Electronic circuits and communication fundamentals (ECCF)

May-2018(Choice based)

Q1. A) Draw input and output characteristics of BJT. State significance of DC load line.

Solution: **(5)**

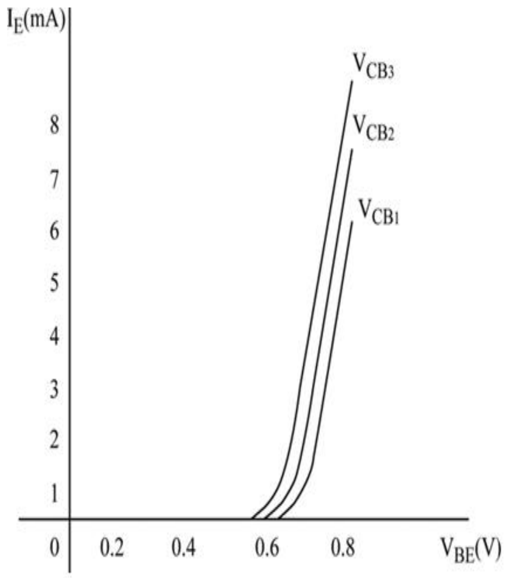


Fig1:Input characteristics of CB configuration

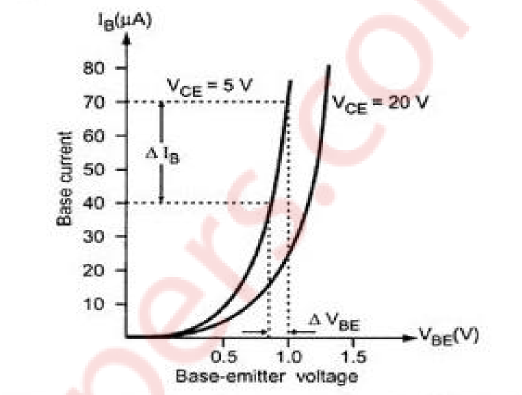


Fig3: Input characteristics of CE configuration

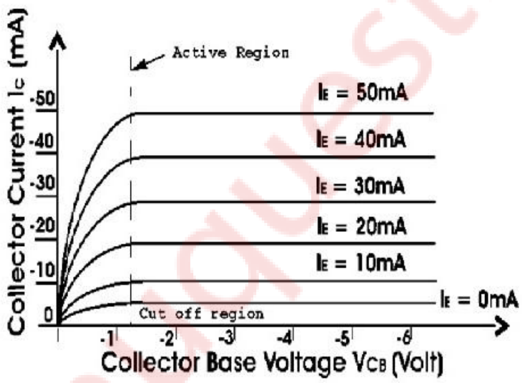


Fig2: Output characteristics of CB configuration

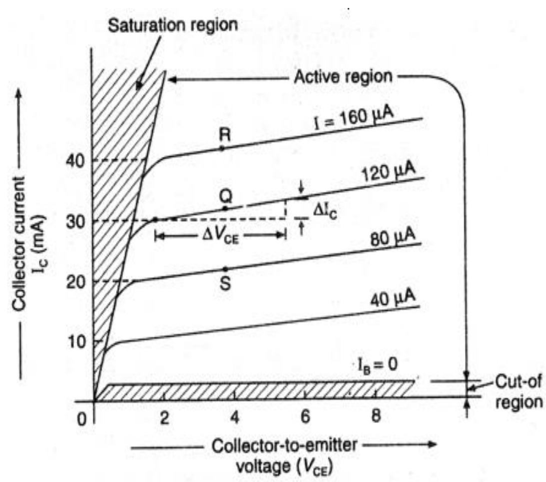


Fig4: Output characteristics of CE configuration

Significance of DC load line:

- When the transistor is given the bias and no signal is applied at its input, the load line drawn under such conditions, can be understood as DC condition.
- Here there will be no amplification as the signal is absent. This condition off amplifier is known as Quiscent condition, where only Dc voltage are applied.

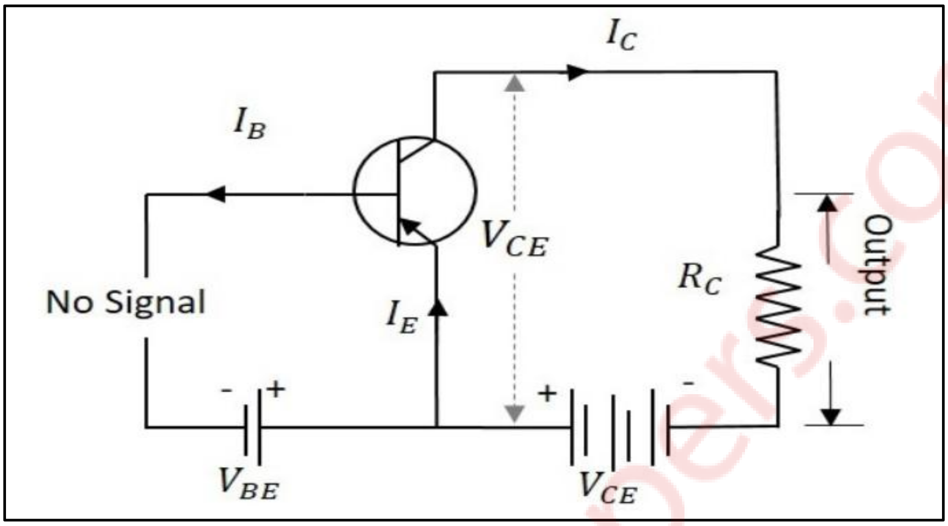


Fig1: Circuit diagram for DL

- The value of collector emitter voltage at any given time will be

  VCE=VCC−IC RC

- As V_{CC} and R_C are fixed values, the above one is a first degree equation and hence will be straight line on the output characteristics. This line is called as D.C Load line. This line can be drawn on o/p characteristics of CE.

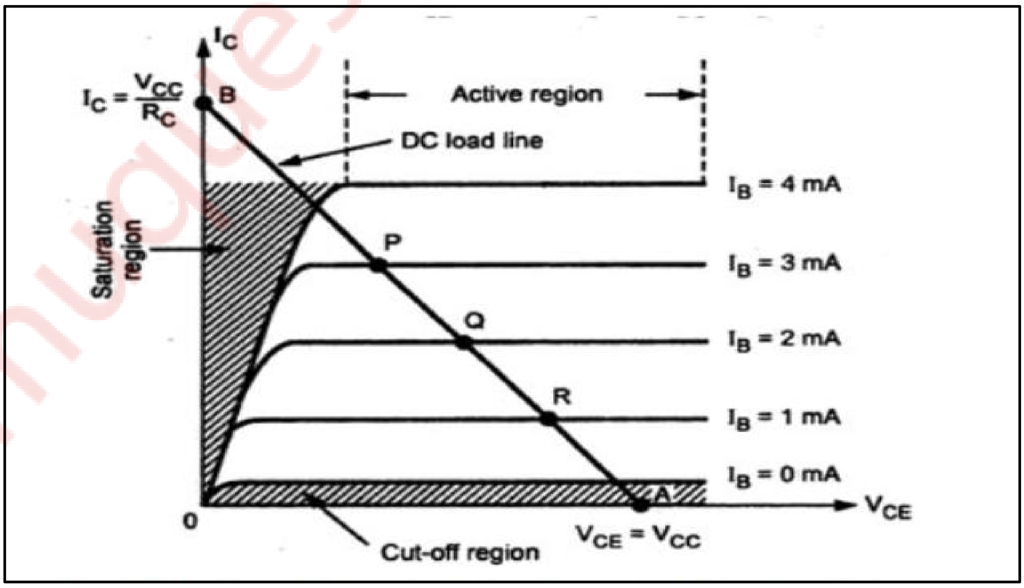


Fig2: Common emitter output characteristics with dc load line

- A point can be obtained on DC load line which is called Q-point.

- Normally whatever signals we want to amplify will be of the order milli volts or less. If we directly input these signals to the amplifier they will not get amplified as transistor needs voltages greater than cut in voltages for it to be in active region. Only in active region of operation transistor acts as amplifier.
- So we can establish appropriate DC voltages and currents through BJT by external sources so that BJT operates in active region and superimpose the AC signals to be amplified.
- The DC voltage and current are so chosen that the transistor remains in active region for entire AC signal excursion. All the input AC signals variations happen around Q-point.

Q1.B) For am AM DSBFC modular with carrier frequency fc= 100kHz and maximum modulating signal frequency fm=5kHz, determine

Frequency limits for the upper and lower side bands **(5)**

Bandwidth

Draw the frequency spectrum

<u>Solution:</u>

Given: fc= 100kHz

fm= 5kHz

Sol:

Upper side bands= fc + fm

=100 + 5

=**<u>105 kHz</u>**

Lower side bands= fc - fm

=100 - 5

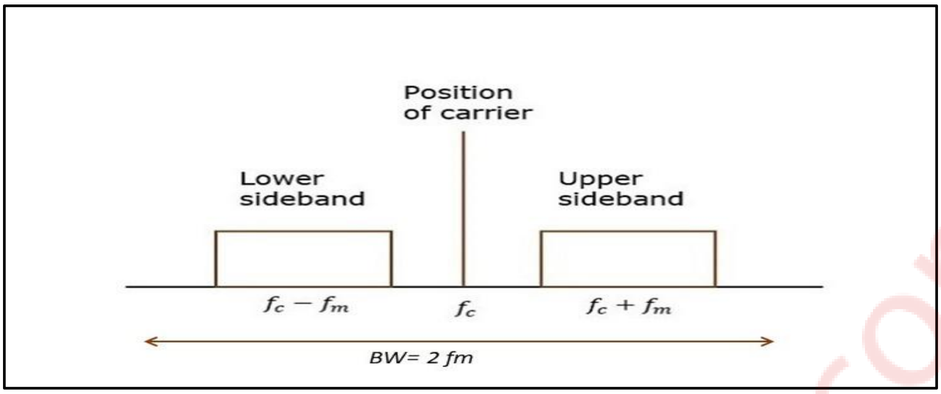
=**<u>95 kHz</u>**

Bandwidth= 2fm

= 2 x 5

= **<u>10 kHz</u>**

Frequency spectrum



Q1.C) Write short note on zero crossing detector using op-amp with waveform.

Solution: **(5)**

- The zero crossing detector(ZCD) circuit is an important application of the op-amp comparator circuit.
- A zero crossing detector is a comparator with the reference level set at zero.
- It is used for detecting the zero crossings of AC signals. It can be made from an operational amplifier with an input voltage at its positive input
- It can also be called as the sine to square wave converter.

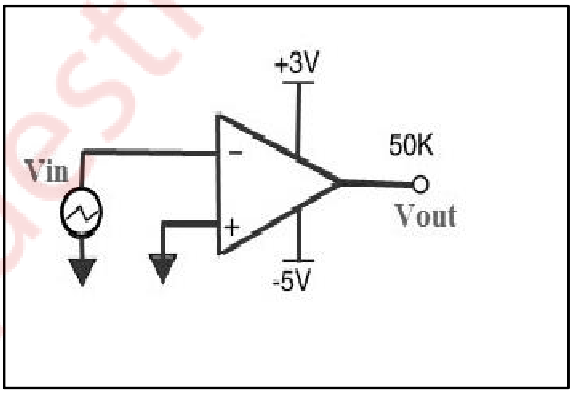


Fig1: Zero crossing detector

- Anyone of the inverting or non-inverting comparators can be used as a zero-crossing detector. The only change to be brought in is the reference voltage with which the input voltage is to be compared, must be made zero (Vref = 0V). An input sine wave is given as Vin.

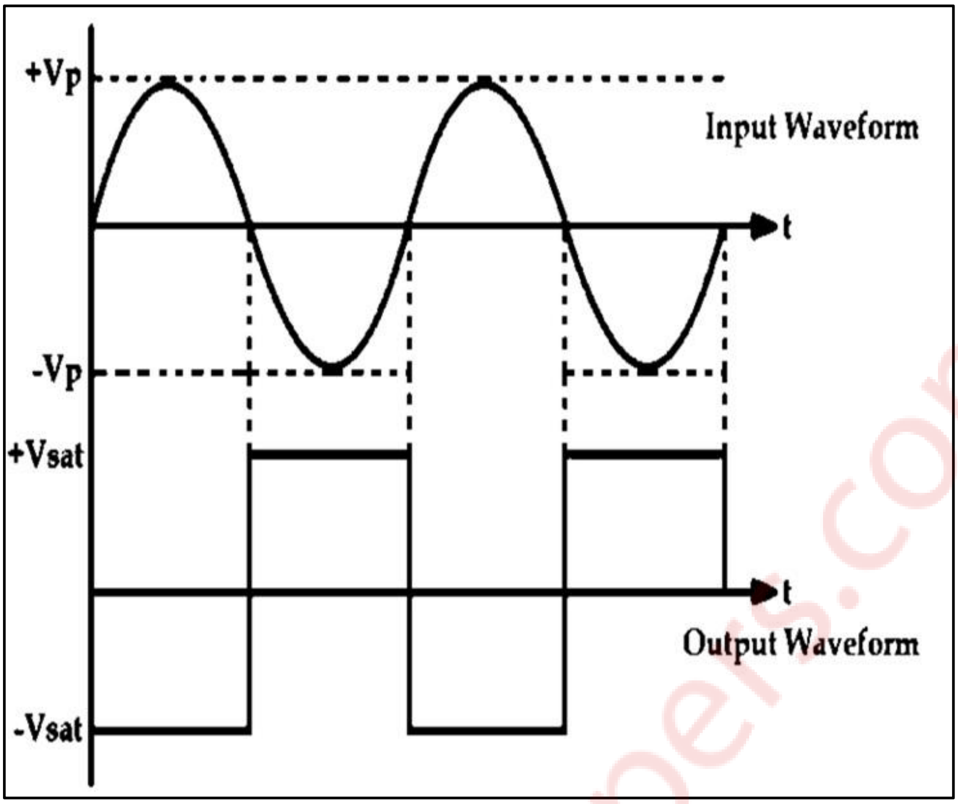


Fig2: Waveform of zero crossing detector

- A zero crossing detector (ZCD) can be built using a 741 operational amplifier IC. One input must be set to zero for the reference voltage, while a sine wave voltage is applied to the other input. As shown in the fig of waveform, When the input sine wave passes through zero in a negative direction, the output voltage is driven into positive saturation. Similarly, as the input passes zero in a positive direction, the output is driven into negative saturation. This arrangement is also known as a sine to square wave converter.
- ZCDs are useful tools for reducing or eliminating electrical noise. Noise produced during switching is proportional to the amplitude of the AC voltage at the switching point; therefore, switching should take place at the voltage's zero crossing to minimise this noise. Effective zero crossing detection can facilitate this function.
- The application of Zero Crossing Detector are:

  ZCD as Phasemeter

  ZCD as Time Marker Generator

Q1.D) Compare class A and class C Amplifier.

Solution: **(5)**

Class A Amplifier:

- If the collector current flows all the time during full cycle of input signal, the power amplifier is called as class A amplifier.
- The operating point Q lies at the centre of the load line
- The class A power amplifier is biased such that no part of the signal is cut off.
- As the output waveform is same as the input waveform there is least distortion.

- The transformer coupled class A power amplifier has maximum collector efficiency 50%. It means that the maximum 50% DC supply power is converted into AC power.
- The efficiency of class A amplifier is less than 50% due to power losses in the output transformer.
- The maximum power dissipation in the transistor occurs under zero signal condition.
- Low power output and low collector efficiency.

Class C Amplifier

- If the collector current flows for less than half cycle of the input signal, the power amplifier is called as class C amplifier.
- The negative bias is given to the base of transistor therefore the collector current does not flow when the positive half cycle of the signal starts.
- The class C amplifiers are never used for power amplification.
- As they amplify narrow band of frequencies near the resonant frequency, they are used as tuned amplifiers.
- The maximum collector efficiency of class C power amplifier is nearly 100%.
- As the power losses are very small in the high Q resonant circuit, narrow pulse will compensate all the losses in the class C power amplifier.

Q2.A) Explain Superheterodyne receiver with suitable diagram.

Solution: **(5)**

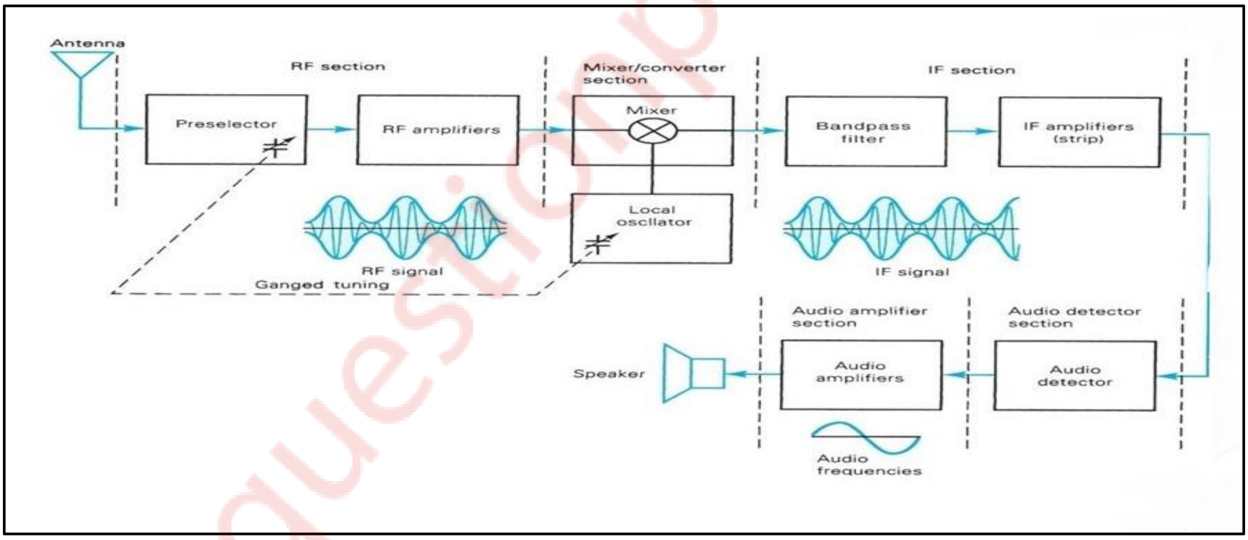


Fig1: Block diagram of Superheterodyne receiver

The Various block of superheterodyne receiver are as follows:

- Preselector (antenna tank circuit) and RF amplifiers:
  The preselector circuit is a tuned circuit with a variable capacitor. By varying capacitor, the tank circuit can be tuned to carrier frequency of desired radio station.
  Once the desired signal is selected it is either given to mixer directly (no BPF amplifier) or it given to Rf amplifier. The use of RF amplifier can provide following advantages:
  Greater gain
  Image frequency rejection is high
  S/N ratio is high

Better selectivity
Better coupling between antenna and transmitter, etc.

- Mixer:
  It is a non-linear active device which mixes or heterodynes received signal frequency (fs) with local oscillator frequency(fo).
  At the output of the mixer, along with all other frequencies we also get fo-fs.
  This frequency is known as intermediate frequency (IF) i.e fi= fo-fs.
  The tank circuit at the o/p of the mixer we get only fi.

- Local oscillator:
  It is normally a LC oscillator (an oscillator frequency of which depends upon the values of L and C in its tank circuit) the frequency deciding capacitance of which is ganged with the tuning capacitor of antenna tuning circuit or preselector.
  Capacitor of preselector (Ca) and local oscillator (Co) are ganged such that whenever the receiver is tuned to a new signal frequency fs, fo is always 455kHz more than fs. It means that IF or fi= fo - fs

  = fs- (455-fs)

  fi = 455kHz.

- Detector:
  At the output of IF amplifier we get amplitude modulated signal with fixed frequency fi.
  Detector takes out or extracts AF information from AM signal at its input. This detector is also known as secondary audio detector .

- AF Amplifier:
  It provides maximum audio frequency power to frequency.

Q2. B) Implement summing operational amplifier using inverting configuration of Op-amp.

<u>Solution:</u> **(5)**

Summing amplifier is basically an op-amp that can combine numbers of input signal to a single output that is weighted um of the applied inputs.

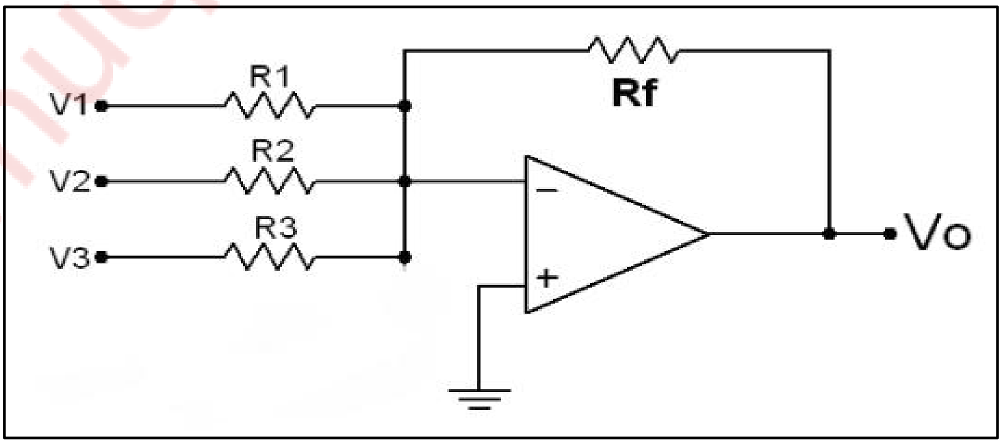


Fig1: Summing inverting amplifier

In this simple summing amplifier circuit, the output voltage, (Vo) now becomes proportional to the sum of the input voltages, V1, V2, V3, etc. Then we can modify the original equation for the inverting amplifier to take account of these new inputs thus:

$$I =- (I_1 + I_2 + I_3)$$

As,

$$I = \frac{v}{R}$$

$$\frac{v_0}{R_f} =- \left(\frac{v_1}{R_1} + \frac{v_2}{R_2} + \frac{v_3}{R_3}\right)$$

$$v_0 =- \left[\left(\frac{R_f}{R_1}\right) v_1 + \left(\frac{R_f}{R_2}\right) v_2 + \left(\frac{R_f}{R_3}\right) v_3\right]$$

The above equation is the equation for summing amplifier.

Case 1: If $R_1 = R_2 = R_3 = R_f$ then $v_0 =- (v_1 + v_2 + v_3)$

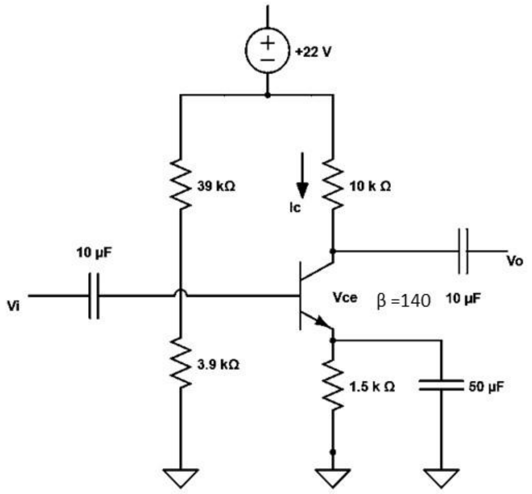
Therefore, the above circuit works as inverting adder.

Case 2: IF $\frac{R_f}{R_1} \neq \frac{R_f}{R_2} \neq \frac{R_f}{R_3}$ then i/p will be amplified by different scaling factor. Then such an amplifier will be known as scaling amplifier/ scaling adder.

Case 3: If $R_1 = R_2 = R_3 = R_f$ or if $\frac{R_f}{R} = \frac{1}{3}$ then $v_0 =- \left(\frac{v_1+v_2+v_3}{3}\right)$ Such an amplifier will be known as Averaging amplifier.

Q2. C)For the emitter bias network of figure below, determine :

(a) I_b (b) I_c (c) V_{ce} (d) V_c (e) E_{th} (f) R_{th} **(10)**



Solution:

Eth = (Vcc *R2) /(R1 + R2)

= (22 * 3.9) / (39 + 3.9)

= 2 V

Rth = R1 || R2

= (39 * 3.9) /(39+ 3.9)

= 3.5454 kΩ

Ib = (Eth – Vbe) / Rth +(1+β) Re

= (2-0.7) / 3.5454 +(1+140) 1.5

= 6.0452 A

Ic = β* Ib

=140 * 6.0452

=0.85 mA

Vce = Vcc – Ic(Rc + Re)

= 22 - 0.85(10 + 1.5)

= 12.22 V

Vc = Vcc – IcRc

=22 -0.85 *10

= **13.5 V**

Q3 A) Explain generation of DSBSC using balanced Modulator along with its frequency and power spectrum.

Solution: **(10)**

- A balanced modulator can be constructed using the non-linear devices like diodes and transistors. The balanced modulator using the diodes is given in Fig 1.

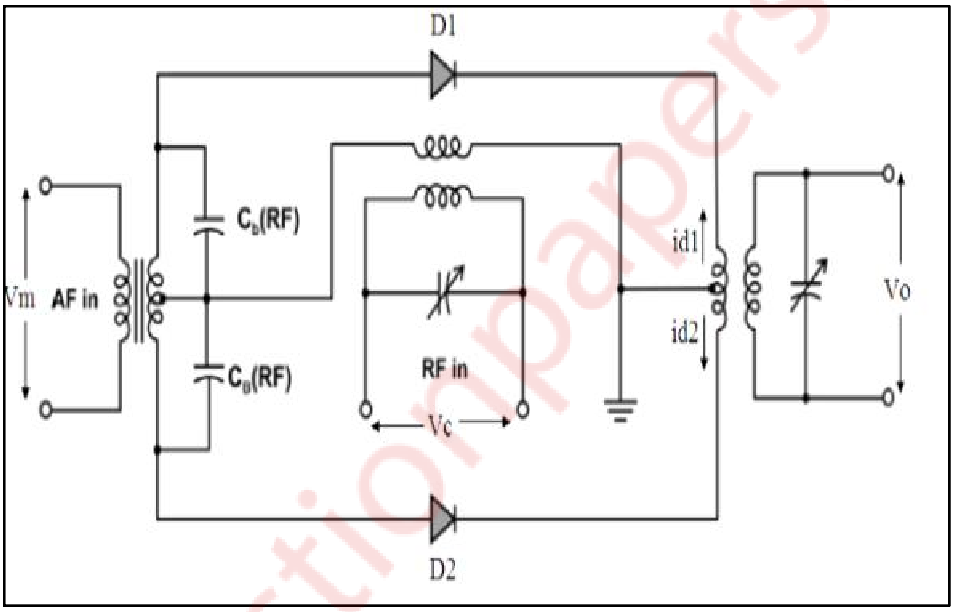


Fig 1: DSBSC Circuit

- The diodes use the nonlinear resistance property for generating modulated signals. Both the diodes receive the carrier voltage in phase; whereas the modulating voltage appears 180° out of phase at the input of diodes, since they are at the opposite ends of a center-tapped transformer. The modulated output currents of the two diodes are combined in the center-tapped primary of the output transformer.
- They therefore subtract, as indicated by the direction of the arrows in the Fig 1. If this system is made completely symmetrical, the carrier frequency will be completely canceled. No system can of course be perfectly symmetrical in practice, so that the carrier will be heavily supressed rather than completely removed. The output of the balanced modulator contaits the two sidebands and some of the miscellaneous components which are taken care of by tuning the output transformer secondary winding. The final output consists only of sidebands.
- As indicated the input voltage will be (Vc + Vm) at the input of diode D1 and (Vc - Vm) at the input of diode D2.
- If perfect symmetry is assumed the proportionality constants will be the same for both diodes and may be called a, b, and c as before.

Generation of DSBSC signal using balanced modulator

The primary current of the output transformer is

i1=id1−id2.

Where,

id1=a+b(vc+vm)+c(vc+vm)^2 and

id2=a+b(vc−vm)+c(vc−vm)^2

Thus, we get,

i1=id1−id2=2bvm+4cvmvc

The modulating and carrier voltage are represented as,

vm=Vmsinωmt   and vc=Vcsinωct

Substituting for vm and vc and simplifying, we get,

$$i_1 = 2bV_m t\, sinWmt + 4c\frac{mv_c}{2}cos\,(\omega_c - \omega_m)t - 4c\frac{mv_{\mathbb{C}}}{2}cos\,(\omega_c + \omega_m)t$$

The output voltage v0 is proportional to primary current i1 and assume constant of proportionality as α, which can be expressed as,

v0=αi1

$$v_0 = 2\alpha b\, sin\,\omega_m t + 4\alpha c\frac{mv_c}{2}cos\,(\omega_c - \omega_m)t - 4\alpha c\frac{mv_c}{2}cos\,(w_c + \omega_m)t$$

Let P=2αbVm and $Q = 2\alpha c\frac{mv_c}{2}$

Thus we have,

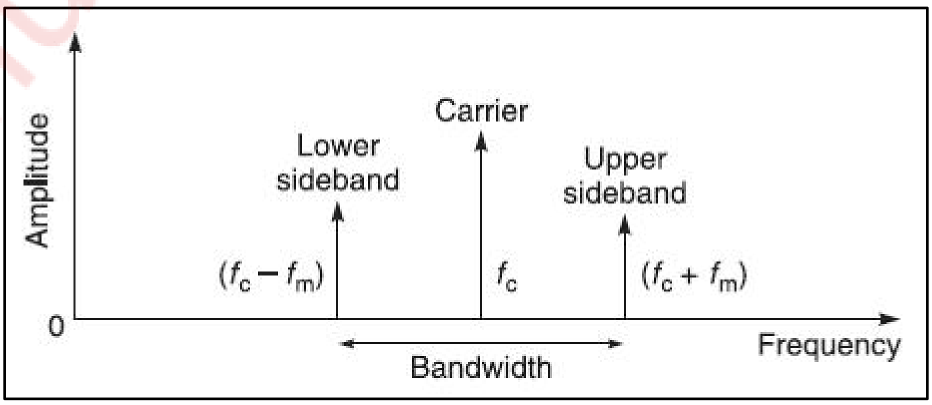
v0=Psinωmt+2Qcos(ωc−ωm)t−2Qcos(ωc+ωm)t

The above equation shows that carrier has been cancelled out , leaving only two sidebands and the modulating frequencies.

The modulating frequencies from the output is eliminated by the tuning of the output transformer, which results in the below equation of the generated DSBSC wave.

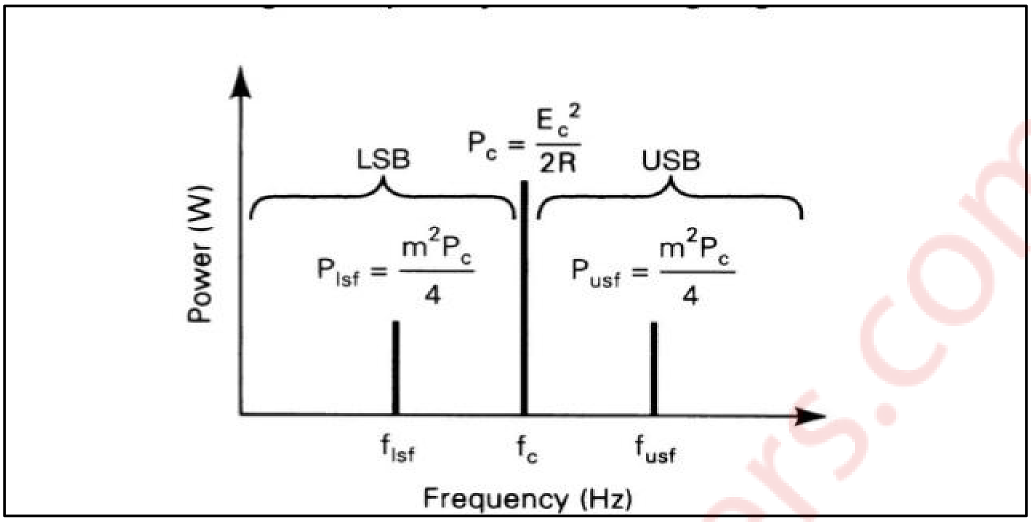
v0=2Qcos(ωc−ωm)t−2Qcos(ωc+ωm)t

Frequency spectrum band:



Frequency spectrum shows that BW of DSBSC is 2fm

Power spectrum band:



Q3 B) With suitable waveforms explain how Op-amp can be used as differentiator.

Solution: **(10)**

- Differentiator is a circuit which provides an o/p waveform whose value at any instant of time is equal to the rate of input at that point in time.
- Differentiator is a circuit which produce o/p voltage which is derivative of i/p voltage.

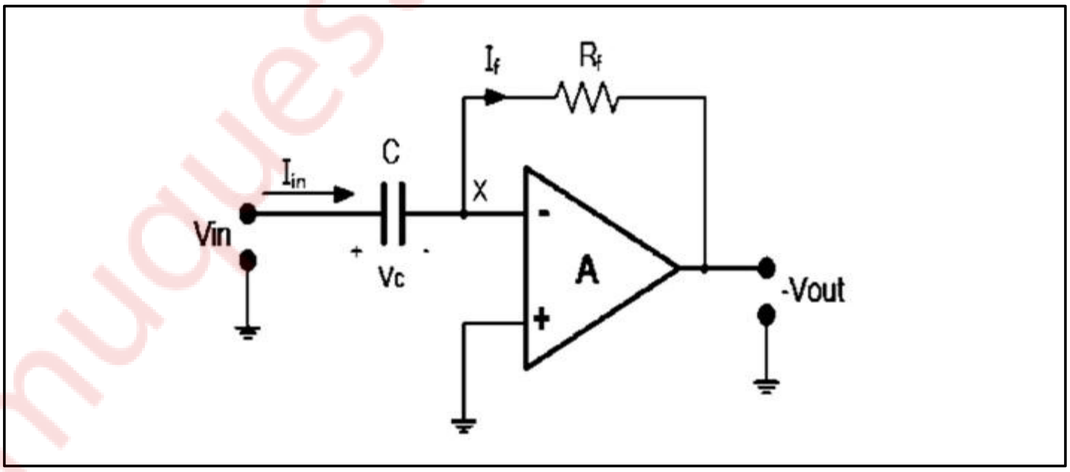


Fig : Op-amp Differentiator circuit

- The input signal to the differentiator is applied to the capacitor. The capacitor blocks any DC content so there is no current flow to the amplifier summing point, X resulting in zero output voltage. The capacitor only allows AC type input voltage changes to pass through and whose frequency is dependant on the rate of change of the input signal.

- At low frequencies the reactance of the capacitor is "High" resulting in a low gain (Rf/Xc) and low output voltage from the op-amp. At higher frequencies the reactance of the capacitor is much lower resulting in a higher gain and higher output voltage from the differentiator amplifier.
- However, at high frequencies an op-amp differentiator circuit becomes unstable and will start to oscillate. This is due mainly to the first-order effect, which determines the frequency response of the op-amp circuit causing a second-order response which, at high frequencies gives an output voltage far higher than what would be expected. To avoid this the high frequency gain of the circuit needs to be reduced by adding an additional small value capacitor across the feedback resistor Rf.
- Since the node voltage of the operational amplifier at its inverting input terminal is zero, the current, i flowing through the capacitor will be given as:

  $I_{in} = I_f$

  And $I_f = -\frac{Vout}{R_f}$
- The charge on the capacitor

  $Q = C \times V_{in}$
- Thus rate of change of this charge is: $\frac{dQ}{dt} = C\frac{dv_{in}}{dt}$

  But dQ/dt is the capacitor current ,I

  $I_{IN} = C\frac{dv_{in}}{dt} = I_F$

  Therefore $-\frac{Vout}{R_f} = C\frac{dv_{in}}{dt}$

  From which we have ideal voltage output for the op-amp differentiator is given as:

  $\mathbf{V_{OUT} = -R_F\, C\frac{dv_{in}}{dt}}$

  <u>Op-amp differentiator waveform</u>
- If we apply a constantly changing signal such as a Square-wave, Triangular or Sine-wave type signal to the input of a differentiator amplifier circuit the resultant output signal will be changed and whose final shape is dependant upon the RC time constant of the Resistor/Capacitor combination.

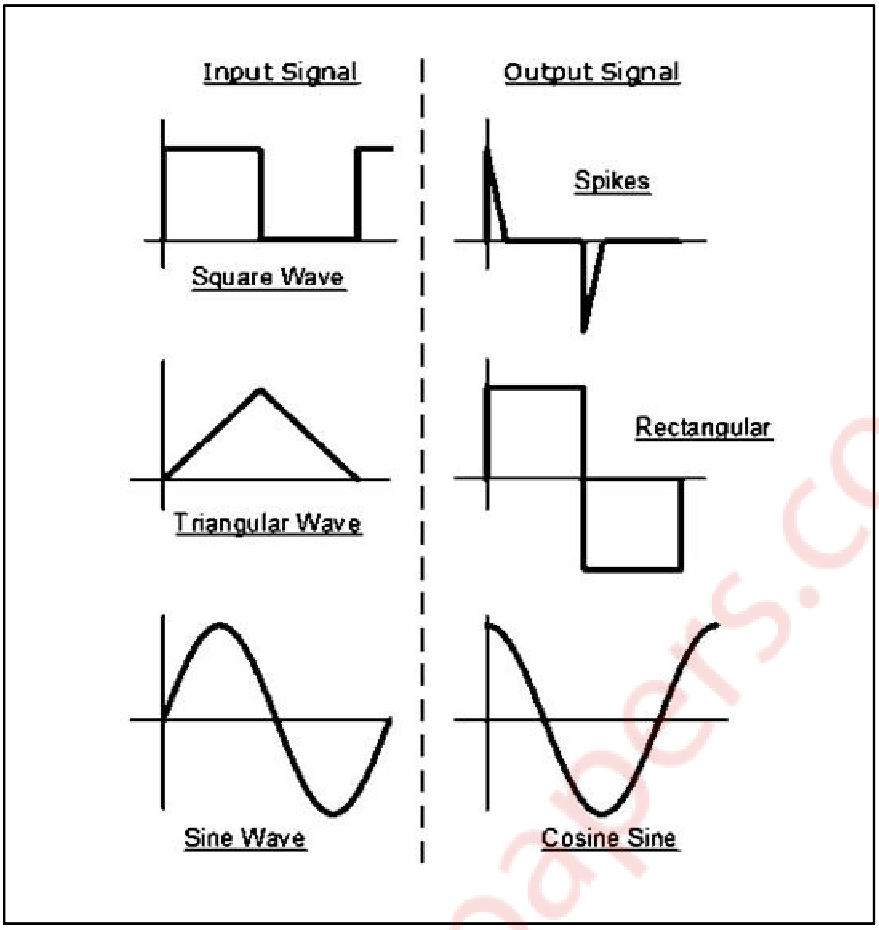


Fig2: Op-amp differentiator waveform

Q4 A) For an AM DSBFC envelope with Vmax= 20V and Vmin = 4V; determine

i. Peak amplitude of USF AND LSF **(10)**

ii. Peak amplitude of carrier

iii. Peak change in the amplitude of the envelope

iv. Modulation coefficient

v. Draw the AM envelope.

<u>Solution:</u>

Modulation coefficient: m = (vmax-vmin) / (vmax+vmin)

= (20-4) / (20+4)

m = 16 / 24

m =0.667

Peak amplitude of USF and LSF = ¼(Vmax - Vmin)

=1/4(20-4)

=4 V

Peak change in the amplitude of the envelope: Em =1/2(Vmax-Vmin)

=1/2(20-4)

=1/2(16)

=8 V

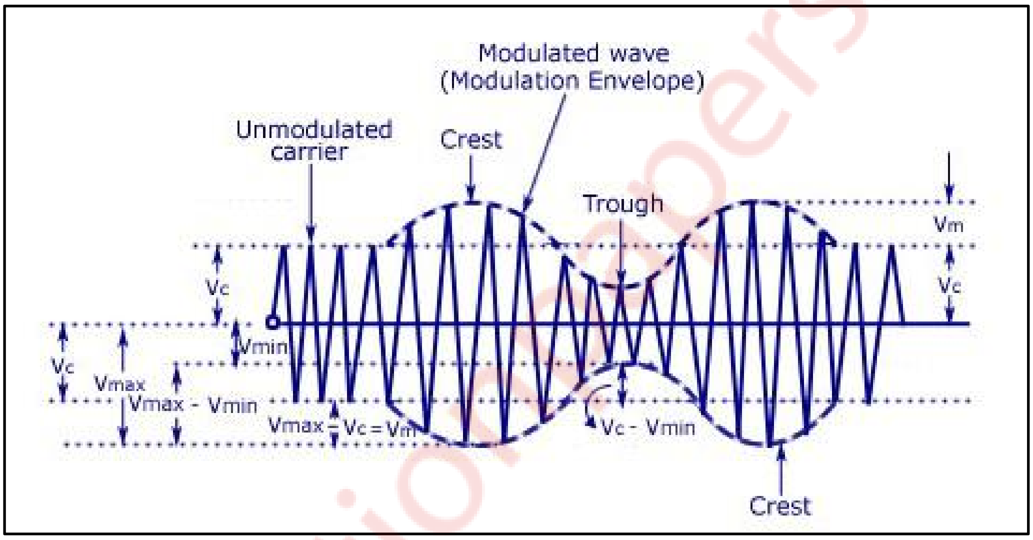
Peak amplitude of carrier : Ec=1/2(Vmax+Vmin)

=1/2(20+4)

=1/2(24)

=12 V

Draw the AM envelope:



Q4.B) Differentiate between TDM and FDM.

Solution:

(10)

TDM	**FDM**
TDM means Time-Division Multiplexing.	FDM means Frequency-Division Multiplexing.
TDM requires Synchronization.	FDM does not requires Synchronization.
TDM requires sync pulse for its operation.	FDM requires Guard bands for its operation.
TDM is more efficient and is widely used technique in multiplexing.	FDM is less efficient compared to TDM.
TDM is the transmission technique in which different signal are transmitted over a common channel and each signal occupies entire range of bandwidth in the time domain.	FDM is the transmission technique in which different signal are transmitted over a common channel and each signal occupies different slot within that bandwidth of the frequency domain.
TDM is not sensitive for Cross Talk (Noise Immunity).	FDM suffers from the cross talk immunity due to Bandpass Filter.

TDM can be used for both Analog and Digital signals.	FDM can be used for Analog signals only.
TDM is used in Pulse code modulation.	FDM is used in TV and RADIO broadcasting.

Q4. C) State Shannon's Theorem and explain its significance.

Solution: **(5)**

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.

Q5 A) Draw PAM, PWM and PPM waveforms in time domain using a sinusoidal signal and explain in brief.

Solution: **(10)**

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.

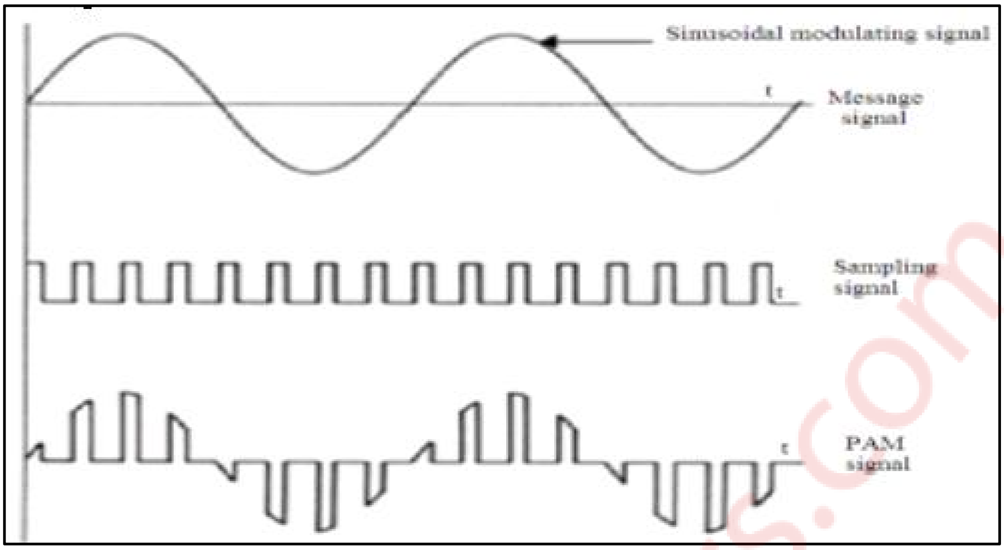


Fig1 : PAM

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.

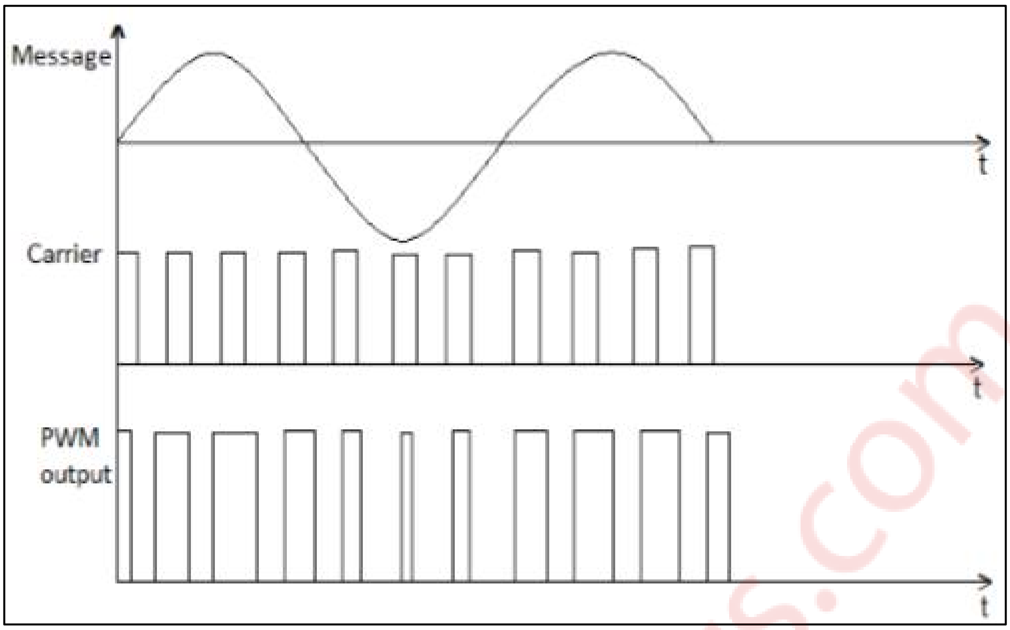


Fig 2: PWM

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.

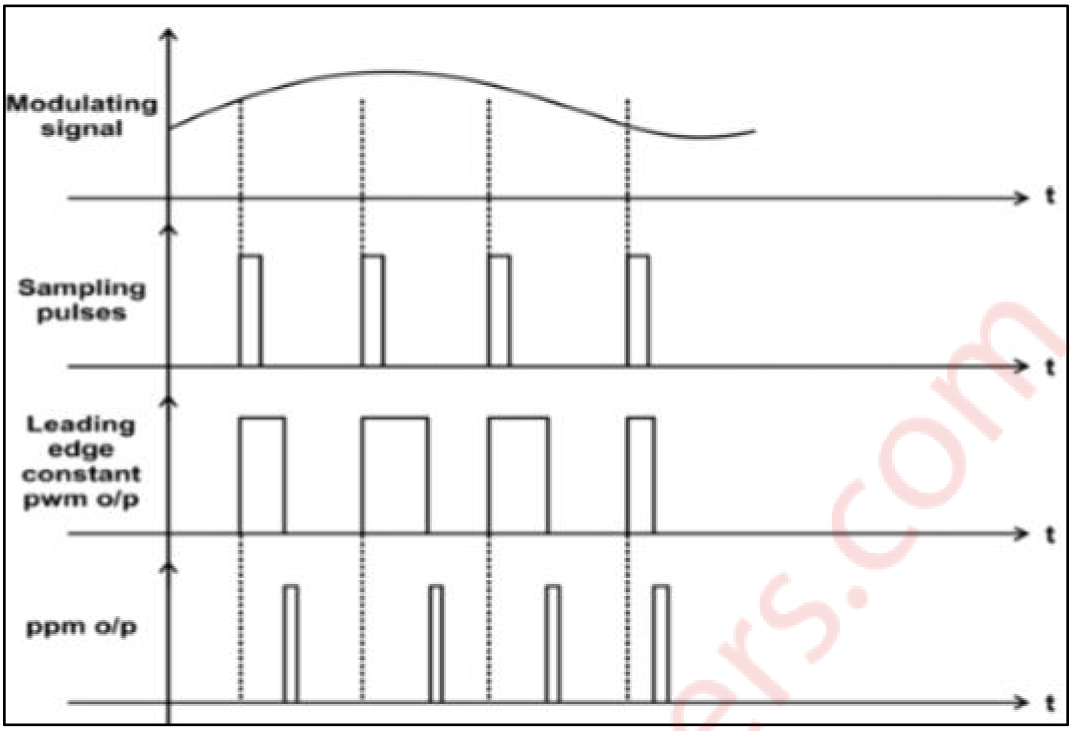


Fig3: PPM

Q5 B) Define and explain in brief Amount of information, average information, information rate and channel capacity of a communication system.

Solution: **(10)**

1)Information:

Source of communication system is information, which can be analog or digital.

Information theory is a mathematical approach to the study of coding of information along with the quantification, storage, and communication of information.

Conditions of occurrence of events

If we consider an event, there are three conditions of occurrence.

If the event has not occurred, there is a condition of **uncertainty**.

If the event has just occurred, there is a condition of **surprise**.

If the event has occurred, a time back, there is a condition of having some **information**.

These three events occur at different times. The difference in these conditions help us gain knowledge on the probabilities of the occurrence of events.

2)Average information:

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

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.

3)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

4)Channel Capacity:

The channel capacity, C, is defined to be the maximum rate at which information can be transmitted through a channel. The fundamental theorem of information theory says that at any rate below channel capacity, an error control code can be designed whose probability of error is arbitrarily small.

Data rate(channel capacity) depends upon 3 factors:

The bandwidth available

Number of levels in digital signal

The quality of the channel – level of noise

Two theoretical formulas to calculate the data rate:

Nyquist for a noiseless channel

Shannon for a noisy channel.

Noiseless Channel : Nyquist Bit Rate –
For a noiseless channel, the Nyquist bit rate formula defines the theoretical maximum bit rate

$$BitRate = 2 * Bandwidth * \log_2(L)$$

Where,

bandwidth is the bandwidth of the channel

L is the number of signal levels used to represent data

BitRate is the bit rate in bits per second.

Noisy Channel : Shannon Capacity –
In reality, we cannot have a noiseless channel; the channel is always noisy. Shannon capacity is used, to determine the theoretical highest data rate for a noisy channel:

Capacity = bandwidth *$\log_2$(1+ SNR)

Where,

bandwidth is the bandwidth of the channel

SNR is the signal-to-noise ratio

and capacity is the capacity of the channel in bits per second.

Q6 A) State significance of modulation in communication.

Solution: **(5)**

Modulation is a process in which we select a higher frequency (normally referred as carrier frequency or RF) which is a sine wave. It is a process of mixing a signal with a sinusoid to produce a new signal.

Mixing of low frequency signal with high frequency carrier signal is called modulation.

$$f(t) = A\,sin\,(\omega t + \phi)$$

Where, A= Amplitude

W=frequency

ϕ= phase

In the process, either amplitude or frequency or phase of high frequency(RF) is changed with respect to instantaneous amplitude of AF.

If amplitude is changed we get amplitude modulation(AM). If frequency is changed, we get FM and if phase is changed, we get phase modulation.

Why do we need modulation?

For transmission or reception of any signal with frequency 'f' the height of the antenna required is at λ/4 where λ is the wavelength. If we calculate height of antenna for AF, it comes out to be very large and not possible practically eg. For AF= 15kHz, height of antenna can be calculated as

$\lambda = c/f = 3x\ 10^8 /15\ x\ 10^8$=5000 m

Hence AF cannot be transmitted directly.

Suppose an imaginary antenna capable of transmitting AF is available and being used by all radio station. Then all radio station will transmit same frequency hand at the same time. Therefore , AF signals will interfere each other and what we receive is unwanted signals. Hence AF cannot be transmitted directly.

From the above two points it is clear that the main difficulty is the height of antenna . this can be overcome only if high frequency(HF) is transmitted. But with transmission of HF the very purpose of communication i.e. to transmit AF is lost.

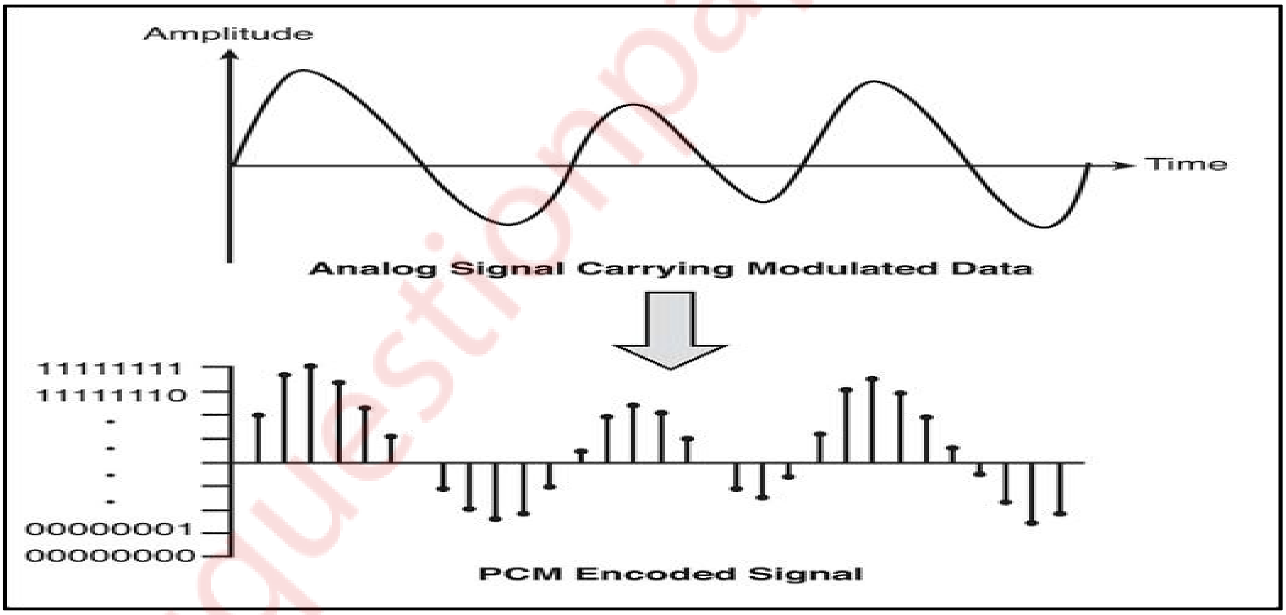
Therefore to overcome above difficulties we have to use modulation.

Q6 B) Write a note on pulse code modulation with waveforms.

Solution: **(5)**

Pulse code modulation is a method that is used to convert an analog signal into a digital signal, so that modified analog signal can be transmitted through the digital communication network.

In this type of modulation, the signal to be transmitted (i.e. an analog signal which is band limited with frequency fm) is divided into different levels.



These levels are known as Quantization levels. Sampling new sample the analog signal with sampling frequency fs >=2fm.

At the o/p of sampler we get voltage of analog signal present at sampling instant. These sampled voltage are given to quantisor via TDM multiplexer. The function of quantiser is to sound off sampled voltage to nearest integer value belonging to quantisation levels.

Quantised levels are converted to digital.

This makes PCM as Digital modulation.

PCM is in binary form, so there will be only two possible states high and low levels(0 and 1). We can get analog signal back by demodulation process. The Pulse Code Modulation process is done in three steps Sampling, Quantization, and Coding.

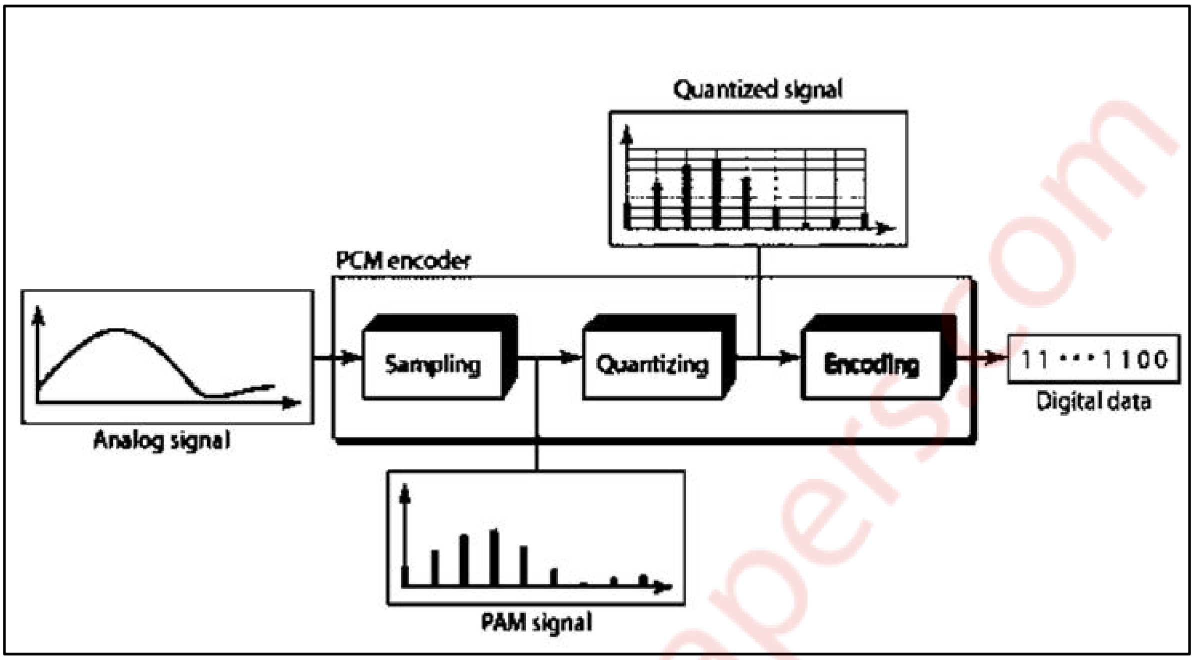


Fig1: Block diagram of PCM

Sampling

Sampling is a process of measuring the amplitude of a continuous-time signal at discrete intervals, which converts the continuous signal into a discrete signal. For example, conversion of a sound wave to a sequence of samples.

Quantization

In quantization, an analog sample with an amplitude that converted into a digital sample with an amplitude that takes one of a specific defined set of quantization values. Quantization is done by dividing the range of possible values of the analog samples into some different levels, and assigning the center value of each level to any sample in quantization interval.

Coding

The encoder encodes the quantized samples. Each quantized sample is encoded into an 8-bit code word by using A-law in the encoding process.

Q6 C)Explain and give ideal values of following parameters of an op-amp:

i) CMRR **(10)**

ii) slew rate

iii) offset voltage

iv) Input Resistance

v) Output Impedence

Solution:

CMRR(Common Mode Rejection Ratio):

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 741IC, CMRR is 90dB.

Slew rate:

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.

Offset voltage:

It is the voltage that must be applied between two input terminals of Op-amp to make o/p offset voltage zero.

Lowest values are 15uV for an ideal precision op-amp & the maximum value if 6mV dc.

In an ideal op amp, if no voltage is applied to the inverting and noninverting input pins, the op amp will output a voltage of 0, since there is no difference at all of the voltage applied to the 2 input pins.

Vio = V1 – V2

Vio can be positive or negative.

Input Resistance:

This is the common mode voltage that can be applied to both input terminals without disturbing performance of Op-amp

For IC741 -> +- 13 V

Output Impdedence:

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

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