Electronic circuits and communication fundamentals (ECCF)

May-2019(Choice based)

Q1A) Explain with diagram Input and output characteristic of Common base configuration.

Solution: **(5)**

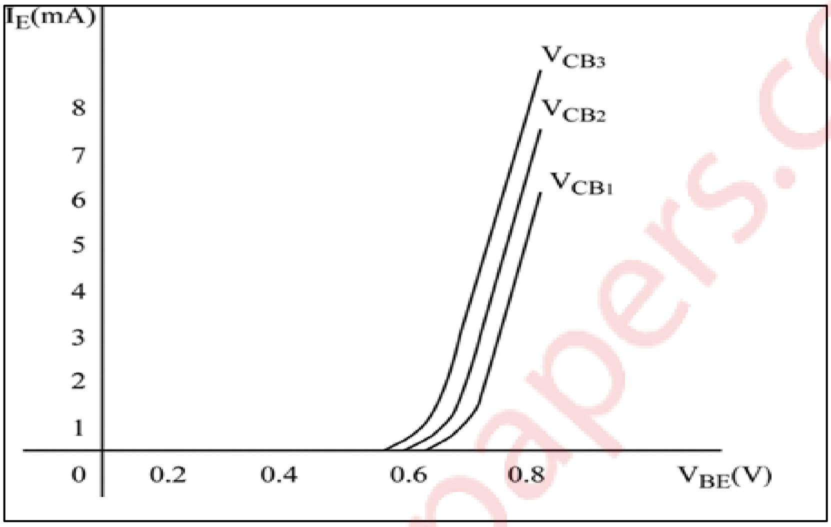


Figure1: Input characteristics of Common base configuration

Input characteristics are obtained between the input current and input voltage at constant output voltage. It is plotted between $\mathbf{V_{EB}}$ and $\mathbf{I_E}$ at constant $\mathbf{V_{CB}}$ in CB configuration.

DC input resistance (static)= R_i

AC input resistance (dynamic)= r_i

$R_i = \frac{V'_{EB}}{I'E} \; |v_{cB} = cons \tan t$

$r_i = \frac{\Delta v_{EB}}{\Delta IE} \; |\Delta V_{CB} = 0$

Input characteristics is a curve of emitter base voltage (V_{EB}) with respect to emitter current (I_E) at constant base collector voltage (V_{BC}). Emitter base voltage is shown in X axis of characteristics. Emitter current is shown on the y axis. This figure shows the input characteristics of common base configuration.

From fig we can conclude:

- As V_{BC} increases, slope increases. This is due to Early effect. Hence early effect provides slope. Therefore at input a Forward biased diode is present, Ri is very small.

- With the all collector-base voltage (V_{BC}) shape of graph remain same that means emitter current is totally independent of base-collector (V_{BC}) voltage. This leads to the conclusion that the emitter current is independent of collector voltage.

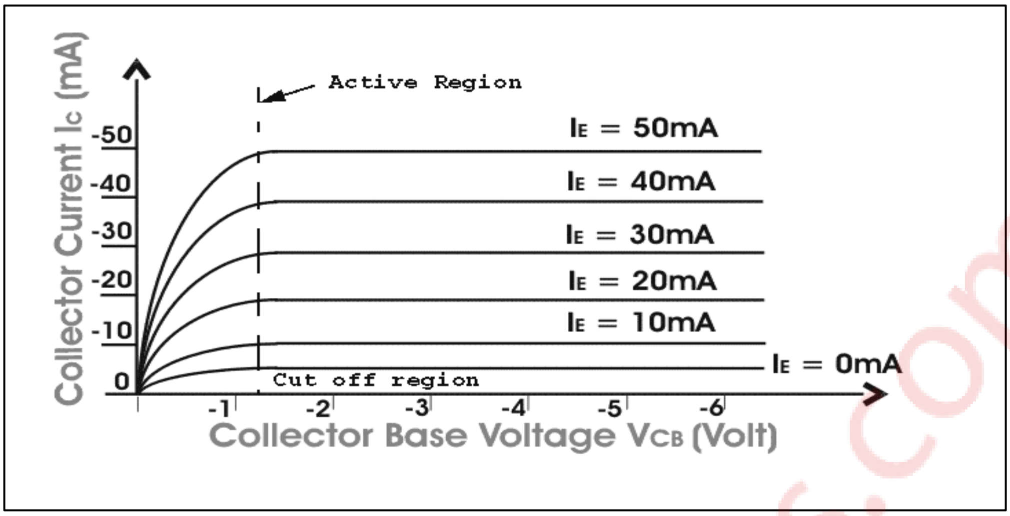


Figure2: Output characteristics of Common base configuration

Output characteristics are obtained between the output voltage and output current at constant input current. It is plotted between $\mathbf{V_{CB}}$ and $\mathbf{I_C}$ at constant $\mathbf{I_E}$ in CB configuration.

From fig we can conclude:

- Collector current (Ic) varies with V_{BC} only on the starting or when the collector base voltage (V_{BC}) is below 1v. Transistor never operated below this voltage.

- After voltage (V_{BC}) increase above 1-2 V, you can see collector current (Ic) becomes a straight horizontal line. That mean collector current becomes constant above 1-2 V. It means collector current is independent of collector base voltages and depends upon the emitter current only. This proves that the emitter current almost flows to collector current. The transistor is always operated on this region.

- The large change in collector base voltage there is a small change in collector current. That means output resistance of the circuit is very high.

Q1B) List the ideal Characteristic of op-amp.

Solution: **(5)**

An ideal op-amp is usually considered to have the following characteristics:

- Infinite open-loop gain G = v_{out} / v_{in}

- Infinite input impedance R_{in}, and so zero input current

- Zero input offset voltage

- Infinite output voltage range
- Infinite bandwidth with zero phase shift and infinite slew rate
- Zero output impedance R_{out}
- Zero noise
- Infinite common-mode rejection ratio (CMRR)
- Infinite power supply rejection ratio.

Q1C) Calculate the percent power saving an SSB signal if the AM wave is modulated to a depth of (a)100 % and (b) 50%

Solution: **(5)**

(a) 100 %

$P_t = P_c\,(1 + \frac{m^2}{2})$

$P_t = P_c\,(1 + \frac{1^2}{2})$

$P_t = 1.5\ P_c$

$P_{SB} = Pc\ \frac{m^2}{4}$

$P_{SB} = 0.25\ Pc$

% Power saving = (1.5 – 0.25) / 1.5

=0.83

= 83.3 %

(b) 50%

$P_t = P_c\,(1 + \frac{m^2}{2})$

$P_t = P_c\,(1 + \frac{0.5^2}{2})$

$P_t = 1.125\ P_c$

$P_{SB} = Pc\ \frac{m^2}{4}$

$P_{SB} = 0.0625\ Pc$

% Power saving = (1.125 – 0.0625) / 1.125

=0.944

= 94.4 %

Q1 D) Define the term Information theory. Give definitions for Information Rate and Entropy.

Solution: **(5)**

Information theory:

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

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

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.

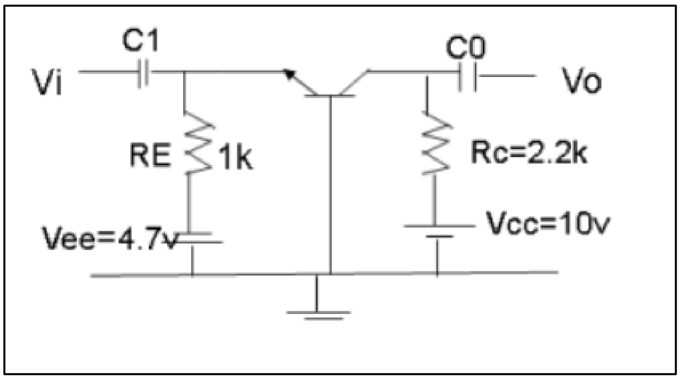
Shannon, in fact, defined entropy as a measure of the average information content associated with a random outcome.

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.

Q2A) For the circuit shown in Figure below calculate V_{CB}, I_E and I_B if β=100 **(10)**



Solution:

$I_E = (V_{EE} – V_{BE}) / R_E$

$= (4.7 – 0.7) / 1$

I_E = 4 mA

$I_B = I_E /(\beta+1)$

$=4 / (100+1)$

I_B =39.60 µA

$V_{CB} = V_{CC} – I_C R_C$

$= V_{CC} – \beta I_B R_C$

$=10 – (100) (39.60) (2.2)$

$=10 – 8.712$

V_{CB} =1.288 V

Q2B) Explain how op-amp can be used as a 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.

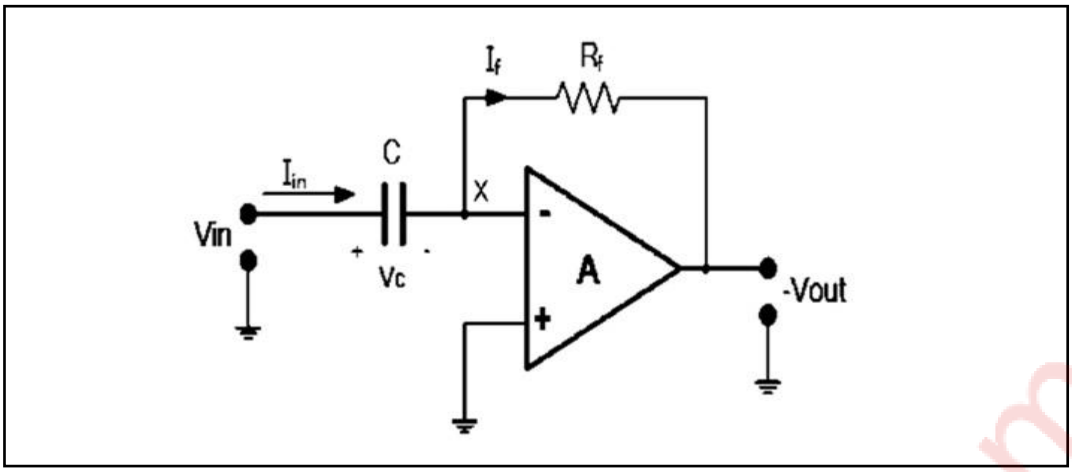


Fig: Op-amp Differentiator circuit

- 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

  $Q = C \text{ x } V_{in}$

  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:

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

Op-amp differentiator waveform

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.

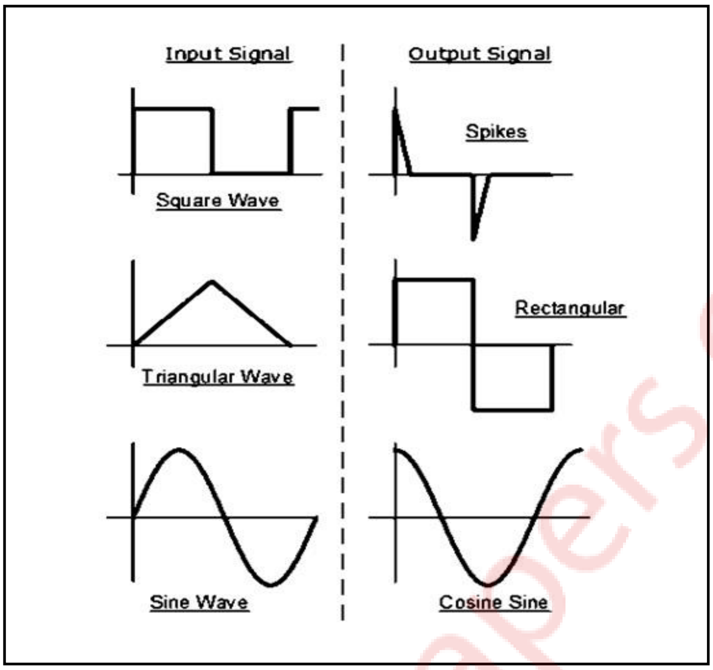


Fig2: Op-amp differentiator waveform

Q3A) What do you mean by Zero Crossing detector? Explain with diagram

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.

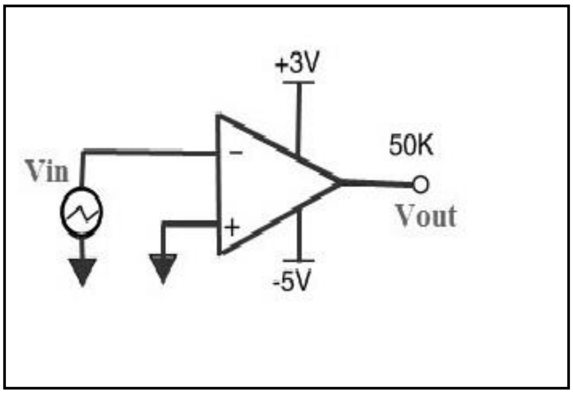


Fig1: Zero crossing detector

- 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.

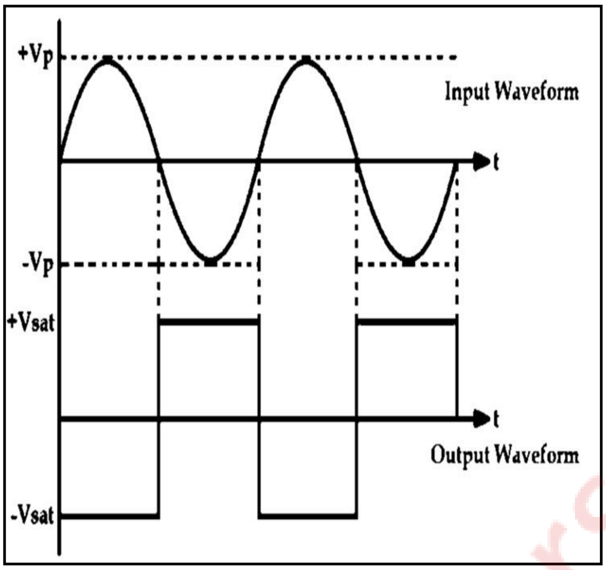


Fig2: Waveform of zero crossing detector

- 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

Q3B) Write Short note on generation of FM by Armstrong method.

Solution: **(5)**

Armstrong method of FM generation is the indirect method because the modulating signal directly varies the phase of the carrier, which indirectly changes the frequency.

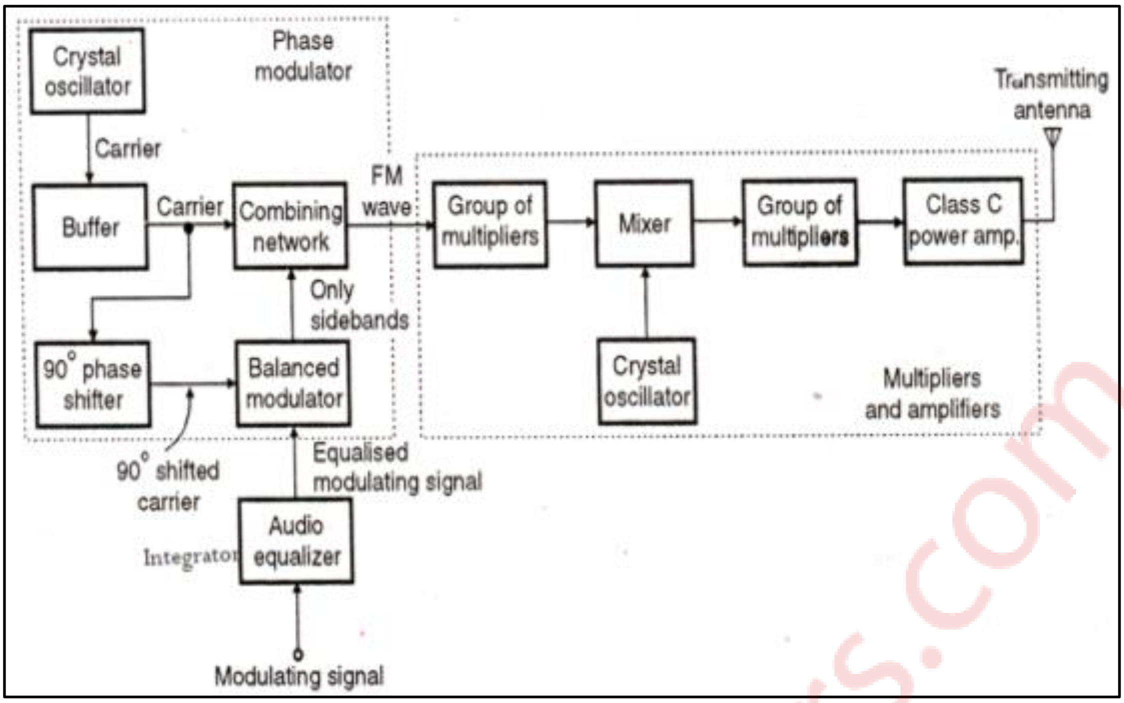


Figure1: Armstrong Frequency modulation system

- The source of carrier for the Armstrong transmitter is the crystal oscillator. A relatively low frequency sub-carrier (fc) is phase shifted by 90° and is fed to a balanced modulator, where it is mixed with the input modulating signal (fm).
- A double sideband suppressed carrier wave is produced at the output from the balanced modulator, and this is combined with the original carrier in the combing network to generate a narrow band frequency modulated waveform.

- The phase diagrams illustrate the working of this modulation system.
  Figure (a)shows the phasor diagram for the original carrier voltage (Vc),
  (b) shows phasor of the phase shifted carrier (V′c),
  (c) shows phasors for side frequency components of the suppressed carrier voltage 〖(V〗_usb)
  and (Vlsb). Since the suppressed carrier voltage (V′c) is 90° out of phase with (Vc), the upper and lower sidebands combine to produe a component (Vm) which is always perpendicular to (Vc).
  (d) shows phasor addition of (Vc), (Vusb) and (Vlsb) which is the resultant of the combining network.

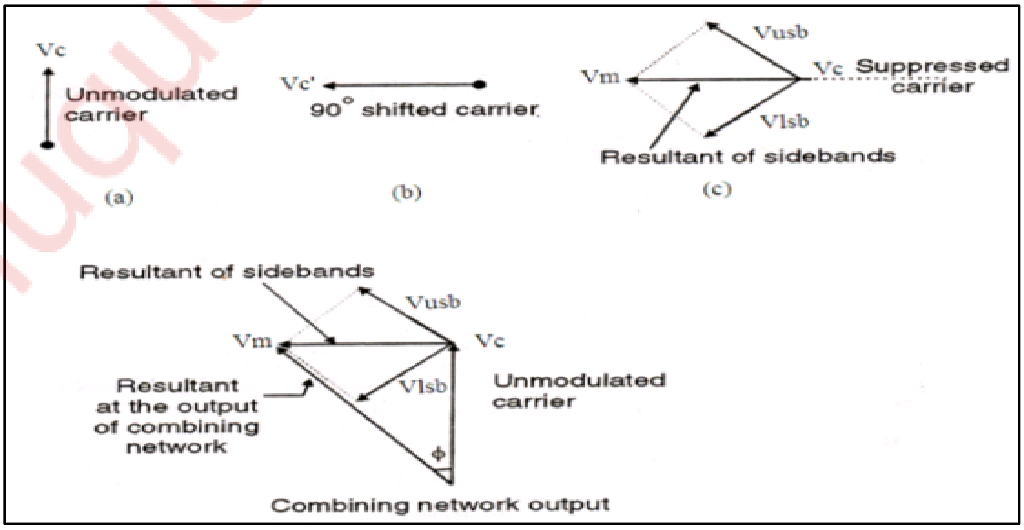


Figure2: Phasor diagrams (a) carrier phasor, (b) phase shifted carrier, (c) sideband phasors, (d) resultant phasor

- It is seen that the output from the combing network is a signal whose phase is varied by fm and magnitude is directly proportional to the magnitude of Vm.
- The modulation index at the output of the combining network is inadequate to produce a wideband FM and therefore must be multiplied and amplified before transmitting.
- A combination of multipliers and mixers are thus placed to develop the desired transmit carrier frequency with 75 kHz frequency deviation.
- The outcome of the mixer block is the change in the center frequency, while the outcome of the multiplier block is the multiplication of the center frequency and the frequency deviation equally. Hence a narrow band FM with small frequency deviation is transformed into a wide band FM with large frequency deviation.
- In the Armstrong method of FM generation, the phase of the carrier is directly modulated in the combing network through summation, generating indirect frequency modulation. The magnitude of the phase deviation is directly proportional to the amplitude of the modulating signal but independent of its frequency.
- Very high frequency stability is achieved through Armstrong method since the crystal oscillator is used as carrier frequency generator.

Q3C) Use op-amp IC741 to realize the expression V0=5V1+2V2−3V3 **(5)**

Solutions:

Taking LCM of 5,2,3

2	5	2	3
3	5	1	3
5	5	1	1
	1	1	1

LCM= 2 X 3 X 5

=30

Assume R_F = 30 kΩ

For inverting amplifier, V0 $=-\frac{R_F}{R_1}\times v_{in}$

For V0 = 5 V1

$\frac{R_F}{R_1}$ = 5

$\frac{30}{R_1}$ = 5

R1=6 kΩ

For V0 = 2 V2

$\frac{R_F}{R2}$ = 2

$\frac{30}{R_2} = 2$

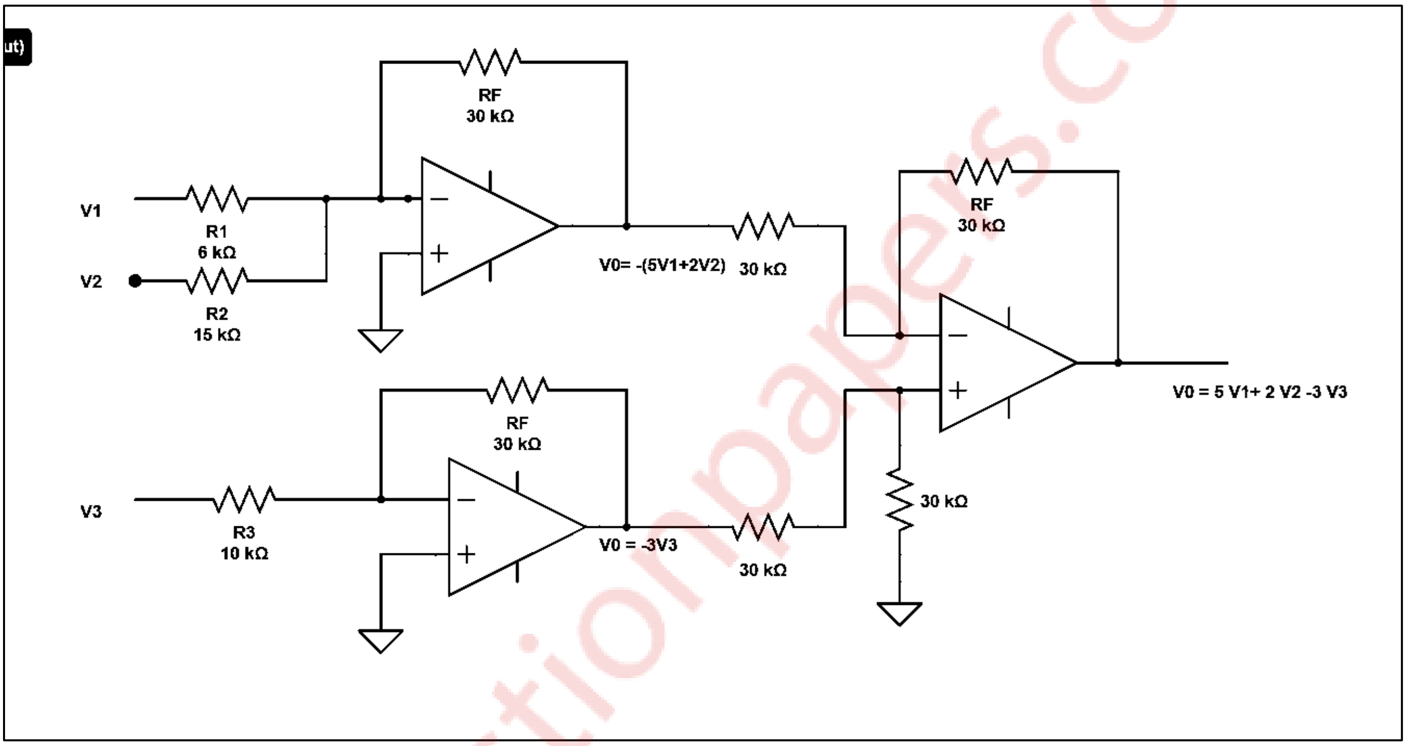
R2=15 kΩ

For V0 = m 3 V3

$\frac{R_F}{R_3} = 3$

$\frac{30}{R_3} = 3$

R3= 10 kΩ



Q3D) What is a Nyquist criteria? What is its significance.

Solutions: **(5)**

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.

Q4A) Explain Delta Modulation with neat diagram and waveforms after each block.

Solutions: **(10)**

Delta modulation (DM) is a Differential Pulse Code Modulation (DPCM) in which the difference between samples at k and k-1 sampling time is encoded into just a single bit.

Delta modulation transmits only one bit per sample. Here, the present sample value is compared with the previous sample value and this result whether the amplitude is increased or decreased is transmitted.

Figure 1. shows the transmitter. It is also known as Delta modulator.

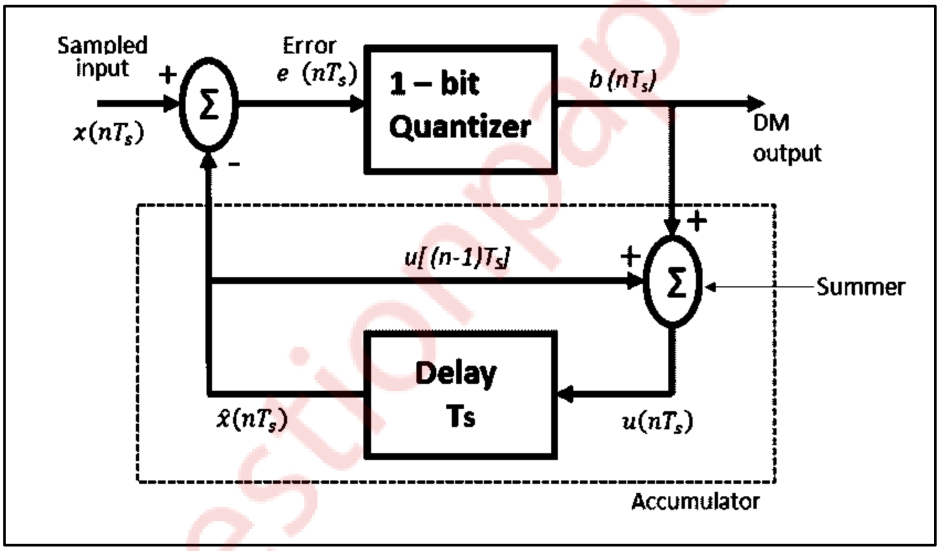


Figure1: Delta Modulation Transmitter

It consists of a 1-bit quantizer and a delay circuit along with two summer circuits.

The summer in the accumulator adds quantizer output (±Δ) with the previous sample approximation. This gives present sample approximation. i.e.,

$U(nT_s)=u[(n-1)T_s]+b(nT_s)$

The previous sample approximation $u[(n-1)T_s]$ is restored by delaying one sample period T_s.

The samples input signal $x(nT_s)$ and staircase approximated signal $\hat{x}(nT_s)$ are subtracted to get error signal $e(nT_s)$.

Thus, depending on the sign of $e(nT_s)$, one bit quantizer generates an output of +Δ or -Δ .
If the step size is +Δ, then binary '1' is transmitted and if it is -Δ, then binary '0' is transmitted.

Figure2, shows the receiver. At the receiver end also known as delta demodulator, it comprises of a low pass filter(LPF), a summer, and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator.

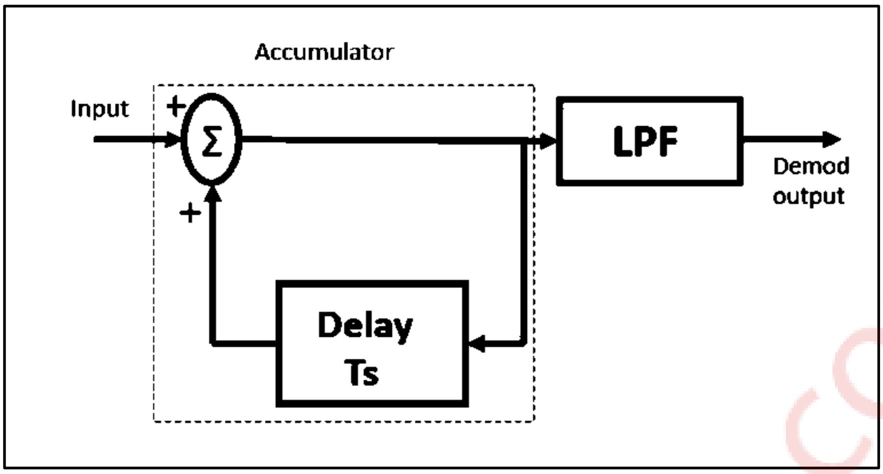


Figure2: Delta Modulation Demodulator

The accumulator generates the staircase approximated signal output and is delayed by one sampling period T_s.
It is then added to the input signal.

If the input is binary '1' then it adds +Δ step to the previous output (which is delayed).

If the input is binary '0' then one step 'Δ' is subtracted from the delayed signal.

Also, the low pass filter smoothens the staircase signal to reconstruct the original message signal x(t) .

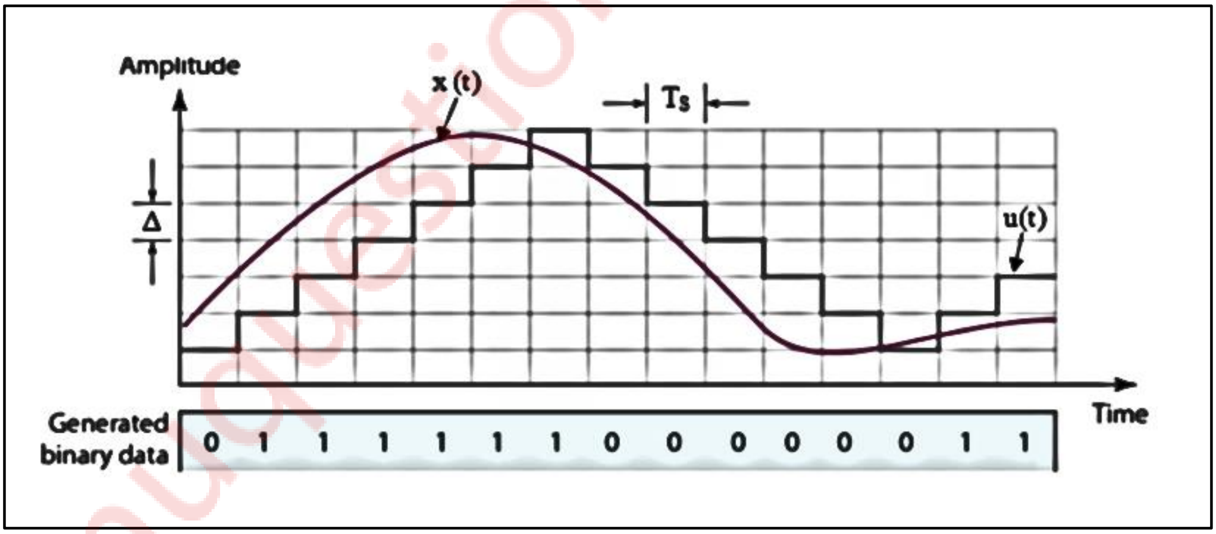


Figure3: Delta Modulation waveform

Figure3 shows the analog signal x(t) and its staircase approximated signal by the delta modulator.

Input signal x(t) is approximated to step signal by the delta modulator. This step size is kept fixed.

The difference between the input signal x(t) and staircase approximated signal is confined to two levels, i.e., +Δ and -Δ.

Now, if the difference is positive, then approximated signal is increased by one step, i.e., 'Δ'. If the difference is negative, then approximated signal is reduced by 'Δ'.

When the step is reduced, '0' is transmitted and if the step is increased, '1' is transmitted.

Hence, for each sample, only one binary bit is transmitted.

Q4B) An AM signal appears a 50Ω load and has the following equation $v(t)=12(1+\sin 12.566\times10^{3}t)\sin 18.85\times10^{8}t$ volts **(10)**

1. Sketch the envelope of this signal in time domain

2. Calculate modulation index, sideband frequencies, total power and bandwidth

Solution:

Given:

R= 50Ω

$V(t) = 12(1+\sin 12.566\times10^{3}t)\sin 18.85\times10^{8}t$ volts

Compare the above expression with standard AM

$$e_{AM} = E_C[1 + m\,sin\,\omega mt]\,sin\,\omega ct$$

Therefore, $E_C = E_M = 12$ V

m=1

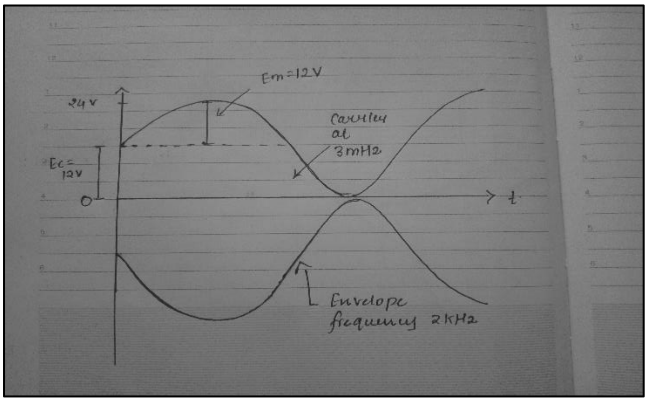
$f_C = 18.85 \text{ x } 10^{6} / 2\pi$

= 3 mHz

$f_M = 12.566 \text{ x } 10^{3} / 2\pi$

= 2 mHz

Sketch of signal in time domain:



m=1

f_{USB} = fc + fm

= 3002 kHz

F_{LSB} = fc- fm

= 2998 kHz

$P_t = P_c\,(1 + \frac{m^2}{2})$

$P_t = P_c\,(1 + \frac{1^2}{2})$

$P_t = 1.5\;\; P_c$

$=1.5\, \frac{\left[\frac{E_C^2}{\sqrt{2}}\right]}{R}$

$=1.5 \times (12)^2 / 2 \times 50$

P_t = 2.16 W

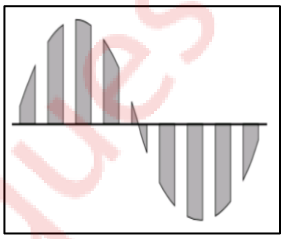
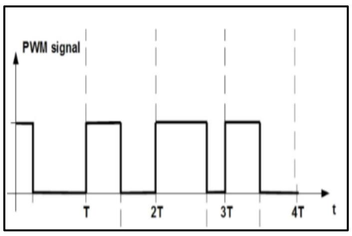
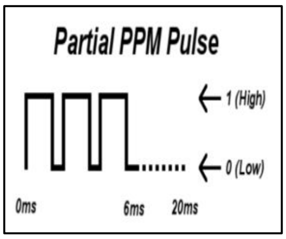
BW= 2fm

BW=4 kHz

Q5A) Compare PAM, PWM and PPM pulse modulation techniques. (10)

Solution:

Parameter	**PAM**	**PWM**	**PPM**
Type of Carrier	Train of Pulses	Train of Pulses	Train of Pulses
Variable Characteristic of the Pulsed Carrier	Amplitude	Width	Position

Bandwidth Requirement	Low	High	High
Noise Immunity	Low	High	High
Information Contained in	Amplitude Variations	Width Variations	Position Variations
Power efficiency (SNR)	Low	Moderate	High
Transmitted Power	Varies with amplitude of pulses	Varies with variation in width	Remains Constant
Complexity of generation and detection	Complex	Easy	Complex
Transmitter power	Instantaneous transmitter power varies with the amplitude of the pulses	Instantaneous transmitter power varies with the amplitude and width of the pulses	Instantaneous transmitter power remains constant with the width of the pulses
Output Waveforms			

Q5B) Explain the generation of DSBSC using Balance modulator.

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.

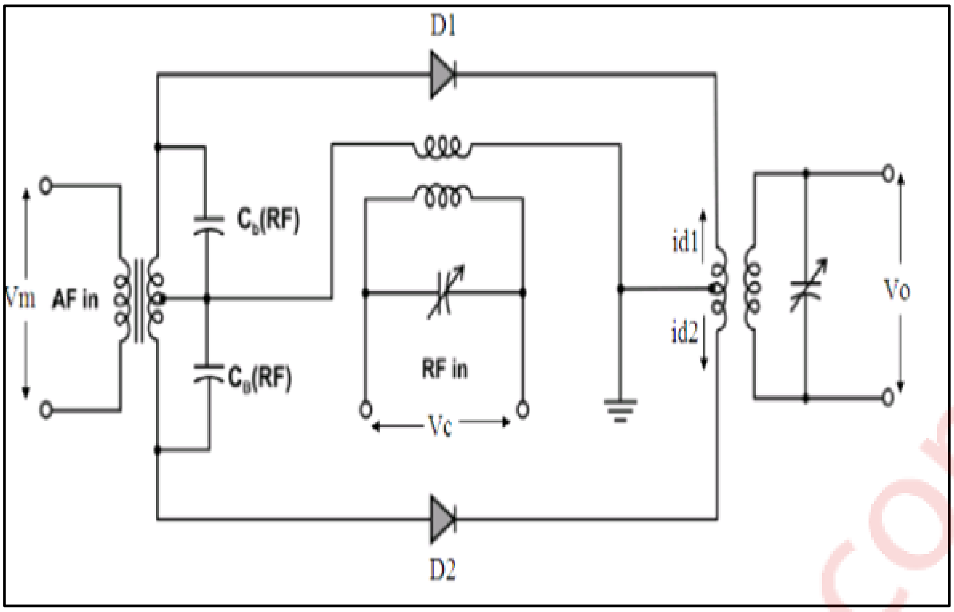


Fig 1: DSBSC Circuit

- 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,

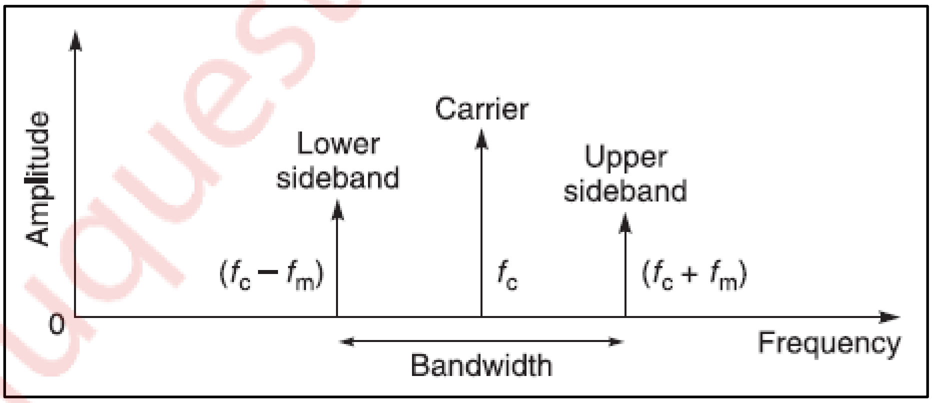
v0=Psinωmt+2Qcos(ωc−ωm)t−2Qcos(ωc+ωm)t

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.

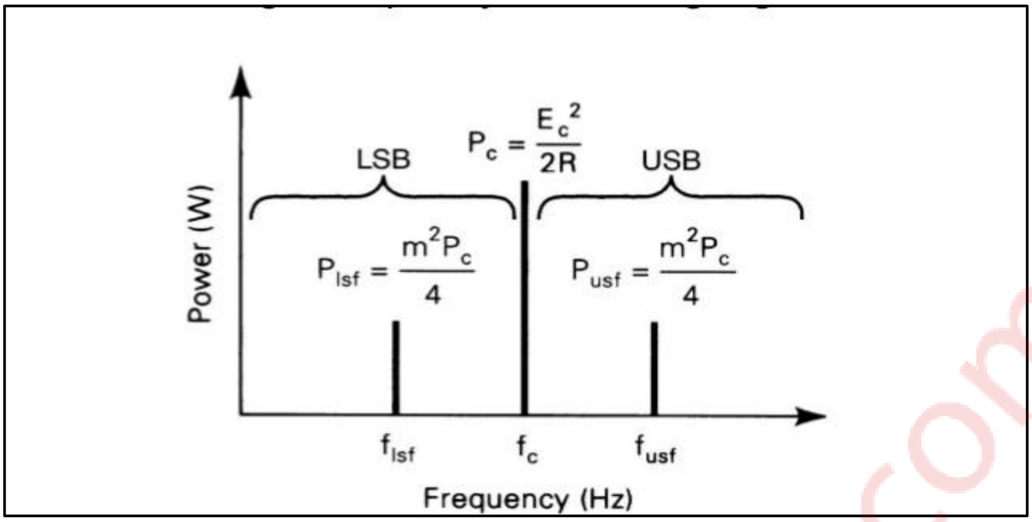
v0=2Qcos(ωc−ωm)t−2Qcos(ωc+ωm)t

Frequency spectrum band:



Frequency spectrum shows that BW of DSBSC is 2fm

Power spectrum band:



Q6A) What do you mean by multiplexing? Explain TDM. (10)

Solution:

Multiplexing:

Multiplexing (or muxing) is a way of sending multiple signals or streams of information over a communications link at the same time in the form of a single, complex signal; the receiver recovers the separate signals, a process called demultiplexing (or demuxing).

Networks use multiplexing for two reasons:

To make it possible for any network device to talk to any other network device without having to dedicate a connection for each pair. This requires shared media.

To make a scarce or expensive resource stretch further -- e.g., to send many signals down each cable or fiber strand running between major metropolitan areas, or across one satellite uplink.

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.

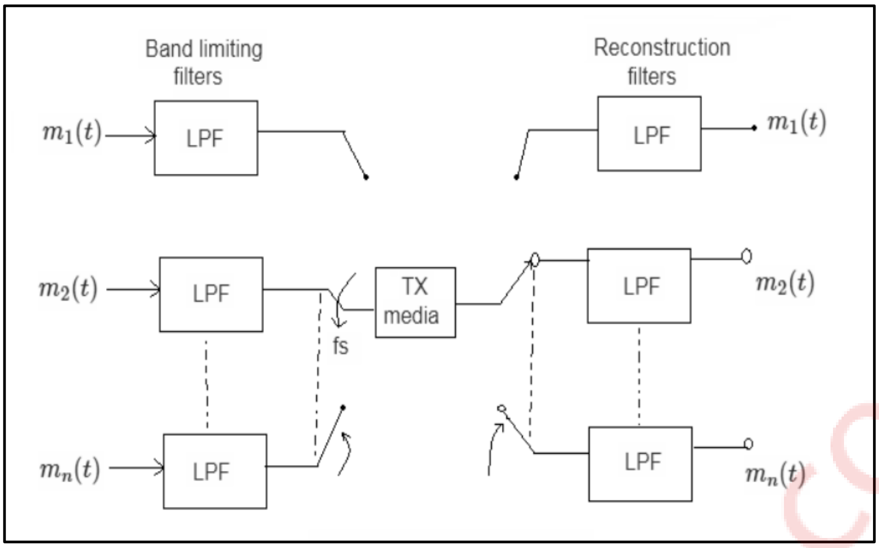


Fig1: Time Division Multiplexing

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.

Q6B) List down various parameters of op-amp with their practical values for IC 741 .Explain common mode gain and differential mode gain.

Solution: **(10)**

Parameter	**IC 741 values**
Differential input Resistance	2MΩ
Input capacitance	1-4 pF
Open Loop Voltage Gain	200,000
CMRR	90 dB
Output Voltage Swing	±13 to ±15 V
Output Resistance	75 Ω
Input Voltage Range	Input Voltage Range
Power Supply Rejection Ratio	30µV/V
Power Consumption	85 mW
Gain-Bandwidth Product	1MHz
Average Temperature Coefficient of Offset Parameters	$12pA/C^0$
Supply Current	2.8 mA
Slew Rate	15 µA

Common mode gain:

Common Mode gain is the response of a circuit where the input changes relative to some common point, such as ground. A single transistor amplifier is an example. An op-amp can also be an example, if one input is grounded.

common mode component, which can be written as **V_cm = (V_1+V_2)/2**, which is average of two signals at inverting and non-inverting component.

So common mode gain = **V_out/V_cm**

Differential mode gain:

Differential Mode gain is the response of a circuit where the input is measured across two pins, and the difference between those two pins controls the output, rather than the common mode value of those pins relative to ground. An op-amp in balanced or bridge mode is an example.

Gain of op-amp or any amplifier is V_{out}/V_{in}.

differential mode Gain = $V_{out}/(V_1 - V_2)$

where V_1 = is the voltage at non-inverting terminal and V_2 is the voltage at inverting terminal.

So differential mode component can be written as $V_d = V_1 - V_2$
