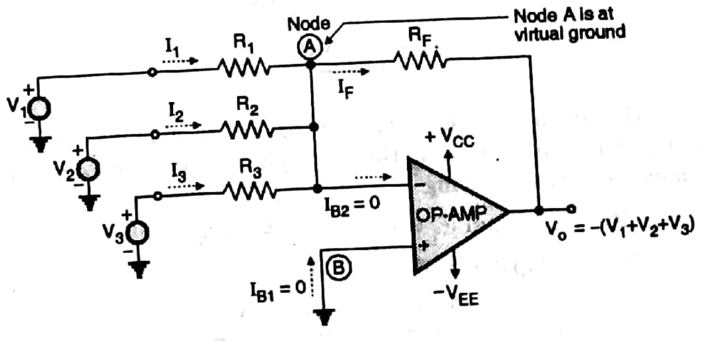


Fig. 3.16 : An inverting adder

V_1, V_2 and V_3 are three input signals applied simultaneously to the inverting terminal of the OP-AMP through resistors R_1, R and R_3 respectively. V_1, V_2 and V_3 are measured with respect t ground. R_F is the feedback resistor connected between the outpu terminal and the inverting input terminal of OP-AMP. The non inverting input terminal is connected to ground. So th configuration of Fig. 3.16 is basically an inverting amplifier with three inputs. Let the currents through the resistors R_1, R_2 and R_3 be I_1, I_2 and I_3 respectively.

Expression for the output voltage :

Apply KCL at node A of Fig. 3.16 to write,

$$I_1 + I_2 + I_3 = I_{B2} + I_F \quad ...(1)$$

But as R_i of the OP-AMP is ideally infinite, $I_{B2} = 0$ and $V_A = V_B = 0$ due to virtual ground concept.

Hence, $$I_1 + I_2 + I_3 = I_F \quad ...(2)$$

On the input side, $$I_1 = \frac{V_1 - V_A}{R_1} = \frac{V_1}{R_1} \text{ as } V_A = 0 \quad ...(3)$$

Similarly, $$I_2 = \frac{V_2 - V_A}{R_2} = \frac{V_2}{R_2} \text{ and } I_3 = \frac{V_3 - V_A}{R_3} = \frac{V_3}{R_3} \quad ...(4)$$

And on the output side,

$$I_F = \frac{V_A - V_o}{R_F} = -\frac{V_o}{R_F} \quad ...(5)$$

Substituting these values in Equation (2),

$$\left.\begin{aligned} \frac{V_1}{R_1} + \frac{V_2}{R_2} + \frac{V_3}{R_3} &= -\frac{V_o}{R_F} \\ \text{OR} \\ V_o &= -\left[\frac{R_F}{R_1}V_1 + \frac{R_F}{R_2}V_2 + \frac{R_F}{R_3}V_3\right] \end{aligned}\right\} \quad ...(6)$$

In Equation (6) if we substitute $R_F = R_1 = R_2 = R_3 = R$ then ,

$$V_o = -(V_1 + V_2 + V_3) \quad ...(7)$$

Thus output voltage is the negative sum of the input voltage. Therefore this circuit is called as "Inverting adder" or "Inverting summing amplifier".

Q. 20 Explain how operational amplifier can be used for taking average of three signals. May 16

Ans. :

The inverting adder circuit of Fig. 3.17 can be used as an averaging circuit by setting $R_1 = R_2 = R_3 = R$ and $R_F = R / 3$.

$$V_o = \frac{-R_F}{R}(V_1 + V_2 + V_3)$$

$$= \frac{-R/3}{R}(V_1 + V_2 + V_3)$$

$$\therefore V_o = \frac{-(V_1 + V_2 + V_3)}{3}$$

Thus the magnitude of output voltage is equal to the average of the three input voltages. This principle can be extended for n number of inputs by setting

$$R_F = R / n \text{ and } R_1 = R_2 = ... = R_n = R.$$

Then the output voltage is given by,

$$V_o = \frac{-(V_1 + V_2 + V_3 + ... + V_n)}{n}$$

Non-Inverting Adder :

An adder circuit which can produce the addition of its input signals without inversion (without negative sign) is called as the non-inverting adder or non-inverting summing amplifier. Such a configuration with two inputs V_a and V_b is shown in Fig. 3.17 the input signals and their resistors are connected to the non-inverting (+) input terminal of the OP-AMP. In short this is a non-inverting amplifier with two inputs.

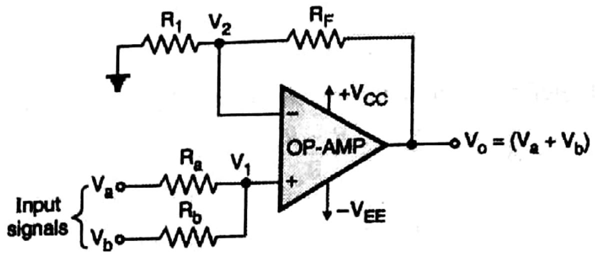
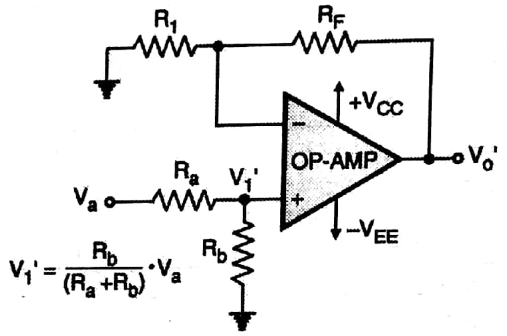


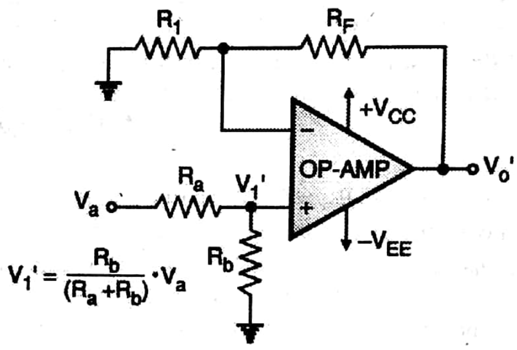
Fig. 3.17 : A non-inverting summing amplifier

Expression for output voltage (V_o) :

Assuming the input impedance of OP-AMP is very large. Therefore the current entering into the non-inverting terminal is zero.



(a) Equivalent circuit for $V_b = 0$



(b) Equivalent circuit for $V_a = 0$

Fig. 3.18

Let us use the superposition theorem to obtain the voltage V_1. So consider only one input at a time making the other input zero. Consider V_a only and make $V_b = 0$ i.e. short circuit it to ground. The equivalent circuit is as shown in Fig. 3.18(a).

The voltage at the non-inverting terminal due to only V_a is given by,

$$\therefore V_1' = V_a \cdot \frac{R_b}{(R_a + R_b)}$$

Assume $R_a = R_b = R \therefore V_1' = \frac{V_a}{2}$

Now consider V_b only and make $V_a = 0$. The equivalent circuit is as shown in Fig. 3.18(b).

$$\therefore V_1'' = \frac{R_a}{(R_a + R_b)} \cdot V_b = \frac{V_b}{2}$$

$$\therefore \; V_1 = V_1' + V_1'' \; \therefore \; V_1 = \frac{V_a + V_b}{2}$$

As the amplifier is non-inverting type, its gain is given by,

$$A_{VF} = 1 + \frac{R_F}{R_1}$$

If $R_F = R_1 = R$ then $A_{VF} = 1 + 1 = 2$

The output voltage, $V_o = A_{VF} \times V_1 = 2 \times \frac{(V_a + V_b)}{2}$

$$\therefore \; V_o = V_a + V_b$$

This expression for output voltage shows that the output voltage is the algebraic addition of the two input voltages. The positive sign of output voltage indicates that the inversion does not take place in this adder circuit and therefore it is called as the non-inverting adder.

Q. 21 Design a circuit with OP-AMP to produce the output V_o given by,

$V_o = (V_{s1} + V_{s3}) - (V_{s2} + V_{s4})$ May 13

Ans. :

The required circuit is shown in Fig.3.19

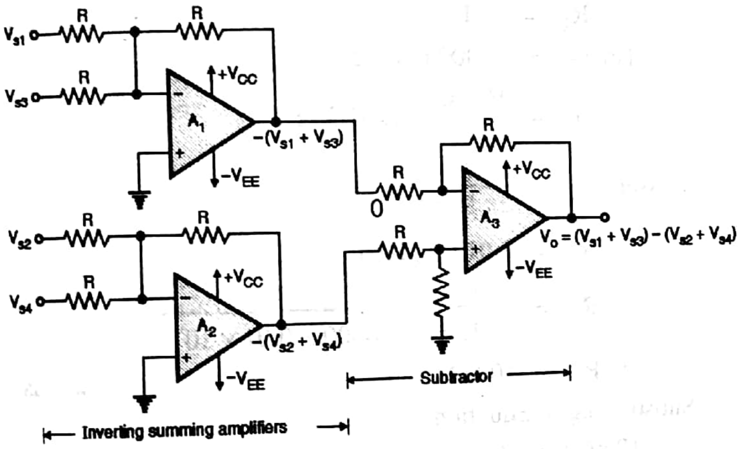


Fig. 3.19

Q.22 Give the circuit diagram with values of components to realize following relation.

$V_o = 2\,V_1 + 5\,V_2 + 7\,V_3 - V_4$ (Use OP-AMP 741)

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Ans. :

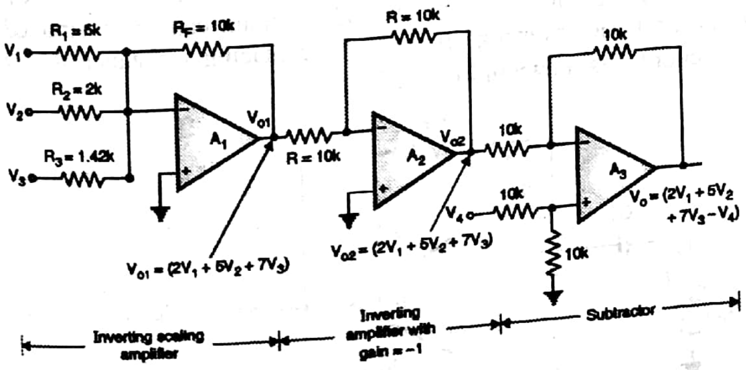


Fig. 3.20

The given expression can be realised by using three blocks namely the inverting scaling amplifier, a unity gain inverting amplifier and a subtractor.

1. Inverting scaling amplifier :

For the inverting scaling amplifier the expression for output voltage is

$$V_{o1} = \left[\frac{R_F}{R_1} V_1 + \frac{R_F}{R_2} V_2 + \frac{R_F}{R_3} V_3\right] \qquad ...(1)$$

Comparing it with the given expression,

$$\frac{R_F}{R_1} = 2, \quad \frac{R_F}{R_2} = 5 \quad \text{and} \quad \frac{R_F}{R_3} = 7$$

Let $R_F = 10\ k\Omega$

$\therefore \; R_1 = 5\ k\Omega, \; R_2 = 2\ k\Omega$ and $R_3 = 1.42\ k\Omega$

2. Unity gain inverting amplifier :

The gain of this block is $A_{VF} = -\frac{R_F}{R_1}$

want it to be -1.

$\therefore \; -R_F/R_1 = -1 \qquad \therefore \; R_F = R_1$

Choose $R_F = R_1 = 10\ k\Omega$

3. Subtractor :

For a subtractor, all the resistors should be of the same value. Hence choose all the resistors of $10\ k\Omega$ value.

Q. 23 A difference amplifier is to be designed to amplify the difference between two voltages by a factor of 20. The inputs each approximately equal to 2 V. Determine suitable resistor values for the circuit shown in Fig. 3.21 using a 741 opamp. Dec. 16

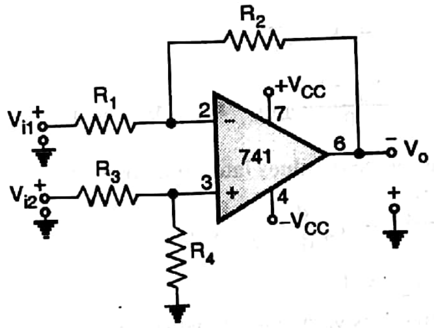


Fig. 3.21

Ans. : The gain of the difference amplifier is given by,

$$\text{Gain} = \frac{R_2}{R_1}$$

$$\therefore \; 20 = \frac{R_2}{R_1} \text{ or } R_2 = 20\,R_1.$$

Let $R_1 = 1\ k\Omega$ ...Ans.

$\therefore \; R_2 = 20\ k\Omega$ ...Ans.

For proper operation of the circuit,

$R_4 = R_2 = 20\ k\Omega$ ...Ans.

and $R_3 = R_1 = 1\ k\Omega$ ...Ans.

Q. 24 Sketch an op-amp integrating circuit together with the circuit waveforms. Explain in brief the circuit operation. Dec. 13

Ans. : The ideal integrator circuit is as shown in Fig. 3.22. This circuit is obtained by replacing the feedback resistor R_F in the inverting amplifier configuration by "C_F". Assume that the OP-AMP used here is an ideal one. V_2 is at virtual ground potential.

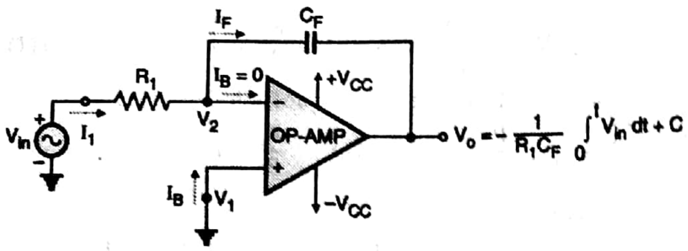


Fig. 3.22 : An ideal integrator circuit

Expression for the output voltage :

Referring to the Fig. 3.22, apply the Kirchhoff's current law at node V_2 to write,

$$I_1 = I_B + I_F \quad ...(1)$$

Due to high input impedance R_i of the OP-AMP, I_B will be negligible as compared to I_F.

$$\therefore \quad I_1 \approx I_F \quad ...2)$$

The basic relation between the current through and voltage across a capacitor is as follows :

$$I_C = C \cdot \frac{d V_c}{dt} \quad ...(3)$$

But in case of an integrator, $I_F = I_{CF}$. Therefore Equation (3) gets modified as :

$$I_1 = C_F \frac{dV_c}{dt} \quad ...(4)$$

But, $I_1 = \frac{V_{in} - V_2}{R_1}$ and $V_c = (V_2 - V_0)$.

Substituting these values into Equation (.4)

$$\frac{V_{in} - V_2}{R_1} = C_F \frac{d}{dt}(V_2 - V_0). \quad ...(5)$$

$$V_2 = 0 \quad ...(6)$$

Substituting this value into Equation (5),

$$\frac{V_{in}}{R_1} = C_F \frac{d}{dt}(-V_o) \quad ...(7)$$

The output voltage can be obtained by integrating the above equation as :

$$V_o = -\frac{1}{R_1 C_F} \int_0^t V_{in}\, dt + C \quad ...(8)$$

Where C is the constant of integration and it is proportional to the output voltage V_o at t = 0 seconds.

Q. 25 Find R_1 and R_2 in the lossy integrator so that the peak gain is 20 dB and the gain is 3 dB down from its peak when ω = 10,000 rad/s. Use capacitance of 0.01μF. Dec. 14

Ans.:

Given :

Maximum gain = 20 dB, 3 dB frequency ω_a = 10000 rad/s,

c_f = 0.01 μF.

Step 1 : Draw the circuit diagram :

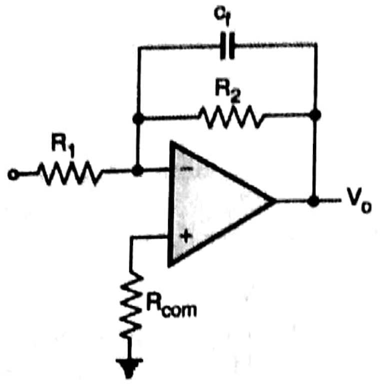


Fig. 3.24

Step 2 : Find the component values :

Maximum gain is nothing but dc gain.

$\therefore$ DC gain = 20 dB = 20 $\log_{10} |A_V|$ $\quad \therefore |A_V| = 10$

$$\text{But } |A_V| = \frac{R_2}{R_1}$$

$$\therefore R_2 = 10 R_1 \quad ...(1)$$

$$\text{Also } \omega_a = 10000 \text{ rad/sec.}$$

$$\therefore f_a = \frac{10000}{2\pi} = 1591.55 \text{ Hz}$$

$$\text{But } f_a = \frac{1}{2\pi R_2 c_f}$$

$$c_f = 0.01\ \mu\text{F given}$$

$$\therefore R_2 = \frac{1}{2\pi f_a c_f} = \frac{1}{10000 \times 0.01 \times 10^{-6}}$$

$$\therefore R_2 = 10\ k\Omega \quad \text{...Ans.}$$

Substituting in Equation (1),

$$10\ k\Omega = 10 R_1$$

$$\therefore R_1 = 1\ k\Omega \quad \text{...Ans.}$$

Q. 26 With suitable waveforms explain how op-amp can be used as differentiator ? Dec. 16

Ans. :

The differentiator or differentiating amplifier is as shown in Fig. 3.25. This circuit will perform the mathematical operation of differentiation on the input voltage waveform and the output is a derivative of the input signal. The differentiator can be constructed from the basic inverting amplifier by replacing resistance R_1 by capacitor C_1 as shown in Fig. 3.25.

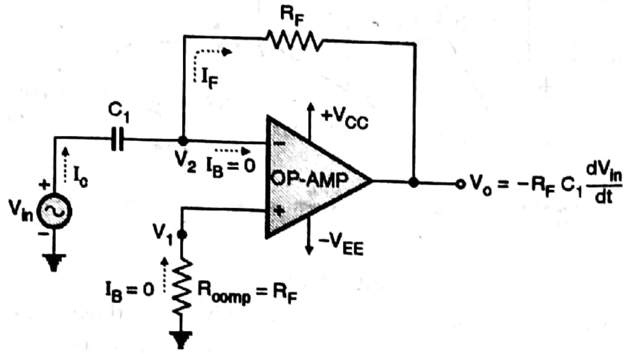


Fig. 3.25 : Basic or Ideal differentiator circuit

The resistance R_{comp} connected from the non-inverting terminal to ground provides the bias compensation.

Expression for the output voltage :

Assume that an ideal OP-AMP is being used. Hence $V_2 = 0, I_B = 0$

$$V_2 = 0, I_B = 0$$

$$I_C = I_B + I_F \quad ...(1)$$

But since $I_B \approx 0$ the above equation gets modified to,

$$I_C = I_F \quad ...(2)$$

$$I_C = C\frac{dV_c}{dt} \quad ...(3)$$

$$I_C = C\frac{dV_c}{dt} \quad ...(4)$$

The voltage across C_1 is given by,

$$V_c = V_{in} - V_2 \quad ...(5)$$

Substituting this into Equation (3),

$$I_f = C_1\frac{d}{dt}(V_{in} - V_2) \quad ...(6)$$

Now let us obtain the expression for the current I_F.

$$I_F = \frac{V_2 - V_o}{R_F} \quad ...(7)$$

But I_C and I_F are equal because $I_B = 0$ as shown in Fig. 3.25. Therefore, equating the Equations (6) and (7),

$$C_1\frac{d}{dt}(V_{in} - V_2) = \frac{V_2 - V_o}{R_F} \quad ...(8)$$

Using the concept of virtual ground,

$$V_1 = V_2 = 0 \quad ...(9)$$

Substituting $V_1 = V_2 = 0$ into Equation (8)

$$C_1\frac{d}{dt}(V_{in}) = \frac{-V_o}{R_F}$$

$$\therefore\ V_o = -R_F C_1\frac{d}{dt}(V_{in}) \quad ...(10)$$

Thus the output is $-R_F C_1$ times the time derivative of the input voltage.

Q. 27 With suitable waveforms explain how op-amp can be used as differentiator ? **Dec. 16**

Ans. :

The expression for instantaneous output voltage of an ideal differentiator is ;

$$V_o = -R_F C_1\frac{d}{dt}(V_{in})$$

Assume $R_F C_1 = 1$ $\quad \therefore\ V_o = -\frac{dV_{in}}{dt}$

Output for a step input :

The step input is mathematically expressed as,

$$V_{in} = A \quad ...t \geq 0$$

$$= 0 \quad ...t < 0$$

Substituting this

$$V_o = -\frac{d}{dt}(A) = 0$$

Thus output voltage is zero when the step input voltage is constant at "A", i.e. when the rate of change is zero as shown in Fig. 3.19.2(a).

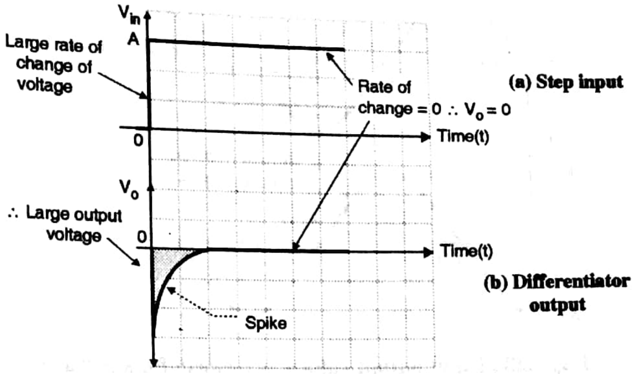
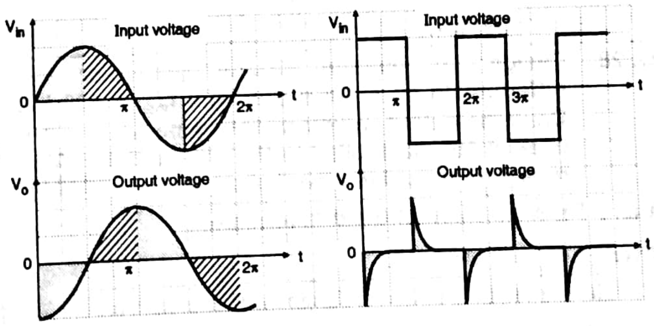


Fig. 3.26

But the step input changes suddenly from 0 to A at instant t = 0. So there is a large rate of change of voltage associated at this instant of time. Therefore a large output voltage is produced at t = 0 and this voltage is negative as the differentiator is an inverting differentiator.

Output voltage for a square wave input :

The square waveform of Fig. 3.27(b) is actually made up of many step signal. So we have to extend the same principle that we have used for step input voltage. Thus the output voltage is in the form of spikes corresponding to the rising and falling edges of the square wave and the output voltage is zero when the input is constant at ± A. These waveforms are as shown in Fig. 3.27(b).



(a) (b)

Fig. 3.27 : Input and output voltage waveforms of a differentiator

Output voltage for a sinewave input :

Let the sinusoidal input signal be expressed as,

$$V_{in} = A \sin \omega t$$

The corresponding output voltage, assuming $R_F C_1 = 1$ is given by,

$$V_o = \frac{-A \cdot d(\sin \omega t)}{dt}$$

$$\therefore\ V_o = \frac{-A \cos \omega t}{\omega} = -\frac{A}{\omega}\cos \omega t$$

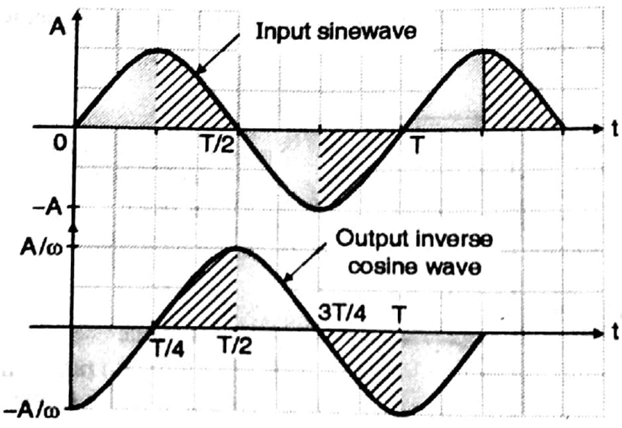


Fig. 3.28 : Input output voltage waveform for sinewave

t	0	T/4	T/2	3T/4	T
V_o	$-A/\omega$	0	A/ω	0	$-A/\omega$

Steps to Design a Practical Differentiator :

Following are the steps to be followed for design of a practical differentiator .

Steps to be followed :

Step 1 : Select f_a equal to the highest frequency of the input signal.

Step 2 : Assume the value of C_1 and calculate R_F from the expression of f_a .

Step 3 : Select $f_b = 20\ f_a$ and calculate the values of R_1 and C_F so that $R_1 C_1 = R_F C_F$.

Q. 28 Design an op-amp differentiator that will differentiate an input signal with f_{max} = 100 Hz. Draw the output waveform for a sine wave of 1 V peak at 100 Hz applied to the differentiator. Also repeat it for square wave input. **May 15**

Ans. : Part I : Design of the differentiator :

Given : $f_{max} = 1$ kHz.

The steps to be followed to design a differentiators are :

Step 1 : Select "f_a" and calculate R_F :

f_a = 1 kHz as the highest input frequency is 1 kHz.

Let $C_1 = 0.1\ \mu F$.

But $f_a = \dfrac{1}{2\pi R_F C_1}$

$\therefore \quad R_F = \dfrac{1}{2\pi f_a C_1} = \dfrac{1}{2\pi \times 1 \times 10^3 \times 0.1 \times 10^{-6}}$

$\therefore \quad R_F = 1.59\ k\Omega$ ...Ans.

Step 2 : Calculate R_1 and C_F :

$f_b = 20\ f_a = 20$ kHzBut $f_b = \dfrac{1}{2\pi R_1 C_1}$

$\therefore \quad R_1 = \dfrac{1}{2\pi f_b C_1} = \dfrac{1}{2\pi \times 20 \times 10^3 \times 0.1 \times 10^{-6}}$

$\therefore \quad R_1 = 79.5\ \Omega$ ...Ans.

Let us use the standard value of 82 Ω.

As $R_1 C_1 = R_F C_F$

$C_F = \dfrac{82 \times 0.1 \times 10^{-6}}{1.59 \times 10^3} = 0.00515\ \mu F$ ...Ans.

Let C_F be 0.005 μF. And finally;

$R_{comp} = R_F = 1.59\ k\Omega \approx 1.5\ k\Omega$.

Thus the component values are as follows :

$R_1 = 82\ \Omega$; $\quad R_{comp} = 1.5\ k\Omega$

$C_F = 0.005\ \mu F$; $\quad R_F = 1.59\ k\Omega$

$C_1 = 0.1\ \mu F$

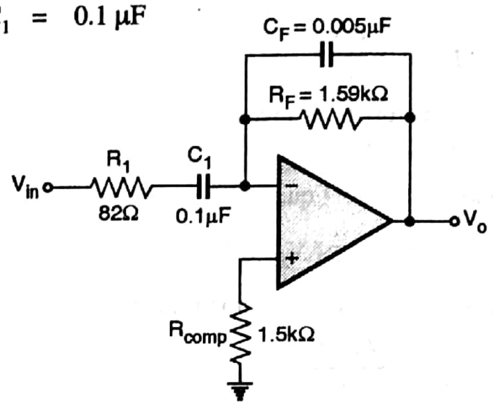


Fig. 3.29: Designed circuit

Part II : To draw the waveforms :

For an inverting differentiator,

$$V_o = -R_F C_1 \frac{d V_{in}}{dt} \qquad ...(1)$$

But, $V_{in} = 1 \sin(2\pi \times 100\ t)$

$= 2 \sin(200\pi t)$ ...(2)

values of R_F and C_1 are :

$R_F = 1.59\ k\Omega$ and

$C_1 = 0.005 \times 10^{-6}$ F

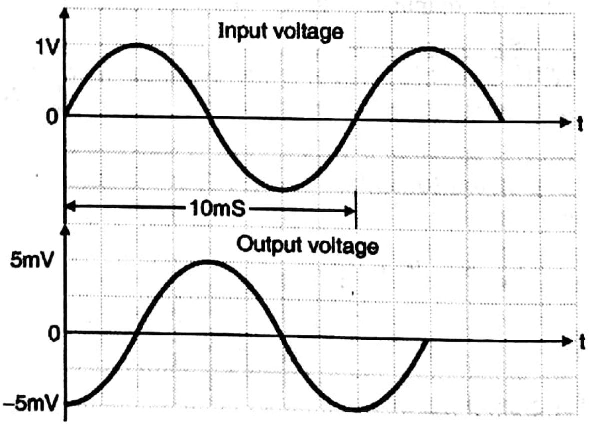


Fig. 3.30: Waveforms for a differentiator

Substituting these values into Equation (1)

$V_o = -1.59 \times 10^{+3} \times 0.005 \times 10^{-6} \cdot \dfrac{d}{dt}[1 \sin(200\pi\ t)]$

$= 7.95 \times 10^{-6} \times 200\ \pi\ [\cos(200\ \pi\ t)]$

$= -5 \times 10^{-3} \cos(200\ \pi\ t)$ ...Ans.

This is a cosine wave with a peak voltage of 5 mV and frequency of 100 Hz. The input and output waveforms are

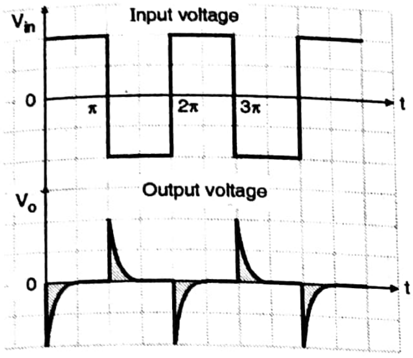


Fig. 3.31: Input and output voltage waveforms of a differentiator for a square wave input

Q. 29 Draw and explain op-amp inverting comparator. Draw input and output waveforms for $V_{ref} > 0$ and also for $V_{ref} < 0$. **May 15**

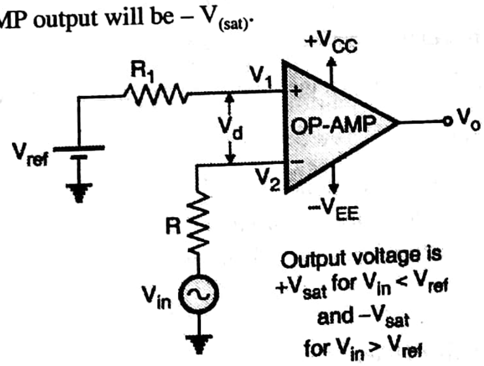
Ans. : Assuming the OP-AMP to have a very high input resistance. Hence the currents flowing through the resistors R_1 and R of Fig. 3.32(a) is very small. Hence the voltage drop across them is close to zero.

$\therefore \quad V_1 = V_{ref}$ and $V_2 = V_{in}$

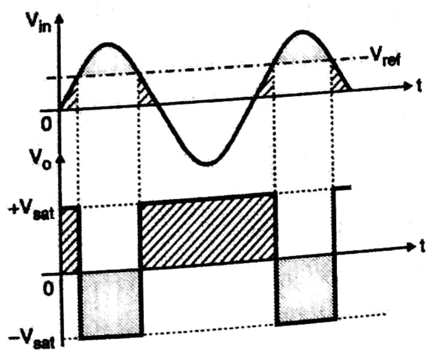
$\therefore$ Differential input voltage $V_d\,v = V_1 - V_2 = (V_{ref} - V_{in})$

When V_{in} is less than V_{ref}, the voltage at the non-inverting (+) terminal is higher than the voltage at the inverting (–) terminal. This makes the differential input voltage V_d positive and the OP-AMP output will swing to $+V_{(sat)}$.

When V_{in} is greater than V_{ref}, the voltage at the inverting (–) terminal is greater than the voltage at the non-inverting (+) terminal. Therefore the differential input voltage V_d is negative and the OP-AMP output will be $-V_{(sat)}$.



(a) Inverting comparator



(b) Input and output voltage waveforms

Fig. 3.32

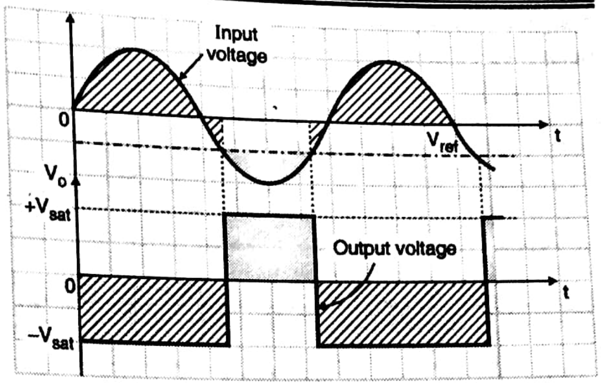


Fig. 3.33 : Waveforms of an inverting comparator for a negative reference voltage

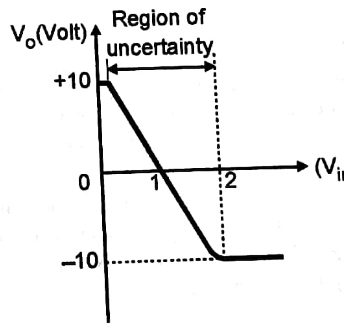
If the reference voltage is made negative ($-V_{ref}$) then the input and output voltage waveforms gets modified as shown in Fig. 3.33.

Transfer Characteristics of an Inverting Comparator :

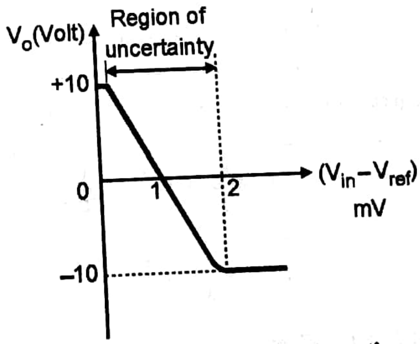
Fig. 3.34(a) shows the ideal transfer characteristic of an inverting comparator for a positive V_{ref}. It shows that the output voltage is equal to $+V_{sat}$ for V_{in} less than V_{ref}.

It switches instantaneously from $+V_{sat}$ to $-V_{sat}$ when V_{in} becomes just slightly greater than V_{ref}.

The differential voltage needed for switching the output state is very small, ideally it is zero volt. The commercial OP-AMPs have the transfer characteristic as shown in Fig. 3.34(b). The change in output state takes place when the differential voltage is about 2 mV. The state change take place gradually and there is a region of uncertainty where we cannot directly define the output.



(a) Ideal transfer characteristic of inverting comparator for positive V_{ref}

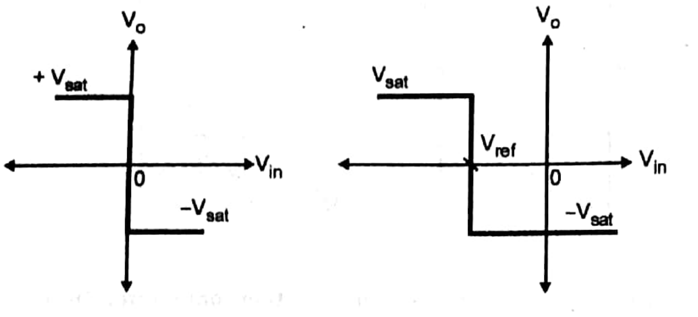


(b) Practical transfer characteristic of an inverting comparator

Fig. 3.34

Transfer Characteristics for Zero and Negative Reference Voltage :

Assuming the comparator to be ideal, the transfer characteristics of an inverting comparator for zero and negative reference voltage are shown in Fig. 3.35



(a) For $V_{ref} = 0$ (b) For negative V_{ref}

Fig. 3.35 : Transfer characteristics of an inverting comparator

Q. 30 With respect to op-amp explain the ideal characteristics and concept of virtual ground. Explain how op-amp can be used as an averaging amplifier in inverting configuration. Also draw neat circuit diagrams to :

1. **Convert sine wave to square wave using op-amp.**
2. **Detect the crossing of zero's in the generated square wave.** **Dec. 15**

Ans. :

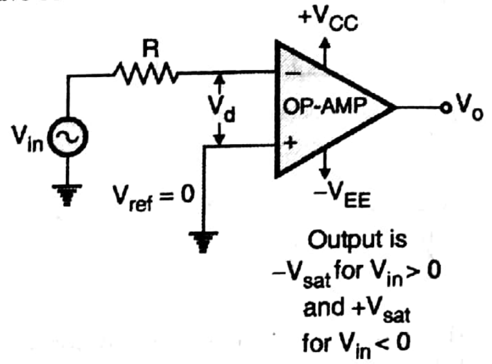
Zero crossing detector is as shown in Fig. 3.36(a). As you can see, zero crossing detector is nothing but the basic comparator circuit with a zero reference voltage applied to the non-inverting terminal.

Circuit operation :

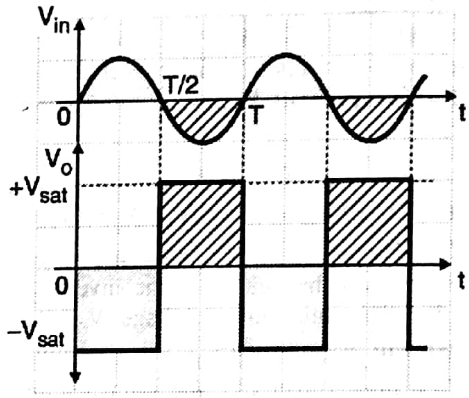
Consider the waveforms shown in Fig. 3.36(b). When the input sinewave crosses zero and becomes positive at instant t = 0, the differential input voltage V_d becomes negative and the output voltage will swing to $-V_{(sat)}$.

When the sinewave again crosses zero and becomes negative at instant t = T/2, the differential input voltage becomes positive and the output voltage will swing to $+V_{(sat)}$ as now the (+) terminal is more positive than the (–) terminal.

The zero crossing detector thus switches its output from one state to the other everytime when the input voltage crosses the zero. The zero crossing detector is also known as a sinewave to square wave converter.



(a) **Zero crossing detector**



(b) Input and output voltage waveforms

Fig. 3.36

Demerit of zero crossing detector :

The noise present on the input side can cause false switching.

Applications :

1. Square wave generators.
2. In the mains supply synchronizing circuit.
3. Microprocessor based triggering circuit for thyristors.

Chapter 4 : Analog Communication

Q. 1 Classification of analog and digital modulation. **Dec. 12**

Ans. :

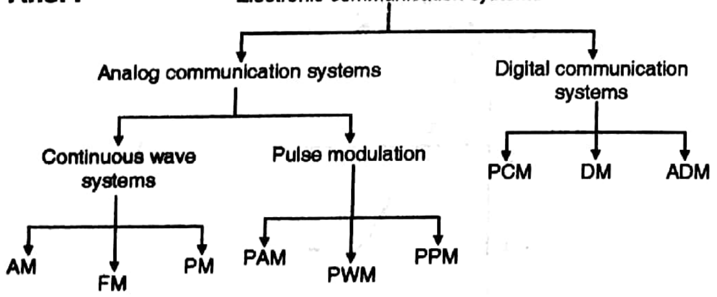


Fig. 4.1 : Classification based on analog or digital communication

Q. 2 Explain the necessity and significance of modulation in communication. **May 16**

Ans. : In the process of modulation, the baseband signal is "translated" i.e. shifted from low frequency side to high frequency side of the frequency spectrum. This frequency shift is proportional to the frequency of carrier. The modulation process has the following advantages :

Advantages of modulation :

1. Reduction in the height of antenna
2. Avoids mixing of signals
3. Increases the range of communication
4. Multiplexing becomes possible
5. Improves quality of reception

Q. 3 Explain the concept of amplitude modulation.

May 16

Ans. : To transmit the information or message signal over a band pass communication channel such as satellite channel or radio channel, some type of modulation is essential. The modulation process is associated with shift of frequency range from low frequency side to high frequency side of the spectrum. Modulation is the process by which some characteristics of a carrier is varied in accordance with a modulating wave.

The three characteristics of the carrier are amplitude, frequency and phase. So one of these characteristics is varied in proportion with the modulating signal (i.e. message or information signal). The message signal is referred to as the modulating signal and the result of modulation process is referred to a **modulated signal**.

Depending on which characteristics of the carrier is being changed, the modulation systems are classified as shown in Fig. 4.2.

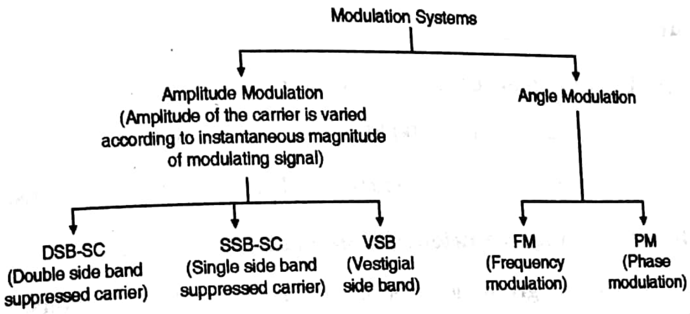


Fig. 4.2 : Classification of modulation systems

Q. 4 Explain amplitude modulation for more than one modulating signal in the following cases :

1. **Mathematical equation**
2. **AM waveform**
3. **AM amplitude and power spectrum**
4. **Modulation coefficient**
5. **Transmission power**

Dec. 15

Ans. : **Amplitude modulation (AM)** or Amplitude Modulation with Full Carrier (AM-FC) is the process of changing the amplitude of a high frequency sinusoidal carrier signal in proportion with the instantaneous value of modulating signal.

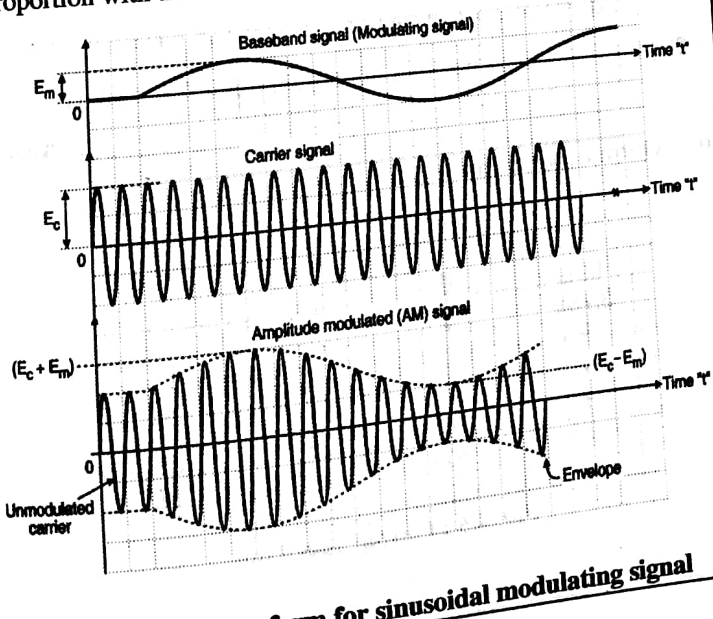


Fig. 4.3 : AM waveform for sinusoidal modulating signal

Observations :

1. The frequency of the sinusoidal carrier is much higher than that of the modulating signal.
2. In AM the instantaneous amplitude of the sinusoidal high frequency carrier is changed in proportion to the instantaneous amplitude of the modulating signal. This is the principle of AM.
3. The time domain display of AM signal is as shown in Fig. 4.3. This AM signal is transmitted by a transmitter. The information in the AM signal is contained in the amplitude variations of the carrier of the envelope shown by dotted lines in Fig. 4.3.
4. The frequency and phase of the carrier remain constant.
5. AM is used in the applications such as radio transmission, TV transmission etc.

Assumptions : Let the modulating signal be sinusoidal and be represented as,

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

where "e_m" is the instantaneous amplitude of the modulating signal, E_m is the peak amplitude, $\omega_m = 2\pi f_m$ and f_m = Frequency of the modulating signal.

Let the carrier signal also be sinusoidal at a much higher frequency than that of the modulating signal. The instantaneous carrier signal e_c is given by,

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

where E_c = Peak carrier amplitude,

f_c = Carrier frequency and $\omega_c = 2\pi f_c$.

The AM wave is expressed by the following expression,

$$e_{AM} = A \cos (2\pi f_c t)$$

where A = Envelope of AM wave

Where A represents the instantaneous value of the envelope. The modulating signal either adds or gets subtracted from the peak carrier amplitude E_c as shown in Fig. 4.3. Hence we can represent the instantaneous value of envelope as,

$$A = E_c + e_m$$

$$= E_c + E_m \cos (2\pi f_m t)$$

Hence the AM wave is given by,

$$e_{AM} = A \cos (2\pi f_c t)$$

$$= [E_c + E_m \cos (2\pi f_m t)] \cos (2\pi f_c t)$$

$$\therefore e_{AM} = E_c \left[1 + \frac{E_m}{E_c} \cos (2\pi f_m t) \right] \cos (2\pi f_c t)$$

Let $m = E_m / E_c$ be the modulation index.

$$\therefore e_{AM} = E_c [1 + m \cos (2\pi f_m t)] \cos (2\pi f_c t)$$

This expression represents the time domain representation an AM signal.

Modulation Index or Modulation Factor and Percenta Modulation :

In AM wave the modulation index (m) is defined as the ra of amplitudes of the modulating and carrier waves as follows :

$$m = \frac{E_m}{E_c}$$

When $E_m \le E_c$ the modulation index "m" has values between 0 and 1 and no distortion is introduced in the AM wave. But if $E_m > E_c$ then m is greater than 1. This will distort the shape of AM signal. The distortion is called as "over modulation."

The modulation index is also called as modulation factor, modulation coefficient or degree of modulation. However if modulation index is expressed as percentage it is called as "percentage modulation."

$\therefore$ % Modulation $= \frac{E_m}{E_c} \times 100$; "m" is a dimensionless quantity.

Frequency Spectrum of the AM Wave (Frequency Domain Description) :

The frequency spectrum is a graph of amplitude on Y axis versus frequency on X axis. The frequency spectrum of AM wave tells us about which frequency components are present in the AM wave and what are their amplitudes. So consider the equation for AM wave.

$$e_{AM} = (E_c + E_m \cos\omega_m t) \cos\omega_c t$$

$$E_c = \left[1 + \frac{E_m}{E_c} \cos \omega_m t\right] \cos\omega_c t$$

As per the definition of the modulation index, $m = E_m / E_c$.

$$\therefore e_{AM} = E_c (1 + m \cos\omega_m t) \cos\omega_c t$$

Simplifying we get,

$$e_{AM} = E_c \cos\omega_c t + m E_c \cos\omega_m t \cos\omega_c t$$

For the second term in the above expression use the following standard identity :

$$2 \cos A \cos B = \cos (A + B) + \cos (A - B)$$

Therefore simplified as follows :

$$e_{AM} = \underbrace{E_c \cos\omega_c t}_{\text{Carrier}} + \underbrace{\frac{m E_c}{2} \cos (\omega_c + \omega_m) t}_{\text{Upper sideband}} + \underbrace{\frac{m E_c}{2} \cos (\omega_c - \omega_m) t}_{\text{Lower sideband}}$$

Observations : The expression for the AM wave shows that it consists of three terms :

1. First term is nothing else but the unmodulated carrier signal.
2. The second term is a sinusoidal signal at frequency ($f_c + f_m$). This is called as the upper sideband (USB). Its amplitude is $\frac{m E_c}{2}$.
3. The third term represents a sinusoidal signal at frequency ($f_c - f_m$). It is called as the lower sideband (LSB). Its amplitude is $\frac{m E_c}{2}$.

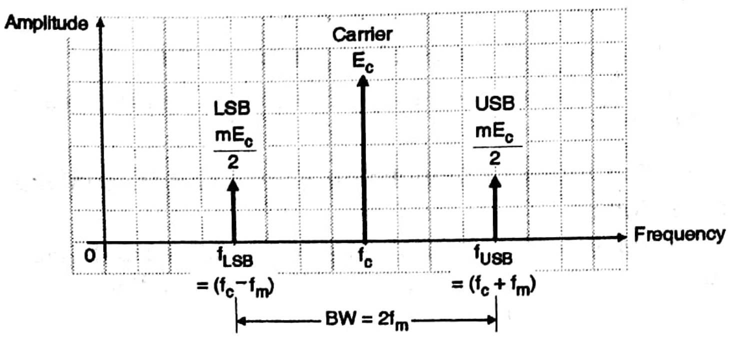


Fig. 4.4 : Single sided frequency spectrum of AM wave

Hence the frequency spectrum of an A.M. wave is as shown in Fig. 4.4 . it is a single sided spectrum i.e. the spectrum plotted for only the positive values of frequency.

Q. 5 : One input to a conventional AM modulator is a 500 kHz carrier with an amplitude of 20 V_p. The second input is a 10 kHz modulating signal that is sufficient amplitude to cause a change in the output wave of ± 7.5 V_p. Determine :

1. **Side frequencies and modulation index.**
2. **Peak amplitude of the modulated carrier and the upper and lower side frequency voltages.**
3. **Maximum and minimum amplitudes of the envelope.**
4. **Expression for the modulated wave.**
5. **Draw the output spectrum and output envelope.** Dec. 07, Dec. 13, Dec. 16

Ans. :

(a) Upper and lower sideband frequencies :

$f_{USB} = f_c + f_m = 500 \text{ kHz} + 10 \text{ kHz} = 510 \text{ kHz}$ ...Ans.

$f_{LSB} = f_c - f_m = 500 \text{ kHz} - 10 \text{ kHz} = 490 \text{ kHz}$ ...Ans.

(b) Modulating coefficient and percentmodulation :

From the given description the peak modulating voltage $V_m = 7.5$ V and peak carrier voltage $E_c = 20$ V.

$\therefore$ Modulating coefficient $m = \frac{E_m}{E_c} = \frac{7.5}{20} = 0.375$ ...Ans.

% Modulation $= m \times 100\% = 0.375 \times 100 = 37.5\%$ ...Ans.

(c) Amplitude of sidebands :

Peak amplitude of modulated carrier $= E_c + E_m = 20 + 7.5$

$= 27.5$ V ...Ans.

Amplitude of sidebands $= \frac{m E_c}{2} = 0.375 \times \frac{20}{2} = 3.75$ V ...Ans.

(d) Expression for modulated wave :

$$e_{AM} = E_c [1 + m \sin (2\pi f_m t)] \sin (2\pi f_c t)$$

$$= 20 [1 + 0.375 \sin (2 \pi \times 10 \times 10^3 t)] \sin (2\pi \times 500 \times 10^3 t)$$

...Ans.

(e) Output spectrum :

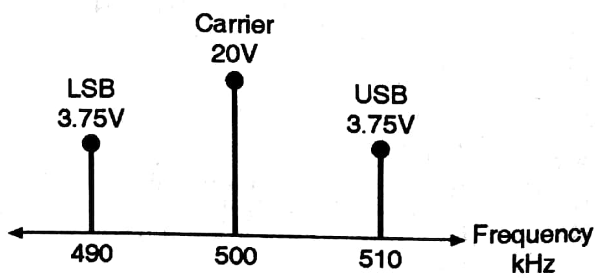


Fig. 4.5 : Output spectrum

Q. 6 The antenna current of an AM broadcast transmitter, modulated to a depth of 40% by an audio sine wave is 11 A. It increases to 12 A as result of simultaneous modulation by another audio sine wave. What is the modulation index due to this second wave? May 11

Ans. : Given : $I_{t1} = 11$ A, $m_1 = 0.4$, $I_{t2} = 12$ A

To find : Modulation index due to second wave.

Step 1 : Find I_c :

$$\left[\frac{I_{t1}}{I_c}\right]^2 = 1 + \frac{m_1^2}{2}$$

$$\therefore I_c = \frac{I_{t1}}{\left[1 + \frac{m_1^2}{2}\right]^{1/2}} = \frac{11}{\left[1 + \frac{(0.4)^2}{2}\right]^{1/2}} = 10.58 \text{ A}$$

Step 2 : Find m_t :

After modulating with the second signal.

$$I_{t2}^2 = I_c^2\left[1 + \frac{m_t^2}{2}\right]$$

$$m_t = \sqrt{2\left[\left(\frac{I_{t2}}{I_c}\right)^2 - 1\right]} = \sqrt{2\left[\left(\frac{12}{10.58}\right)^2 - 1\right]}$$

$$\therefore m_t = 0.756$$

Step 3 : Find m_2 :

$$m_t = \left[m_1^2 + m_2^2\right]^{1/2}$$

$$\therefore \quad m_2 = \left[m_t^2 - m_1^2\right]^{1/2} = \left[(0.756)^2 - (0.4)^2\right]^{1/2}$$

$$\therefore \quad m_2 = 0.6415 = 64.15\%$$ ...Ans.

Q. 7 : When a broadcast AM transmitter is 50% modulated, its antenna current is 12 A. What will be the current, when the modulation depth is increased to 0.9 ? Dec. 16

Ans. : Step 1 : Find I_C :

Given : $I_{t1} = 12$ A, $m_1 = 0.5$

$$I_{t1} = I_c\left[1 + \frac{m_1^2}{2}\right]^{1/2} \qquad \therefore 12 = I_c\left[1 + \frac{(0.5)^2}{2}\right]^{1/2} \quad ...(1)$$

$$\therefore I_c = 11.32 \text{ Amp}$$

Step 2 : Find I_{t2} :

Given : $m_2 = 0.9$

$$I_{t2} = I_c\left[1 + \frac{m_2^2}{2}\right]^{1/2} = 11.32\left[1 + \frac{(0.9)^2}{2}\right]^{1/2}$$

$$\therefore I_{t2} = 13.42 \text{ Amp}$$ ...Ans.

Q. 8 Explain amplitude modulation for more than one modulating signal in following cases :

1. **Mathematical equation**
2. **AM waveform**
3. **Am amplitude and power spectrum**
4. **Modulation coefficient**
5. **Transmission power**

May 14

Ans. :

Assuming only one sinusoidal modulating signal is present. But in practice more than one modulating signals will be present. This is due to the fact that a practical modulating signal is a non-sinusoidal complex signal which consists of a mixture of many sinusoidal modulating signal in different proportions.

Let us assume that there are two modulating signals.

$$x_1(t) = E_{m1} \cos \omega_{m1} t$$

and $$x_2(t) = E_{m2} \cos \omega_{m2} t$$

The total modulating signal will be the sum of these two in the time domain.

$$\therefore \text{Total modulating signal} = x_1(t) + x_2(t) = E_{m1} \cos\omega_{m1} t + E_{m2} \cos\omega_{m2} t$$

Let the carrier be $c(t) = E_c \cos\omega_c t$

The instantaneous value of the envelope of AM wave is,

$$A = E_c + x_1(t) + x_2(t)$$

$$= E_c + E_{m1} \cos \omega_{m1} t + E_{m2} \cos\omega_{m2} t$$

Therefore the AM wave is given by,

$$e_{AM} = A \cos\omega_c t$$

It can be proved that the expression for the DSBFC AM wave is given by,

$$e_{AM} = E_c \cos \omega_c t + \frac{m_1 E_c}{2} \cos(\omega_c + \omega_{m1}) t + \frac{m_1 E_c}{2} \cos(\omega_c - \omega_{m1}) t + \frac{m_2 E_c}{2} \cos(\omega_c + \omega_{m2}) t + \frac{m_2 E_c}{2} \cos(\omega_c - \omega_{m2}) t$$

Where m_1 and m_2 are the modulation indices corresponding to the two modulating signals $x_1(t)$ and $x_2(t)$ respectively.

There are two USB components at frequencies $(f_c + f_{m1})$ and $(f_c + f_{m2})$ and two LSB components at frequencies $(f_c - f_{m1})$ and $(f_c - f_{m2})$.

The frequency spectrum of AM wave is as shown in Fig. 4.6.

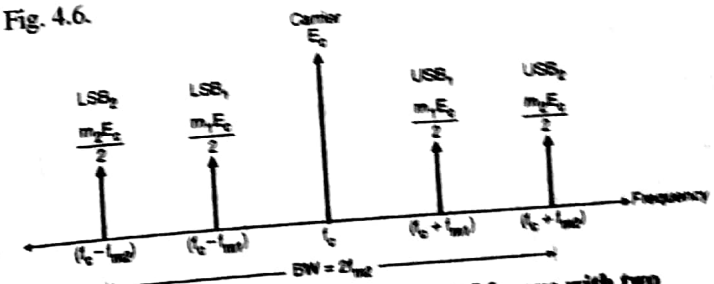


Fig. 4.6 : Frequency spectrum of AM wave with two modulating signals

Total Power (P_t) :

The total power in AM wave with a complex modulating signal is given by,

$$P_t = P_c\left[1 + \frac{m_1^2}{2} + \frac{m_2^2}{2} .... + \frac{m_n^2}{2}\right]$$

Effective modulation index (m_t) :

In general the total modulation index due to the simultaneous modulation by "n" modulating signals is given by,

$$m_t = [m_1^2 + m_2^2 + \ldots\ldots m_n^2]^{1/2}$$

Q. 9 Draw the block diagram of AM transmitter and explain each block in brief. Dec. 07

Ans. :

The AM transmitters can be of two types :

1. High level modulated transmitter or
2. Low level modulated transmitter.

Let us see their operation one by one.

Low Level Modulated AM Transmitter :

The block diagram of a low level AM transmitter is as shown in Fig. 4.7.

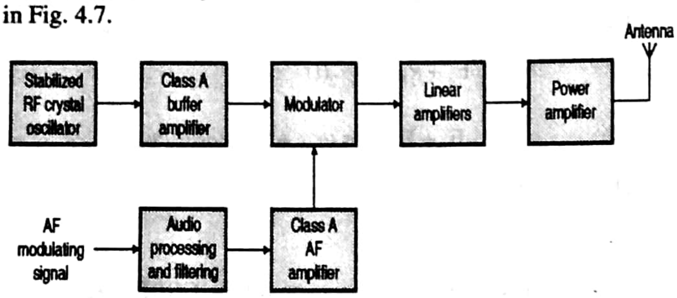


Fig. 4.7 : Low level AM transmitter

The RF oscillator produces the carrier signal. The RF oscillator is stabilized in order to maintain the carrier frequency deviation within a prescribed limit. The carrier frequency is equal to the transmitter frequency and it should remain very stable.

The amplified modulating signal is applied to the modulator along with the carrier. At the output of the modulator we get the AM wave. This AM signal is then amplified using a chain of linear amplifiers to raise its power level. The linear amplifiers can be class A, AB or B type amplifiers. The linear amplifiers are used in order to avoid the waveform distortion in AM wave. However these amplifiers posses a low efficiency. The amplitude modulated signal is then transmitted using transmitting antenna.

The transistorized modulator circuits can be used for low level modulator due to the low power which is to be handled. The low level transmitter does not require a large AF modulator power so its design is simplified. However the overall efficiency is much lower compared to high level modulation. This is due to the use of less efficient linear amplifiers.

Q. 10 Write short note on : High level AM transmitter. May 06

Ans. :

Many of the AM transmitters use the high level modulation technique. The block diagram of high level AM transmitter is as shown in Fig. 4.8.

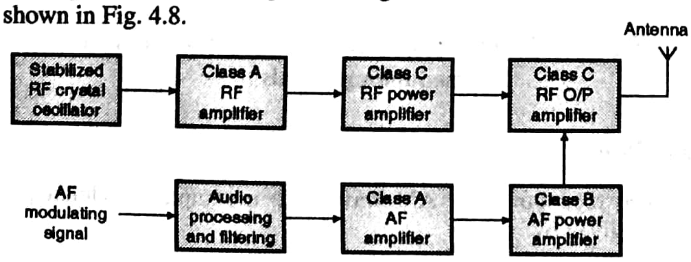


Fig. 4.8: High level AM transmitter

Operation :

Here the carrier generated by the stabilized crystal oscillator is first amplified to the adequate power level using class C RF power amplifiers. The modulating signal also is amplified to a high power level before modulation takes place. If we want 100 % modulation then the power of modulating signal must be 33 % of the total power. So if 1500 W total power is to be transmitted, the modulating power will be 500 W. This highlights the need to amplify the modulating signal to an adequate power level.

The modulation takes place in the last class C RF amplifier. The modulator output is AM wave which can be directly transmitted. The collector modulated transistorized circuit or plate modulated vacuum tube modulator is used as modulator stage. The advantage of high level modulation is its high efficiency due to the use of highly efficient class C amplifiers. The disadvantage is that a large AF power amplifier is needed to raise the modulating signal to the adequate power level.

Q. 11 Differentiate between low level modulation and high level modulation. May 08

Ans. :

Sr. No.	Parameter	High level modulation	Low level modulation
1.	Modulation takes place at	High power level.	Low power level.
2.	Types of amplifiers	Highly efficient class C amplifiers are used.	Linear amplifiers (A, AB or B) are used after modulation.
3.	Efficiency	Very high.	Lower than high level modulators.
4.	Devices used	Vacuum tubes or transistors for medium power applications.	Transistors, JFET, OP-AMPs.
5.	Design of AF power amplifier	Complex due to very high power involved.	Easy as it is to be done at low power.
6.	Applications	High power broadcast transmitters.	Sometimes used in TV transmitters (IF modulation).
7.	Power handling capacity	High.	Low.

Q. 12 Amplitude modulation is a waste of power and bandwidth. Justify. May 03

Ans.:

The AM signal is also called as "Double Sideband Full Carrier (DSBFC)" signal. The three main disadvantages of this technique are :

1. Power wastage takes place (carrier does not contain any information).
2. AM needs larger bandwidth.
3. AM wave gets affected due to noise.

These are explained as follows :

The carrier signal in the DSBFC system does not convey any information. The information is contained in the two sidebands only. But the sidebands are images of each other and hence both of them contain the same information. Thus all the information can be conveyed by only one sideband.

Power wastage due to DSBFC transmission :

As the total power transmitted by an AM wave is given by :

$$P_t = P_c + P_{USB} + P_{LSB} \quad ...(1)$$

$$\therefore \quad P_t = P_c + \frac{m^2}{4} P_c + \frac{m^2}{4} P_c \quad ...(2)$$

Out of the three terms in Equation (1), carrier component does not contain any information and one sideband is redundant.

So out of the total power $P_t = \left[1 + \frac{m^2}{2}\right] P_c$ the wasted power is given by :

$$\text{Power wastage} = P_c + \frac{m^2}{4} P_c = \left[1 + \frac{m^2}{4}\right] P_c \quad ...(3)$$

Bandwidth requirement of DSBFC :

The BW of DSBFC system is $2f_m$. This is due to the simultaneous transmission of both the sidebands, out of which only one is sufficient to convey all the information. Thus the BW of DSBFC is "double" than actually required. Therefore DSBFC is a "bandwidth inefficient" system.

Effect of Noise :

When the AM wave travels from the transmitter to receiver over a communication channel, noise gets added to it. The noise will change the amplitude of the envelope of AM in a random manner.

As the information is contained in the amplitude variations of the AM wave, the noise will contaminate the information contents in the AM.

Hence the performance of AM is very poor in presence of noise.

The waveform of AM wave contaminated by noise is shown in Fig.

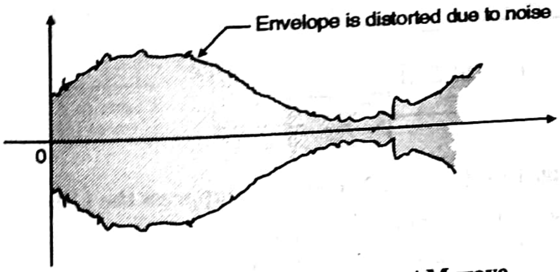


Fig. 4.9 : Effect of noise on AM wave

Advantages of AM :

1. AM transmitters are less complex.
2. AM receivers are simple, detection is easy.
3. AM receivers are cost efficient. Hence even a common person can afford to buy it.
4. AM waves can travel a longer distance.
5. Low bandwidth.

Applications of AM :

1. Radio broadcasting.
2. Picture transmission in a TV system.

Q. 13 What is DSBSC wave ? Explain its generation using balanced modulator. Dec. 14, May 15

Ans.:

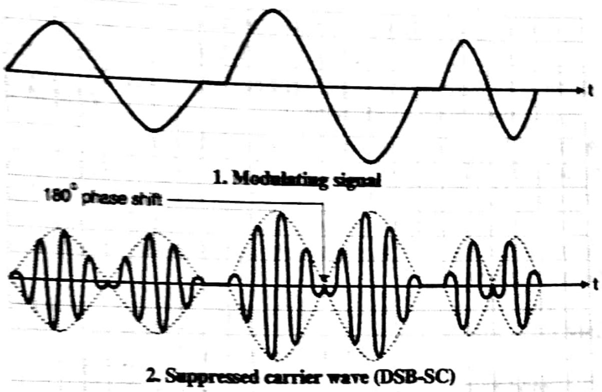


Fig. 4.10: Time domain display of DSB-SC signal

From the AM wave (DSB-FC) if only the carrier component is suppressed then the remaining signal is called as DSBSC (double sideband suppressed carrier) signal.

If m = 1 then the percent power saving in DSBSC is given by,

$$\% \text{ Power saving} = \frac{[m^2/2]\, P_c \times 100}{\left[1 + \frac{m^2}{2}\right] P_c} = \frac{0.5}{1.5} \times 100 \times = 66.667\%$$

Thus DSB-SC is a power efficient system as compared to DSB-FC.

However as both the sidebands are being transmitted, the BW of DSB-SC system is still $2 f_m$ i.e. same as that of AM wave.

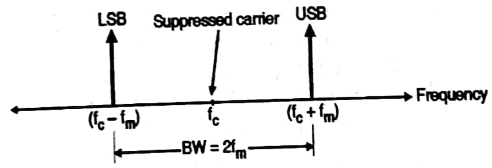


Fig. 4.11 : Frequency spectrum of DSB-SC signal

Q. 14 Give advantages of SSB over full carrier DSB amplitude modulated wave. Dec. 14

Ans. : The advantages of SSB over DSB-FC signal are :

1. Less bandwidth requirement as SSB requires a BW of f_m. This will allow more number of signals to be transmitted in the same frequency range.
2. Lots of power saving. This is due to the transmission of only one sideband component. At 100 % modulation, the percent power saving is 83.33 %.
3. Reduced interference of noise. This is due to the reduced bandwidth. As the bandwidth increases, the amount of noise added to the signal will increase.

Q. 15 Give and disadvantages of SSB over full carrier DSB amplitude modulated wave. Dec. 14

Ans. : Even though the SSB system has many advantages it has the following disadvantages :

1. The generation and reception of SSB signal is complicated as discussed later on.
2. The SSB transmitter and receiver need to have an excellent frequency stability. A slight change in frequency will hamper the quality of transmitted and received signal. SSB therefore is not generally used for the transmission of good quality music. It is used for speech transmission.

Q. 16 Explain the generation of DSBSC using balanced modulator. **Dec. 16**

Ans. : The nonlinear modulators are also known as the balanced modulators. The balanced modulators are used to suppress the unwanted carrier in an AM wave. The carrier and modulating signals are applied to the inputs of the balanced modulator and we get the DSB signal with suppressed carrier at the output of the balanced modulator.

Thus the output consists of the upper and lower sidebands only.

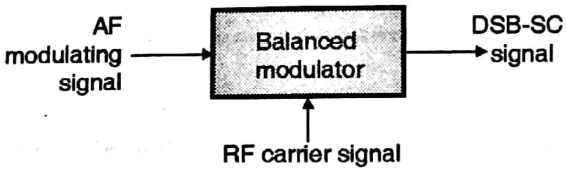


Fig. 4.12: Block diagram of balanced modulator

Principle of operation :

The principle of operation of a balanced modulator states that if two signals at different frequencies are passed through a "nonlinear resistance" then at the output we get an AM signal with suppressed carrier. The device having a nonlinear resistance can be a diode or a JFET or even a bipolar transistor.

Balanced Modulator using Diodes :

Fig. 4.13 shows the balanced modulator using diode as nonlinear device. The modulating signal x(t) is applied at the primary of a center tapped AF transformer. At its secondary windings.

Here it produce two equal voltages with a phase shift of 180°.

Thus the two diodes receive the modulating signal x(t) of same magnitude but opposite phase. The carrier is applied to the center tap of the secondary. So input voltage to D_1 is given by,

$$v_1 = \cos\omega_c t + x(t)$$

And the input voltage to D_2 is given by,

$$v_2 = \cos\omega_c t - x(t)$$

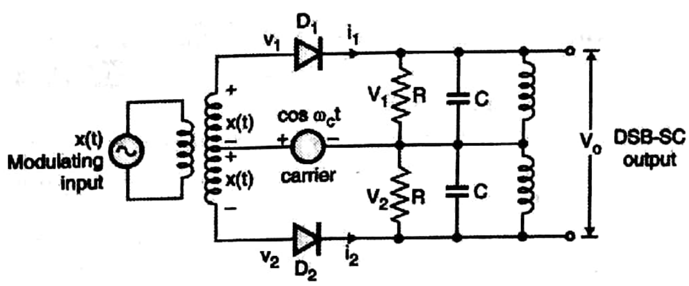


Fig. 4.13: Balanced modulator using diodes

The parallel RLC circuits on the output side form the tuned band pass filters.

Analysis : The diode current i_1 and i_2 are given by,

$$i_1 = a v_1 + b v_1^2 = a[x(t) + \cos\omega_c t] + b[x(t) + \cos\omega_c t]^2$$

$$= a\,x(t) + a\cos\omega_c t + b\,x^2(t) + 2b\,x(t)\cos\omega_c t + b\cos^2\omega_c t$$

Similarly

$$i_2 = a v_2 + b v_2^2 = a[\cos\omega_c t - x(t)] + b[\cos\omega_c t - x(t)]^2$$

$$= a\cos\omega_c t - a\,x(t) + b\,x^2(t) - 2b\,x(t)\cos\omega_c t + b\cos^2\omega_c t$$

The output voltage is given by,

$$v_o = v_1 - v_2 = i_1 R - i_2 R$$

Substituting the expression for i_1 and i_2,

$$v_o = R[2a\,x(t) + 4b\,x(t)\cos\omega_c t]$$

$$\therefore v_o = \underbrace{2aR\,x(t)}_{\text{Modulating signal}} + \underbrace{4bR\,x(t)\cos\omega_c t}_{\text{DSB-SC signal}}$$

So the output voltage contains a modulating signal term and the DSB-SC signal. The modulating signal term is eliminated by the bandpass filters and the second term is allowed to pass through to the output by the LC band pass filter section.

$$\therefore \text{Final output} = 4bR\,x(t)\cos\omega_c t = K\,x(t)\cos\omega_c t$$

Thus the diode balanced modulator produces the DSB-SC signal at its output.

Q. 17 Explain generation of SSB using phase shift method. **Dec. 15, May 16**

Ans. :

The block diagram for the phase shift method of SSB generation is as shown in Fig. 4.14. This system is used for the suppression of lower sideband. This system uses two balanced modulators M_1 and M_2 and two 90° phase shifting networks as shown in the Fig. 4.14.

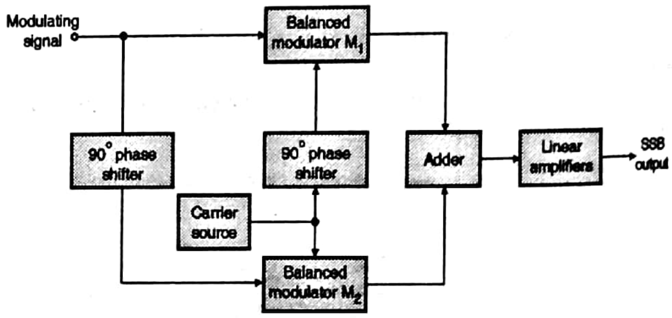


Fig. 4.14: Phase shift method to suppress the LSB

Operation : The operation of the phase shift method is as follows

The balanced modulator M_1 has two inputs, the modulating signal without any phase shift and the RF carrier with a 90° phase shift. The other balanced modulator M_2 receives the modulating signal with a 90° phase shift and carrier without any phase shift.

At the output of both the balanced modulators we DSB-SC signal consisting of both sidebands. The carrier is completely removed. The upper sidebands (USBs) at the outputs of both the balanced modulators lead the carrier by 90°. But LSB at the output of M_1 leads the carrier by 90° and the LSB at the output of M_2 lags behind the carrier by 90°. Thus the LSBs are out of phase.

So when the outputs of M_1 and M_2 are applied to the adder, the LSBs are cancelled out and the output of the adder consists of only the upper sideband. The linear amplifiers will follow the adder. They are class B or AB type amplifiers used to amplify the USB without introducing any distortion.

Mathematical proof of sideband suppression :

The balanced modulator M_1 are

Inputs to M_1 $\begin{cases} \cos\omega_m t & \text{.....Modulating signal as it is} \\ \cos(\omega_c t + 90°) & \text{......90° phase shifted carrier} \end{cases}$

And the inputs to balanced modulator M_2 are

Inputs to M_2 $\begin{cases} \cos(\omega_m t + 90°) & \text{....90° shifted modulating signal} \\ \cos\omega_c t & \text{....carrier as it is} \end{cases}$

So the output of $M_1 = \cos(\omega_c t) \cdot \cos\omega_m t$

$$= \underbrace{\frac{1}{2}\cos[\omega_c t + \omega_m t + 90°]}_{\text{USB with 90° advance}} + \underbrace{\frac{1}{2}\cos[\omega_c t - \omega_m t + 90°]}_{\text{LSB with 90° delay}}$$

And output of $M_2 = \cos\omega_c t \cdot \cos(\omega_m t + 90°)$

$$= \underbrace{\frac{1}{2}\cos[\omega_c t + \omega_m t + 90°]}_{\text{USB with 90° advance}} + \underbrace{\frac{1}{2}\cos[\omega_c t - \omega_m t - 90°]}_{\text{LSB with 90° delay}}$$

Output of the adder = Output of M_1 + Output of M_2

$$= \cos(\omega_c t + \omega_m t + 90°)$$

The LSBs in the outputs of M_1 and M_2 are 180° out of phase with respect to each other. Hence they are cancelled out when added. So the adder output contains only the upper sideband.

Suppression of the upper sideband :

Here the modulating and the carrier signals are applied to the upper balanced modulator M_1 directly (without any phase shift). Whereas both these signals are 90° phase shifted and then applied to the lower balanced modulator M_2.

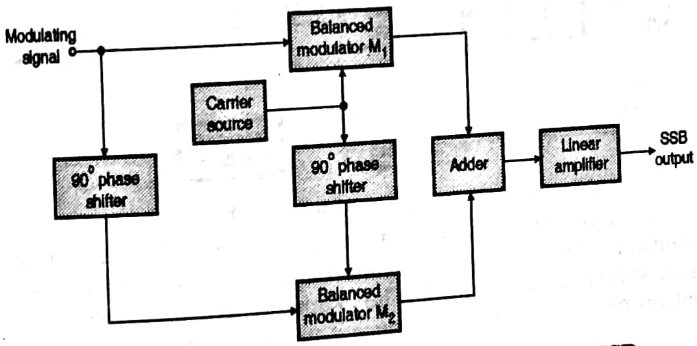


Fig. 4.15: Phase shift method to suppress the USB

Mathematical proof for USB suppression :

1. Inputs to M_1 are : $\cos\omega_m t$ and $\cos\omega_c t$
2. Inputs to M_2 are : $\cos(\omega_m t + 90°)$ and $\cos(\omega_c t + 90°)$
3. Output of $M_1 = \cos\omega_m t \cos\omega_c t$

$$= \frac{1}{2}\cos(\omega_c t + \omega_m t) + \frac{1}{2}\cos(\omega_c t - \omega_m t)$$

$$= \underbrace{\frac{1}{2}\cos(\omega_c + \omega_m)t}_{\text{USB with 0° phase shift}} + \underbrace{\frac{1}{2}\cos(\omega_c - \omega_m)t}_{\text{LSB with 0° phase shift}}$$

4. Output of $M_2 = \cos(\omega_c t + 90°) \cdot \cos(\omega_m t + 90°)$

$$= \underbrace{\frac{1}{2}\cos(\omega_c t + \omega_m t + 180°)}_{\text{USB with 180° phase shift}} + \underbrace{\frac{1}{2}\cos(\omega_c t - \omega_m t)}_{\text{LSB with 0° phase shift}}$$

5. Adder output $= \cos(\omega_c - \omega_m)t$

This output is obtained by adding outputs of M_1 and M_2. The USB in the output of M_1 and that in the output of M_2 are 180° out of phase. So in the adder they cancel each other and only the LSB is obtained.

Advantages of phase shift method :

The advantages of the phase shift method are :

1. It can generate the SSB at any frequency so the frequency up converter stage is not required.
2. It can use the low audio frequencies as modulating signal. (In filter method this is not possible.)
3. It is easy to switch from one sideband to the other.

Disadvantage :

The disadvantage is that the design of the 90° phase shifting network for the modulating signal is extremely critical. This network has to provide a correct phase shift of 90° at all the modulating frequencies which is practically extremely difficult to achieve.

Q. 18 Compare the following amplitude modulated systems for transmission/reception efficiencies.

1. DSB with carrier

2. DSB / SC

3. SSB.

Dec. 03

Sr. No.	Parameter	DSB-FC	DSB-SC	SSB
1.	Carrier suppression	N.A.	Fully	Fully
2.	Sideband suppression	N.A.	N.A.	One S.B. completely
3.	Bandwidth	$2 f_m$	$2 f_m$	f_m
4.	Transmission efficiency	Minimum	Moderate	Maximum
5.	No. of modulating inputs	1	1	1
6.	Application	Radio broadcasting	Radio broadcasting	Point to point mobile communication
7.	Power requirement to cover same area	High	Medium	Very small
8.	Complexity	Simple	Simple	Complex

Q. 19 In relation with FM, explain : Maximum frequency deviation.

May 03

Ans. : **Frequency deviation** δ represents the maximum departure of the instantaneous frequency f_i (t) of the FM wave from the carrier frequency f_c.

Since $\delta = k_f E_m$, the **frequency deviation** is proportional to the amplitude of modulating voltage (E_m) and it is independent of the modulating frequency f_m.

Maximum and minimum frequency of FM wave :

The maximum frequency of FM wave is,

$$f_{max} = f_c + \delta$$

The minimum frequency of a FM wave is $f_{min} = (f_c - \delta)$.

Q. 20 Discuss the factors that influence modulation index of an FM wave. Dec. 14. May 15

Ans. : The modulation index of an FM wave is defined as :

$$m_f = \frac{\text{Maximum frequency deviation}}{\text{Modulating frequency}}$$

$$\therefore m_f = \frac{\delta}{f_m}$$

The modulation index (m_f) is very important in FM because it decides the bandwidth of the FM wave (discussed later on).

The modulation index also decides the number of sidebands having significant amplitudes.

In AM the maximum value of the modulation index m is 1. But for FM the modulation index can be greater than 1. The modulation index m_f is measured in radians.

Deviation Ratio :

In FM broadcasting the maximum value of deviation is limited to 75 kHz. The maximum modulating frequency is also limited to 15 kHz. The modulation index corresponding to the maximum deviation and maximum modulating frequency is called as the "deviation ratio".

$$\text{Deviation ratio} = \frac{\text{Maximum deviation}}{\text{Maximum modulating frequency}}$$

Percentage Modulation of FM Wave :

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

$$\therefore \quad \%\text{ Modulation} = \frac{\text{Actual frequency deviation}}{\text{Maximum allowed deviation}}$$

Q. 21 In relation with FM, explain : Bandwidth. May 03

Ans. : Theoretically the bandwidth of the FM wave is infinite. But practically it is calculated based on how many sidebands have significant amplitude.

The simplest method to calculate the bandwidth is as follows

$$BW = 2 f_m \times \text{Number of significant sidebands}$$

With increase in modulation index, the number of significant sidebands increase. This will increase the bandwidth. The bandwidth of FM is higher than that of AM.

Carson's Rule : The second method to find the practical bandwidth is a rule of thumb (Carson's rule). It states that the bandwidth of FM wave is equal to twice the sum of the deviation and the highest modulating frequency.

$$BW = 2[\delta + f_{m\,(max)}]$$

The Carson's rule gives correct results if the modulation index is greater than 6.

Q. 22 Explain narrow band and wideband FM. May 04. Dec. 04. Dec. 05

Ans. : The FM systems are basically classified into two types :

1. **Narrowband FM (NBFM) :**

A narrowband FM is the FM wave with a small bandwidth. The modulation index m_f of narrowband FM is small as compared to one radian. Hence the spectrum of narrowband FM consists of the carrier and upper sideband and a lower sideband. For small values of m_f the values of the J coefficients are,

$$\left.\begin{aligned} J_0(m_f) &= 1, J_1(m_f) = m_f/2 \\ J_n(m_f) &= 0 \text{ for } n > 1 \end{aligned}\right\}$$

Therefore a narrowband FM wave can be expressed mathematically as follows,

$$e_{FM} = s(t) = \underbrace{E_c \sin \omega_c t}_{\text{Carrier}} + \underbrace{\frac{m_f E_c}{2} \sin(\omega_c + \omega_m) t}_{\text{USB}} - \underbrace{\frac{m_f E_c}{2} \sin(\omega_c - \omega_m) t}_{\text{LSB}}$$

The (–) sign associated with the LSB magnitude represents a phase shift of 180°.

Practically the narrowband FM systems have m_f less than 1. The maximum permissible frequency deviation is restricted to about 5 kHz.

This system is used in FM mobile communications such as police wireless, ambulances, taxicabs etc.

Generation of NBFM :

Fig. 4.16 shows the generation of NBFM using balanced modulator.

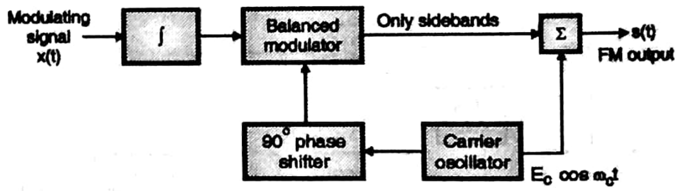


Fig. 4.16: Generation of narrow band FM

The modulating signal is integrated and applied to a balanced modulator. The other input to the balanced modulator is a 90° phase shifted carrier signal. The output of balanced modulator contains only the sidebands because it suppresses the carrier.

These sidebands are added with the carrier to obtain the NBFM signal.

Phasor diagram of NBFM :

Fig. 4.17 shows the phasor diagram of NBFM.

The carrier phasor is represented by $E_c \sin \omega_c t$. It is drawn along the horizontal axis and remains fixed always.

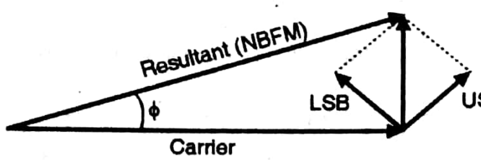


Fig. 4.17 : Phasor diagram of NBFM

The upper sideband is represented by $\frac{1}{2}$ $m_f E_c \sin(\omega_c + \omega_m)$. It is drawn as a phasor rotating in counter clockwise direction at an angular velocity ω_m. Similarly LSB is represented by $\frac{1}{2} m_f E_C \sin(\omega_c - \omega_m)$. This phasor rotates in the clockwise direction at an angular velocity of ω_m.

The resultant of the two sidebands is always perpendicular to the carrier phasor and the resultant phasor that represents the NBFM signal makes an angle ϕ with the carrier vector.

2. Wideband FM (WBFM) :

For large values of modulation index m_f, the FM wave ideally contains the carrier and an infinite number of sideband located symmetrically around the carrier. Such a FM wave has infinite bandwidth and hence called as wideband FM. The modulation index of wideband FM is much higher than 1. The maximum permissible deviation is 75 kHz and it is used in the entertainment broadcasting applications such as FM radio, TV etc.

Q. 23 The maximum deviation allowed in a FM broadcast system is 75 kHz. If the modulating signal is a single tone sinusoidal of frequency 15 kHz, find the bandwidth of the FM signal. How does the bandwidth change if the modulating frequency is doubled ? Dec. 14

Ans. : Given : $\Delta f = 75$ kHz, $f_m = 15$ kHz

To find : 1. Bandwidth of FM signal.

2. Bandwidth of FM signal if modulating frequency is doubled.

Step 1 : Calculate bandwidth :

$$BW = 2[\Delta f + f_m] = 2[75 + 15] = 180 \text{ kHz} \quad \text{...Ans.}$$

Step 2 : Calculate BW if modulating frequency is doubled :

If $f_m = 30$ kHz

$$\therefore BW = 2[\Delta f + f_m] = 2[75 + 30] = 210 \text{ kHz} \quad \text{...Ans.}$$

Q. 24 Write short note on generation of FM by Armstrong method. Dec. 15. May 16. Dec. 16

Ans. :

In the direct methods of generation of FM, LC oscillators are to be used. The crystal oscillators cannot be used.

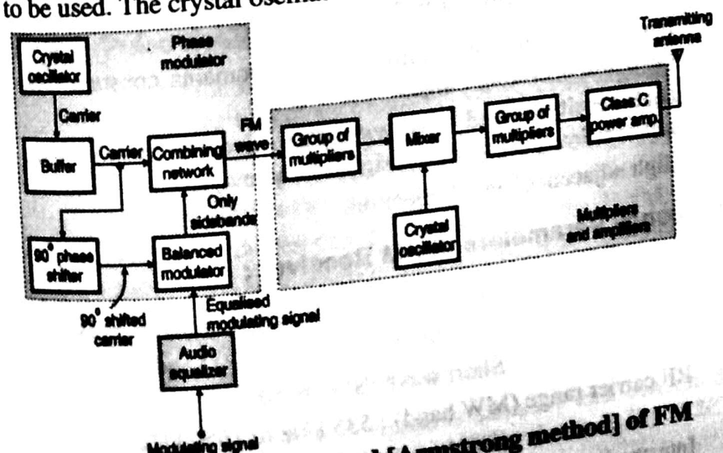


Fig. 4.18 : Indirect method [Armstrong method] of FM generation

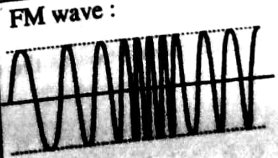
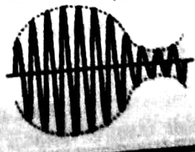
The LC oscillators are not stable enough for the communications or broadcast purpose. Thus the direct methods cannot be used for the broadcast applications. The alternative method is to use the indirect method called as the Armstrong method of FM generation.

In this method the FM is obtained through phase modulation. A crystal oscillator can be used hence the frequency stability is very high. The Armstrong method uses the phase modulation to generate frequency modulation. This method can be understood by dividing it into four parts.

Part 1 : How to obtain FM from phase modulator ?
Part 2 : Implementation of phase modulator.
Part 3 : Combining parts 1 and 2 to obtain the indirect method.
Part 4 : Use of frequency multipliers and amplifiers.

Q. 25 Compare AM with FM. Dec. 03. 4 Marks. May 04. Dec. 05. Dec. 09. May 14

Ans. :

Sr. No.	FM	AM
1.	Amplitude of FM wave is constant. It is independent of the modulation index.	Amplitude of AM wave will change with the modulating voltage.
2.	Hence transmitted power remains constant. It is independent of m_f.	Transmitted power is dependent on the modulation index.
3.	All the transmitted power is useful.	Carrier power and one sideband power are useless.
4.	FM receivers are immune to noise.	AM receivers are not immune to noise.
5.	It is possible to decrease noise further by increasing deviation.	This feature is absent in AM.
6.	Bandwidth = $2[\delta + f_m]$. The bandwidth depends on modulation index.	BW = $2f_m$. It is not dependent on the modulation index.
7.	BW is large. Hence wide channel is required.	BW is much less than FM.
8.	Space wave is used for propagation. So radius of transmission is limited to line of sight.	Ground wave and sky wave propagation is used. Therefore larger area is covered than FM.
9.	Hence it is possible to operate several transmitters on same frequency.	Not possible to operate more channels on the same frequency.
10.	FM transmission and reception equipment are more complex.	AM equipments are less complex.
11.	The number of sidebands having significant amplitudes depends on modulation index m_f.	Number of sidebands in AM will be constant and equal to 2.
12.	The information is contained in the frequency variation of the carrier.	The information is contained in the amplitude variation of the carrier.
13.	FM wave : 	AM wave : (L-46) 

Sr. No.	FM	AM
14.	**Applications :** Radio, TV broadcasting, police wireless, point to point communications.	**Applications :** Radio and TV broadcasting.

Q. 26 With neat diagram and waveforms, explain the principle of operation of super heterodyne receiver. Dec. 16

Ans. :

The problems in the TRF receiver are solved in this receiver by converting every selected RF signal (station) to a fixed lower frequency called as the "intermediate frequency (IF)". This frequency contains the same modulation as the original carrier. The IF signal is then amplified and detected to get back the modulating signal. The intermediate frequency is lower than the lowest frequency that is to be received i.e. 530 kHz.

As the "IF" is lower than the lowest RF signal frequency, the possibility of oscillations and instability is minimized. Also the required value of Q for constant BW does not depend on the frequency of desired input signal, because the "IF" is constant and same for all the incoming RF signals. Thus the superheterodyne receiver solves all the problems associated with the TRF receiver. The radio and TV receivers operate on the principle of **superheterodyning.** The block diagram of a superheterodyne radio receiver is shown in Fig. 4.19

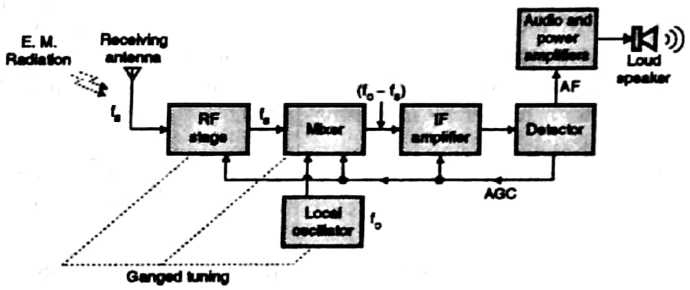


Fig. 4.19 : The superheterodyne receiver

Operation : The DSBFC or AM signal transmitted by the transmitter travels through the air and reaches the receiving antenna. This signal is in the form of electromagnetic waves. It induces a very small voltage (few μV) into the receiving antenna.

RF stage : The RF stage is an amplifier which is used to select the desired signal and reject other out of many, present at the antenna. It also reduces the effect of noise. At the output of the RF amplifier we get the desired signal at frequency "f_s".

Mixer : The mixer receives signals from the RF amplifier at frequency (f_s) and from the local oscillator at frequency (f_o) such that $f_o > f_s$.

Intermediate frequency (IF) : The mixer will mix these signals to produce signals having frequencies f_s, f_o, $(f_o + f_s)$ and $(f_o - f_s)$. Out of these the difference of frequency component i.e. $(f_o - f_s)$ is selected and all others are rejected. This frequency is called as the intermediate frequency (IF).

$$\therefore \text{ I.F. } = (f_o - f_s)$$

This frequency contains the same modulation as the original signal f_s.

In order to maintain a constant difference between the local oscillator frequency and the incoming frequency, ganged tuning is used. This is simultaneous tuning of RF amplifier, mixer and local oscillator and it is achieved by using ganged tuning capacitors (Tuning control knob in radio set).

This intermediate frequency signal is then amplified by one or more IF amplifier stages. IF amplifiers satisfy most of the gain (and hence sensitivity) and the bandwidth requirements of the receiver. Therefore the sensitivity and selectivity of this receiver do not change much with changes in the incoming frequency. The amplified IF signal is detected by the detector to recover the original modulating signal. This is then amplified and applied to the loudspeaker.

AGC means automatic gain control. This circuit controls the gains of the RF and IF amplifiers automatically to maintain a constant output voltage level even when the signal level at the receiver input is fluctuating. This is done by feeding a controlling dc voltage to the RF and IF amplifiers. The amplitude of this dc voltage is proportional to the detector output.

Q. 27 With neat diagram and waveforms, explain the principle of operation of super heterodyne receiver. Dec. 16

Ans. :

The waveforms drawn at various points in the previous section are in the time domain.

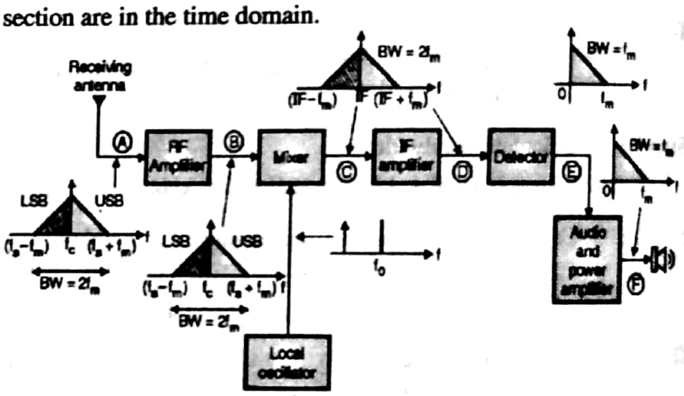


Fig. 4.20 :Superheterodyne receiver with centre frequencies and bandwidth of each block

Advantages of Superheterodyning :

1. No variation in bandwidth. The BW remains constant over the entire operating range.
2. High sensitivity and selectivity. The selectivity also remains constant irrespective of the signal frequency.
3. High adjacent channel rejection.

Frequency Parameters of AM Receiver :

The AM receiver has the following frequency parameters.

Two frequency bands : Medium wave (MW) band

Short wave (SW) band

RF carrier range (MW band) : 535 kHz to 1650 kHz

(SW band) : 5 to 15 MHz

Intermediate frequency IF : 455 kHz

IF bandwidth B : 10 kHz

Chapter 5 : Analog Pulse Modulation

Q. 1 Draw the PAM, PWM and PPM waveforms in time domain assuming a sinusoidal modulating signal. Explain them in brief. Dec. 14. May 15

Ans. :

In pulse modulation, the carrier is in the form of train of periodic rectangular pulses. Pulse modulation can be either analog or digital. In the analog pulse modulation, the amplitude, width or position of the rectangular carrier pulses is changed in accordance with the modulating signal. This will result in PAM (pulse amplitude modulation), PWM (pulse width modulation) or PPM (pulse position modulation) respectively.

PAM, PWM and PPM the examples of analog pulse modulation. The pulse modulation can be digital as well. The well known examples of digital pulse modulation are pulse code modulation (PCM), delta modulation (DM), adaptive delta modulation (ADM), etc.

The classification of the pulse modulation system is as follows :

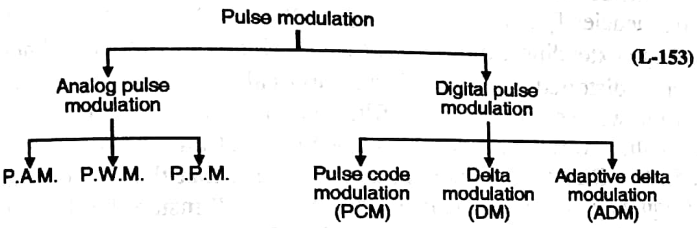


Fig. 5.1 pulse modulation system

1. Pulse Amplitude Modulation (PAM) :

The amplitude of a constant width, constant position rectangular carrier is varied in proportion with the instantaneous magnitude of the modulating signal as shown in Fig. 5.2(c).

2. Pulse Width Modulation (PWM) :

The width of carrier pulses is made to vary in proportion with the instantaneous magnitude of the modulating signal as shown in Fig. 5.2(d). PWM is also called as pulse duration modulation (PDM) or pulse length modulation (PLM).

3. Pulse Position Modulation (PPM) :

In PPM the amplitude and width of the pulses is kept constant but the position of each pulse is varied in accordance with the amplitudes of the sampled values of the modulating signal. The position of the pulses is changed with respect to the position of reference pulses.

The PPM pulses can be derived from the PWM pulses as shown in Fig. 5.2(e). Note that with increase in the modulating voltage the PPM pulses shift further with respect to reference.

4. Pulse Code Modulation (PCM) :

The analog message signal is sampled and converted to a fixed length, serial binary number as shown in Fig. 5.2(f).

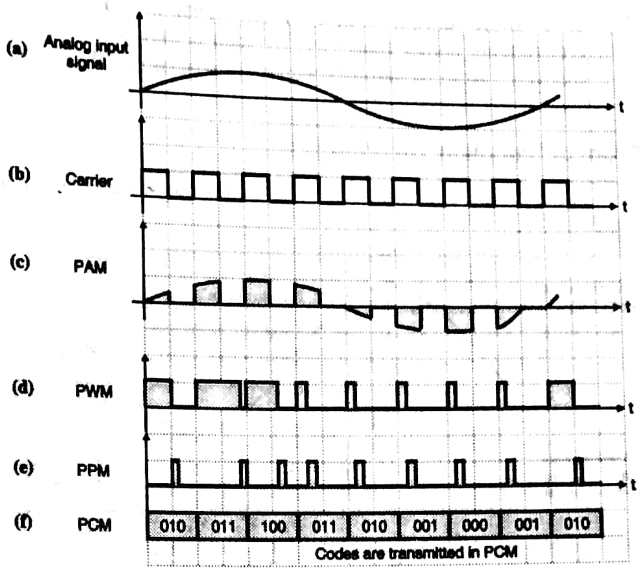


Fig. 5.2 : Pulse modulation

The PAM, PWM and PPM are called as the analog pulse communication systems whereas PCM, delta modulation (DM) are the examples of digital pulse communication systems.

Q. 2 Explain sampling technique principles. Dec. 12

Ans. :

In the pulse modulation and digital modulation systems, the signal to be transmitted must be in the discrete time form.

If the message signal is coming from a digital source (e.g. a digital computer) then it is in the proper for a digital communication system to be processed.

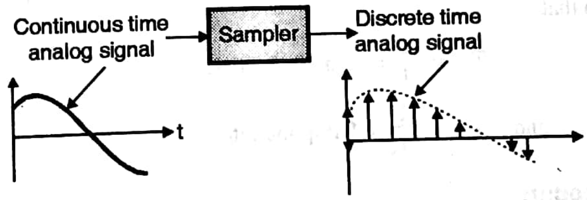


Fig. 5.3 : Sampling process

But this is not always the case. The message signal can be analog in nature. In such a case it has to be first converted into a discrete time signal. Thus using the sampling process we convert a continuous time signal into a discrete time signal.

For the sampling process to be of practical utility it is necessary to choose the sampling rate properly. The sampling process should satisfy the following requirements :

1. Sampled signal should represent the original signal faithfully.
2. Hence it should be able to reconstruct the original signal from its sampled version.

Q. 3 State sampling theorem. What happens if the sampling is done at less than 2 f_{max}. May 14

Ans. : Statement :

1. If a finite energy signal x (t) contains no frequencies higher than "W" Hz (i.e. it is a band limited signal) then it is

completely determined by specifying its values at the instants of time which are spaced (1/2W) seconds apart.

2. If a finite energy signal x (t) contains no frequency components higher than "W" Hz then it may be completely recovered from its samples which are spaced (1/2W) seconds apart.

Proof of Sampling Theorem :

Let x (t) be a continuous time analog signal as shown in Fig. 5.4.

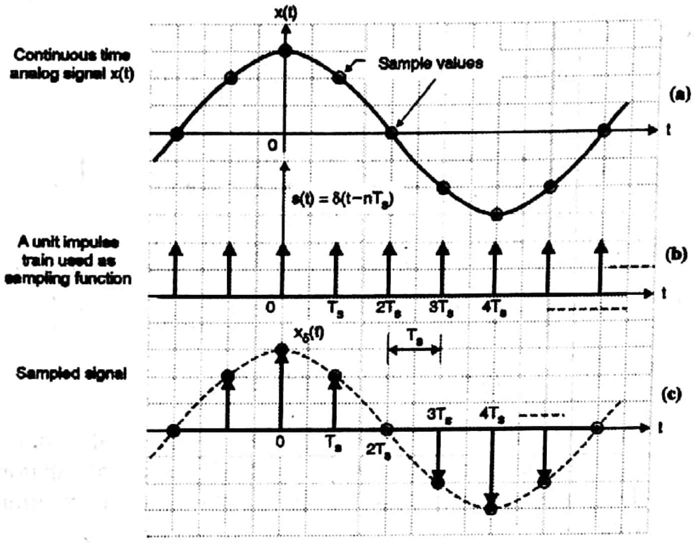


Fig. 5.4 : Sampling of a continuous time signal x (t)

Let x (t) be a signal with finite energy and infinite duration. Let x (t) be a strictly bandlimited signal. That means it does not contain any frequency components above "W" Hz.

Let s (t) be the sampling function as shown in Fig. 5.4. It is a train of unit impulses, spaced by a period of T_s seconds. This sampling function samples the original signal at a rate of "f_s" samples per second. Therefore "T_s" represents the sampling period such that,

$$T_s = \frac{1}{f_s} = \text{Sampling period}$$

$$\text{and} \quad f_s = \frac{1}{T_s} = \text{Sampling rate.}$$

Procedure :

Step 1 : Represent the sampling function s (t) mathematically.

Step 2 : Represent the sampled signal x_δ (t) mathematically.

Step 3 : Obtain the Fourier transform of the sampled signal.

Step 4 : Prove that the sampled signal x_δ (t) completely represents x (t).

Step 5 : Represent x (t) as summation of sinc functions (interpolation).

Step 6 : Graphical representation of the interpolation process.

Step 7 : Actual recovery of x (t) using an ideal low pass filter.

If the sampling is done at less than 2 f_{max}

If the signal x (t) is not strictly bandlimited and / or if the sampling frequency f_s is less than 2W, then an error called **aliasing** or **foldover error** is observed. The adjacent spectrums overlap if f_s < 2W. The signal x (t) is not strictly bandlimited. The spectrum of signal x (t) is shown in Fig. 5.5(b). The spectrum X_δ (f) of the discrete time signal x_δ (t) is shown in Fig. 5.5(b) which is nothing but the sum of X (f) and infinite number of frequency shifted replicas of it as explained earlier.

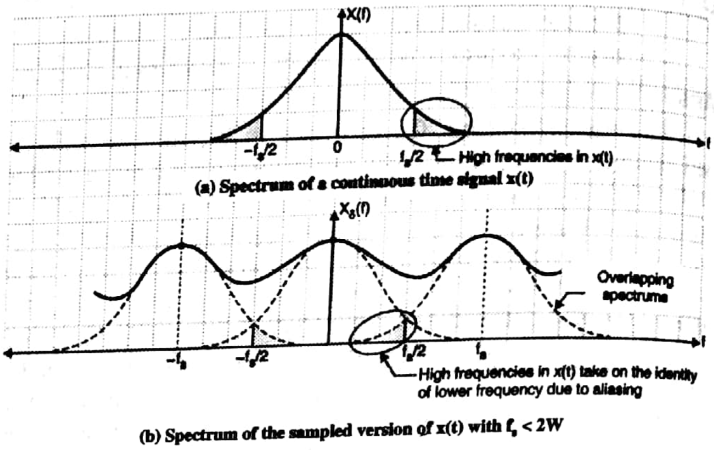


Fig. 5.5

Consider the two replicas of X (f) which are centered about the frequencies f_s and $-f_s$. By using a reconstruction filter with its pass-band extending from $-f_s/2$ to $+f_s/2$ then its output will not be an undistorted version of the original signal x (t). Some distortion will be present in the filter output. The distortion occurs due to the overlapping of the adjacent spectrums as shown in Fig. 5.5(b). Due to this overlapping, it is seen that the portions of the frequency shifted replicas are "folded over" inside the desired spectrum. Due to this "fold over", high frequencies in X (f) are reflected into low frequencies in X_δ (f).

Q. 4 Explain the following term : Aliasing effects. Dec. 04

Ans. :

Aliasing : This phenomenon of a high frequency in the spectrum of the original signal x (t), taking on the identity of lower frequency in the spectrum of the sampled signal x_δ (t) is called as aliasing or fold over error.

Effect of aliasing : Due to aliasing some of the information contained in the original signal x (t) is lost in the process of sampling.

Eliminate aliasing :

Aliasing can be completely eliminated under the following action :

Use a bandlimiting low pass filter and pass the signal x (t) through it before sampling as shown in Fig. 5.6(a).

This filter has a cutoff frequency at f_c = W, therefore it will strictly bandlimit the signal x (t) before sampling takes place. This filter is also called as antialiasing filter or prealias filter.

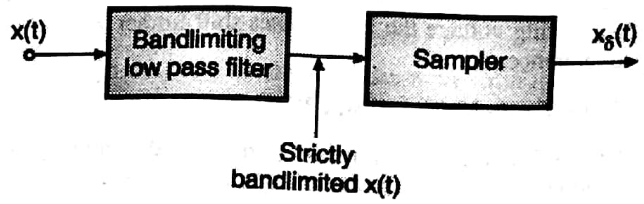


Fig. 5.6(a) : Use of a bandlimiting filter to eliminate aliasing

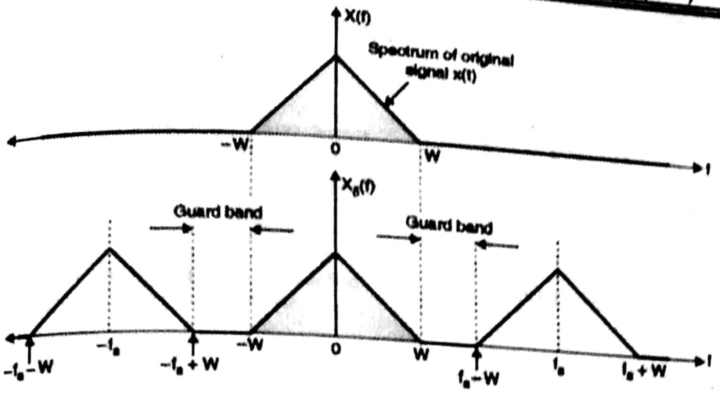


Fig. 5.6(b) : Spectrum of a sampled signal for $f_s > 2W$

Increase the sampling frequency f_s to a great extent i.e. $f_s >>> 2W$. Due to this, even though x (t) is not strictly bandlimited, the spectrums will not overlap. A guard band is created between the adjacent spectrums as shown in Fig. 5.6(b).

Thus aliasing can be prevented by :

1. Using an antialiasing or prealiasing filter and
2. Using the sampling frequency $f_s > 2W$.

Q. 5 Explain Nyquist criteria. May 16

Ans. : The minimum sampling rate of "2W" samples per second for a signal x (t) having maximum frequency of "W" Hz is called as "Nyquist rate". The reciprocal of Nyquist rate i.e. 1/2W is called as the Nyquist interval.

Nyquist rate = 2W Hz

Nyquist interval = 1/2W seconds

Effect of Nonideal Filter :

As mentioned earlier the reconstruction filter is a low pass filter. It is expected to pass all the frequencies in the range of (– W to + W) Hz. This is because the original signal x (t) is bandlimited to "W" Hz. Therefore the frequency response of a reconstruction filter should be as shown in Fig. 5.7. This is the frequency response of an ideal low pass filter.

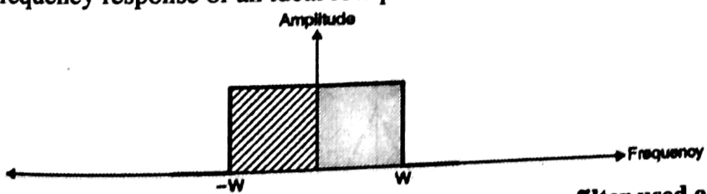


Fig. 5.7 : Frequency response of an ideal low pass filter used as a reconstruction filter

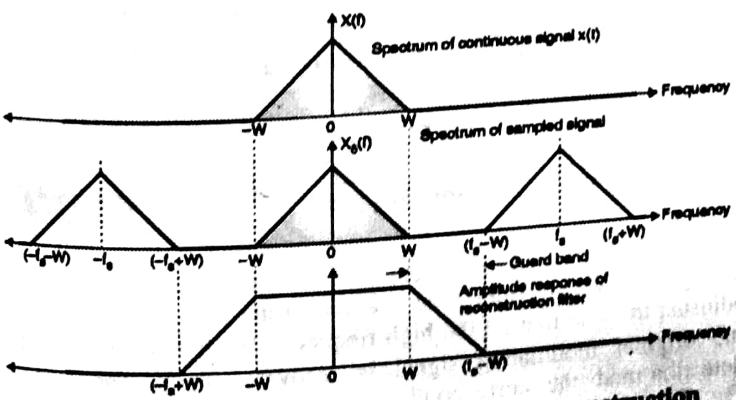


Fig. 5.8 : Amplitude response of a practical reconstruction filter

But it is not possible to practically realize an ideal low pass filter. Therefore a practical low pass filter with a frequency response as shown in Fig. 5.8 is used.

It is possible to use the practical low pass filter without introducing any distortion due to the presence of the guard bands between the adjacent frequency spectrums as shown in Fig. 5.8. That is why it is necessary to have $f_s > 2W$.

Q. 6 State and prove the sampling theorem for band pass filters. Dec. 10, May 12

Ans. : The sampling theorem for the bandpass signals can be stated as follows :

A bandpass signal x (t), having a maximum bandwidth of 2W Hz can be completely represented in its sampled form and recovered back from the sampled form if it is sampled at a rate which is at least twice the maximum bandwidth. (i.e. $f_s \geq 4W$.)

Quadrature Sampling of Bandpass Signals :

Consider a scheme called "quadrature sampling" for the uniform sampling of bandpass signals. This scheme is actually a natural extension of the sampling of low pass signals. The scheme is as follows :

In this scheme, it do not sample the bandpass signal directly. Instead, before sampling it represent the bandpass signal x (t) in terms of its "in-phase" and "quadrature" components, x_I (t) and x_Q (t) respectively.

The in-phase and quadrature components can be obtained by multiplying the bandpass signal x (t) by cos $(2\pi f_c t)$ and sin $(2\pi f_c t)$ respectively and then by suppressing the sum frequency components by means of low pass filters as shown in Fig. 5.9(a).

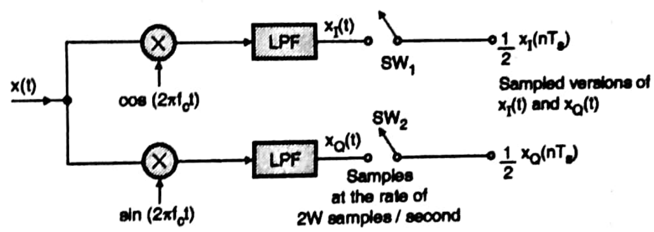


Fig. 5.9(a) : Generation of in-phase and quadrature samples from the bandpass signal x (t)

If x_I (t) = In-phase component and x_Q (t) = Quadrature component. Then we can express the bandpass signal x (t) in terms of x_I (t) and x_Q (t) as follows :

$$x(t) = x_I(t) \cos(2\pi f_c t) - x_Q(t) \sin(2\pi f_c t) \quad ...(1)$$

Under the assumption of $f_c > W$, it is found that x_I (t) and x_Q (t) both are "low pass signals" extending from – W to + W.

Q. 7 Draw the PAM, PWM and PPM waveforms in time domain assuming a sinusoidal modulating signal. Explain them in brief. Dec. 14, May 15

Ans. : In the PAM system, the amplitude of the pulsed carrier is changed in proportion with the instantaneous amplitude of the modulating signal x(t). So the information is contained in the amplitude variation of PAM signal. The carrier is in the form of train of narrow pulses as shown in Fig. 5.10. If you compare the PAM system with the sampling process, you will find that these two processes are identical. The PAM signal is then sent by either wire or cable or it is used to modulate a carrier.

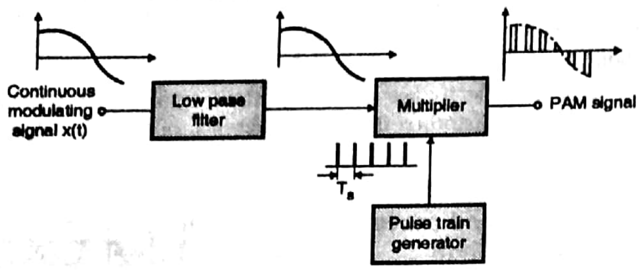


Fig. 5.10 : Generation of PAM

Types of PAM :

There are two types of PAM :

1. Natural PAM 2. Flat top PAM

Q. 8 Explain generation of PAM. Dec. 12

Ans. :

The continuous modulating signal x(t), is passed through a low pass filter. The LPF will bandlimit this signal to f_m. That means all the frequency components higher than the frequency f_m are removed. Bandlimiting is necessary to avoid the "aliasing" effect in the sampling process.

The pulse train generator generates a pulse train at a frequency f_s, such that $f_s \geq 2f_m$. Thus the Nyquist criteria is satisfied.

The rectangular narrow carrier pulses generated by the pulse train generator would carry out the uniform "sampling" in the multiplier block, to generate the PAM signal as shown in Fig. 5.11. The samples are placed T_s seconds away from each other.

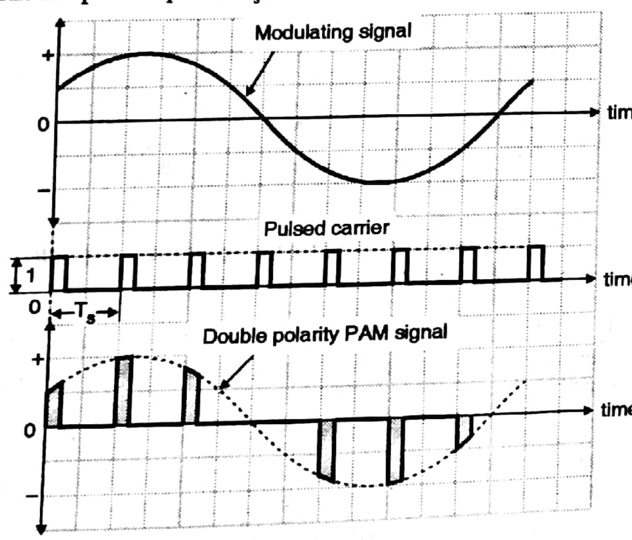
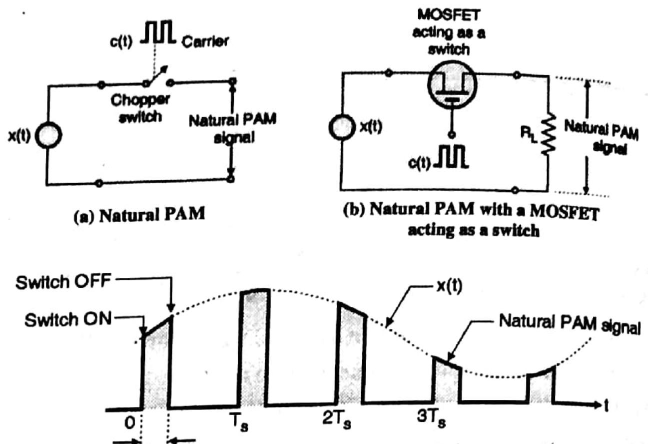


Fig. 5.11 : Waveform of natural PAM

The "information" in the modulating signal is contained in the "Amplitude variations" of the pulsed carrier.

Natural PAM is sometimes called as chopper sampled PAM because the waveform of the sampled signal appears to be chopped off from the continuous time signal x(t). The chopper arrangement is as shown in Fig. 5.12 where the chopper switch is being operated by the pulsed carrier "c (t)".

Circuit arrangement for natural PAM :



(a) Natural PAM

(b) Natural PAM with a MOSFET acting as a switch

(c) Natural PAM signal

Fig. 5.12

The natural PAM signal is same as the naturally sampled signal. Hence the spectrum of natural PAM signal is same as that of the naturally sampled signal. It is given by,

Spectrum of natural PAM signal,

$$S(f) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} \text{sinc}(n f_s \tau)\, X(f - n f_s) \quad ...(1)$$

Conclusions :

1. The term X (f – n f_s) represents the shifted version of the frequency spectrum X (f). The spectrum S (f) consists of X (f) and its shifted replicas as shown in Fig. 5.13.
2. These shifted replicas are observed at frequencies f = ± f_s, ± 2 f_s, ± 3 f_s ...etc.
3. The spectrum of x (t) is periodic in f_s and weighted by the sinc function. Therefore the amplitude of the spectrum of natural PAM signal reduces on both sides of Y axis as shown in Fig. 5.13.

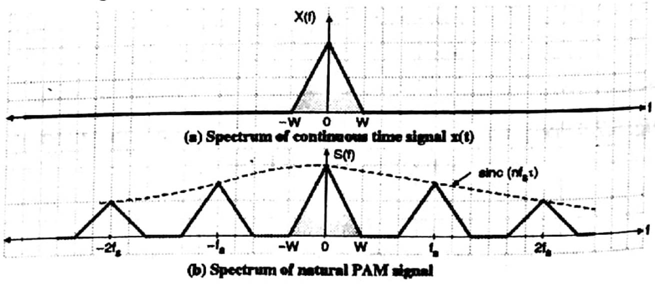


Fig. 5.13

Q. 9 Explain detection of PAM. Dec. 12

Ans. :

The PAM signal can be detected (demodulated) by passing it through a low pass filter. The low pass filter cutoff frequency is adjusted to f_m so that all the high frequency ripple is removed and the original modulating signal is recovered back. The PAM detection and the corresponding waveforms are as shown in Fig. 5.14. From the waveforms, it is seen that the demodulated output signal is close to the original modulating signal x (t).

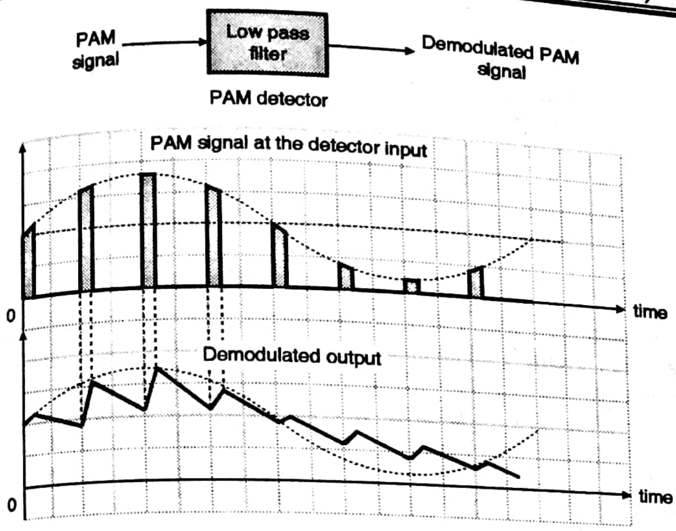


Fig. 5.14 : Detection of PAM and waveforms

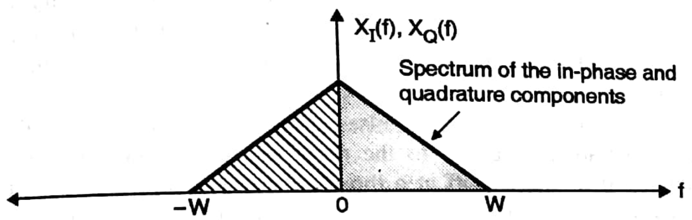


Fig. 5.15(a) : Spectrum of the in-phase and quadrature components of x (t)

Then both the in-phase and quadrature components are separately sampled at a rate of 2W samples per second by the switches SW_1 and SW_2 as shown in Fig. 5.15(a) to obtain the sampled versions of x_I (t) and x_Q (t).

In order to reconstruct the original bandpass signal from its quadrature sampled version, first reconstruct the in-phase component $x_I(t)$ and quadrature component x_Q (t) from their respective sampled versions x_I (nT_s) and x_Q (nT_s) by means of reconstruction filters. Then multiply x_I (t) and x_Q (t) by cos $(2 \pi f_c t)$ and sin $(2 \pi f_c t)$ respectively and add the result. The reconstruction process of x (t) is shown in Fig. 5.15(b).

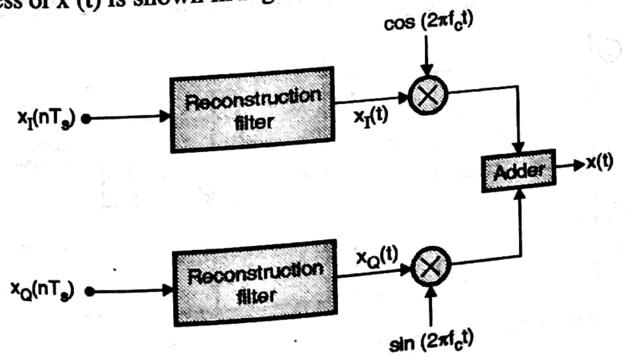
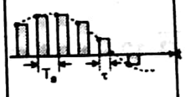


Fig. 5.15(b) : Reconstruction of the bandpass signal x(t)

Q. 10 Give Difference between Natural PAM and Flat top PAM Techniques.

Ans. :

Table 5.1 : Comparison of PAM techniques

Sr. No.	Parameter	Natural PAM	Flat top PAM
1.	Nature of the sampling function	Train of finite duration pulses	Train of finite duration pulses
2.	Circuit arrangement	Uses a chopper	Uses a sample and hold circuit
3.	Practical realizability	Practically realizable	Practically realizable
4.	Waveforms	 (D-452)	
5.	Sampling rate	Satisfies Nyquist criteria	Satisfies Nyquist criteria
6.	Mathematical representation in time domain	$s(t) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} x(t)\, \text{sinc}(nf_s\tau)\, e^{j2\pi n f_s t}$	$s(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\, h(t - nT_s)$
7.	Frequency spectrum	$S(f) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} \text{sinc}(nf_s\tau)\, X(f - nf_s)$	$S(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)\, H(f)$
8.	Signal power	Increases with increase in the pulse width τ.	Increases with increase in pulse width τ.
9.	Bandwidth requirement	Increases with the reduction in pulse width.	Increases with the reduction in pulse width.
10.	Effect of noise	Moderate.	Moderate.

Q. 11 With proper waveforms explain principles of PWM system of modulation. **Dec. 03**

Ans. : The other type of a pulse analog modulation is the pulse width modulation (PWM). In PWM, the width of the carrier pulses varies in proportion with the amplitude of modulating signal. The waveforms of PWM are as shown in Fig. 5.16. As seen from the waveforms, the amplitude and the frequency of the PWM wave remains constant. Only the width changes. That is why the "information" is contained in the width variation. This is similar to FM. As the noise is normally "additive" noise, it changes the amplitude of the PWM signal.

At the receiver, it is possible to remove these unwanted amplitude variations very easily by means of a limiter circuit.

As the information is contained in the width variation, it is unaffected by the amplitude variations introduced by the noise. Thus the PWM system is more immune to noise than the PAM signal.

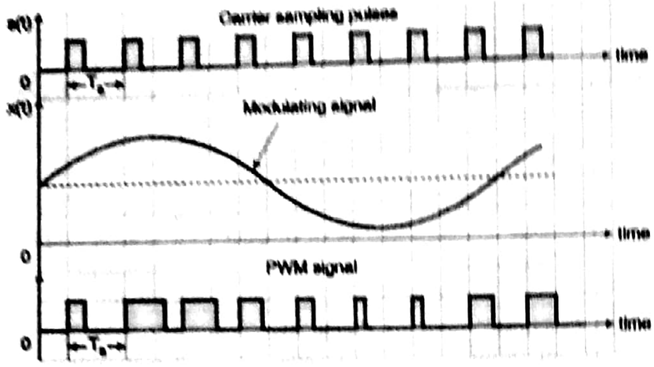


Fig. 5.16 : PWM signal [Trail edge modulated signal]

Q. 12 Explain pulse width modulation.

Dec. 09, May 12

Ans. :

The block diagram of Fig. 5.17(a) can be used for the generation of PWM as well as PPM.

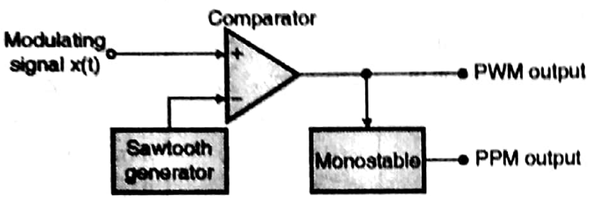


Fig. 5.17(a) : PWM and PPM generator

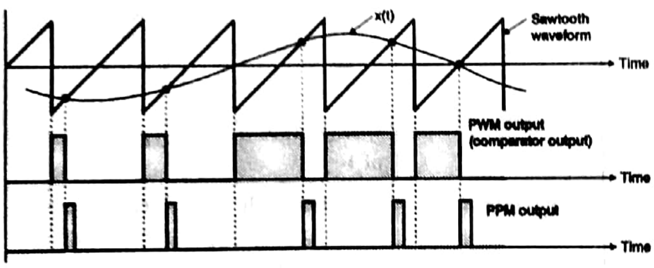


Fig. 5.17(b) : Waveforms

A sawtooth generates a sawtooth signal of frequency f_s, therefore the sawtooth signal in this case is a sampling signal. It is applied to the inverting terminal of a comparator. The modulating signal x (t) is applied to the non-inverting terminal of the same comparator. The comparator output will remain high as long as the instantaneous amplitude of x (t) is higher than that of the ramp signal. This gives rise to a PWM signal at the comparator output as shown in Fig. 5.17(b).

The leading edges of the PWM waveform coincide with the falling edges of the ramp signal. Thus the leading edges of PWM signal are always generated at fixed time instants. However the occurrence of its trailing edges will be dependent on the instantaneous amplitude of x (t). Therefore this PWM signal is said to be trail edge modulated PWM.

Q. 13 Explain pulse width demodulation.

Dec. 09, May 12

Ans. :

The block diagram of PWM detector is as shown in Fig. 5.18.

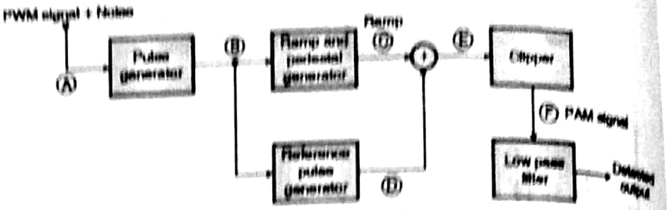


Fig. 5.18 : PWM detection circuit

Operation : The PWM signal received at the input of the detection circuit is contaminated with noise. This signal is applied to pulse generator circuit which regenerates the PWM signal. Thus some of the noise is removed and the pulses are squared up. The regenerated pulses are applied to a reference pulse generator. It produces a train of constant amplitude, constant width pulses. These pulses are synchronized to the leading edges of the regenerated PWM pulses but delayed by a fixed interval. The regenerated PWM pulses are also applied to a ramp generator. At the output of it we get a constant slope ramp for the duration of the pulse. The height of the ramp is thus proportional to the widths of the PWM pulses. At the end of the pulse a sample and hold amplifier retains the final ramp voltage until it is reset at the end of the pulse.

The constant amplitude pulses at the output of reference pulse generator are then added to the ramp signal. The output of the adder is then clipped off at a threshold level to generate a PAM signal at the output of the clipper. A low pass filter is used to recover the original modulating signal back from the PAM signal.

The waveforms for this circuit are as shown in Fig. 5.19.

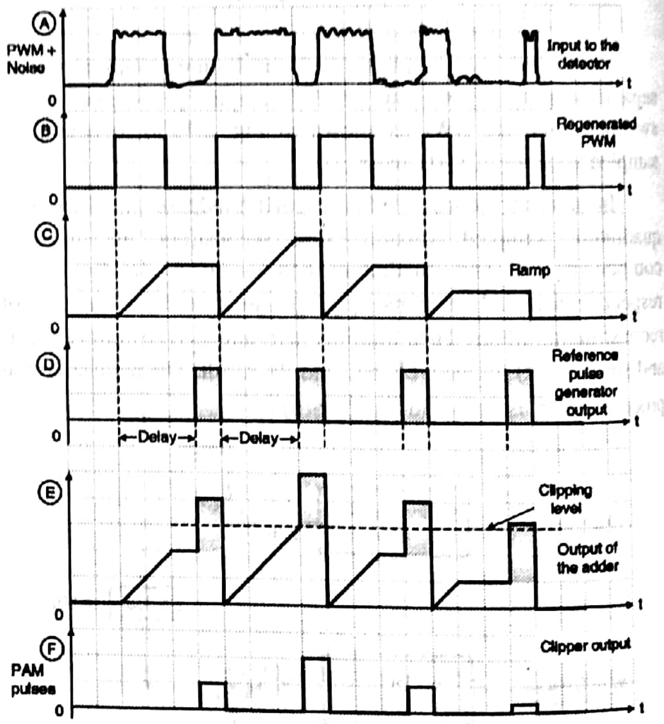


Fig. 5.19 : Waveforms for PWM detection circuit

Frequency Spectrum of PWM Wave :

Fig. 5.20 shows the spectrum of a PWM signal for a sinusoidal modulating signal with a frequency f_m. The spectrum shows that the modulating frequency f_m and many of its sidebands are present.

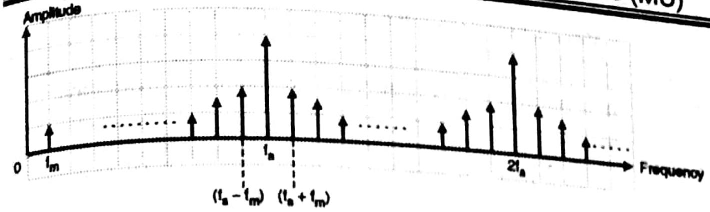


Fig. 5.20 : Spectrum of PWM signal

Q. 14 List the advantages of PWM. May 03

Ans. :

1. Less effect of noise i.e. very good noise immunity.
2. Synchronization between the transmitter and receiver is not essential. (Which is essential in PPM).
3. It is possible to reconstruct the PWM signal from a noise contaminated PWM, as discussed in the detection circuit. Thus it is possible to separate out signal from noise (which is not possible in PAM).

Q. 15 List the disadvantages of PWM. May 03

Ans. :

1. Due to the variable pulse width, the pulses have variable power contents. So the transmitter must be powerful enough to handle power corresponding to the maximum width pulse. The average power transmitted can be as low as 50% of this maximum power.
2. In order to avoid any waveform distortion, the bandwidth required for the PWM communication is large as compared to BW of PAM.

Q. 16 With proper waveforms explain principle of PPM system of modulation. Dec. 03

Ans. :

In PPM the amplitude and width of the pulsed carrier remains constant but the position of each pulse is varied in proportion with the amplitudes of the sampled values of the modulating signal. The position of the pulses is changed with respect to the position of reference pulses.

The PPM pulses can be derived from the PWM pulses as shown in Fig. 5.21. with increase in the modulating voltage the PPM pulses shift further with respect to reference.

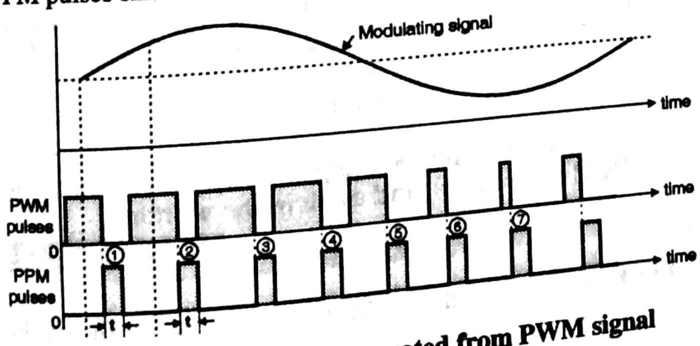


Fig. 5.21 : PPM pulses generated from PWM signal

The vertical dotted lines drawn in Fig. 5.21 are treated as reference lines to measure the shift in position of PPM pulses. The leading edge of each PPM pulse coincides with the trailing pulse of a PWM pulse. The PPM pulses marked 1, 2 and 3 etc. in Fig. 5.21 go away from their respective reference lines. This is corresponding to increase in the modulating signal amplitude. Then as the modulating voltage decreases the PPM pulses 4, 5, 6, 7 come progressively closer to their respective reference lines.

Q. 17 Explain the following term : Explain how PPM is obtained from PWM. Dec. 04

Ans. :

The PPM signal can be generated from PWM signal .The same block diagram has been repeated in Fig. 5.22 as shown. The PWM pulses obtained at the comparator output are applied to a monostable multivibrator. The monostable is negative edge triggered. Hence corresponding to each trailing edge of PWM signal, the monostable output goes high. It remains high for a fixed time decided by its own RC components.

Thus as the trailing edges of the PWM signal keep shifting in proportion with the modulating signal x(t), the PPM pulses also keep shifting as shown in Fig. 5.22.

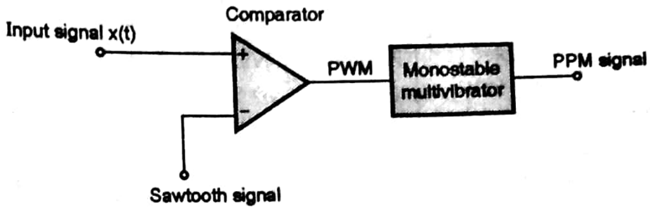


Fig. 5.22 : Generation of PPM signal

All the PPM pulses have the same width and amplitude. The information is conveyed via changing position of the pulses.

Demodulation of PPM :

The PPM demodulator block diagram is as shown in Fig. 5.23.

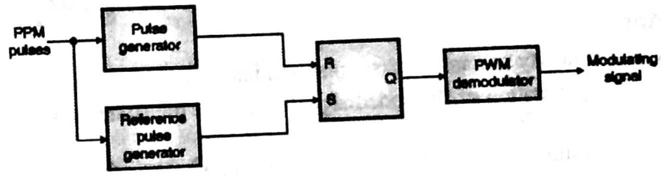


Fig. 5.23 : PPM demodulator circuit

The operation of the demodulator circuit is explained as follows :

The noise corrupted PPM waveform is received by the PPM demodulator circuit. The pulse generator develops a pulsed waveform at its output of fixed duration and apply these pulses to the reset pin (R) of a SR flip-flop. A fixed period reference pulse is generated from the incoming PPM waveform and the SR flip-flop is set by the reference pulses. Due to the set and reset signals applied to the flip-flop, we get a PWM signal at its output. The PWM signal can be demodulated using the PWM demodulator.

Q. 18 List the advantages of PPM. May 03

Ans. :

1. Due to constant amplitude of PPM pulses, the information is not contained in the amplitude. Hence the noise added to PPM signal does not distort the information. Therefore it has good noise immunity. This is same as that explained for PWM.
2. It is possible to reconstruct PPM signal from the noise contaminated PPM signal. This is also possible in PWM but not possible in PAM.
3. Due to constant amplitude of pulses, the transmitted power always remains constant. It does not change as it used to, in PWM.

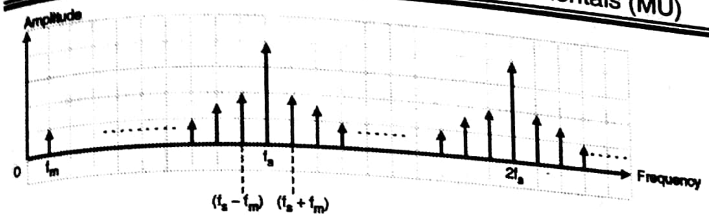


Fig. 5.20 : Spectrum of PWM signal

Q. 14 List the advantages of PWM. May 03

Ans. :

1. Less effect of noise i.e. very good noise immunity.
2. Synchronization between the transmitter and receiver is not essential. (Which is essential in PPM).
3. It is possible to reconstruct the PWM signal from a noise contaminated PWM, as discussed in the detection circuit. Thus it is possible to separate out signal from noise (which is not possible in PAM).

Q. 15 List the disadvantages of PWM. May 03

Ans. :

1. Due to the variable pulse width, the pulses have variable power contents. So the transmitter must be powerful enough to handle power corresponding to the maximum width pulse. The average power transmitted can be as low as 50% of this maximum power.
2. In order to avoid any waveform distortion, the bandwidth required for the PWM communication is large as compared to BW of PAM.

Q. 16 With proper waveforms explain principle of PPM system of modulation. Dec. 03

Ans. :

In PPM the amplitude and width of the pulsed carrier remains constant but the position of each pulse is varied in proportion with the amplitudes of the sampled values of the modulating signal. The position of the pulses is changed with respect to the position of reference pulses.

The PPM pulses can be derived from the PWM pulses as shown in Fig. 5.21. with increase in the modulating voltage the PPM pulses shift further with respect to reference.

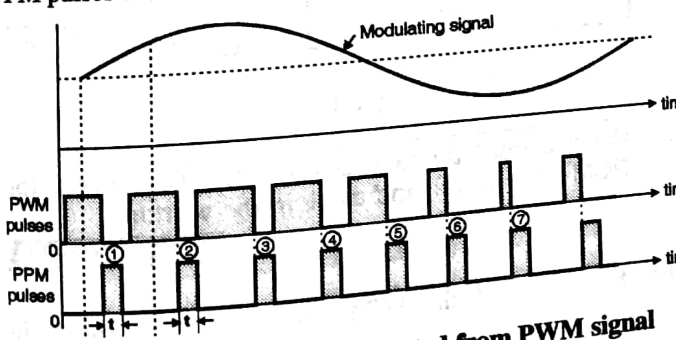


Fig. 5.21 : PPM pulses generated from PWM signal

The vertical dotted lines drawn in Fig. 5.21 are treated as reference lines to measure the shift in position of PPM pulses. The leading edge of each PPM pulse coincides with the trailing pulse of a PWM pulse. The PPM pulses marked 1, 2 and 3 etc. in Fig. 5.21 go away from their respective reference lines. This is corresponding to increase in the modulating signal amplitude. Then as the modulating voltage decreases the PPM pulses 4, 5, 6, 7 come progressively closer to their respective reference lines.

Q. 17 Explain the following term : Explain how PPM is obtained from PWM. Dec. 04

Ans. :

The PPM signal can be generated from PWM signal .The same block diagram has been repeated in Fig. 5.22 as shown. The PWM pulses obtained at the comparator output are applied to a monostable multivibrator. The monostable is negative edge triggered. Hence corresponding to each trailing edge of PWM signal, the monostable output goes high. It remains high for a fixed time decided by its own RC components.

Thus as the trailing edges of the PWM signal keep shifting in proportion with the modulating signal x(t), the PPM pulses also keep shifting as shown in Fig. 5.22.

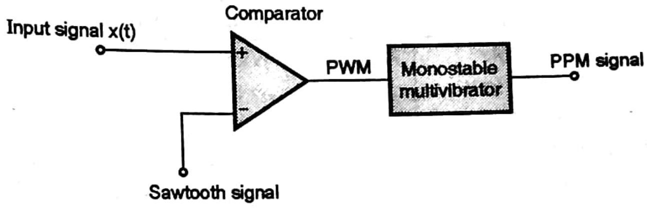


Fig. 5.22 : Generation of PPM signal

All the PPM pulses have the same width and amplitude. The information is conveyed via changing position of the pulses.

Demodulation of PPM :

The PPM demodulator block diagram is as shown in Fig. 5.23.

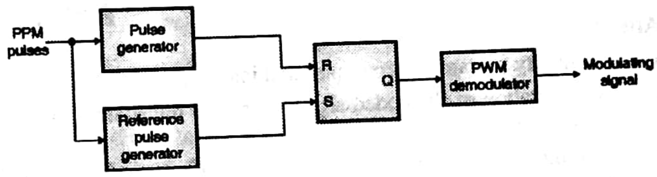


Fig. 5.23 : PPM demodulator circuit

The operation of the demodulator circuit is explained as follows :

The noise corrupted PPM waveform is received by the PPM demodulator circuit. The pulse generator develops a pulsed waveform at its output of fixed duration and apply these pulses to the reset pin (R) of a SR flip-flop. A fixed period reference pulse is generated from the incoming PPM waveform and the SR flip-flop is set by the reference pulses. Due to the set and reset signals applied to the flip-flop, we get a PWM signal at its output. The PWM signal can be demodulated using the PWM demodulator.

Q. 18 List the advantages of PPM. May 03

Ans. :

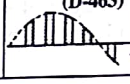
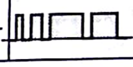
1. Due to constant amplitude of PPM pulses, the information is not contained in the amplitude. Hence the noise added to PPM signal does not distort the information. Therefore it has good noise immunity. This is same as that explained for PWM.
2. It is possible to reconstruct PPM signal from the noise contaminated PPM signal. This is also possible in PWM but not possible in PAM.
3. Due to constant amplitude of pulses, the transmitted power always remains constant. It does not change as it used to, in PWM.

Q. 19 List the disadvantages of PPM. **May 03**

Ans. :

1. As the position of the PPM pulses is varied with respect to a reference pulse, a transmitter has to send synchronizing pulses to operate the timing circuits in the receiver. Without them the demodulation won't be possible to achieve.
2. Large bandwidth is required to ensure transmission of undistorted pulses.

Q. 20 Compare various pulse modulation techniques. **Dec. 16**

Sr. No.	Parameter	PAM	PWM	PPM
1.	Type of carrier	Train of pulses	Train of pulses	Train of pulses
2.	Variable characteristic of the pulsed carrier	Amplitude	Width	Position
3.	Bandwidth requirement	Low	High	High
4.	Noise immunity	Low	High	High
5.	Information is contained in	Amplitude variations	Width variation	Position variation
6.	Transmitted power	Varies with amplitude of pulses	Varies with variation in width	Remains constant
7.	Need to transmit synchronizing pulses	Not needed	Not needed	Necessary
8.	Complexity of generation and detection	Complex	Easy	Complex
9.	Similarity with other modulation systems	Similar to AM	Similar to FM	Similar to PM
10.	Output waveforms	 (D-463)		

Chapter 6 : Digital Pulse Modulation

Q. 1 Explain the following :Advantages of digital communication. **May 04**

Ans. :

1. Immunity to transmission noise and interference.
2. Regeneration of the coded signal along the transmission path is possible.
3. Communication can be kept "private" and "secured" through the use of encryption.
4. It is possible to use a uniform format for different kinds of baseband signals.
5. It is possible to store the signal and process it further.
6. Digital signals are better suited for processing and multiplexing.
7. Digital transmission systems are more immune to noise.
8. Measurement and evaluation of digital signals is simpler.
9. It is possible to evaluate error performance of digital systems.

Q. 2 Explain the following :Disadvantages of digital communication. **May 04**

Ans. :

1. The required bandwidth is increased due to digital technology.
2. System complexity is increased.
3. In order to convert the analog signal to digital prior to transmission and then from digital to analog at the receiver, we need to use the additional encoders and decoder circuits.
4. Synchronization is necessary for the digital systems (between transmission and receiver clocks).
5. Digital transmission systems are not compatible to the older analog transmission systems.

Q. 3 Explain PCM in detail. **Dec. 12**

Ans. :

Fig. 6.1 shows the simplified block diagram of a PCM system. It consists of a transmitter and receiver. The transmitter converts the message signal x (t) into a series of coded pulses and sends it over the communication channel.

The transmitter is also called as an encoder. The receiver performs exactly in the reverse way as compared to the transmitter. It will convert the received encoded PCM pulses back into the message signal.

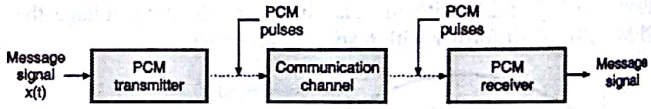


Fig. 6.1

Q. 4 Draw a neat block diagram and waveforms for PCM transmitter and explain the working. **May 11, Dec. 11**

Ans. :

Block diagram of the PCM transmitter is as shown in Fig. 6.2.

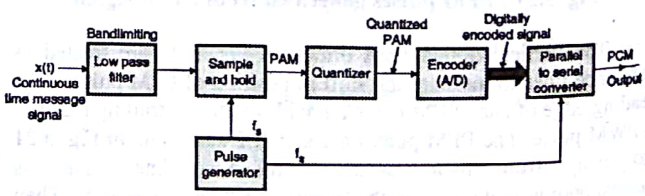


Fig. 6.2 : PCM transmitter (Encoder)

Operation of PCM transmitter :

Operation of the PCM transmitter is as follows :

The analog signal x (t) is passed through a bandlimiting low pass filter, which has a cut-off frequency f_c = W Hz. This will ensure that x (t) will not have any frequency component higher than "W". This will eliminate the possibility of aliasing. The band limited analog signal is then applied to a sample and hold circuit where it is sampled at adequately high sampling rate. Output of sample and hold block is a flat topped PAM signal. These samples are then subjected to the operation called "Quantization" in the "Quantizer". Quantization process is the process of approximation as will be explained later on. The quantization is used to reduce the effect of noise. The combined effect of sampling and quantization produces the quantized PAM at the quantizer output.

The quantized PAM pulses are applied to an encoder which is basically an A to D converter. Each quantized level is converted into an N bit digital word by the A to D converter. The value of N can be 8, 16, 32, 64 etc. The encoder output is converted into a stream of pulses by the parallel to serial converter block. Thus at the PCM transmitter output we get a train of digital pulses. A pulse generator produces a train of rectangular pulses with each pulse of duration "τ" seconds. The frequency of this signal is "f_s" Hz. This signal acts as a sampling signal for the sample and hold block. The same signal acts as "clock" signal for the parallel to serial converter. The frequency "f_s" is adjusted to satisfy the Nyquist criteria.

Waveforms :

The waveforms at various points in the PCM transmitter are as shown in Fig. 6.3.

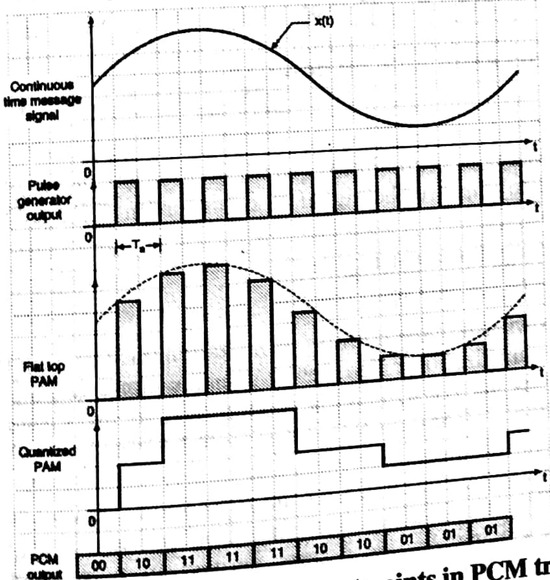


Fig. 6.3 : Waveforms at different points in PCM transmitte

Shape of the PCM Signal (A to D Conversion Concept) :

Fig. 6.4 shows input to and output of a PCM system. It is important to understand that the output is in the form of binary codes. Each transmitted binary code represents a particular amplitude of the input signal. Hence the "information" is contained in the "code" which is being transmitted.

The range of input signal magnitudes is divided into 8-equal levels. Each level is denoted by a three bit digital word between 000 and 111. Input signal x (t) is sampled. If the sample is in the 5th - window of amplitude then a digital word 101 is transmitted. If the sample is in the 2nd - window then the transmitted word is 010 and so on. In this example it converted the amplitudes into 3 bit codes, but in practice the number of bits per word can be as high as 8, 9 or 10.

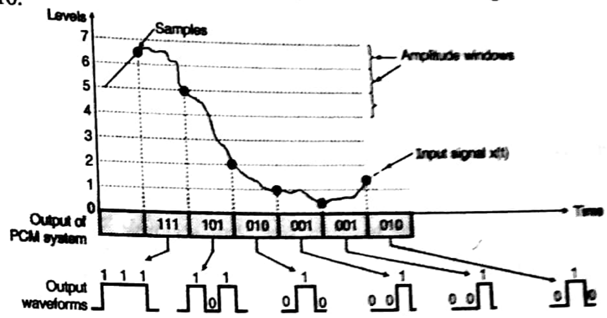


Fig. 6.4 : Input and output waveforms of a PCM system

Q. 5 Explain PCM Transmission Path.

Ans. :

The path between the PCM transmitter and PCM receiver over which the PCM signal travels is called as PCM transmission path and it is as shown in Fig. 6.5.

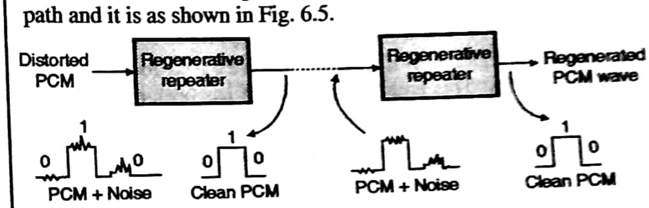


Fig. 6.5 : PCM transmission path

The most important feature of PCM system lies in its ability to control the effects of distortion and noise when the PCM wave travels on the channel. PCM accomplishes this capacity by means of using a chain of regenerative repeaters as shown in Fig. 6.5. Such repeaters are spaced close enough to each other on the transmission path. The regenerative repeater performs three basic operations namely equalization, timing and decision making. So each repeater actually reproduces the clean noise free PCM signal from the PCM signal distorted by the channel noise. This improves the performance of PCM in presence of noise.

Q. 6 Draw a neat block diagram and waveforms for PCM receiver and explain the working. May 11, Dec. 11

Ans. :

Fig. 6.6 shows the block diagram of a PCM receiver.

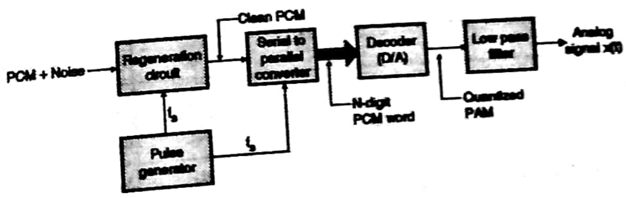


Fig. 6.6 : PCM receiver (Decoder)

Operation of PCM receiver :

A PCM signal contaminated with noise is available at the receiver input. The regeneration circuit at the receiver will separate the PCM pulses from noise and will reconstruct the original PCM signal. The pulse generator has to operate in synchronization with that at the transmitter. Thus at the regeneration circuit output we get a "clean" PCM signal. The reconstruction of PCM signal is possible due to the digital nature of PCM signal. The reconstructed PCM signal is then passed through a serial to parallel converter.

Output of this block is then applied to a decoder. The decoder is a D to A converter which performs exactly the opposite operation of the encoder. The decoder output is the sequence of a quantized multilevel pulses. The quantized PAM signal is thus obtained, at the output of the decoder. This quantized PAM signal is passed through a low pass filter to recover the analog signal, x (t). The low pass filter is called as the reconstruction filter and its cut off frequency is equal to the message bandwidth W.

Q. 7 Write short note on quantization. **May 12**

Ans. :

Quantization is a process of approximation or rounding off. The sampled signal in PCM transmitted is applied to the quantizer block. Quantizer converts the sampled signal into an approximate quantized signal which consists of only a finite number of predecided voltage levels. Each sampled value at the input of the quantizer is approximated or rounded off to the nearest standard predecided voltage level. These standard levels are known as the "quantization levels". Refer to Fig. 6.7 to understand the process of quantization. The quantization process takes place as follows :

The input signal x (t) is assumed to have a peak to peak swing of V_L to V_H volts. This entire voltage range has been divided into "Q" equal intervals each of size "s".

"s" is called as the step size and its value is given as,

$$s = \frac{V_H - V_L}{Q} \qquad ...(1)$$

In Fig. 6.7, the value of Q = 8

At the center of these ranges, the quantization levels q_0, q_1,...q_7 are placed. Thus the number of quantization levels is Q = 8. The quantization levels are also called as decision thresholds. x_q (t) represents the quantized version of x (t). We obtain x_q (t) at the output of the quantizer. When x (t) is in the range Δ_0, then corresponding to any value of x (t), the quantizer output will be equal to "q_0". Similarly for all the values of x (t) in the range Δ_1, the quantizer output is constant equal to "q_1".

Thus in each range from Δ_0 to Δ_7, the signal x (t) is rounded off to the nearest quantization level and the quantized signal is produced. The quantized signal x_q (t) is thus an approximation of x (t). The difference between them is called as **quantization error or quantization noise.** This error should be as small as possible.

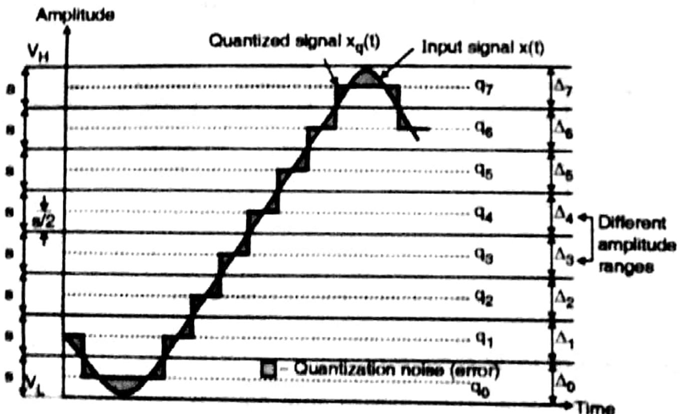


Fig. 6.7 : Process of quantization

To minimize the quantization error we need to reduce the step size "s" by increasing the number of quantization levels Q.

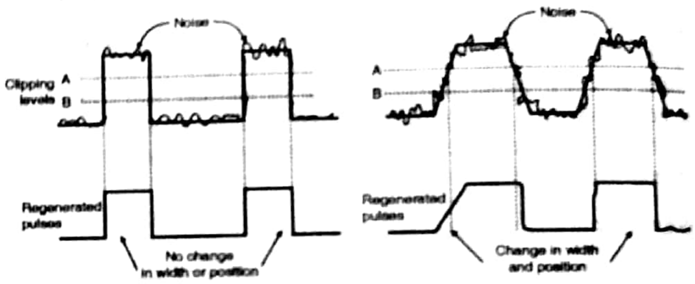
Q. 8 Why is PCM more noise resistant than other forms of pulse modulation ? **May 03 Dec 03**

Ans. :

Figs. 6.8(a) and 6.8(b) shows the effect of noise on the transmitted pulses. In Fig. 6.8(a) Due to the noise superimposed on the pulses, only the PAM system will be affected. However the PWM, PPM and PCM systems will remain unaffected. The regeneration of the pulses is achieved by using a clipper circuit with reference levels A and B.

In Fig. 6.8(b). Here the sides of the transmitted pulse are not perfectly vertical. In practice the transmitted pulses usually have slightly sloping sides (edges). As the noise is superimposed on them, the width and the position of the regenerated pulses is changed. This is going to distort the information contents in the PWM and PPM signals. But PCM is still unaffected as it does not contain any information in the width or the position of the pulses.

Thus PCM has much better noise immunity as compared to PAM, PWM and PPM systems.



(a) (b)

Fig. 6.8 : Effect of noise on PCM

Q. 9 Write short note on companding. **May 08, May 10, Dec 10**

Ans. :

Companding is non-uniform quantization. It is required to be implemented to improve the signal to quantization noise ratio of weak signals.

The quantization noise is given by,

$$N_q = s^2 / 12$$

This shows that in the uniform quantization once the step size is fixed, the quantization noise power remains constant. But the signal power is not constant. It is proportional to the square of signal amplitude. Hence signal power will be small for weak signals, but quantization noise power is constant. Therefore the signal to quantization noise ratio for the weak signals is very poor. This will affect the quality of signal. The remedy is to use **companding.** Companding is a term derived from two words, compression and expansion.

Companding = Compressing + Expanding

Practically it is difficult to implement the non-uniform quantization because it is not known in advance about the changes in the signal level. The weak signals are amplified and strong signals are attenuated before applying them to a uniform quantizer.

This process is called as "compression" and the block that provides it is called as a "compressor". At the receiver exactly opposite process is followed which is called expansion. The circuit used for providing expansion is called as an "expander".

The compression of signal at the transmitter and expansion at the receiver is combined to be called as "companding". The process of companding is shown in the block diagram form in Fig. 6.9.

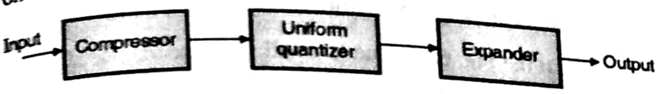


Fig. 6.9 : Model of companding

Types of Companding :

There are two possible types of companding :

1. Analog companding
2. Digital Companding

Q. 10 State Advantages, Disadvantages and Applications of PCM .

Ans. :

Advantages of PCM :

1. Very high noise immunity.
2. Due to digital nature of the signal, repeaters can be placed between the transmitter and the receivers. The repeaters actually regenerate the received PCM signal. This is not possible in analog systems. Repeaters further reduce the effect of noise.
3. It is possible to store the PCM signal due to its digital nature.
4. It is possible to use various coding techniques so that only the desired person can decode the received signal. This makes the communication secure.
5. The increased channel bandwidth requirement for PCM is balanced by the improved SNR.
6. There is a **uniform format** used for the transmission of different types of base band signals. Hence it is easy to integrate all these signals together and send them on the common network.
7. It is easy to drop or reinsert the message sources in a PCM-TDM system.

Disadvantages of PCM :

1. The encoding, decoding and quantizing circuitry of PCM is complex.
2. PCM requires a large bandwidth as compared to the other systems.

Applications of PCM :

Some of the applications of PCM are as follows :

1. In digital telephone systems.
2. In the space communication, space craft transmits signals to earth. Here the transmitted power is very low (10 to 15W) and the distances are huge (a few million km). Still due to the high noise immunity, only PCM systems can be used in such applications.

Q. 11 Discuss delta modulation and adaptive delta modulation. Dec. 16

Ans. :

In PCM system, N number of binary digits are transmitted per quantized sample. Hence the signaling rate and transmission channel bandwidth of the PCM system are very large.

These disadvantages can be overcome by using the delta modulation.

Principle of operation :

Delta modulation transmits only one bit per sample instead of N bits transmitted in PCM. This reduces its signaling rate and bandwidth requirement to a great extent.

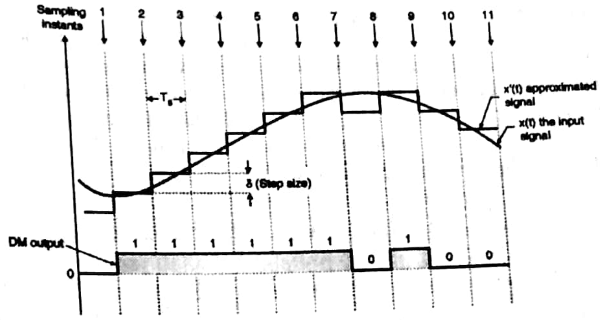


Fig. 6.10 : D.M. Waveforms

In the basic or linear D.M., a staircase approximated version of the sampled input signal is produced as shown in Fig. 6.10. The original signal and its staircase representation are then compared to produce a difference signal. And this difference signal is quantized into only two levels namely $\pm\delta$ corresponding to positive and negative difference respectively.

That means if the approximated signal x′ (t) lies below x (t) at the sampling instant, then the approximated signal is increased by "δ". Whereas if x′ (t) is greater than x (t) at the sampling instant, then x′ (t) is decreased by "δ".

D.M. output :

In Fig. 6.10, the D.M. output is 1 if the staircase signal x′ (t) is increased by "δ" i.e. at sampling instants 1, 2, 3, 4, 5 and 6. Whereas D.M. output is 0 if x′ (t) is decreased by "δ" i.e. at sampling instants 7, 9 and 10. In delta modulation, the present sample value x (t) is compared with the approximate value x′ (t) and the result of this comparison is transmitted. Thus we are sending the information of whether the present sample value is higher than or lower than the approximate value. Note that the actual sampled value is not being transmitted.

Q. 12 Draw neat block diagram of Delta modulator and explain its working. May 07, Dec. 11

Ans. :

The block diagram of a delta modulator transmitter is as shown in the Fig. 6.11.

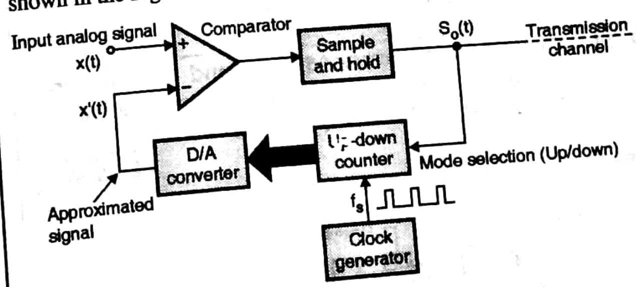


Fig. 6.11 : D.M. transmitter

The operation of the circuit is as follows :

x (t) is the analog input signal and x′ (t) is the quantized (approximated) version of x (t). Both these signals are applied to a comparator. The comparator output goes high if x (t) > x′ (t) and i goes low if x (t) < x′ (t). Thus the comparator output is either 1 or 0. The sample and hold circuit will hold this level (0 or 1) for the entire clock cycle period. The output of the sample and hold circuit is transmitted as the output of the DM system. Thus in DM, the information which is transmitted is only whether x (t) > x′ (t) or vice versa. one bit per clock cycle is being sent. This will reduce the bit rate and hence the BW. The transmitted signal is also used to decide the mode of operation of an up/down counter. The counter output increments by 1 if $S_o(t) = 1$ and it decrements by 1 if $S_o(t) = 0$, at the falling edge of each clock pulse. This is as shown in the waveform in the Fig. 6.12.

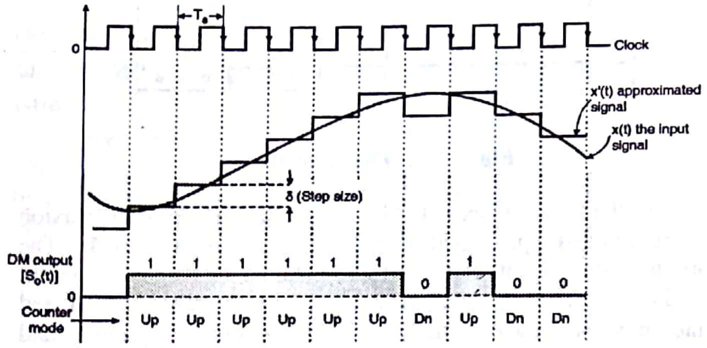


Fig. 6.12 : D.M. waveforms

The counter output is converted into analog signal by a D to A converter. Thus it produce the approximated signal x′ (t) at the output of the D to A converter.

Q. 13 Draw a neat block diagram of a delta demodulator system **May 03**

Ans. :

The block diagram of the D.M. receiver is as shown in Fig. 6.13. Compare it with the transmitter block diagram, you will find that it is identical to the chain of blocks producing the signal x′ (t) i.e. the approximated signal. The original modulating signal can be recovered back by passing this signal through a low pass filter.

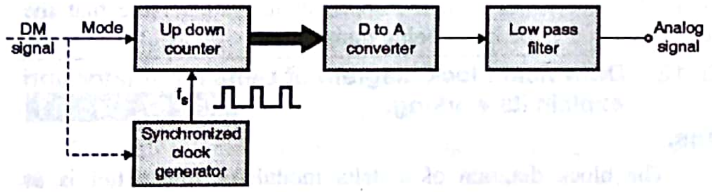


Fig. 6.13 : D.M. receiver

Q. 14 Explain slope overload error and hunting error (granular noise) in Delta Modulation. Derive the condition to avoid slope overload error. **Dec. 07**

Ans. :

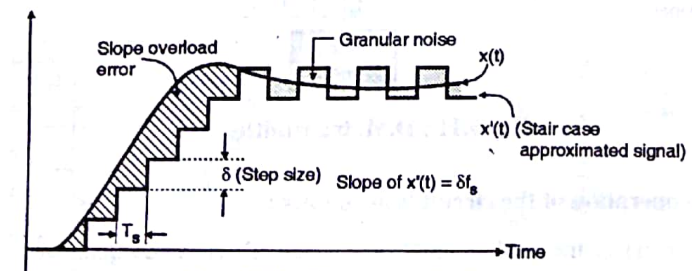


Fig. 6.14 : Distortions in D.M.

The DM system is subjected to two types of quantization error or distortions :

1. Slope overload distortion :

Look at the Fig. 6.14. Due to small step size (δ), the slope of the approximated signal x′ (t) will be small.

$$\text{The slope of } x'(t) = \frac{\delta}{T_s} = \delta f_s \quad ...(1)$$

If slope of the input analog signal x (t) is much higher than that of x′ (t) over a long duration then x′ (t) will not be able to follow the variations in x (t), at all. The difference between x (t) and x′ (t) is called as the slope overload distortion. Thus the slope overload error occurs when slope of x (t) is much larger than slope of x′ (t).

The slope overload error can be reduced by increasing slope of the approximated signal x′ (t). Slope of x′ (t) can be increased and hence the slope overload error can be reduced by either increasing the step size "δ" or by increasing the sampling frequency, f_s. However with increase in δ the granular noise increases and if f_s is increased, signaling rate and bandwidth requirements will go up. Thus reducing the slope overload error is not easy.

2. Granular noise :

When the input signal x (t) is relatively constant in amplitude, the approximated signal x′ (t) will fluctuate above and below x (t) as shown in Fig. 6.14. The difference between x (t) and x′ (t) is called as granular noise. The granular noise is similar to the quantization noise in the PCM system. It increases with increase in the step size δ. To reduce the granular noise, the step size should be as small as possible.

However this will increase the slope overload distortion. In the linear delta modulator the step size δ is not variable. If it is made variable then the slope overload distortion and granular noise both can be controlled. A system with a variable step size is known as the adaptive delta modulator (ADM).

D.M. Bit Rate (Signaling Rate) :

D.M. bit rate (r) = Number of bits transmitted / second

= Number of samples/sec × Number of bits/sample = $f_s \times 1 = f_s$

Thus the D.M. bit rate is (1/N) times the bit rate of a PCM system, where N is the number of bits per transmitted PCM codeword. Hence the channel bandwidth for the D.M. system is reduced to a great extent as compared to that for the PCM system.

Q. 15 Give advantages of delta modulation. **May 08**

Ans. :

1. Low signaling rate and low transmission channel bandwidth, because in delta modulation, only one bit is transmitted per sample.
2. The delta modulator transmitter and receiver are less complicated to implement as compared to PCM.

Q. 16 Derive the condition to avoid slope overload error. **May 03, Dec. 06**

Ans. :

Consider a sinusoidal signal x (t) = A cos (ω_m t) applied to a delta modulator with a step size δ. Show that the slope overload distortion will occur if

$$A > \frac{\delta}{\omega_m T_s} = \frac{\delta}{2\pi}\left(\frac{f_s}{f_m}\right)$$

where T_s is the sampling period.

Let the input signal be sinusoidal with amplitude A volts and frequency f_m Hz as shown in Fig. . 6.15

The given signal is $x(t) = A \cos \omega_m t$

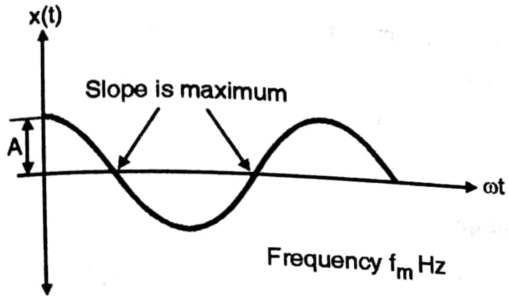


Fig. 6.15 : Input signal x (t)

The slope of this signal will be maximum when derivative of x (t) with respect to time is maximum.

$$\therefore \quad \text{Slope of } x(t) = \frac{dx(t)}{dt} = -A\,\omega_m \sin \omega_m t$$

The maximum value of the slope of x (t) $= -A\,\omega_m$ ...(1)

Slope of the staircase approximated signal $x'(t) = \frac{\delta}{T_s}$ ...(2)

To avoid the slope overload distortion, it is necessary that the maximum slope of x (t) be less than the slope of x′ (t).

$$\therefore \left|\frac{dx(t)}{dt}\right|_{max} \leq \frac{\delta}{T_s}$$

$$\therefore \quad A\,\omega_m \leq \frac{\delta}{T_s}$$

$$\therefore \quad A \leq \frac{\delta}{\omega_m T_s}$$

This is the condition for avoiding the slope overload distortion. Therefore the slope overload distortion will occur if this condition is not satisfied i.e.

$$\text{if } A > \frac{\delta}{\omega_m T_s}$$

Maximum Output Signal to Noise Ratio :

It can be proved that the maximum signal to noise ratio of a D.M. system is given by,

$$\frac{S}{N_q} = \frac{3}{8\pi^2 f_m^2 f_M T_s^3} \quad ...(3)$$

where f_M = Cutoff frequency of the low pass filter in the D.M. receiver.

Q. 17 Explain adaptive delta modulation. May 09, Dec. 09, Dec. 10, Dec. 12

Ans. :

In the ADM system, the step size is not constant. Rather when the slope overload occurs the step size becomes progressive larger and therefore x′ (t) will catch up with x (t) more rapidly.

Whenever the slope of input signal is large, the step size of the staircase approximated signal x′ (t) is increased. On the other hand when the input signal is varying slowly the step size is reduced. Thus the step size is adapted as per the level of input signal.

Types of ADM :

There are various types of ADM systems available depending on the type of scheme used for adjusting the step size. In one type a discrete set of values is provided for the step size whereas in another type a continuous range of step size variation is provided.

Q. 18 How is adaptive delta modulation better than linear delta modulation ? Draw block diagram of adaptive delta modulation and explain each block in detail. May 06

Ans. :

In the ADM system, the step size is not constant. Rather when the slope overload occurs the step size becomes progressive larger and therefore x′ (t) will catch up with x(t) more rapidly.

The ADM transmitter is as shown in Fig. 6.16.

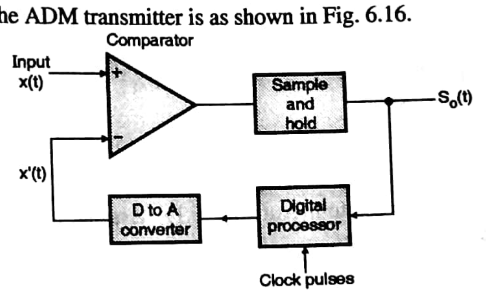


Fig. 6.16 : ADM transmitter

If you compare this block diagram with that of the linear delta modulator, then you will find that except for the counter being replaced by the digital processor, the remaining blocks are identical..

Operation :

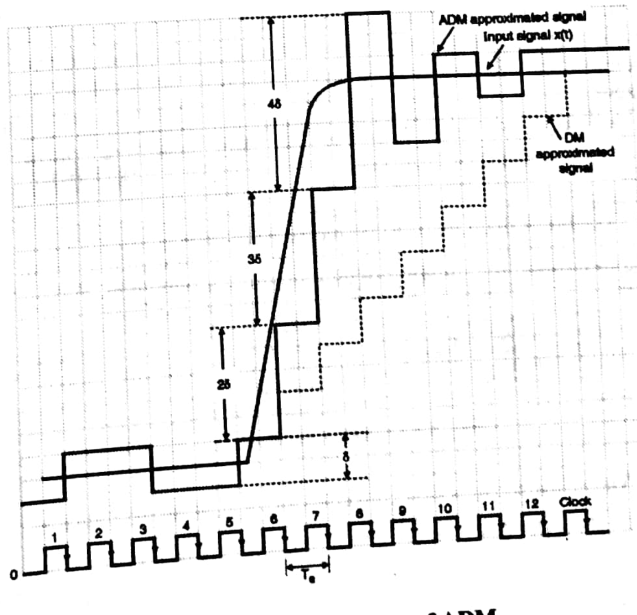


Fig. 6.17 : Waveforms of ADM

In response to the k^{th} clock pulse trailing edge, the processor generates a step which is equal in magnitude to the step generated in response to the previous i.e. $(k-1)^{th}$ clock edge.

If the direction of both the steps is same, then the processor will increase the magnitude of the present step by "δ". If the directions are opposite then the processor will decrease the magnitude of the present step by "δ".

$S_o(t)$ in the Fig. 6.16, i.e. the output of the ADM system is given as,

$$S_o(t) = +1 \text{ if } x(t) > x'(t) \text{ just before the } k^{th} \text{ clock edge.}$$

and $$S_o(t) = -1 \text{ if } x(t) < x'(t) \text{ just before the } k^{th} \text{ clock edge.}$$

Then the step size at the sampling instant k is given by,

$$\underset{\substack{\text{Step size}\\ \text{at } k^{th} \text{ clock}\\ \text{edge}}}{\delta(k)} = \underset{\substack{\text{Step size}\\ \text{at } (k-1)^{th}\\ \text{clock edge}}}{[\delta(k-1)]} \; \underset{\substack{\text{Output at}\\ k^{th} \text{ edge}}}{S_o(k)} + \underset{\substack{\text{Basic}\\ \text{step}\\ \text{size}}}{\delta} \quad ...(1)$$

$$\underset{\substack{\text{Output at } (k-1)^{th}\\ \text{clock edge}}}{S_o(k-1)}$$

example :

Refer to the waveforms of Fig. 6.17. Assume k = 6, i.e. consider the 6^{th} clock edge.

$$\therefore \quad k-1 = 5 \qquad \therefore \delta(k-1) = \delta(5) = \delta$$

$$S_o(k) = S_o(6) = +1 \quad S_o(k-1) = S_o(5) = +1$$

Substitute in Equation (6.16),

$$\delta(6) = \delta + \delta = 2\delta \quad ...(2)$$

In Fig. 6.17, the step size at the 6^{th} clock edge is 2 δ.

As shown in Fig. 6.17, due to variable step size, the slope overload error is reduced. But quantization error is increased. Due to the adjustable step size, the slope overload problem is solved. Hence ADM system has a low bit rate than the PCM system. Therefore the BW required is also less than a comparable PCM system.

Q. 19 Explain the receiver for the adaptive delta modulation system. May 10

Ans. :

The block diagram of ADM receiver is shown in Fig. 6.18.

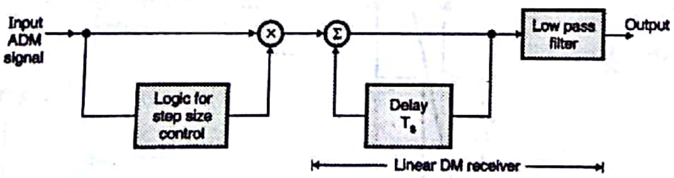


Fig. 6.18 : ADM receiver

The ADM signal is first converted into a D.M. signal with the help of the step size control logic and then applied to a D.M. receiver. At the output of low pass filter we get the original signal back.

Advantages of Adaptive Delta Modulation :

The advantages of ADM over DM are as follows :

1. Reduction in slope overload distortion and granular noise.
2. Improvement in signal to noise ratio.
3. Wide dynamic range due to variable step size.
4. Better utilization of bandwidth as compared to delta modulation.
5. Low signaling rate.
6. Simplicity of implementation.

Disadvantages :

For a relatively constant magnitude input signal x (t), the ADM will produce a high granular noise.

Q. 20 Compare PCM, DM and ADM. May 13

Ans. :

Table 6.1 : Comparison of PCM, DM and ADM

Sr. No.	Parameter	PCM	DM	ADM
1.	Number of bits per sample	N can be 4, 8, 16, 32, 64 etc.	N = 1	N = 1
2.	Step size	Depends on the number of Q levels.	Step size is fixed	Step size is variable
3.	Distortions / errors	Quantization error	Slope overload and granular noise	Granular noise
4.	Signaling rate and bandwidth	Highest	Low, if the input is slow varying	Lowest
5.	System complexity	Complex	Simple	Simple
6.	Feedback from output	No feedback	Feedback is present	Feedback is present
7.	Noise immunity	Very good	Very good	Very good
8.	Use of repeaters	Possible	Possible	Possible

Chapter 7 : Multiplexing

Q 1 What is multiplexing in communication system ? Draw block diagram of TDM-PCM system and explain. Dec. 13, May 14

Ans. : The concept of a simple multiplexer is illustrated in Fig. 7.1.

The multiplexer receives a large number of different input signals. Multiplexer has only one output which is connected to the single communication channel.

The multiplexer combines all input signals into a single composite signal and transmits it over the communication medium.

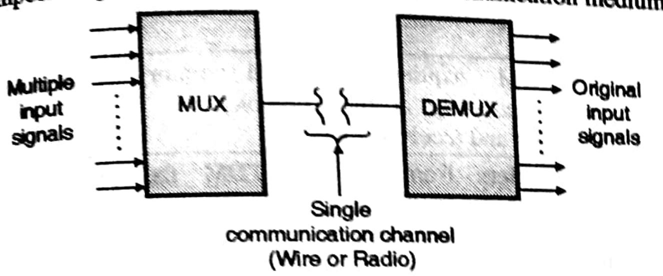


Fig. 7.1 : Concept of multiplexing

Sometimes the composite signal is used for modulating a carrier before transmission. At the receiving end, of communication link, a demultiplexer is used to separate out the signals into their original form. The operation of demultiplexer is exactly opposite to that of a multiplexer. Demultiplexing is the process which is exactly opposite to that of multiplexing.

Q. 2 Explain with suitable example what do you understand by signal multiplexing ? May 16

Ans. : There are three basic types of multiplexing. They are :

1. Frequency division multiplexing (FDM)
2. Time division multiplexing (TDM).
3. Wavelength division multiplexing (WDM).

Q. 3 Define and explain various multiplexing techniques used in communication systems. May 09, Dec. 10

Ans. : The multiplexing techniques can be broadly classified into two categories namely analog and digital. Analog multiplexing can be either FDM or WDM and digital multiplexing is TDM.

Fig. 7.2 shows the classification of multiplexing techniques.

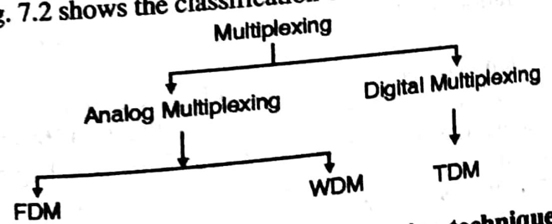


Fig. 7.2 : Classification of multiplexing techniques

Generally the FDM and WDM systems are used to deal with the analog information whereas the TDM systems are used to handle the digital information. In FDM many signals are transmitted simultaneously where each signal occupies a different frequency slot within a common bandwidth. In TDM the signals are not transmitted at a time, instead they are transmitted in different time slots.

Q. 4 Explain FDM. Dec. 03, Dec. 06, May 10, Dec. 12, May 13

Ans. : The operation of FDM is based on sharing the available bandwidth of a communication channel among the signals to be transmitted. That means many signals are transmitted simultaneously with each signal occupying a different frequency slot within thetotal available bandwidth. Each signal to be transmitted modulates a different carrier. The modulation can be AM, SSB, FM or PM. The modulated signals are then added together to form a composite signal which is transmitted over a single channel. The spectrum of composite FDM signal is shown in Fig. 7.3. Generally the FDM systems are used for multiplexing the analog signals.

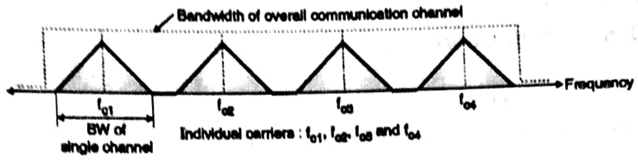


Fig. 7.3 : Spectrum of FDM signal

Q. 5 What do you understand by signal multiplexing? Explain TDM and FDM with suitable examples. Dec. 14, May 15

Ans. : The process called multiplexing is used in order to utilize common transmission channel or medium to transmit more than one signals simultaneously. TDM is a digital multiplexing process.

In TDM all the signals to be transmitted are not transmitted simultaneously. Instead, they are transmitted one-by-one. Thus each signal will be transmitted for a very short time. One cycle or frame is said to be complete when all the signals are transmitted once on the transmission channel. The TDM principle is illustrated in Fig. 7.4

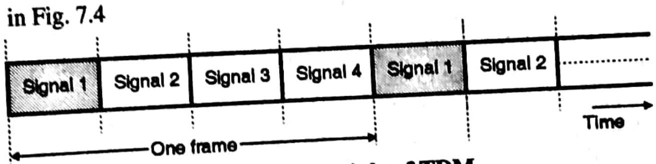


Fig. 7.4 : Principle of TDM

As shown in the Fig. 7.4 one transmission of each channel completes one cycle of operation called as a "Frame". The TDM system can be used to multiplex analog or digital signals, however it is more suitable for the digital signal multiplexing. The data flow of each source (A, B or C) is divided into units (say A_1, A_2 or B_1, C_1 etc.). Then one unit from each source is taken and combined to form one frame. The size of each unit such as A_1, B_1 etc. can be 1 bit or several bits.

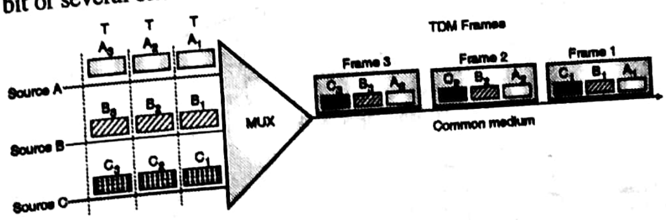


Fig. 7.5 : TDM system

Fig. 7.6 shows the frames of TDM signal. For 3 inputs being multiplexed, a frame of TDM will consist of 3 units i.e. one unit from each source. Similarly for n number of inputs, each TDM frame will consist of n units.

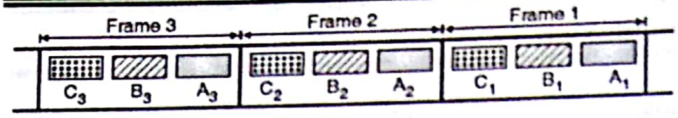


Fig. 7.6 : TDM frames

The TDM signal in the form of frames is transmitted on the common communication medium.

Data rate : For a TDM, the data rate of the multiplexed signal is always n times the data rate of individual sources, where n is the number of sources. So if three sources are being multiplexed, then the data rate of the TDM signal is three times higher than the individual data rate. Naturally the duration of every unit (A_1 or B_1 etc.) in TDM signal is n times shorter than the unit duration before multiplexing.

Q. 6 Write short note on : PAM/TDM systems. **Dec. 04**

Ans. : The TDM system combines the concepts of PAM and TDM both. The TDM system is as shown in Fig. 7.7

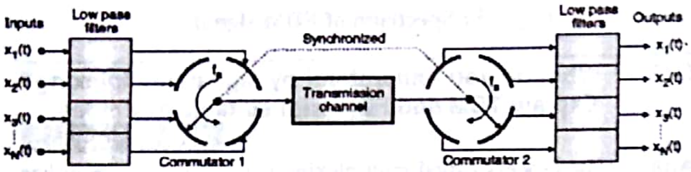
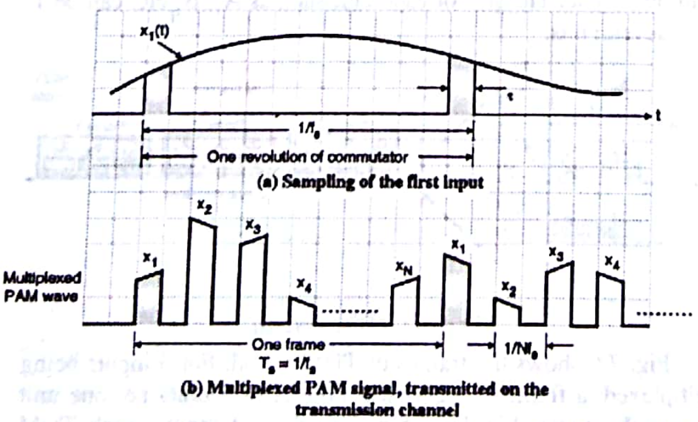


Fig. 7.7 : PAM/TDM system

The operation of the system is as follows :

The multiplexer here is a single pole rotating switch or commutator. It can be a mechanical switch or an electronic switch. It rotates at f_s rotations per second. As the switch arm rotates, it is going to make contact with the position 1, 2, 3 or N for a short time. To these contacts are connected the N analog signals which are to be multiplexed. Thus the switch arm will connect these N input signals one by one to the communication channel.

The waveform of a TDM signal which is being transmitted is as shown in Fig. 7.8 It shows that the rotary switch samples each channel during each of its rotations. Each rotation corresponds to one frame. Hence 1 frame is completed in T_s seconds where $T_s = 1/f_s$. At the receiver, there is one more rotating switch or commutator used for demultiplexing. This switch must rotate at the same speed as that of the commutator 1 at the transmitter and its position must be synchronized with commutator 1 in order to ensure proper demultiplexing. The same principle of multiplexing can be used for multiplexing more number of signals.



(a) Sampling of the first input

(b) Multiplexed PAM signal, transmitted on the transmission channel

Fig. 7.8

Q. 7 Compare TDM and FDM. **May 07, May 11, Dec. 11**

Ans. :

Sr. No.	FDM	TDM
1.	The signals which are to be multiplexed are added in the time domain. But they occupy different slots in the frequency domain.	The signals which are to be multiplexed can occupy the entire bandwidth but they are isolated in the time domain.
2.	FDM is usually preferred for the analog signals.	TDM is preferred for the digital signals.
3.	Synchronization is not required.	Synchronization is required.
4.	The FDM requires a complex circuitry at the transmitter and receiver.	TDM circuitry is not very complex.
5.	FDM suffers from the problem of crosstalk due to imperfect band pass filters.	In TDM the problem of crosstalk is not severe.
6.	Due to wideband fading in the transmission medium, all the FDM channels are affected.	Due to fading only a few TDM channels will be affected.
7.	Due to slow narrowband fading taking place in the transmission channel only a single channel may be affected in FDM.	Due to slow narrowband fading all the TDM channels may get wiped out.

Q. 8 With block diagram explain TDM-PCM system. **Dec. 15**

Ans. :

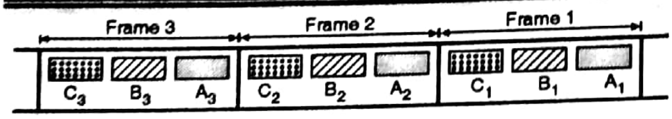
When a large number of PCM signals are to be transmitted over a common channel, multiplexing of these PCM signals is required. Fig. 7.9 shows the basic time division multiplexing scheme for PCM voice channels called as the **T_1 digital system.** This system is used to convey a number of voice signals over telephone lines using wideband coaxial cable. Thus the communication medium used is a coaxial cable.

Operation of the T_1 system :

The operation of the PCM-TDM system shown in Fig. 7.9 is as follows :

This system has been designed to multiplex 24 voice channels marked as S_1 to S_{24}. Each signal is bandlimited to 3.3 kHz, and the sampling is done at a standard rate of 8 kHz. This sampling rate is higher than the Nyquist rate. The sampling is done by the commutator switch SW_1. These voice signals are selected one by one and connected to a PCM transmitter by the commutator switch SW_1, as it completes its rotation. The commutator switch remains in contact with each voice channel for a short time. Thus it samples each of the 24 channels. Each sampled signal is then applied to the PCM transmitter which converts it into a digital signal by the process of A to D conversion and companding. Each sampled voice signal is converted into an 8-bit PCM word.

The resulting digital waveform is transmitted over a co-axial cable. This waveform is called as the PCM-TDM signal. Periodically, after every 6000 ft., the PCM-TDM signal is regenerated by amplifiers called "Repeaters". They eliminate the



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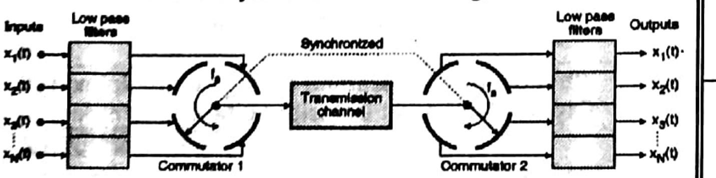

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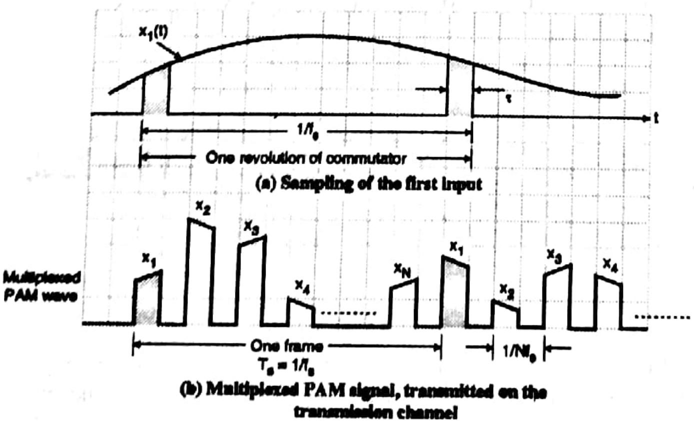

Fig. 7.8

Q. 7 Compare TDM and FDM. May 07, May 11, Dec. 11

Ans. :

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Ans. :

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The resulting digital waveform is transmitted over a co-axial cable. This waveform is called as the PCM-TDM signal. Periodically, after every 6000 ft., the PCM-TDM signal is regenerated by amplifiers called "Repeaters". They eliminate the distortion introduced by the channel and remove the superimposed noise and regenerate a clean noise free PCM-TDM signal at their output. This ensures that the received signal is free from the distortions and noise.

At the destination the signal is companded, decoded and demultiplexed, using a PCM receiver. The PCM receiver output is connected to different low pass filters via the commutator switch SW_2. The LPF outputs are applied to the destination receivers (subscribers). Synchronization between the transmitter and receiver commutators SW_1 and SW_2 is essential in order to ensure proper communication.

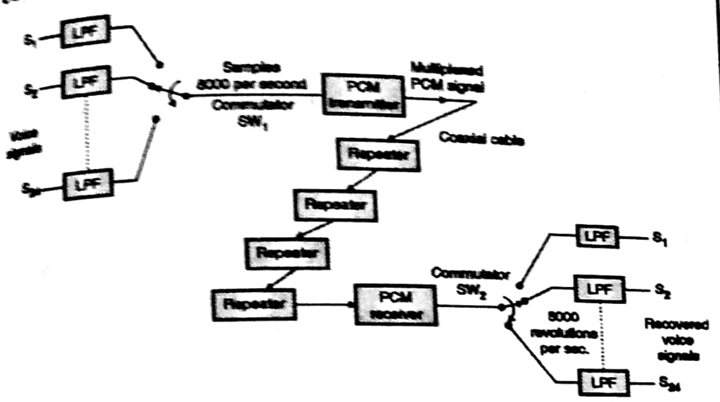

Fig. 7.9 : Block diagram of a basic PCM-TDM system

Bits/Frame :

The commutators sweep continuously from S_1 to S_{24} and back to S_1 at the rate of 8000 revolutions per second (Sampling rate = 8000 samples/sec.). This will generate 8000 samples per second of each signal (S_1 to S_{24}). Each sample is then encoded (converted) into an eight bit digital word. One complete revolution of commutator switches corresponds to generation of one frame which consists of all 24 voice channels.

Thus the digital signal generated during one complete sweep (revolution) of the commutator is given by :

$$1 \text{ Frame} \equiv 1 \text{ revolution} = 24 \text{ channels}$$

$$= 24 \times 8 \text{ bits} = 192 \text{ bits}$$

One frame of PAM-TDM is shown in Fig. 7.10. Each voice signal from S_1 to S_{24} is encoded into eight bits. One frame corresponds to one revolution which is the time taken to transmit each signal once. Hence 1-frame corresponds to one-revolution of the commutator.

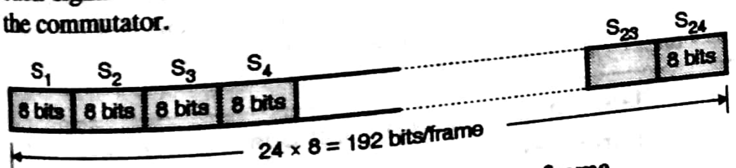

Fig. 7.10 : One frame and bits per frame

Q. 9 State advantages and disadvantages of TDM system.

Ans. :

Advantages of TDM :

1. Full available channel bandwidth can be utilized for each channel.
2. Intermodulation distortion is absent.
3. TDM circuitry is not very complex.
4. The problem of crosstalk is not severe.

Disadvantages of TDM :

1. Synchronization is essential for proper operation.
2. Due to slow narrowband fading, all the TDM channels may get wiped out.

Q. 10 State Advantages, Disadvantages and Applications of FDM.

Ans. : Advantages of FDM :

1. A large number of signals (channels) can be transmitted simultaneously.
2. FDM does not need synchronization between its transmitter and receiver for proper operation.
3. Demodulation of FDM is easy.
4. Due to slow narrow band fading only a single channel gets affected.

Disadvantages of FDM :

1. The communication channel must have a very large bandwidth.
2. Intermodulation distortion takes place.
3. Large number of modulators and filters are required.
4. FDM suffers from the problem of crosstalk.
5. All the FDM channels get affected due to wideband fading.

Applications of FDM :

Some of the important applications of FDM are :

1. Telephone systems.
2. AM (amplitude modulation) and FM (frequency modulation) radio broadcasting.
3. TV broadcasting
4. First generation of cellular phones used FDM.

Q. 11 What do you understand by signal multiplexing? Explain TDM and FDM with suitable examples. Dec.14, May 15

Ans. : Fig. 7.11 shows the block diagram of an FDM transmitter. The signals which are to be multiplexed will each modulate a separate carrier. The type of modulation can be AM, SSB, FM or PM. The modulated signals are then added together to form a complex signal which is transmitted over a single channel.

Operation :

Each signal modulates a separate carrier. The modulator outputs will contain the sidebands of the corresponding signals. The modulator outputs are added together in a linear mixer or adder. The linear mixer is different from the normal mixers. Here the sum and difference frequency components are not produced. But only the algebraic addition of the modulated outputs will take place.

Different signals are thus added together in the time domain but they have their own separate identity in the frequency domain. The composite signal at the output of mixer is transmitted over the single communication channel as shown in Fig. 7.11. This signal can be used to modulate a radio transmitter if the FDM signal is to be transmitted through air.

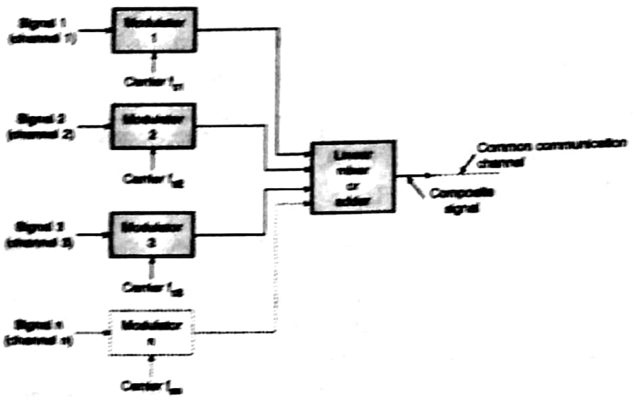

Fig. 7.11 : The FDM transmitter

The block diagram of an FDM receiver is as shown in Fig. 7.12 The composite signal is applied to a group of band pass filters (BPF).

Each BPF has a center frequency corresponding to one of the carriers used in the transmitter i.e. f_{c1}, f_{c2}, ...f_{cn} etc. The BPFs have an adequate bandwidth to pass all the channel information without any distortion. Each filter will pass through only its channel and reject all the other channels. Thus all the multiplexed channels are separated out. The channel demodulator then removes the carrier and recovers the original signal back.

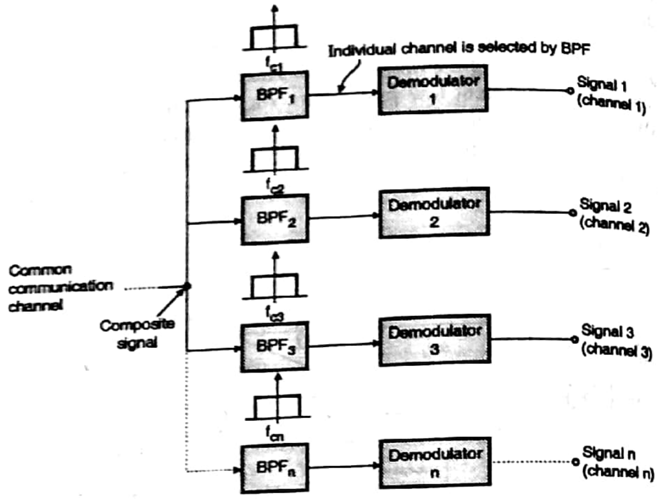

Fig. 7.12 : FDM receiver

Chapter 8 : Information Theory

Q. 1 What is uncertainity ?

Ans. : Uncertainity :

The words uncertainity, surprise and information are all related to each other. Before an event occurs, there is an uncertainty, when the event actually takes place there is an amount of surprise and after the event has taken place there is a gain of information.

The concept of information is related to the "uncertainity". This can be explained using the following sentences :

1. Earth revolves around sun.
2. The rainfall is likely in the evening today.
3. India may win the world cup.

The first sentence does not have any uncertainity or surprise element. Hence the information content in it is minimum. But look at the other two sentences. They are full of uncertainity or element of doubt. Therefore they carry more information.

Extending this concept it say that if a source transmits a message of probability p_k then the information carried by the message will increase as its probability p_k decreases, i.e. as the message becomes less likely. So higher the uncertainity higher the information.

Q. 2 Define information.

Ans. : Information :

The quantitative measure of the information is dependent on our notion of the word "information". To explain what is meant by information assume an example.

Consider a communication system transmitting messages m_1, m_2, m_3..... Let the probabilities of occurrence of these messages be p_1, p_2...etc. respectively. Then the amount of "information" transmitted through the signal m_k is given as,

$$I_k = \log_2\left(\frac{1}{p_k}\right)$$

where, p_k = Probability of occurrence of m_k.

Q. 3 State the units of information.

Ans. : Unit of information :

The information I_k is actually a dimensionless quantity. But by convention the unit attached to it is "bits". This is, when the base of the logarithm is 2.

If there is change in the base of the logarithm, then the unit of information will also change. For the natural logarithm (base "e") the unit of information is "nat", and for base 10 the unit is "Harley" or "decit". The term "bit" was used to represent a binary digit. So hence use the term "bit" as unit of information and represent the binary digit by the term "binit".

Q. 4 State and explain various properties of information.

Ans. :

Properties of Information :

The information contained in an absolutely certain event is zero. That means,

$$I_k = 0 \quad \text{... for } p_k = 1$$

2. The information contents of a message increases with decrease in the value of its probability of occurrence (p_k).

   That means the most unexpected event (which has the least probability of occurrence) will contain maximum information.

3. The occurance of an event, either provides some information or no information, but it is never associated with a loss of information. That means

$$I_k \geq 0 \quad \ldots \text{for } 0 \leq p_k \leq 1$$

4. I_k is a continuous function of p_k.

5. The total information of two or more mutually independent message signals or events is equal to the sum of the information contents of the individual messages. i.e.

$$I_T = I_1 + I_2 + I_3 + \ldots$$

where, I_T = Total information

$I_1, I_2,$ .....are the information in individual messages.

Q. 5 Define and explain the term Information rate.

Ans. : Information Rate (R) :

If the source of the messages generates "r" number of messages per second then the information rate is given as,

$$R = r \times H$$

where, r = Number of messages/sec, and H = Average information/message. (Entropy)

Units of Information rate :

$$R = \left[r \frac{\text{messages}}{\text{sec}} \right] \times \left[H \frac{\text{information}}{\text{message}} \right]$$

∴ R = Average information per second expressed in bits/sec.

Q. 6 What is Shannon's limit ?

Ans. : A noiseless channel is not possible in the real world, so Shannon introduced a theorem called Shannon's capacity theorem to determine the highest possible data rate on the noisy channel. Shannon extended Nyquist's work. He included the effect of noise present on the transmission channel. According to Shannon's theorem, if (S/N) is the signal to noise ratio then the maximum data rate is given by

$$C = R_{max} = B \log_2 \left[1 + \frac{S}{N} \right] \text{ bits/sec}$$

Shannon's theorem puts a limit on the maximum number of levels for a given (S/N) ratio and bandwidth. This expression shows that the maximum data rate for a communication channel is dependent on the channel bandwidth B and signal to noise ratio (S/N). It is important to note that the Shannon's formula does not indicate the signal level. It says no matter how much is the signal level, it is not possible to achieve a data rate (R) which is greater than the capacity of the channel (C).

Q. 7 Define entropy.

Ans. : The "Entropy" is defined as the average information per message. It is denoted by H and its units are bits/message. The entropy must be as high as possible in order to ensure maximum transfer of information. Its prove that the entropy depends only on the probabilities of the symbols that are being produced by the source.

Q. 8 Define channel capacity.

Ans. :

The channel capacity is defined as the maximum data rate at which the digital data can be transmitted over the channel reliably.

The various other concepts related to channel capacity are as follows :

1. Data rate 2. Bandwidth3. Noise 4. Error rate

Q. 9 Explain Shannon's theorem and state its importance.

Ans. : A noiseless channel is not possible in the real world, so Shannon introduced a theorem called Shannon's capacity theorem to determine the highest possible data rate on the noisy channel. Shannon extended Nyquist's work. He included the effect of noise present on the transmission channel.

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Importance of channel bandwidth :

Bandwidth of the communication channel should be higher than the bandwidth of the signal that is to be transmitted over it. This is essential in order to preserve the shape of the signal being transmitted. If the channel bandwidth is less than the signal bandwidth then the signal shape will be distorted when it travels over this channel.

Q. 10 Define Information Theory.

Ans. :

The information theory deals with three basic concepts namely :

1. Amount of source.
2. The capacity of a channel.
3. Use of coding for utilizing channel capacity for information transfer.

Information theory is a broadly based mathematical discipline and it is applicable to various fields such as communications, computer science, statistics and probability. In the field of communications the information theory is used for mathematical modelling and analysis of a communication system. The information theory is used to find answers to many questions related to a communication system.

Q. 11 State Expression for Entropy .

Ans. :

Assume a transmitter is transmitting M different and independent messages m_1, m_2, m_3 .... Let their probabilities of occurrence be p_1, p_2, p_3 ... respectively.

Assume during a long period of transmission a sequence of L messages is generated.

1. if L is very verylarge,expect that in the L message sequence,

p_1 L messages of m_1 are transmitted

p_2 L messages of m_2 are transmitted

p_3 L messages of m_3 are transmitted

.
.
.

p_M L messages of m_N are transmitted.

2. The information conveyed by the message m_1,

$$I_1 = \log_2 [1/p_1]$$

However there are p_1 L number of messages of m_1. Therefore the information conveyed by p_1 L number of messages

$$I_{1(\text{Total})} = p_1 L \log_2 [1/p_1]$$

Similarly the total information conveyed by p_2 L number of m_2 messages is :

$$I_{2(\text{Total})} = p_2 L \log_2 [1/p_2]$$

Similar expression can be written for the remaining messages.

3. The total information of more than one mutually independent message signals is equal to the sum of the information content of individual messages. i.e.

$$I_{(\text{Total})} = I_{1(\text{Total})} + I_{2(\text{Total})} + I_{3(\text{Total})} + \ldots$$

Substitute the values of $I_{1(\text{Total})}$, $I_{2(\text{Total})}$ ...etc.

$$I_{(\text{Total})} = p_1 L \log_2 [1/p_1] + p_2 L \log_2 [1/p_2] + p_3 L \log_2 [1/p_3] + \ldots$$

$$\therefore \quad I_{(\text{Total})} = L\,[p_1 \log_2 (1/p_1) + p_2 \log_2 (1/p_2) + p_3 \log_2 (1/p_3) \quad + \ldots\ldots]$$

4. The "Entropy" is defined as the average information per message interval. It is represented by the symbol "H".

$$H = \frac{I_{(\text{Total})}}{L} = p_1 \log_2 (1/p_1) + p_2 \log_2 (1/p_2) + \ldots\ldots$$

$$\therefore \text{ Entropy : } \quad H = \sum_{k=1}^{M} p_k \log_2 (1/p_k)$$

This is the expression for Entropy.

Q. 12 Prove that the entropy of extremely likely and extremely unlikely message is zero.

Ans. :

1. In case of the "extremely likely" message, there is only one single possible message m_k to be transmitted. Therefore its probability $p_k = 1$. The entropy of a most likely message m_k is given as,

$$H = p_k \log_2(1/p_k) = 1 \log_2(1)$$

$$= \frac{\log_{10} 1}{\log_{10} 2} = 0$$

2. For an extremely unlikely message m_k, its probability $p_k \to 0$

$$\therefore \quad H = p_k \log_2(1/p_k) = 0$$

Q. 13 Define Data rate.

Ans. :

It is defined as the number of bits transmitted by the transmitter per second. It indicates how fast a signal can be transmitted reliably over the given medium.

This capability depends on the following factors :

1. The amount of energy put into transmitting each signal.
2. Distance to be travelled.
3. Noise.
4. Channel bandwidth

Q. 14 Define Channel bandwidth.

Ans. :

The bandwidth of the communication medium should be large enough to transmit the digital signal reliably. An inadequate bandwidth will distort the signal and introduce errors into the received signal.

Q. 15 Define Data Rate Limits

Ans. :

In data communication a large data is required to be transferred from one place to the other. It is necessary to transfer it as quickly as possible. In other words the data rate in bits per second over a channel should be as high as possible. The data rate is decided by the following factors :

1. The maximum bandwidth.
2. The signal level.
3. The noise presented by the channel.

Two theorems were developed to calculate the data rate and we can use them on the basis of the type of channel as follows :

1. A noiseless channel : Nyquist theorem
2. A noisy channel : Shannon's theorem

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p_3 L messages of m_3 are transmitted

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.

p_M L messages of m_N are transmitted.

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Substitute the values of $I_{1(Total)}$, $I_{2(Total)}$ ...etc.

$$I_{(Total)} = p_1 L \log_2 [1/p_1] + p_2 L \log_2 [1/p_2] + p_3 L \log_2 [1/p_3] + \ldots$$

$$\therefore \quad I_{(Total)} = L [p_1 \log_2 (1/p_1) + p_2 \log_2 (1/p_2) + p_3 \log_2 (1/p_3) + \ldots\ldots]$$

4. The "Entropy" is defined as the average information per message interval. It is represented by the symbol "H".

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