Electronic circuits and communication fundamentals (ECCF)

Dec-2017(Choice based)

Q1 A)What is the source of the leakage current in a transistor?

If the emitter current of a transistor is 8 mA and IB is 1/100 of IC, determine the levels of IC and IB. (5)

Solution:

- The main source of the leakage current in a transistor are thermally generated minority carrier.

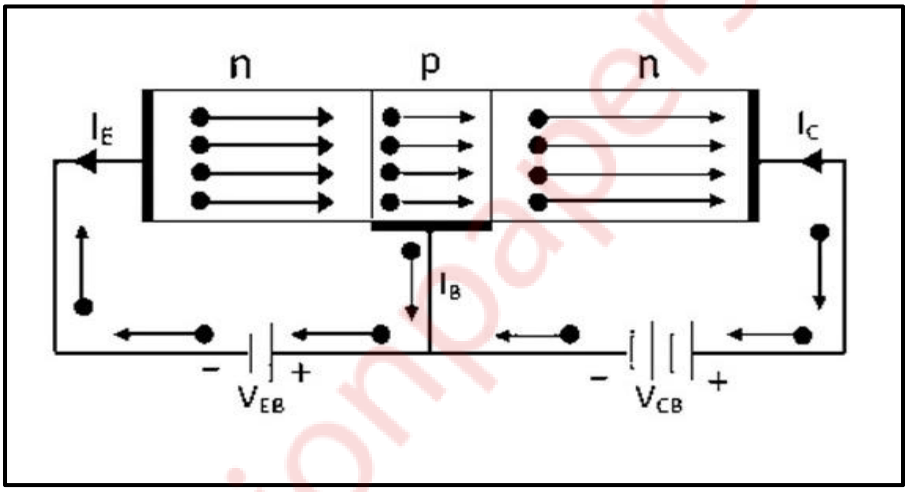


Fig 1: Working of npn transistor

- V_{EB} is used to make J_{BE} Forward Biased(FB) and VBC is used to make JBC Reversed Biased (RB).
- As J_{BE} is FB electron diffuse from emitter(E) to base(B). This creates deficiency of electron in E which is fulfilled by V_{EB}. This constitutes electronic current I_E.
- Some of the diffused electron get recombined with the holes in B. This generates deficiency of holes in B. To fulfill requirement of these holes, covalent bonds are broken, while remaining electron are attracted by positive terminal of V_{EB}. This constitutes electronic current I_B.
- As B is very thin and lightly dopped only few electrons recombine while most of electron drift under the influence of B across J_{BC}.
- Drifted electron in C becomes access and are collected by positive terminal of V_{BC}. These constitutes electronic current I_C.
- Minority carriers generated in base due to thermal agitation also drifts in C and constitutes leakage current (I_{CBO}).

Given: $I_E = 8$ mA

$I_B = 1/100\ I_C$

Sol: As, $I_E = I_B + I_C$

$8 = 1/100\ I_C + I_C$

$I_C = (800/101)$

$\mathbf{I_C = 7.920\ mA}$

$I_B = 1/100\ I_C$

$= 1/100 \times 7.920$

$\mathbf{I_B = 0.0792\ mA}$

Q1 B) Explain the concept of virtual ground in operational amplifiers. (5)

Solution:

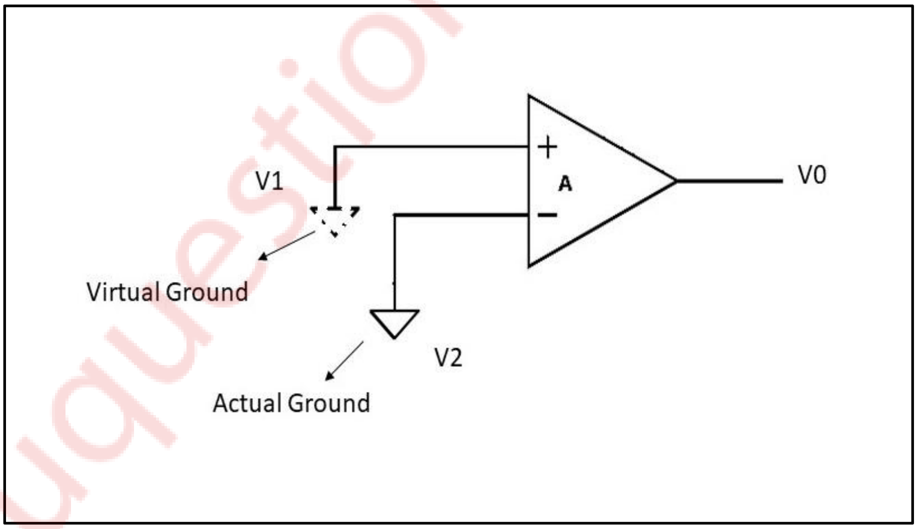


Fig 1: Virtual Ground

In op-amps the term virtual ground means that the voltage at that particular node is almost equal to ground voltage (0V). It is not physically connected to ground.

As Op-amp amplifies differences in input voltages i.e. V1-V2

Which can be written as A= $\frac{v_0}{v_{id}} = \frac{v_0}{v_1 - v_2}$ …………………………..(1)

Where, A= open loop gain

The open loop gain of the op-amp is very high A= ∞, then from eqn.(1)

$\infty = \frac{v_0}{v_1 - v_2}$

For A to be ∞

V1-V2=0

i.e. V1=V2

Now if V2=0 0r negative (-) terminal is grounded then V1 will also be at ground potential.

V1 is not physically grounded but is at ground potential due to very high gain, it is known to be at virtual ground.

Virtual Ground concept is very useful in analysis of an op-amp when negative feedback is employed. It will simplify a lot of calculations and derivations.

Q1 C) Draw the spectrum of amplitude modulated wave and explain its components. (5)

Solution:

Amplitude modulation is a process in which amplitude of high frequency carrier signal is varied according to instantaneous amplitude of low frequency modulating signal.

Let the carrier signal voltage and modulating signal voltage be vc and vm, both represented as,

vc=Vc sinωct

vm=Vm sinωmt

m=Vm/ Vc

Where m= coefficient of modulation.

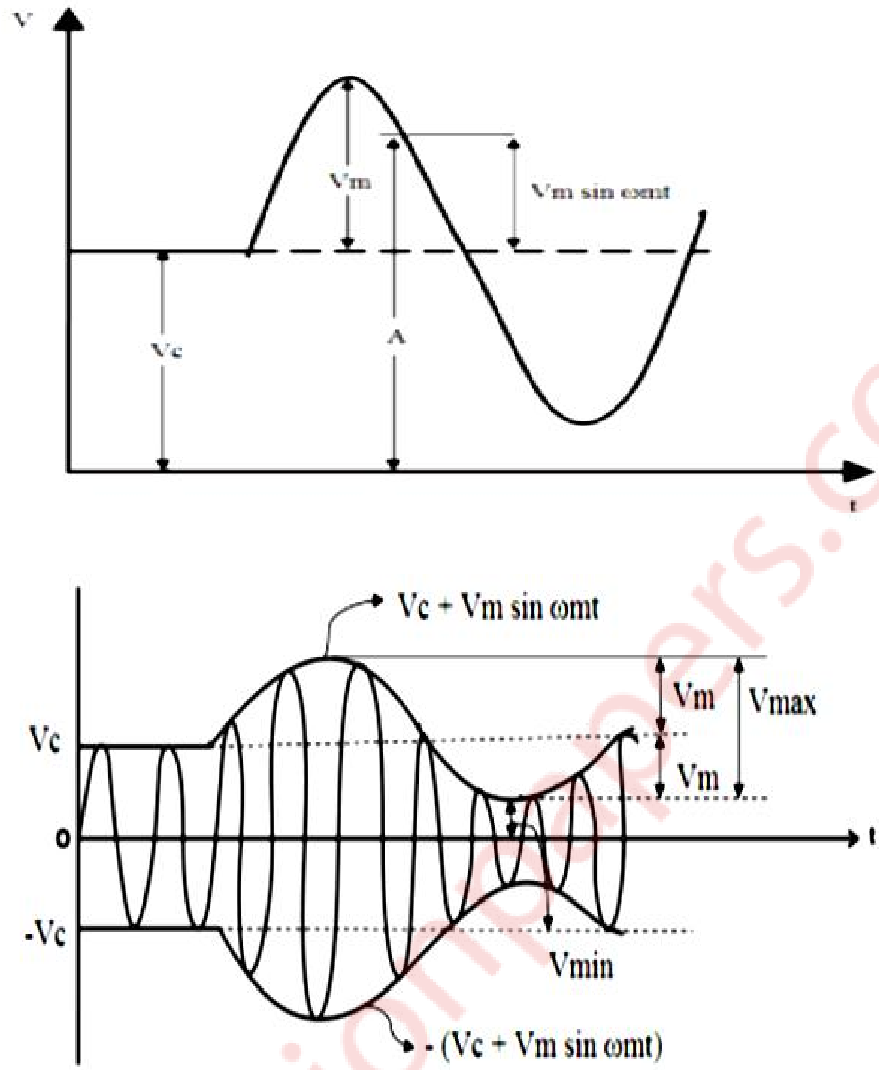


Fig1:Waveform of amplitude Modulation

The modulating signal or message signal contains information, s the high frequency carrier signal which contains no information and the resultant amplitude is modulated signal.

The central frequency has the highest amplitude which is the carrier frequency fc. Adjoining both the sides of the carrier frequencies are the sideband frequencies with lower amplitude. The bandwidth of the amplitude modulated wave is given by, BW=(fc+fm)−(fc−fm)=2fm

Thus we can say that apart from the original carrier signal there are two additional sine waves having frequency above carrier frequency i.e (fc+fm) and the other below the carrier frequency i.e (fc-fm). Therefore, the complete AM signal consists of a three component a carrier wave and two additional frequencies one on each side which are called the sideband frequencies.

The frequency which is above the carrier frequency is called the upper sideband and the frequency below the carrier frequency is called the lower sideband.

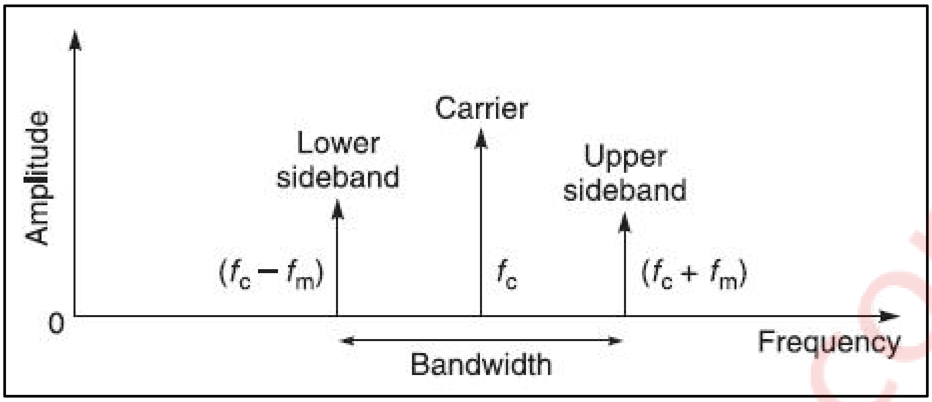


Fig2: Frequency spectrum

Frequency spectrum shows that BW of DSBSC is 2fm

Q1 D) Explain adaptive delta modulation. (5)

Solution:

In digital modulation, we have come across certain problem of determining the step-size, which influences the quality of the output wave.

A larger step-size is needed in the steep slope of modulating signal and a smaller stepsize is needed where the message has a small slope. The minute details get missed in the process. So, it would be better if we can control the adjustment of step-size, according to our requirement in order to obtain the sampling in a desired fashion. This is the concept of **Adaptive Delta Modulation**.

In adaptive delta modulation the step size is not kept constant. Rather ,when slope overload occurs the step size progressively becomes larger, thereby allowing m'(t) to catchup with m(t) more rapidly.

In order to overcome the quantization errors due to slope overload and granular noise, the step size (Δ) is made adaptive to variations in the input signal x(t).

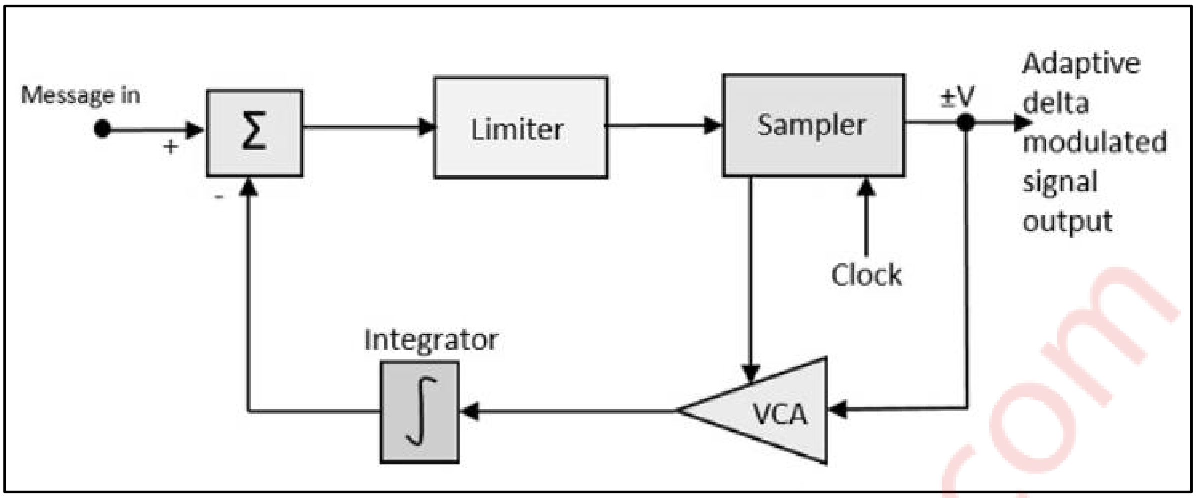


Fig1: Adaptive Delta Modulator

The gain of the voltage controlled amplifier is adjusted by the output signal from the sampler. The amplifier gain determines the step-size and both are proportional.

ADM quantizes the difference between the value of the current sample and the predicted value of the next sample. It uses a variable step height to predict the next values, for the faithful reproduction of the fast varying values.

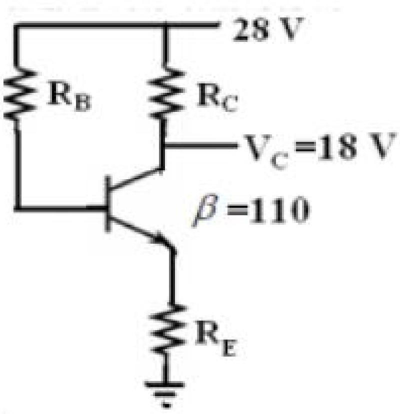
Q2 A) The emmiter bias configuration as shown in following figure has the specifications: (10)

I_{CQ} = ½ $I_{Csat.}$

I_{Csat} = 8 mA

V_C = 18 V and β = 110

Determine R_C, R_E and R_B



Solution:

V_{CC} = 18V and β = 110

$I_{CQ} = ½\ I_{Csat.}$

$= ½ \times 8$

$= 4$ mA

$V_{RC} = (V_{CC} – V_C)$

$= (28-18)$

$= 10V$

$R_C = V_{RC} / I_{CQ}$

$= 10/4mA$

$R_C = 2.5k\Omega$

$I_{Csat} = V_{CC} / (R_C + R_E)$

$8 = 28/(2.5+R_E)$

$R_E = 1\ k\Omega$

$I_B = (Vcc - V_{BE}) / R_B + (\beta + 1) R_E$

$I_B = (28 – 0.7) / R_B + (110+1)\ 1$ ……………………………(1)

$I_B = I_C / \beta$

$= 4 / 110$

$I_B = 36.3636\ \mu A$

Putting I_B value in eqn . (1)

$R_B = 639.8\ k\Omega$

Q2 B) Explain the following parameters and their values for 741 opamp

CMRR, Slew Rate, Gain Bandwidth Product, Input Offset Voltage and Output Resistance. (10)

Solution:

<u>CMRR(Common Mode Rejection Ratio):</u>

It is a figure of merit of Op-amp which decides how far Op-amp is capable of rejecting common mode input signal Vc(noise) and amplifying desired signal Vd i.e. differential input voltage.

It is given as CMRR=$\left|\frac{Ad}{AC}\right|$

Where, Ad -> Differential gain

Ac -> common mode gain

Normally value of CMRR is very high. It is expressed in dB.

CMRR dB = 20 log CMRR

For 741 IC, CMRR is 90dB.

<u>Slew rate:</u>

The maximum rate at which an amplifier respond to an abrupt change of input level.

Slew rate= maximum rate at which amplifier output can change in Volts per microsecond (V/µs)

The slew rate provides a parameter specifying the maximum rate of change of output voltage when $s_R = \frac{\Delta v_0}{\Delta_t}\frac{v}{\mu_s}$

Driven by a large step input signal. If one tried to drive the output at a rate of voltage change greater than slew rate, the output would not be able to change fast enough. In any case, the output would not be amplified duplicate of I/p signal if the Op-amp slew rate to be exceded.

In the case of the 741 IC the slew rate is 0.5V/us, which is very small. This is one reason why the 741 IC is considered not suitable for high frequency applications, such as oscillators, comparators, and filters.

<u>Gain Bandwidth Product</u>

Gain Bandwidth Product describes op-amp's gain in different frequencies. It is defined as the bandwidth of an opamp when voltage gain is one. As shown in the graph, open loop gain falls at the rate of -20dB/decade. This means that if we double the frequency,gain drops by half. And if we half the frequency, gain doubles. This implies that product of gain and bandwidth are almost constant. That is the significance of Gain Bandwidth Product.

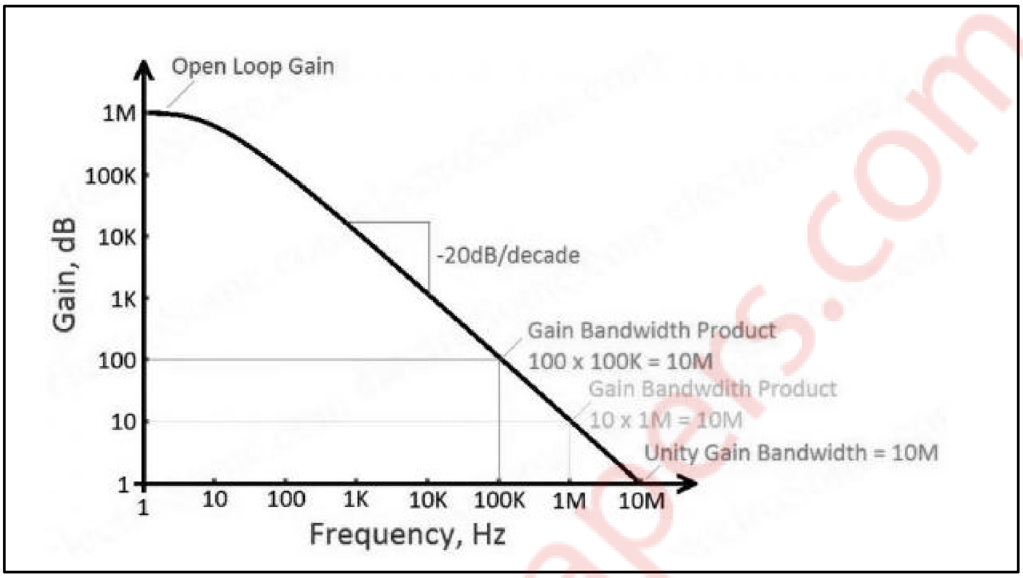


Fig1: Gain Bandwidth Product

For 741 opamp, GB is approximately 1MHz.

<u>Input Offset Voltage</u>

It is the voltage that must be applied between two input terminals of Op-amp to make output offset voltage zero.Since this voltage could be positive or negative its absolute value is listed on the data sheet. For 741C, the maximum value is 6mV.

<u>Output Impedence:</u>

An ideal op amp will have zero output impedance.

When an op amp produces its output signal, we want the op amp to have zero voltage so that the maximum voltage will be transferred to the output load.

Voltage is divided in a circuit according to the amount of impedance present in a circuit. Voltage drops across a component of higher impedance.

In order for the voltage to drop across the output load, that load must be of greater impedance than the output of the op amp. This is why, ideally, we want the output impedance of the op amp to be zero.

Q3 A) Given β = 120 and IE = 3.2 mA for a common-emitter configuration with r0 = ∞Ω, determine. (5)

i) Zi

ii) Av if a load of 2 kΩ is applied.

iii) Ai with the 2 kΩ load.

Solution:

$r_e = 26\text{ m} / I_E$

$= 26\text{ m} / 3.2\text{ m}$

r_e **= 8.125 Ω**

$Z_i = \beta\, r_e$

$= 120 \times (8.125)$

Z_i **= 975 Ω**

$A_v = - R_L / r_e$

$= - 2 / 8.125$

A_v **= - 246.15**

$A_i = I_0 / I_i$

$= \beta$

A_i **= 120**

Q3 B) State and explain Barkhausens criteria for oscillations. (5)

Solutions:

Conditions which are required to be satisfied to operate the circuit as an oscillator are called as "Barkhausen criterion" for sustained oscillations.

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

For an oscillation circuit, there is no input signal "Vs", hence the feedback signal Vf itself should be sufficient to maintain the oscillations.

The Barkhausen criterion states that:

- The loop gain is equal to unity in absolute magnitude, that is, $| \beta A | = 1$ and
- The phase shift around the loop is zero or an integer multiple of 2π: $\angle \beta A = 2 \pi n, n \in 0, 1, 2, \ldots$

The product βA is called as the "loop gain".

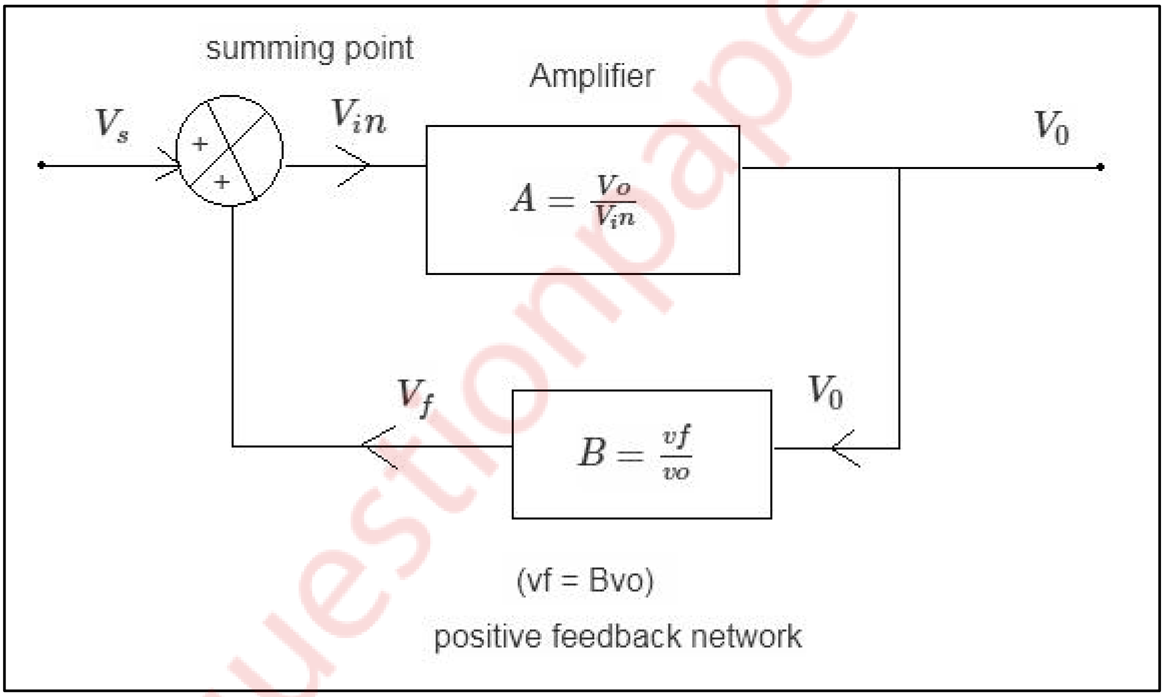


Fig 1: Barkhausens criteria

Q3 C) Explain principle of TDM. (5)

Solution:

Time-division multiplexing (TDM) is a method of putting multiple data streams in a single signal by separating the signal into many segments, each having a very short duration. Each individual data stream is reassembled at the receiving end based on the timing.

According to sampling theorem, a signal is uniquely specified by its value at intervals(1/2 fm) seconds; where fm is frequency of modulating signal. At receiver the complete signal can be reconstructed from the knowledge of the signal at these instant alone.

During this idle period we may transmit the samples of other signals. We can thus interweave the samples of several signals on the channel. At receiving end, the samples can be seperated by a proper synchronous detector. This is known as Time Division Multiplexing.

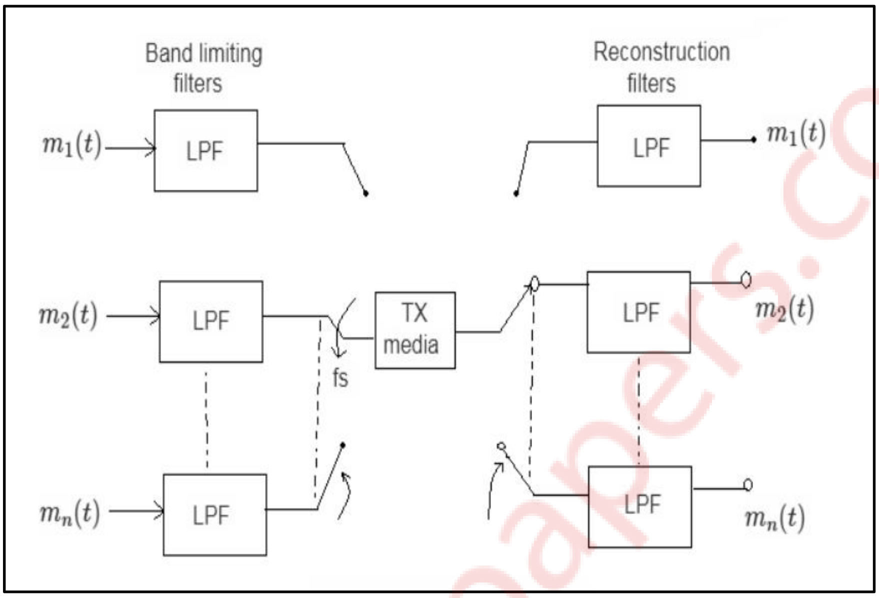


Fig1: Time Division Multiplexing

The switching arrangement at the Tx is provided by the commutator circuit, in each one of its rotation, the commutator extracts or samples, one sample from each message, input m1(t),m2(t)−−−−mn(t)m1(t),m2(t)−−−−mn(t)

Thus, at the output of commutator we get PAM waveform which contain the samples of messages input which are periodically inter placed in time.

These multiplexed message samples are transmitted over the communication channel.

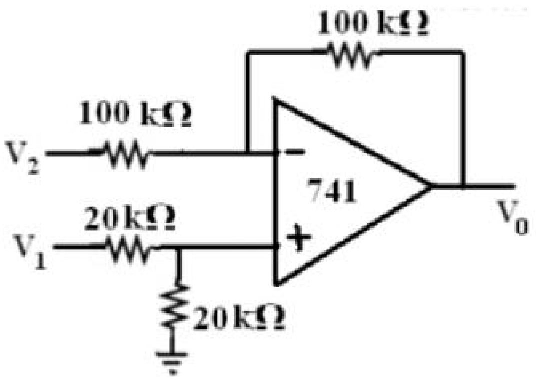
At the recovery end decommutator is used which distributes the pulses to different receiver. the decommutator is again a switching arrangement at the receiving end, similar to that of the transmitting end.

This decommutator is used to separate various received samples and to distribute them to an assembly of LPFs. The LPF then re construct the individual messages, m1(t),m2(t)−−−−mn(t)m1(t),m2(t)−−−−mn(t) at the output.

Here it is necessary that rate of switching of commutator and decommutator must be same and they must be synchronized to each other, this synchronization is achieved by sending a synchronization pulse.

Thus after sending (n-1) pulses (each pulse from different channels) one synchronization pulse is send, thus overall n pulses are sent in time Ts.

Q3 D) Determine the output voltage for the circuit if V1 = 5V, and V2= 3V (5)



Solution:

R1= 100 kΩ

R2 = 20 kΩ

R3 = 100 kΩ

R4= 20 kΩ

The differential amplifier equation when R1= R2 and R3= R4 is

Vout = R3/R1(V2-V1)

= 100/100(3-5)

= -2 V

Q4A) Draw the block diagram of phase cancelleation SSB generation and explain how the carrier and unwanted sidebands are suppressed. (10)

Solutions:

The phasing method of SSB generation uses a phase shift technique that causes one of the side bands to be canceled out. A block diagram of a phasing type SSB generator is shown in fig.

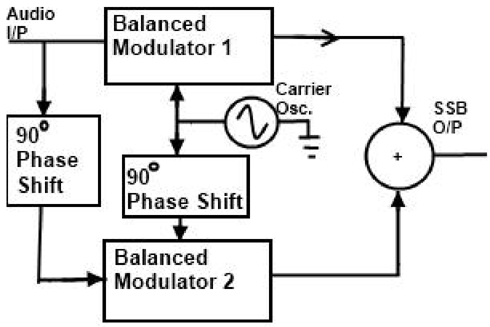


Fig1: SSB Phase shift method

The carrier signal is $V_c \sin 2\pi f_c t$ the modulating signal is $V_m \sin 2\pi f_m t$.

Balanced modulator produces the product of these two signals.

$$(V_m \sin 2\pi f_m t)(V_c \sin 2\pi f_c t)$$

Applying a trigonometric identity.

$$(V_m \sin 2\pi f_m t)(V_c \sin 2\pi f_c t)=1/2[\cos(2\pi f_c-2\pi f_m)t-\cos(2\pi f_c+2\pi f_m)t]$$

these are the sum and different frequencies or the upper and lower side bands.

It is important to remember that a cosine wave is simply a sine wave shifted by 90. A cosine wave has exactly the same shape as a sine wave, but it occurs 90

The 90 phase shifters create cosine waves of the carrier and modulating signal which are multiplied in balanced modulator to produce.

$$(V_m-\cos 2\pi f_m t)(V_c \cos 2\pi f_c t)(V_m-\cos 2\pi f_m t)(V_c \cos 2\pi f_c t)$$

Another common trigonometric identity translates this to

$$(V_m \cos 2\pi f_m t)(V_c \cos 2\pi f_c t)12[\cos(2\pi f_c-2\pi f_m)t+\cos(2\pi f_c+2\pi f_m)t]$$

Now if we add these two expressions together the sum frequencies cancel while the difference frequencies add producing only the lower side band.

$$\cos(2\pi f_c-2\pi f_m)t$$

Carrier and unwanted side band suppression

Carrier and unwanted side band can be suppressed using a spectrum analyzer with a bandwidth narrow enough to see both sidebands and your suppressed carrier.

This task can be done with a separate receiver and a calibrated S meter. Transmit a steady tone on 14.2 MHz USB at low power and then tune the receiver until you can hear your transmission.

If you then switch to LSB on your receiver and tune around you may be able to hear your unwanted sideband. If you measure a signal of S7 then your unwanted sideband is 32dB down.

Q4 B) Draw the PAM, PWM and PPM waveforms in time domain assuming a sinusoidal modulating signal. Explain them in brief. (10)

Solution:

PAM(Pulse Amplitude modulation):

Pulse amplitude modulation is the basic form of pulse modulation. In this type of modulation, the signal is sampled at regular intervals and each sample is made proportional to the amplitude of the modulating signal.

There are two sampling techniques used for sampling the modulating signal in PAM, which are Flat top sampling and Natural sampling.

Fig1 depicts the relationship between the message signal which is a sinusoidal signal, the pulse train or the sampling signal and the resulting PAM signal with the help of the waveform plotted in time domain.

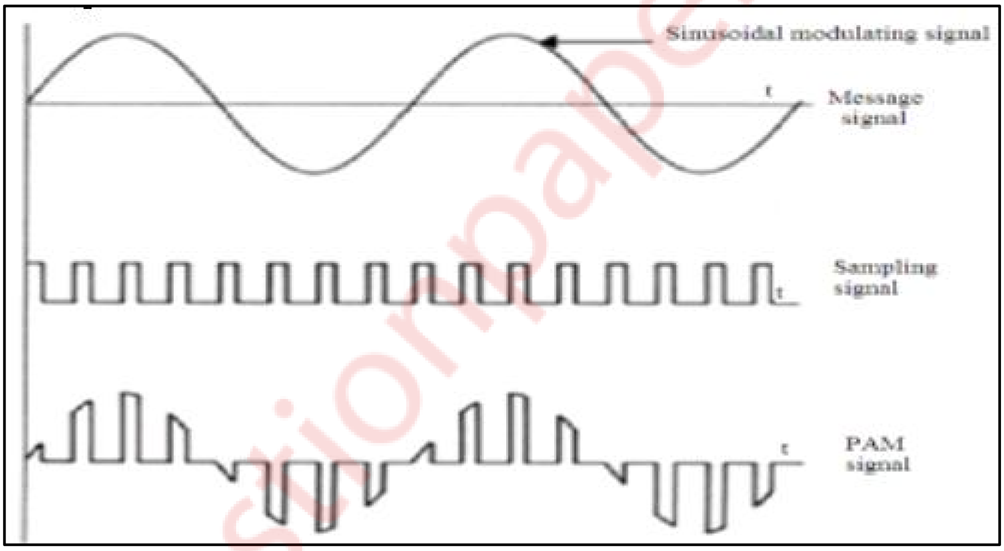


Fig1 : PAM

The pulse amplitude modulated wave in time domain is attained by multiplying the message or modulating wave with the train of pulse or carrier pulse.

Noise affects the amplitude of the waveform and PAM is less immune to noise, since in PAM the information contains amplitude variations.

Demodulation is performed by detecting the level of amplitude of the carrier pulse at each symbol period.

PWM (Pulse Width Modulation):

Pulse width modulation is defined as a process of varying the width of the signal pulse in accordance to the modulating signal variations.

It is also called as Pulse Duration modulation (PDM) or Pulse Length modulation (PLM).

The amplitude and position of the pulse remains constant for PWM signals. Thus PWM is more robust to noise than PAM.

In the absence of the modulating signal the width of the pulse is equal to the original width.

The positive values of the message pulse results in the increase in the width of the PWM signal, and negative values of the message results in the decrease in the width of the PWM signal.

The Fig2 shows the generation of PWM signal from the message and carrier pulse with the help of waveform which is plotted in the time domain.

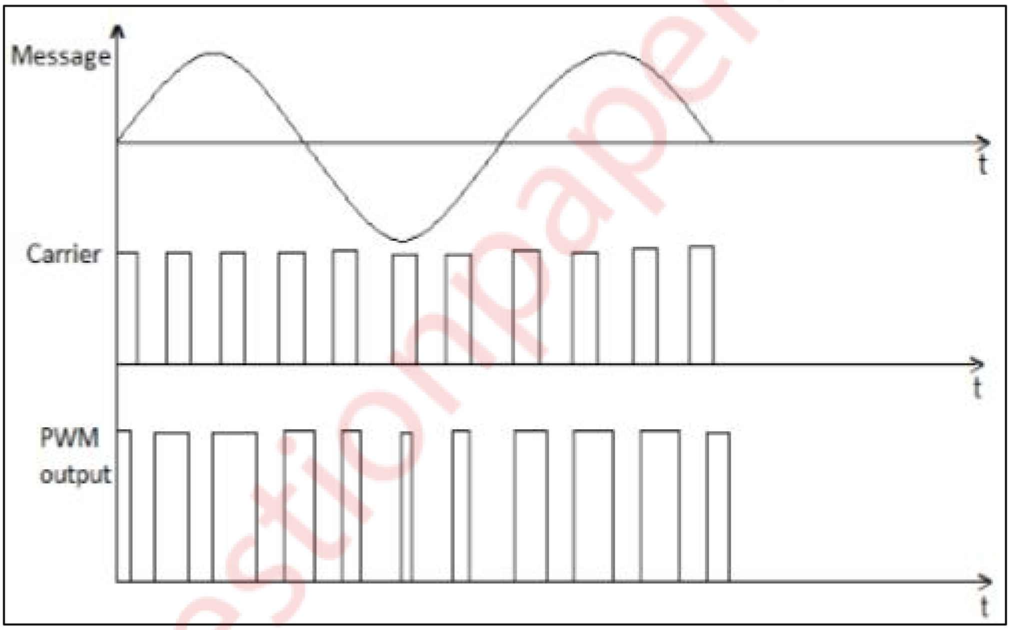


Fig 2: PWM

PPM(Pulse Position Modulation):

Pulse Position modulation is defined as a process of varying the position of the signal pulse, by taking the reference signal, in accordance to the modulating signal variations.

The amplitude and width of the pulse remains constant for PPM signals.

PPM is a kind of modification of the PWM signal.

In the absence of the modulating signal the position of the leading and trailing edge of the pulse is equal to original position.

The positive values of the message pulse results in the proportionate right shift and negative values of the message results in the proportionate left shift.

Fig3 shows the generation of PPM signal from the modulating and sampling pulse. It also shows the generation of PPM signal from PWM signal.

PWM signal is generated using Pulse width generator which helps to trigger the monostable multivibrator, where trial edge of the PWM signal is used for triggering the monostable multivibrator. PWM signal is converted into PPM signal after triggering the monostable multivibrator.

Application of PPM include radio frequency communication, non coherent detection etc.

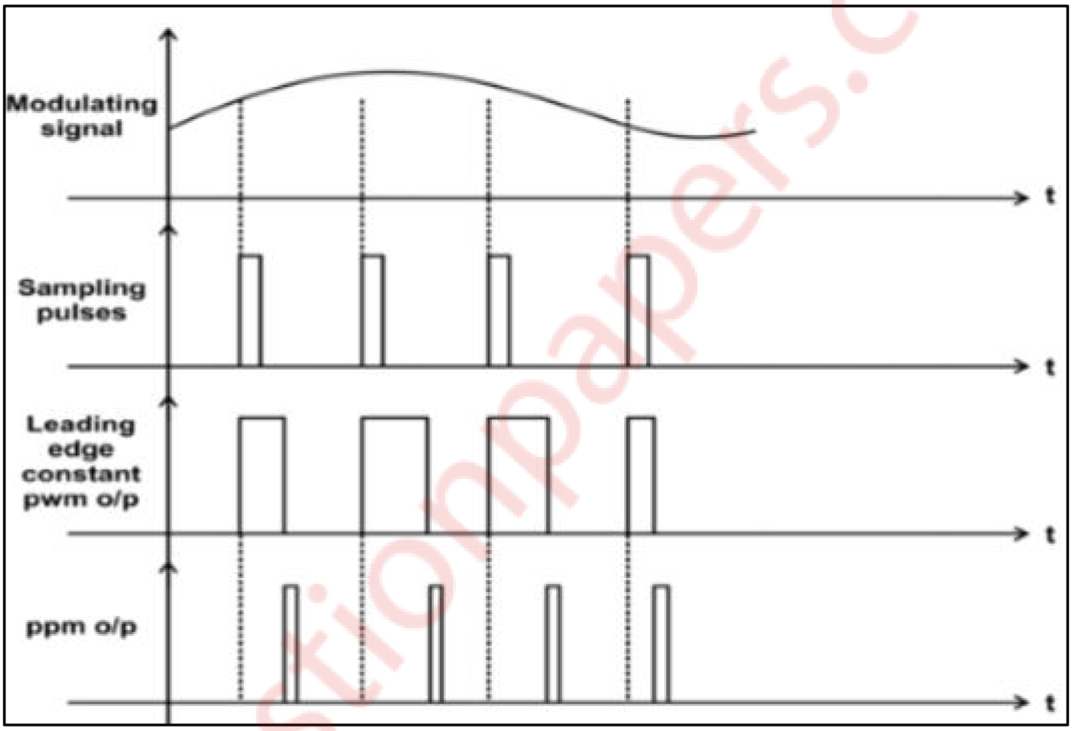


Fig3: PPM

Q5 A) State Shannon's Theorem on channel capacity.

What is the maximum capacity of a perfectly noiseless channel whose bandwidth is 120 Hz, in which the values of the data transmitted may be indicated by any one of the 10 different amplitudes? (10)

<u>Solution:</u>

Shannon's Theorem:

The Shannon-Hartley theorem tells the maximum amount of error-free digital data that can be transmitted over a communications channel (e.g., a copper wire or an optical fiber) with a specified bandwidth in the presence of noise.

Bandwidth is the range of frequencies that a communications channel can carry. The greater the bandwidth of a channel, the larger is its throughput (i.e., data transmission capacity).

The term noise refers to signals in a communication channel that are unrelated to the information that is being transmitted and can reduce the throughput of the channel.

The band width and the noise power place a restriction upon the rate of information that can be transmitted by a channel, it may be shown that in a channel which is disturbed by a white Gaussian noise, one can transmit information at a rate of C bits per second, where C is the channel capacity and is expressed as

$c = B \log_2(1 + S/N)$

Where

B →channel bandwidth in Hz

S → Signal power

N → Noise power.

For noiseless channel:

Capacity =2 bandwidth x $\log_2 M$

= 2 x 120 x $\log_2 10$

= **797.2627 bps.**

Q5 B) With respect to neat diagram explain elements of analog communication system. (10)

Solutions:

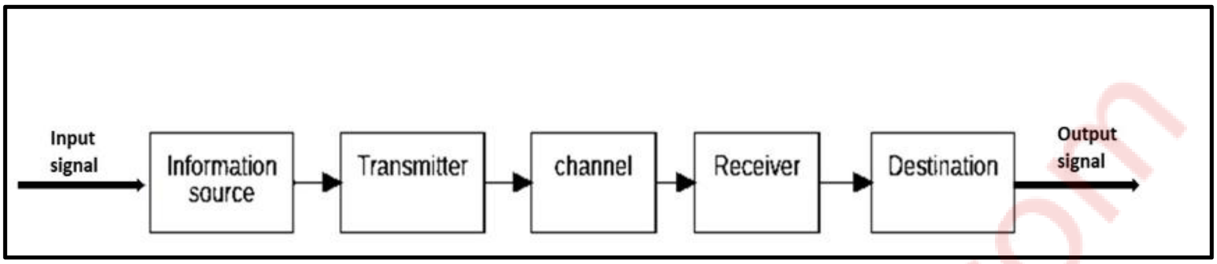


Fig1: Elements of Communication system

The above fig.(a) shows the generic block diagram of a communication system.

The purpose of communication system is to transmit intelligence signal from a source to a destination at some point away from the source.

Any communication system has five basic elements which are information source, transmitter, channel, receiver and destination.

In practical design we are interested in only transmitter, channel and receiver. This is because, we have little control over the other two blocks. Also the communication in electrical form takes place in the three blocks. The function of each blocks are:

A) **<u>Information source:</u>**
The objective of any communication system is to convey information from one point to another.
The information comes from the information source, which originates it.
Information is a very generic word signifying at the abstract level anything which is intend for communication which may include thought, news, feelings, visual scent and so on.

B) **<u>Transmitter</u>**

The objective of the transmitter is to collect the incoming message signal and modify it in a suitable fashion such that it can be transmitted via the chosen channel to receiving point.

The transmitter block involves several operations:

a. Amplification: It involves amplifying the signal amplitude and also adding required power levels.
b. High frequency: Termed as carrier generated by stable oscillator.
c. Modulation: Varying one of the three parameters i.e amplitude, frequency, phase in accordance with variation of message signal.

C) **Channel**

The communication channel is the medium used for transmission of electrical signal from one place to other.

The communication medium can be conducting wires, cables, optical fibres or free space.

Depending on the type of communication medium, two types of communication system exists.

Line communication: The line communication systems use the communication medium like the simple wires or cables or optical fibres. Eg: Telephone, Cable TV.

Radio communication: The radio communication systems use the free space as their communication medium. The transmitted signal is in the form of electromagnetic waves. E.g. Mobile communication, satellite communication.

D) **Receiver**

The receiver block receives the incoming signal from the channel and process it to recreate the original signal. This process is also called as demodulation.

There are variety of receivers in communication system, the type of receiver chosen depends on type of modulation, operating frequency, its range and type of destination required. Most common receiver is superheterodyne receiver.

E) **Destination**

It is the final stage of any communication system which receives the message signal and processes it to comprehend the information present in it.

It would be a loud speaker/ display device/ simply a load etc. depending up on the requirements of the system.

Q6A) What is Nyquist Criteria? What is its significance? (5)

Solution:

Sampling theorem (also known as Nyquist rate) According to this theorem, it is possible to reconstruct a band limited analog signal from periodic samples, as long as the sampling rate is at least twice the frequency of highest frequency component of analog signal. Mathematically it is given as:

$$Fs=2fm$$

In telephony, a sample rate of 8 kHz is use for more AF of 3.4 kHz. This theorem was the key to digitizing the analog signal. Using this, it was possible to turn the human voice into a series of ones and zeroes.

Significance:

Aliasing: To preserve all information in the unsampled signal, we must ensure that the spectrum "islands" do not overlap when replicating the spectrum, if they overlap, we can no longer extract the original signal from the samples this overlapping is known as "Aliasing"

Aliasing allows higher frequencies to disguise themselves as lower frequencies.

To avoid aliasing, you must preserve the following condition.

$1/T \geq 2BW$

This result can be expressed in terms of sampling frequency as

F sampling (Fs) = 2BW

Thus minimum sampling frequency necessary for sampling without aliasing is 2 BW this result is generally known as Nyquist criterion.

Q6 B) Give the proper definition for entropy and information rate. (5)

Solution:

Entropy

The concept of entropy in information theory describes how much information there is in a signal or event.

It is also called as average information.

Shannon, in fact, defined entropy as a measure of the average information content associated with a random outcome.

Entropy can be defined as a measure of the average information content per source symbol. Claude Shannon, the "father of the Information Theory", provided a formula for it as

$$H=-\sum p_i \log_b p$$

Where p_i is the probability of the occurrence of character number i from a given stream of characters and b is the base of the algorithm used. Hence, this is also called as Shannon's Entropy.

Information Rate

The information rate is represented by R and it is given as,

Information Rate : R = rH

where R is the information rate.

H is the Entropy or average information

r is the rate at which messages are generated.

Information rate R is represented in average number of bits of information per second.

It is calculated as follows: R = r(in msg/sec) *H(in bits/msg)= bits / second

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Q6 C) Write short note on op-amp as comparator. **(5)**

Solution:

An comparator is a electronic circuit configuration that compares two voltages and indicates which one is larger. Thus the inputs to a comparator should be different in nature.

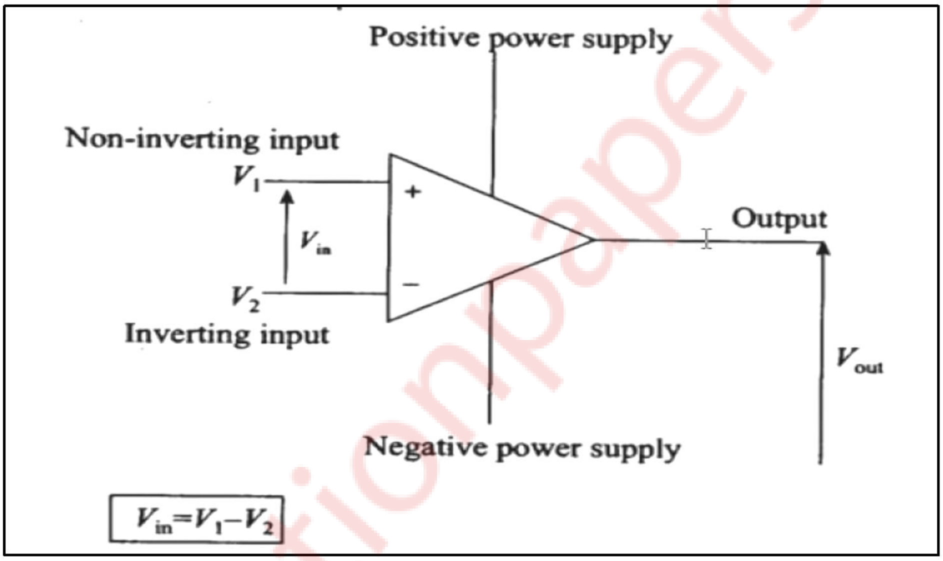

Fig1: Op-amp as comparator

Comparator can be easily configured using op-amp, since the op-amp have high gain and balanced different inputs.

Theoretically, an op-amp is open loop can be used as a comparator. When the input voltage at the non-inverting terminal(+) is greater than voltage at the Inverting (+) is greater than the voltage at the IN terminal, the output at the opamp saturates at the positive extreme.

When the NI input voltage drops below the IN input voltage the output switches to its negative saturation level.

Comparator circuits are most widely used in analog-to-digital converter and in oscillator.

The comparator circuit accepots input of linear voltage and provide a digital output.

Q6 D) Differentiate between Class A and Class C power amplifiers with respect to circuit diagram , operating cycle and power efficiency. (5)

Solution:

Parameters	Class A	Class C
Operating cycle	If the collector current flows all the time during full cycle of input signal, the power amplifier is called as class A amplifier	If the collector current flows for less than half cycle of the input signal, the power amplifier is called as class C amplifier.
Power Efficiency	The efficiency of class A amplifier is less than 50% due to power losses in the output transformer.	The maximum collector efficiency of class C power amplifier is nearly 100%.
Q- point	The operating point Q lies at the centre of the load line	Below X-axis
Distortion	Absent No distortion	Highest
Circuit Diagram