S.E. Sem. III [CMPN]

Electronic Circuits and Communication Fundamentals

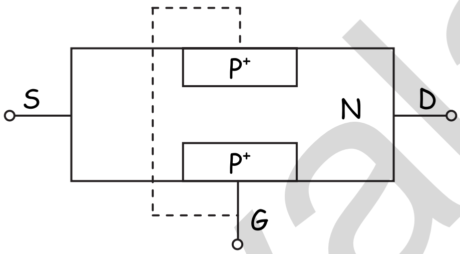
Time : 3 Hrs.] Prelim Question Paper Solution [Marks : 80

Q.1 Solve the following : [20]

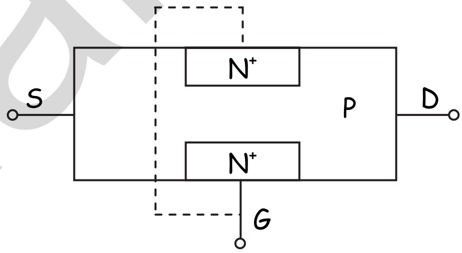
Q.1(a) Explain construction of FET [5]

(A) Consider a pure semiconductor made of silicon or germanium.Isolate certain region by applying a photo sensitive material. The remaining region is lightly doped to get an N–type semiconductor having high resistance. The mask is opened by subjecting to light. Then the corresponding region is heavily doped to get a P^+ material as shown. The two P^+ materials are internally connected to form the common terminal named as gate (G). At the two ends of semiconductor channel, Aluminium metal leads are inserted to form drain (D) and source (S) as shown.

(i) N–channel JFET



(ii) P–channel JFET



Q.1(b) Comment on the following ADC/DAC specifications : [5]

(i) Resolution (ii) Linearity (iii) Accuracy

(iv) Settling Time (v) Stability

(A) **(i) Resolution:** For ADC resolution is the same as that for the DAC.

resolution = 2n

Resolution is also defined as the ratio of a change in value of input voltage, vi needed to change the digital output by 1 LSB. If the full scale input voltage required to cause a digital output of all 1's is VIFS, then resolution can be given as

$$\text{resolution} = \frac{V_{IFS}}{2^n - 1}$$

(ii) Linearity: The relation between the digital input and analog output should be linear. However, practically it is not so due to the error in the values of resistors used for the resistive networks.

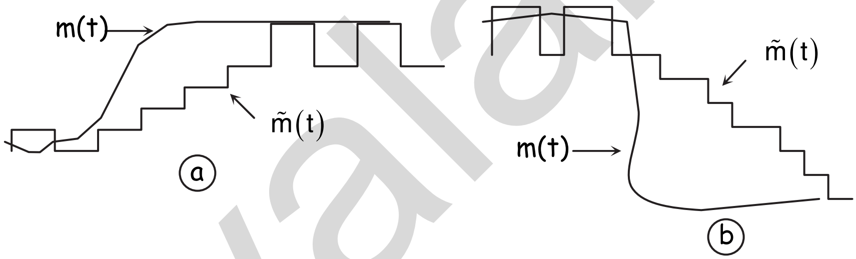
(iii) Accuracy: Accuracy indicates how close the analog output voltage it to its theoretical value. In short, it indicates the duration of actual output from the theoretical value.

(iv) Settling Time: Theoretically, the analog output voltage should change instantaneously in response to the change in its digital input. Practically, the analog output of a D to A converter does not change instantaneously. Due to the resistors and op-amp in the circuit, oscillators are observed at the output. The time required to settle the analog output within $\frac{1}{2}$ LSB of the final value, after the change in digital input is called as setting time. The setting time should be as short as possible.

(v) Stability: Ability of ADC/DAC to maintain unchanged output. Over the time, stability should be as high as possible.

Q.1(c) Explain slope over load error in delta modulation. **[5]**

(A) If m(t) changes too fast then $\tilde{m}(t)$ cannot follow m(t) as shown below.



In fig. (a) slope of m (t) is greater than slope of $\tilde{m}(t)$ hence $\tilde{m}(t)$ is enable to follow m(t). Similarly in (b) slope of m(t) is more negative than slope of $\tilde{m}(t)$. In both cases recovered waveform will be distorted. Then D.M. system is said to be slope overloaded.

Q.1(d) Discuss the factors that influence modulation index of an FM wave. **[5]**

(A) Modulation index is the ratio of maximum frequency deviation to the maximum modulating frequency.

$$Mf = \frac{\delta_{max.}}{f_{m(max.)}} = \frac{KV_m}{f_{m(max)}}$$

It is observed that Mf is directly proportional to V_m and inversely proportional to $f_{m(max)}$.

Q.2(a) Determine the V_{GSQ}, I_{DQ}, V_{DSQ} and A_V for a voltage divider circuit with V_{DD} = 18_V, R_D = 2.2 K, R_1= 2.1 M, R_2 = 330 K and R_S = 1.2 K, I_{DSS} = 10 mA, V_P = −8V and V_{DS} = 40 μs. **[10]**

(A) Calculate V_G:

As $I_G = 0$,

$$V_G = \frac{R_2}{R_1 + R_3} \times V_{DD} = \frac{330}{2100 + 330}$$

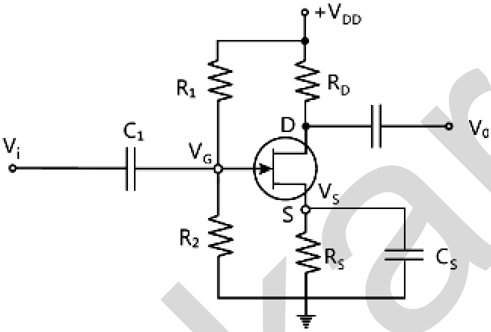


Fig.: Voltage divider biasing

Calculate V_{GSQ}:

$$V_{GSQ} = V_G - I_{DQ} R_S$$
$$= (2.44 - 1.2k) \times I_D$$
$$V_{GSQ} = -2.74\ V$$

Calculate I_D:

Using Schocktty's Equation we get,

$$I_D = I_{DSS}\left[1 - \frac{V_{GS}}{V_P}\right]^2$$

$$I_D = 10 \times 10^{-3}\left[1 - \frac{2.44 - 1.2 \times 10^{-3} I_D}{-8}\right]^2$$

$$I_D = 10 \times 10^{-3}\left[\frac{8 + \left(2.44 - 1.2 \times 10^{-3} I_D\right)}{8}\right]^2$$

$$64 I_D = 10 \times 10^{-3}\ (10.44 - 1.2 \times 10^{-3}\ I_D)^2$$

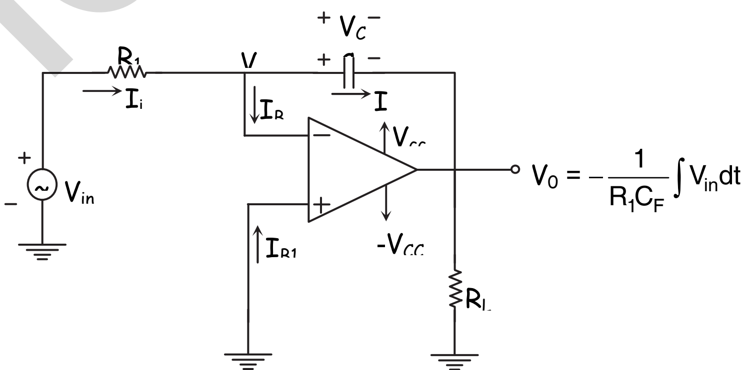
$$I_D = 4.3\ mA$$

Q.2(b) Explain how an op-amp can be used as - (i) Integrator **[10]**
(ii) Differentiator

(A) Op-amp can be used as -

(i) Integrator

The circuit diagram of Op-amp can be used as an Integrator is shown



The expression for output V_0 can be obtained by writing K_cL equation at node V_2.

$$I_{in} = I_{B2} + I_c$$

Since $I_{B2} = 0$

$$I_{in} = I_c$$

$$\frac{V_{in} - V_2}{R_1} = C_F \frac{dV_C}{dt} \qquad (\because\ V_C = V_2 - V_0)$$

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

However, $V_1 = V_2 = 0$, because A is very large.

$$\frac{V_{in}}{R_1} = C_F \frac{dV_0(-)}{dt}$$

Integrate both sides with respect to time t.

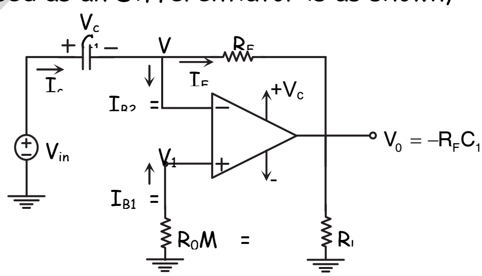
$$\int \frac{V_{in}}{R_1} dt = C_F \int \frac{d}{dt}(-V_0)$$

$$V_0 = -\frac{1}{R_1 C_F} \int V_{in} dt \qquad \ldots (A)$$

The equation (A) shows that output voltage is directly proportional to time constant $R_F C_F$.

(ii) Differentiator

The circuit diagram of Op-amp can be used as an Differentiator is as shown, Apply KCL at node V_2.



$$I_c = IB_2 + I_F \qquad \because\ I_{B2} = 0$$

$$I_c = I_F$$

$$C_1 \frac{dV_c}{dt} = \frac{V_2 - V_0}{R_F}$$

But $V_c = V_{in} - V_2$

$$C_1 \frac{d\left(V_{in} - V_2\right)}{dt} = \frac{V_2 - V_0}{R_F}$$

But $V_1 = V_2 = 0$, because A is very large (virtual ground).

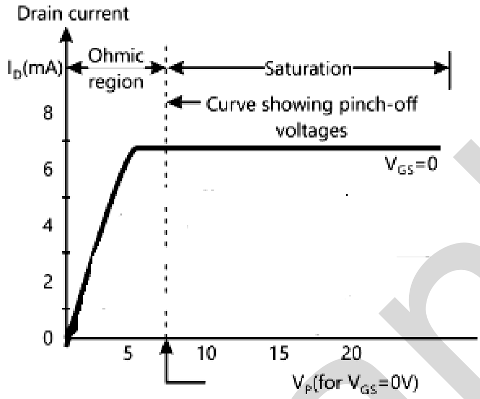
$$C_1 \frac{dV_{in}}{dt} = -\frac{V_0}{R_F}$$

$$V_0 = -R_F C_1 \frac{dV_{in}}{dt} \qquad \ldots (A)$$

Thus the output voltage is equal to the $R_F C_1$ times, the negative instantaneous rate of change of input voltage V_{in} with time. Thus minus sign indicates that a 180° phase shift of the output waveform V_0 with respect to the input signal.

Q.3(a) Sketch a typical drain characteristics for V_{GS} = 0 for an n-channel JEFET. Explain the shape of the characteristics, identify regions and indicate the important current and voltage levels. **[10]**

(A) Drain characteristics of a n-channel JFET are as shown. Drain characteristics is a plot of drain current I_d versus drain to source voltage V_{DS} at different values of gate to source voltage V_{GS}.



Drain characteristics :

The characteristics has been divided into three regions i.e, Cut-off, Saturation and Ohmic region.

Saturation region:

Saturation region is that portion of the characteristics where I_D remains fairly constant and does not change with changes in V_{DS}. This "saturation" is entirely different than the "saturation" in a transistor. In order to use the JFET as an amplifier it is used in the saturation region.

Ohmic region:

The drain current I_D varies with variation in the drain to source voltage V_{DS}. The JFET is therefore said to be operating as a voltage variable resistance in the ohmic region. The resistance offered by the JEFT decreases with decrease in the value of negative gate to source bias voltage i.e. negative V_{GS}. The FET resistance in the ohmic region is given by,

$$RDS = \frac{V_P}{I_{DSS}}$$

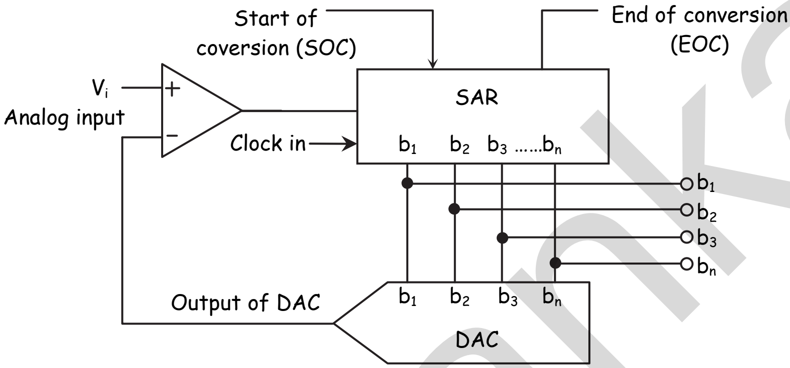
Where V_P = Pinch off voltage and

I_{DSS} = Maximum drain current

Pinch-off voltage:
The pinch-off voltage is the value of V_{DS}, at which the drain current reaches its constant saturation value. Any further increase in V_{DS} does not have any effect on the value of I_D. The pinch-off voltage is denoted by V_P.

Q.3(b) Explain any one technique used of conversion of analog signal to digital with ADC. **[5]**

(A) Successive Approximation ADC



In this technique, the basic idea is to adjust the DAC's input code such output is within $\pm \frac{1}{2}$ LSB of the analog input V_1 to be A/D converted. The code that achieves this represents the desired ADC output.

The successive approximation method uses very efficient code searching strategy called binary search. It completes searching process for n–bit conversion in just n clock periods.

The external clock input sets the internal timing parameters. The control signal start of conversion (SOC) initiates an A/D convers or process and end of conversion signal is activated when the conversion is completed.

The time for one analog to digital conversion must depend on both the clock's period T and number of bits n. It is given as

$T_C = T(nH)$

where, $T_C \rightarrow$ conversion time

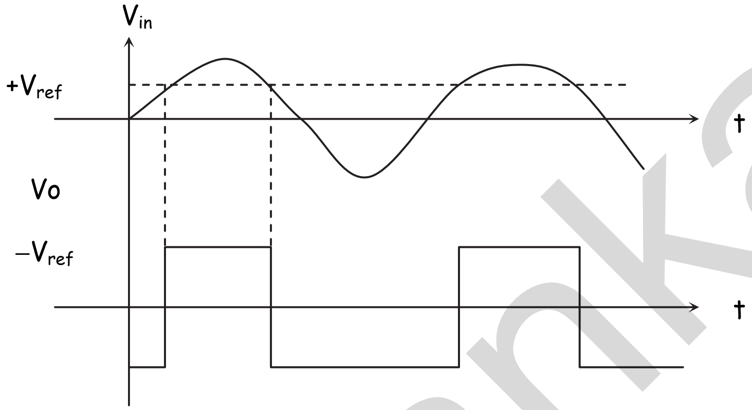
$T \rightarrow$ clock period

$n \rightarrow$ number of bits

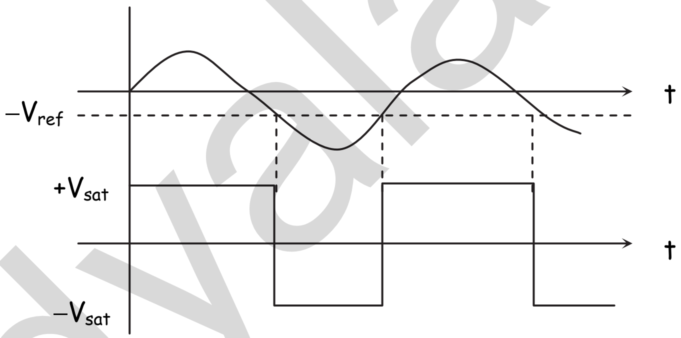
Q.3(c) Draw and explain op-amp inverting comparator. Draw input and output waveforms for V_{ref} > 0 and also for V_{ref} < 0. **[5]**

(A) When the signal is applied to non-inverting terminal and reference voltage is applied to inverting terminal it is referred as non-inverting comparator.

(1) Wave form with positive ref

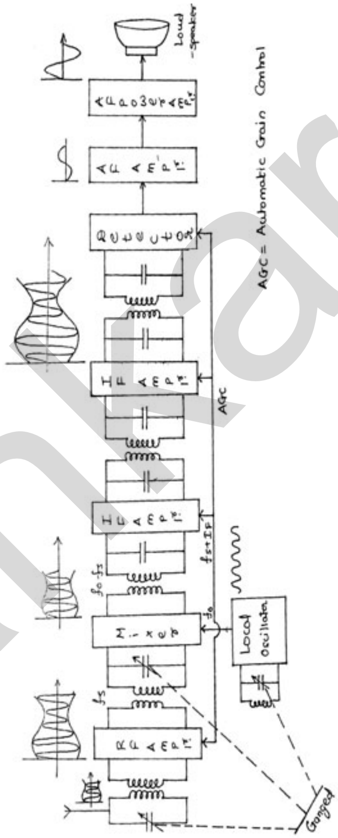


(2) Waveform with negative ref



Q.4(a) Draw and explain superheterodyne receive for amplitude modulation. What are the various characteristics of the receiver? **[10]**

(A) The features of heterodyne receiver is that all incoming radio frequency signals called intermediate frequency by using heterodyne process.



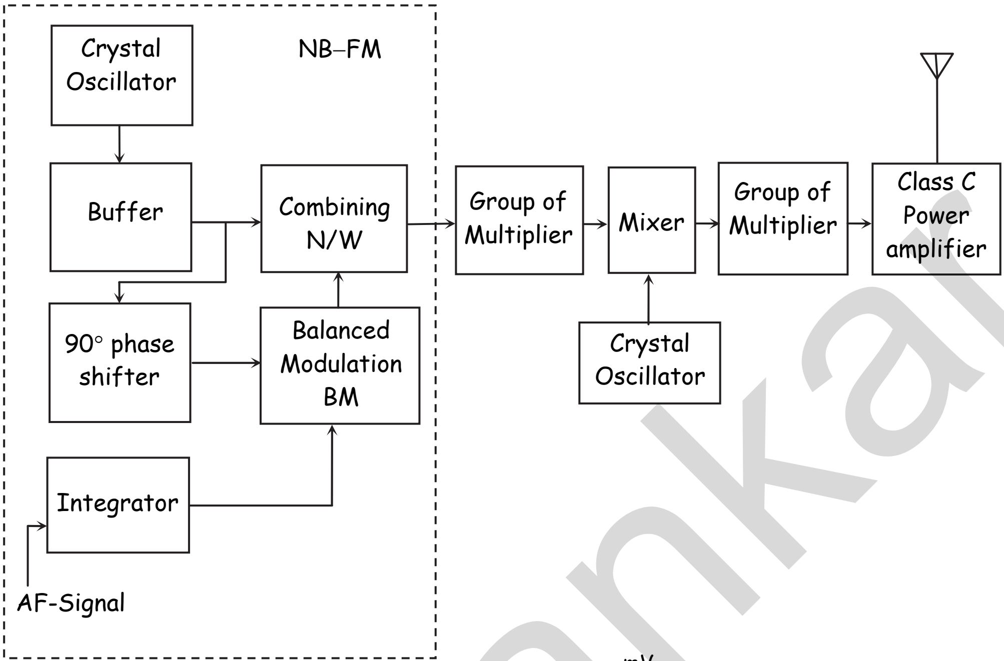
In this receiver, the received signal frequency is mixed with the local oscillator frequency f0. The local oscillator is an LC oscillator which produce sinusoidal oscillations of frequency f0. The frequency of local oscillator depends on the values of L & C of associated tuned circuit. The signal frequency fs and signal f0 gives to the mixer circuit. Mixer is a non-linear circuit. It will produce the output which contain different frequency component such as fs, f0, f0 $\pm$ fs, harmonics of input frequencies and the IC components. The inputs tank circuit of IF amplifier is tuned to f0 $-$ fs which is always 455 KHz and is known as Intermediate frequency If. To get If always = 455 KHz, the local oscillator should be tuned such that whatever may be input signal frequency fs, local oscillator frequency f_0 should be always 455 KHz, more than fs.

AM wave with 455 KHz is amplified to the desired level by 2 or 3. If amplifier and then it is fed to the detector which detects a taken out the AF information from the AM wave. This audio frequency signal is then amplified by voltage and power amplifier so that it can drive the loudspeaker.

Q.4(b) Explain Armstrong method to generate FM wave. **[10]**

(A) In this method high frequency stability can be obtained because we can use crystal oscillator.

Here narrowband (NB) FM is generated indirectly by using phase modulation and then converting NB – FM to a wide band (WB) FM by using mixer and multiplier circuit.



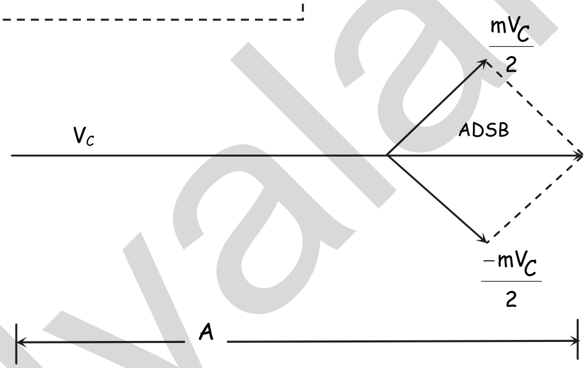


Fig. (a) : Vector diagram of AM wave

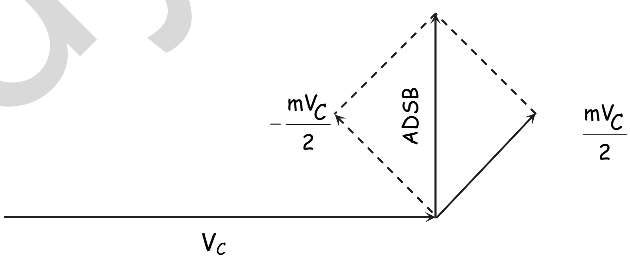


Fig. (b) : ADSB $\Rightarrow$ Amplitude of sideband

Vector diagram explains the principle of operation of Armstrong method. Fig. (a) shows the vector diagram of AM wave. The resultant of two sideband (ADSB) is in phase with carrier. Thus, there is an amplitude variation in the carrier with modulation. But there we require phase or frequency modulation.

To achieve this 90° phase shifted carrier is given to balance modulator. Thus 90° phase shift carrier is amplitude modulated at the output of balance modulator.

The carrier and 90° phase shifted sideband are added in the combining network. Therefore, at the output of combining network we get resultant carrier which is phase modulated w.r.t. unmodulated carrier as shown in figure (b).

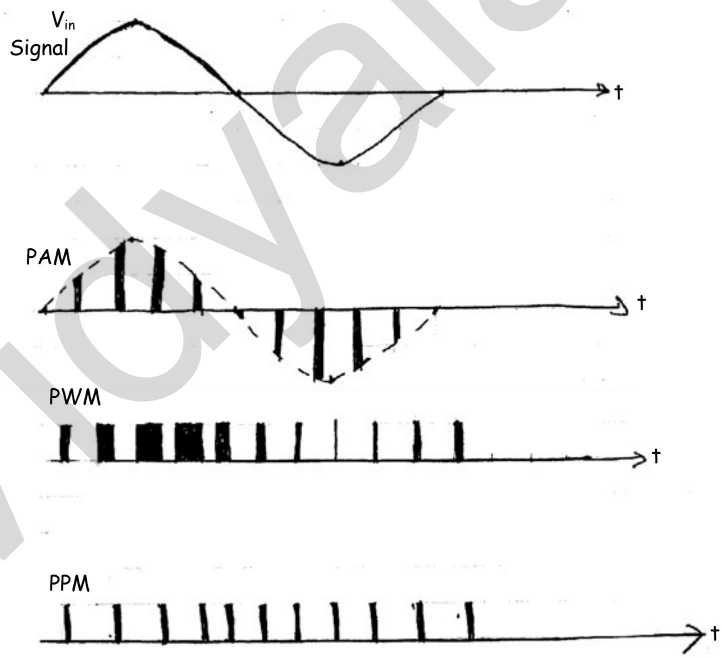
As the modulating signal increases, the resultant of sideband increases.

Therefore, the phase deviation θ will increase as $\theta = \tan{-1}\left(\frac{ADSB}{V_c}\right)$.

Thus, phase modulation is obtained which in a NB-FM. This NBFM converted into WB-FM by using multiplier and mixer circuit. The multiplier circuit will increase the frequency of carrier as well as frequency deviation. However, the mixer circuit only upshift or downshift the carrier signal without affecting the frequency deviation.

Q.5(a) Draw the PAM, PWM and PPM waveform in time domain assuming a sinusoidal modulating signal. Explain them in brief. [10]

(A)



Pulse Amplitude Modulation

- In PAM, the amplitude of the carrier pulses are varied in a accordance with the instantaneous amplitude of modulating signal.

- In this system signal is sampled at regular interval and each sample is made proportional to amplitude of signal at the instant of sampling.
- PAM signals can be detected by passing it through a low pass filter.

Disadvantages :
1) Since noise affect the amplitude of waveform and in AM information contain in amplitude variation, like, AM, PAM is less immunity to noise.
2) Due to change in amplitude of PAM pulses, the transmitted power is not constant.
3) Large bandwidth required to transmit PAM.

Pulse Width Modulation (PWM)
In this system, the width of carrier pulses are varied in accordance with the instantaneous amplitude of modulating signal. The amplitude of pulses are fixed.

Advantage of PWM :
1) Less effect of noise i.e. very good noise immunity.
2) Synchronization between the transmitter and receiver is not essential.

Disadvantages :
1) Due to the variable pulse width, the pulses have variable power contents so transmitter must be powerful enough to handle maximum width pulse.
2) Required large bandwidth.

Pulse Position Modulation (PPM)
The position of pulse are varied in accordance with the modulating signal for increasing AF signal, pulses shift to right and for decreasing AF signal shift to left.

Advantages :
1) It has good noise immunity.
2) Due to constant amplitude and width pulses the transmitted power always remain constant.

Disadvantages :
1) Large bandwidth required
2) Synchronization between transistor and receiver is required.

Q.5(b) An AM signal appears across a 50 Ω load has the following equation [10]

$V(t) = 12\,(1 + \sin 12.566 \times 10^3 t)\,\sin 18.85 \times 10^6\, t$ Volts.

(i) Sketch the envelope of this signal in time domain.

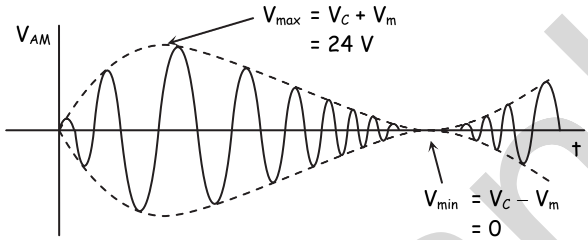
(ii) Calculate the modulation index, side band frequencies, total power and bandwidth.

(A) $V_C = 12$ V

$m = 1 = \dfrac{V_m}{V_C}$

$\therefore\ V_m = V_C$

(i)



(ii) USB $= f_C + f_m = 3002$ KHz

LSB $= f_C - f_m = 2998$ KHz

$$P_T = \frac{V_C^2}{2R}\left[1+\frac{m^2}{2}\right] = \frac{100}{R} \text{ watts}$$

Bandwidth $= 2\, f_m = 4$ KHz

Q.6(a) What do you understand by signal multiplexing? Explain TDM and FDM with suitable examples. [10]

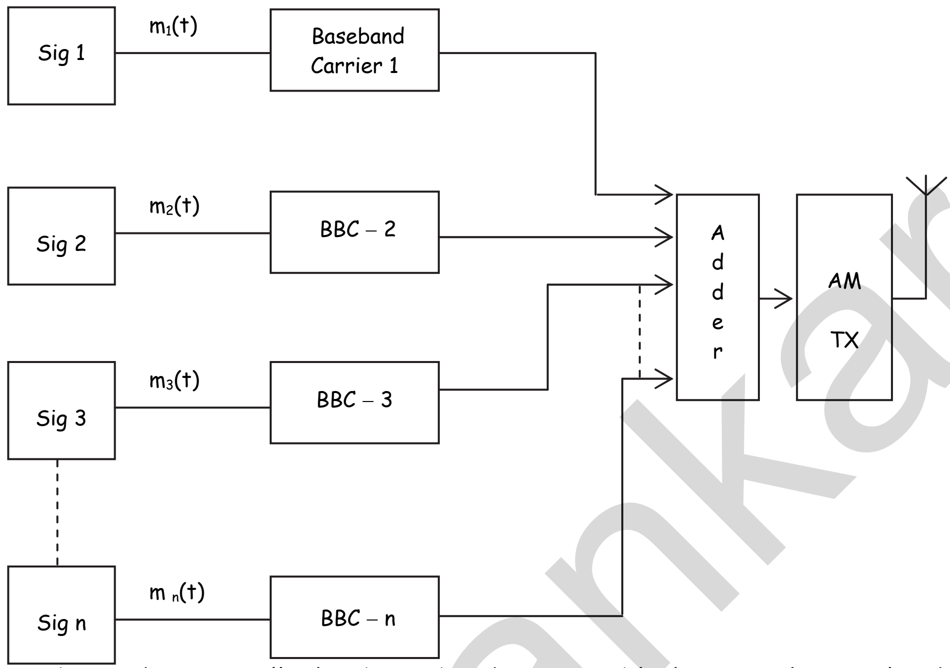
(A) Simultaneous transmission of multiple messages over a channel is called multiplexing. There are 2 type of multiplexing.

(i) Frequency Division Multiplexing.

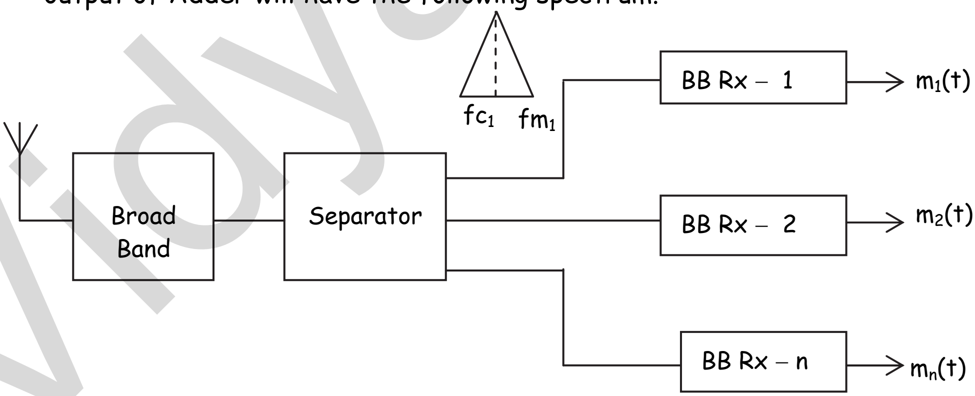
(ii) Time Division Multiplexing.

FDM uses analog modulation system, where as TDM uses pulse modulation system.

FDM consists of simultaneous transmission of message of different channels by shifting them in frequency domain. The bock diagram of FDM T_x & R_x is as shown below.



In above diagram, all the base band carrier blocks are the amplitude modulators. Here, all the signals have the same frequency spectrum. Therefore, all these signals are upshifted in different frequency slot using amplitude modulator. For every BBC Block, the carrier frequency is different. Hence the output of BBC – 1 will be $fc_1 \pm f_{m1}$. The output of BBC – 2 will be $fc_2 \pm fm2$. If all the modulating signals have the same B.W. then output of Adder will have the following spectrum.

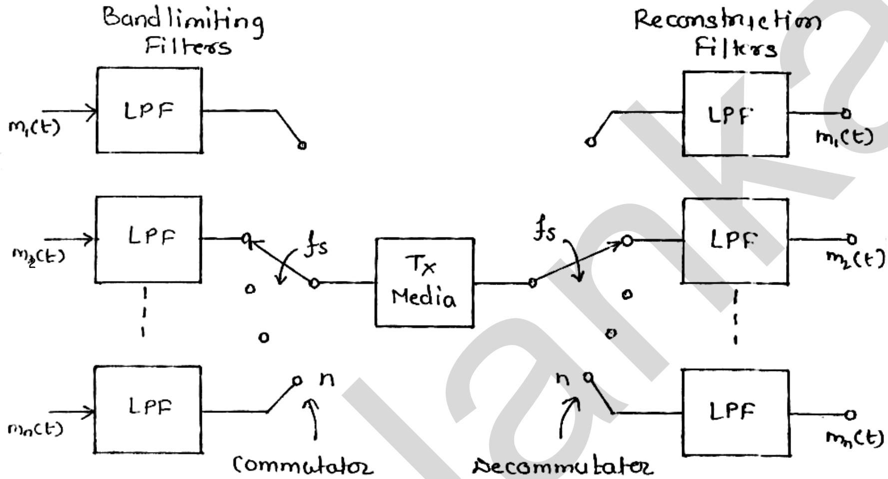


At the receiving station, the FDM signal is received by the broad band Rx. The separator circuit uses approximate mixers & filters, tuned at proper frequencies. The separator has n outputs & each o/p is the modulated signal centered around fc_1, fc_2 ––– fc_n respectively. These outputs are given to BBR_X. These are nothing but AM Rx_s. Hence the output of each block is corresponding modulating signal.

When a signal is sampled by narrow pulses, then the time interval between 2 pulses can be used to transmit the samples of other signals. In this techniques, signals are multiplexed in time domain & hence called as Time Division Multiplexing it is also used to transmit no. of signals on a single transmission media & hence act as alternative to FDM

PAM /TDM system

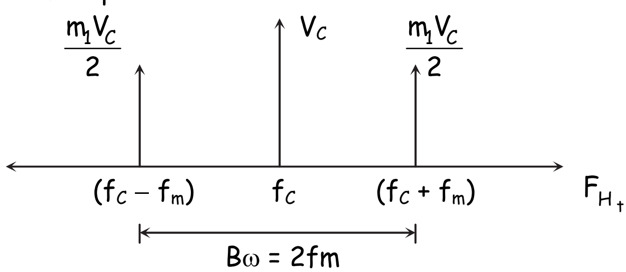
The block diagram of PAM/TDM is as shown in fig.



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 $m_1(t)$, $m_2(t)$ ——$m_n(t)$. Thus, at the output of commutator we get PAM waveform which contain the samples of message inputs which are periodically interplaced in time. These multiplexed message samples are transmitted over the communication channel.

Q.6(b) Draw the spectrum-of an amplitude modulated wave for more than one modulating signal and explain its components. **[5]**

(A) The spectrum of amplitude modulated cable is as shown :



$(f_c - f_m)$ = lower side band

$(f_{Ct}\ f_m)$ = upper side band

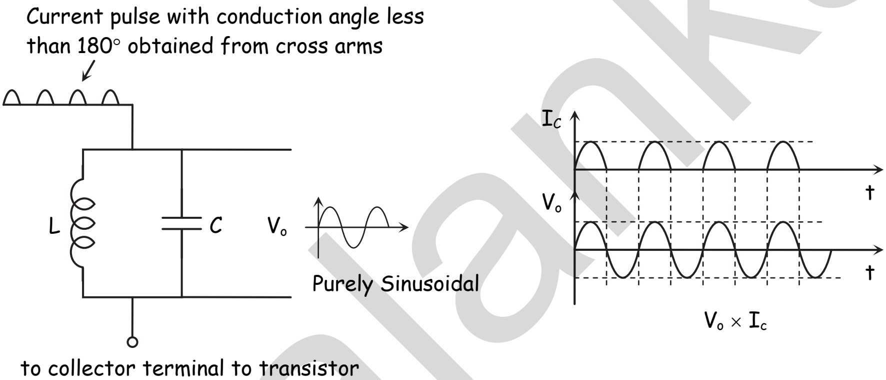
fe = frequency of carrier signal

$\frac{MV_C}{2}$ = Amplitude of side band

V_C = amplitude of carrier signal

Q.6(c) Explain Flywheel effect. **[5]**

(A) When current pulses obtained from class C amplifier having conduction angle less that 180° is applied to tank circuit at resonating frequency, $F_o = \frac{1}{2}\pi\sqrt{LC}$ then the output voltage oscillations obtained are purely sinusoidal, such that the amplitude of voltage oscillations is proportional to the amplitude of current pulse i.e. VaI



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