Important Instructions to examiners:

1) The answers should be examined by key words and not as word-to-word as given in the Model answer scheme.
2) The model answer and the answer written by candidate may vary but the examiner may try To assess the understanding level of the candidate.
3) The language errors such as grammatical, spelling errors should not be given more Importance (Not applicable for subject English and Communication Skills.
4) While assessing figures, examiner may give credit for principal components indicated in the Figure. The figures drawn by candidate and model answer may vary. The examiner may give credit for any Equivalent figure drawn.
5) Credits may be given step wise for numerical problems. In some cases, the assumed constant Values may vary and there may be some difference in the candidate's answers and model answer.
6) In case of some questions credit may be given by judgment on part of examiner of relevant answer based on candidate's understanding.
7) For programming language papers, credit may be given to any other program based on equivalent concept.

Q1. A) Attempt any SIX: 16M

i) State the audibility range.

Ans: Audibility range:

2M

- It is the range of frequencies to which human ear gives response.
- Theoretically it is from 16 Hz to 20KHz
- Human ear is most sensitive from 3 KHz to 4 KHz for all ages.
- Audibility range of ear decreases with age for high frequencies.

ii) Draw neat diagram of balance control circuit.

Ans:- (Diagram-2marks)

Circuit Diagram of Balance control:- **2M**

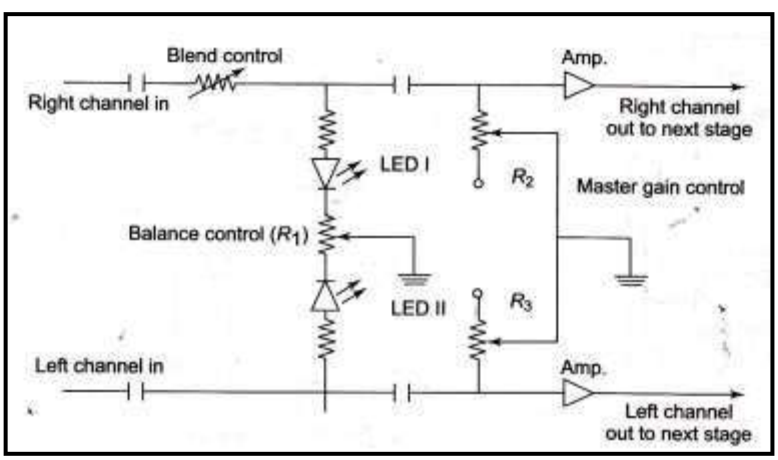


Fig. Balance control

iii) Compare AM with FM.

Ans: Comparison: Any four points ½ M each.

AM	FM
AM signal have low noise immunity	FM is higher noise immunity compared to AM.
AM modifies the amplitude of the carrier frequency	FM modifies the frequency of the carrier
AM is much more simpler compared to FM	FM is much more complex compared to AM
AM waves can be refracted in the Atmosphere resulting in greater range.	FM signals have less range
AM is more prone to signal distortion And degradation	FM signal doesn't degrade as easily as AM
AM usually broadcasts in mono which makes it sufficient for talk radio.	FM can transmit in stereo making it ideal for music
Bandwidth required for Am is Twice the highest modulating frequency (less as compared to FM)	Bandwidth is twice the sum of the modulating frequency and the frequency deviation. (20 times more as compared to AM.)

NOTE: Waveforms to be considered

iv) List any four advantages of CD's.

Ans: (½ marks for each point)

Advantages of CD: **2M**

- Signal to noise ratio is high
- Compact disc is immune to the surface contamination
- Dynamic range is high
- Channel separation is high
- Wow does not exist
- Flutter does not exist
- Total distortion is low
- Frequency response is excellent & covers complete audio range

v) Draw neat diagram showing variable density method and optical recording.

Ans: (each diagram -1Mark)

<u>Variable density method:-</u>

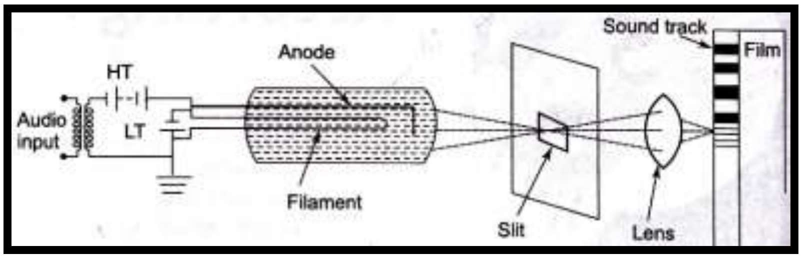


Fig. variable density method

<u>Optical recording:-</u>

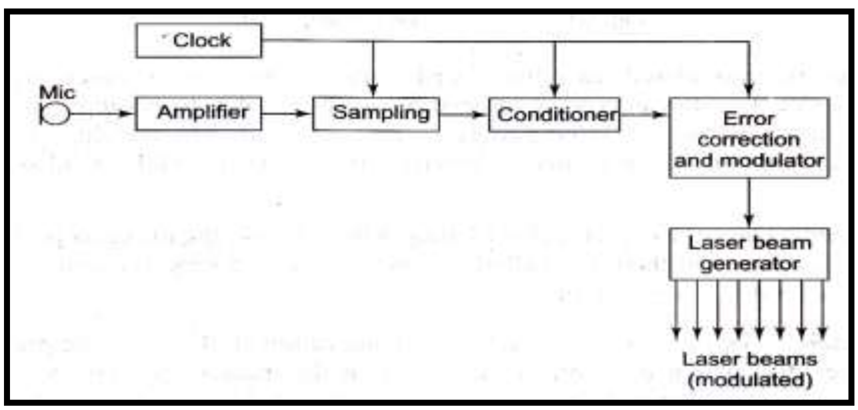


Fig. optical recording

vi) With neat sketch, define the term directivity of mike.

Ans: (Diagram-1Mark, Definition-1Mark)

<u>Directivity of mike</u>:- **1M**

Directivity of microphone is defined with the help of a polar diagram. The angle for half power points in a polar diagram shown above represents directivity of a microphone.

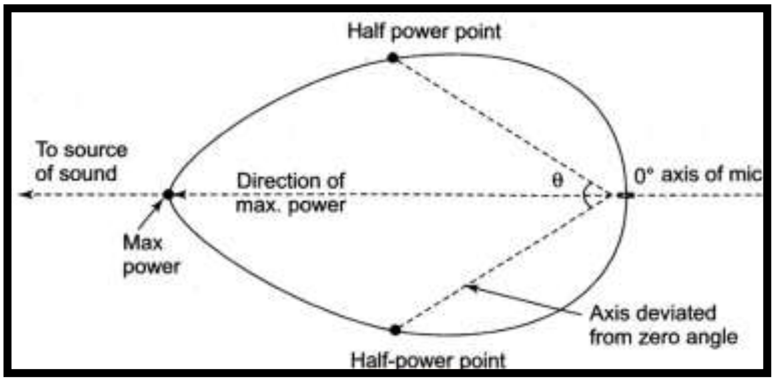
Mathematically it is defined as the ratio of actual output when placed in a direction of maximum response to the output which an omnidirectional microphone in the same direction would have given, keeping the intensity of sound constant.

Therefore, Directivity D = $\frac{E}{E0}$

Where E = actual output in the direction of maximum output

E_0 = output in that direction had the microphone been omnidirectional

Diagram:- **1M**



vii) What are the functions of BASS and treble controls in an amplifier?

Ans: (each function-1Mark)

Function of BASS:- **1M**

- Depth in the sound is given by bass i.e., low notes. The circuit which is used to boost or cut these low notes is called bass control.
- Some people like depth in the sound. So for them bass is to be boosted.
- If audio signal contains hum and external noises, which have low frequency, bass is to be cut.

Function of treble control: **1M**

- Treble means high notes. The circuit used to boost or cut high notes is called as Treble control.
- Some people like sharpness in the sound which is given by treble. Therefore for them, treble is to be boosted.
- If signal contains high frequency noise, then treble is to be cut.

viii) List any four requirements of Hi-Fi amplifier.(any four)

Ans: (each requirement-1/2 mark)

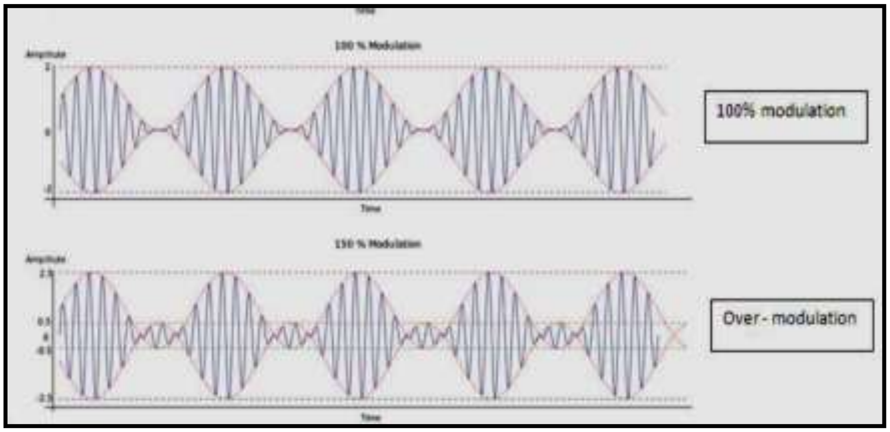
Requirements of Hi-Fi amplifier:-

- Signal to noise ratio should be better than dB.
- Frequency response should be flat within ± 1dB over the frequency range of 40Hz to 15Khz.
- Nonlinear distortion should not be more than 1%.
- The system should posses dynamic range of at least 80dB.
- Stereophonic effect should be provided.
- Environmental conditions should be such as to eliminate the external noise in the listening room and to give desired reverberation time.

B) Attempt any TWO: 8M

i) Draw neat sketch showing AM waveform for m>1 and m=1.

Ans: (for m>1:- 2Marks and for m=1:- 2Marks.)



ii) When the modulating frequency in FM system is 400Hz and modulating voltage is 2.4V, the modulation index is 60. Calculate maximum deviation?

Ans: (4marks for entire correct sum)

NOTE: Even if only formula or final ans is written 1M for each should be given)

Given, Modulating frequency $f_m = 400Hz$

Modulating voltage $V_m = 2.4$ V

Modulation index $m_f = 60$

$$\delta f_m = m_f \cdot f_m$$
$$= 60 * 400$$
$$= 24000 \text{ Hz}$$
$$= 24 \text{ KHz.}$$

Maximum deviation= 24KHz

iii) Draw neat sketch and explain reproduction of sound from films.

Ans: (Diagram-2Marks, Explanation-2marks)

Diagram:- **2M**

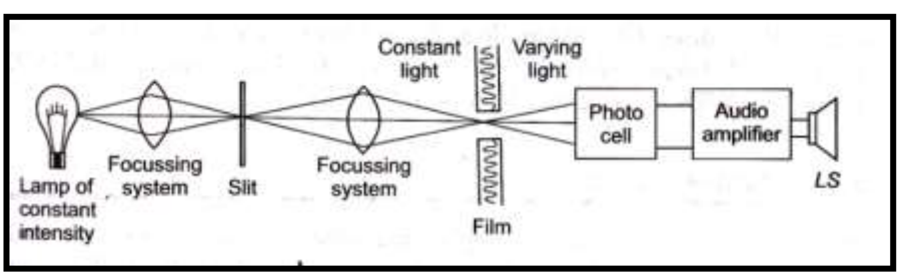


Fig. reproduction of sound

Explanation: **2M**

- A sharply focused narrow beam of light is made to fall on the soundtrack of film.
- As the film moves, light passing through bright and grey shaded portion in case of a variable-density record and through bright portions of variable area in case of a variable-area record,falls on a photocell which converts this light into electrical signals. In both types of recording (variable density as well as variable area), the quantity of light falling on the photocell will depend on the strength of the recorded audio signal.
- The output of the photocell will, therefore, be an audio voltage which can be amplified and fed to a loudspeaker which finally converts it into sound.
- The principle of reproduction is illustrated in above figure.

Q2. Attempt any FOUR: ` 16M

a) Draw neat circuit diagram of three way cross over network and explain its working.

Ans: (Circuit diagram-1Marks, Response curve 1M, Explanation-2Marks)

Diagram:- **2M**

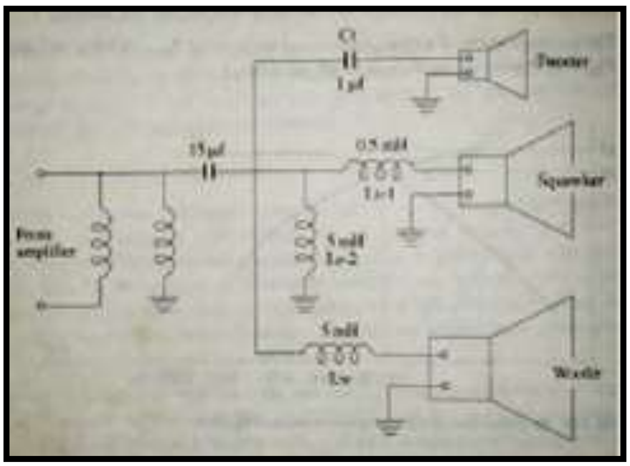


Fig. Three way cross over network

Response curve:

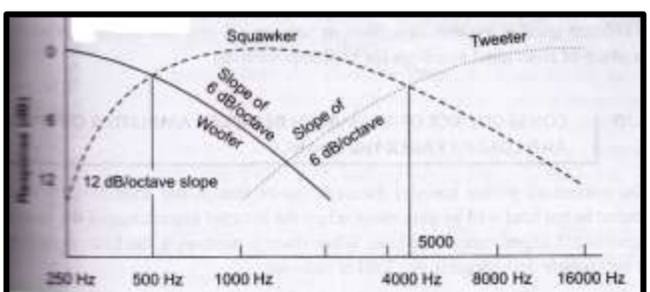


Fig. Response curves for a three-way speaker system

Explanation: **2M**

- When multi-way speaker system is used to get flat frequency response for the entire range of audio frequency it is essential to have a cross over network to divide the incoming signal into separate frequency ranges for each spectrum.
- In absence of cross over network, the speaker will suffer overheating and output will be Distorted when full power at frequencies outside the range in fed to them.
- As well as overall efficiency will be much reduced.
- Ct of 1μf in series with tweeter prevent 100 and mid frequencies reaching the tweeter. L_w of 5mH in series with woofer prevents high and mid frequencies reaching to woofer.
- L_{s1} and L_{s2} allows only mid frequency range to reach to squawker.

b) Draw neat sketch and explain step by step procedure of preparation of CD's on large scale.

Ans: (Diagram-2Marks, Explanation-2Marks)

Diagram:- **2M**

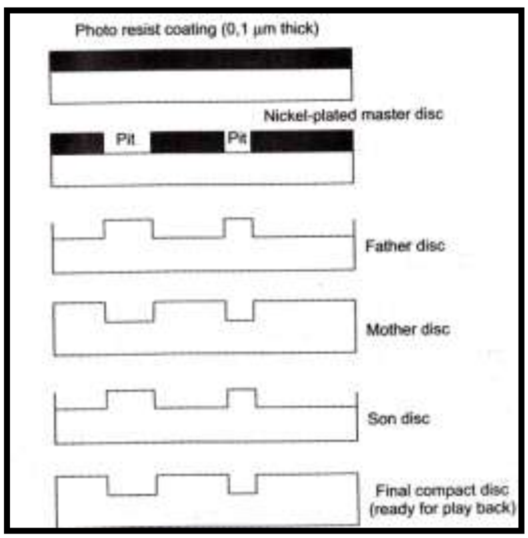


Fig. Preparation of CD

Explanation- **2M**

Master disc:

- The master disc, shown in fig , is the original disc on which audio signal is first recorded.
- The master disc is made of an optically ground glass disc. The glass is polished and is spotlessly clean. Its is coated with a photo- resist compound. The coating is 0.12 mm thick and is distributed uniformly. This is known as Resist master disc.
- When the modulated laser beam strikes the master disc, it reacts with the photo-resist. The disc is now developed by a process akin to photography.
- This results in a microscopic-sized pattern of pits and flats.
- The developed master disc is coated with silver to make it electrically conductive. Flats are also called lands.

Father Disc:

- The next step is nickel plating. After plating, the nickel is peeled off the master disc, and then it is called 'father disc'. It is a negative replica of the glass master disc shown in fig.

Mother disc:

- The father disc is again plated and removal of the plating produces a mother disc which is identical in form with the master disc.
- Generally, ten mother discs are obtained from a single master disc. Mothers are inverted and cannot be used for producing final discs.

Son disc or stamper:

- The mother discs are plated (the third plating in the process) and the plating when removed gives a son disc or stamper which is identical with the father disc.
- Several sons can be obtained from a single mother.
- A son disc is also called a negative nickel-plated stamper.
- The father, the mother and the son (stamper) discs are all produced in the same nickel bath.

Consumer disc or final compact disc:

- Consumer discs for playback are obtained by pressing on the stamper son disc. About 10000 discs can be moulded from one stamper.
- These discs are positive discs. A consumer disc is made of polycarbonate.
- A thin layer of aluminum is added to the disc to make it reflective.
- The consumer disc is protected by adding a transparent layer of lacquer. Recording is done from the center towards the edges.
- A hole is punctured in the centre of the disc. It is then packed in a plastic case.

c) Draw neat block diagram PA system and give function of mixer.

Ans: (Diagram-2Marks, Explanation-2Marks)

Diagram:- **2M**

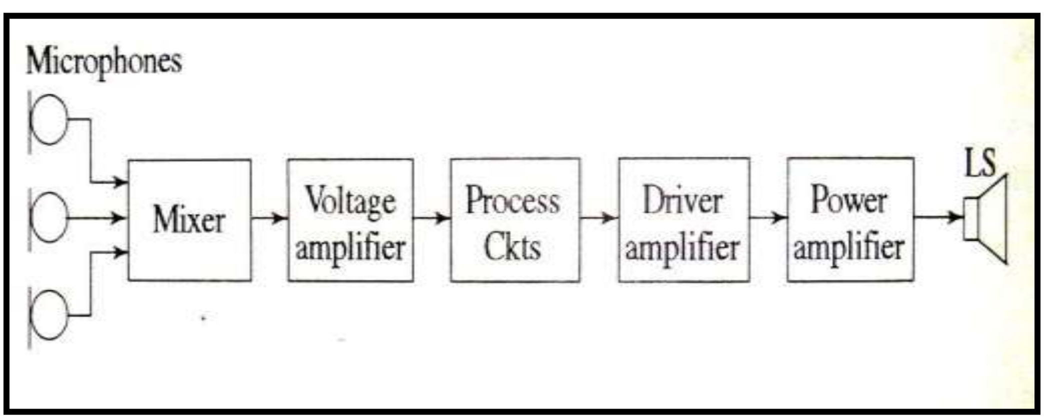


Fig. Block diagram PA system

Mixer- **2M**

The output of microphones is fed to mixer stage. The function of the mixer stage is to effectively isolate different channels from each other before feeding to main amplifier. It may be built in unit or a separate plug-in unit.

Three type of mixers:-

- Simplest – no amplifiers only gain controls (faders) and isolating services resistors.
- Little sophisticated- common amplifiers after isolating resistors.
- Most sophisticated – Has separate pre amplifier for separate channels then after gain control potentiometers and isolation resistor. There is a common amplifier followers
- Function of preamplifier & amplifiers to amplify weak signals.

d) Define and explain the terms: pre-emphasis and de-emphasis.

Ans: (each concept-2Mark)

Concept of pre-emphasis & de-emphasis with respect to audio recording:-

Emphasising low intensity sound before recording is called pre-emphasis the process of de-emphasising the playback circuit to bring originality is called equalization.

Pre-emphasis :- **2M**

Noise signal becomes more significant during quiet passage of music .therefore it is desirable to emphasize a low power notes before recording so that these are at much higher level than noise.

De-emphasis :- **2M**

At the receiver, it is essential that the reproduced sound possess the same proportions of intensities for low & high notes as were present in the original sound, De-emphasis will bring back the originality.

Application:

It is needed to improve signal to noise ratio to maintain high fidelity in the reproduced sound.

OR

Concept of pre-emphasis & de-emphasis with respect to modulation:

(diagram & response: 1 mark, explanation: 1 mark)

Pre-emphasis:-

Pre-emphasis refers to boosting the relative amplitudes of the modulating voltage for higher audio frequencies from 2 to approximately 15 KHz.

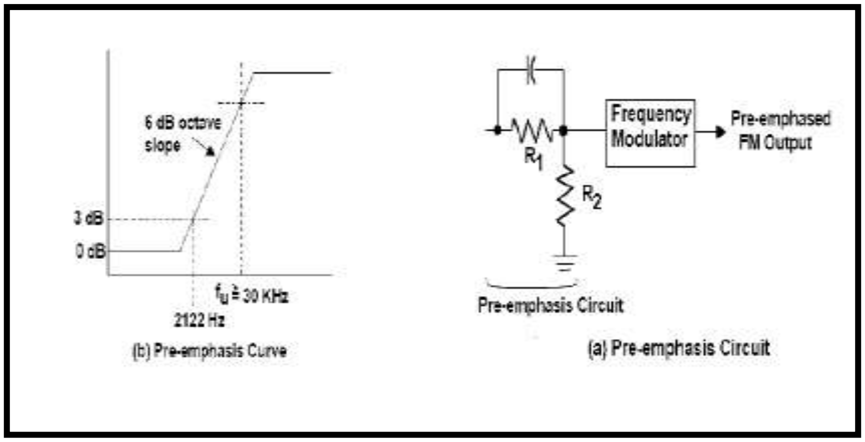


Fig. Pre-emphasis circuit

- At the transmitter, the modulating signal is passed through a simple network which amplifies the high frequency, components more than the low-frequency components.
- The simplest form of such a circuit is a simple high pass filter of the type shown in fig (a). Specification dictate a time constant of 75 microseconds (μs) where t = RC. Any combination of resistor and capacitor (or resistor and inductor) giving this time constant will be satisfactory.
- Such a circuit has a cutoff frequency fco of 2122 Hz. This means that frequencies higher than 2122 Hz will he linearly enhanced.
- The output amplitude increases with frequency at a rate of 6 dB per octave. The pre-emphasis curve is shown in Fig (b).
- This pre-emphasis circuit increases the energy content of the higher-frequency signals so that they will tend to become stronger than the high frequency noise components. This improves the signal to noise ratio and increases intelligibility and fidelity.
- The pre-emphasis circuit also has an upper break frequency fu where the signal enhancement flattens out.
- See Fig (b). This upper break frequency is computed with the expression.

$$fu = R1 + (R2/2\pi R1R1C)$$

- It is usually set at some very high value beyond the audio range. An fu of greater than 30KHz is typical.

De-emphasis:-

De-emphasis means attenuating those frequencies by the amount by which they are boosted.

- However pre-emphasis is done at the transmitter and the de-emphasis is done in the receiver.
- The purpose is to improve the signal-to-noise ratio for FM reception. A time constant of 75μs is specified in the RC or L/Z network for pre-emphasis and de-emphasis.

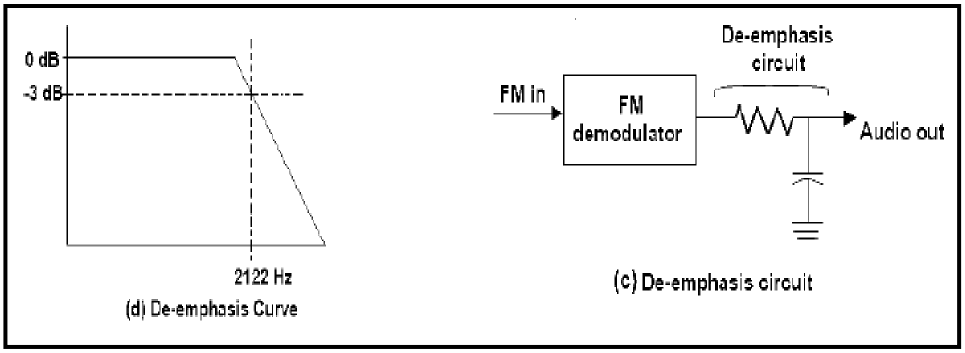


Fig. De-emphasis Circuit

- To return the frequency response to its normal level, a de-emphasis circuit is used at the receiver. This is a simple low-pass filter with a constant of 75 πs. See figure (c).
- It features a cutoff of 2122 Hz and causes signals above this frequency to be attenuated at the rate of 6bB per octave.
- The response curve is shown in Fig (d). As a result, the pre-emphasis at the transmitter is exactly offset by the de-emphasis circuit in the receiver, providing a normal frequency response.
- The combined effect of pre-emphasis and de-emphasis is to increase the high-frequency components during transmission so that they will be stronger and not masked by.

e) Explain the need of modulation.

Ans: <u>Need of modulation</u> – (4marks)

- Suppose we want to transmit, an electric signal having frequency 3 kHz (voice frequency) over an antenna, When it is connected to antenna, it detaches from antenna and travel into space in definite direction. However, there is one difficulty in this process. It is necessary to keep antenna height equal to wavelength (l) of electrical signal connected to it. Now wavelength (l) will be –

$$\text{Height of antenna}(\lambda) = \frac{\text{Velocity of light (C)}}{\text{Frequency}(f)\text{ of the signal}} = \frac{3\times10^8}{3\times10^3} = 10^5 m$$

- It means that we need height of antenna equal to 100 km! This is *practically impossible*! However we can reduce its height by l/2, l/4, l/8 or up to l/16. But even if we reduce it to l/16, it becomes 6.2km, which is still impossible!

- Therefore, we cannot transmit low frequency signals directly. As per equation (1), if we increase frequency of electrical signal then l will reduce and hence the height.

- Hence, there is only one solution on this problem and that is process of modulation. In modulation very high frequency, carrier wave is taken. It is modulated (in either AM or FM style) by modulating signal, which we want to transmit actually. After modulation, low frequency ***RIDES*** over carrier wave. This modulated carrier wave is connected to antenna for transmission. Now suppose we want to transmit 3 kHz signal, with 300 MHz carrier wave. Then actually, 300 MHz signal is transmitted. For this height of antenna will be –

$$\text{Height of antenna}(\lambda) = \frac{\text{Velocity of light (C)}}{\text{Frequency}(f)\text{ of the signal}} = \frac{3\times10^{8}}{300\times10^{6}} = 1m$$

- This height is practically easily possible.

f) Explain generation of FM wave using varactor diode modulator.

Ans: (Diagram-2marks, explanation-2Marks)

<u>Diagram:</u>- **2M**

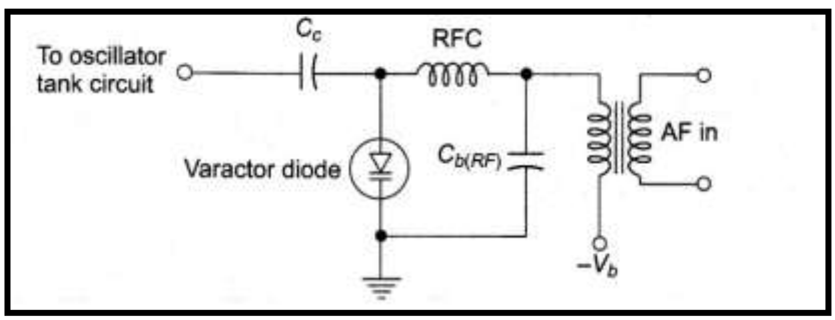


Fig. Generation of FM wave using varactor diode modulator

<u>Explanation</u>:- **2M**

- A varactor diode is a semiconductor diode whose junction capacitance varies linearly with the applied voltage when the diode is reverse-biased.
- It may also be used to produce frequency modulation. Varactor diodes are certainly employed frequently, together with a reactance modulator, to provide automatic frequency correction for an FM transmitter.
- The circuit of fig shows such a modulator. It is seen that the diode has been back- biased to provide the junction capacitance effect, and since this bias is varied by the modulating voltage which is in series with it, the junction capacitance will also vary, causing the oscillator frequency to change accordingly.

Q.3 Attempt any FOUR: 12M

a) A broadcast AM amplifiers radiates 50KW of carrier power. What will be the radiated power at 85% modulation?

Ans: (4Marks)

(i) $P_t = P_c\left(1 + \frac{m_a^2}{2}\right)$ — 1 mark.

where P_t = transmitted total power
P_c = Carrier Power.
m_a = modulation Index.

Given data

$P_c = 50\ KW = 50 \times 10^3\ W$
$m_a = 85\% = 0.85$ } — 1M.

$\therefore P_t = 50 \times 10^3 \left[1 + \frac{(0.85)^2}{2}\right]$ — 1M

$= 68.06 \times 10^3\ W$

$P_t = 68.06\ KW$ — 1M.

Ans:- The radiated power at 85% modulation is 68.06 KW.

b) Derive mathematical expression for power relation in AM.

Ans: (4Marks)

$$P_{total} = \frac{(V_{carrier})^2}{R} + \frac{(V_{LSB})^2}{R} + \frac{(V_{USB})^2}{R}$$ 01M

The above expression is represented in terms of Peak values, but for the power rms values are considered. So

$$V_{c(rms)} = \frac{V_c}{\sqrt{2}} \text{ using concept } V_{rms} = \frac{V_{max}}{\sqrt{2}}$$

$$V_{LSB(rms)} = +\frac{m_a V_c}{2\sqrt{2}} \quad \text{where } V_{LSB} = \frac{m_a V_c}{2}$$ ... Derived in the sideband expressions

$$V_{USB(rms)} = -\frac{m_a V_c}{2\sqrt{2}} \quad \text{where } V_{USB} = -\frac{m_a V_c}{2}$$ ... Derived in the side band expressions

$$P_c = \frac{(V_{carrier})^2}{R} = \frac{(V_c/\sqrt{2})^2}{R} = \frac{V_c^2}{2R}$$ 01M

$$P_{total} = \left(\frac{V_c}{\sqrt{2}}\right)^2 \frac{1}{R} + \left(\frac{m_a V_c}{2\sqrt{2}}\right)^2 \frac{1}{R} + \left(-\frac{m_a V_c}{2\sqrt{2}}\right)^2 \frac{1}{R}$$ 01M

$$P_t = \frac{V_c^2}{2R} + \frac{m_a^2 V_c^2}{8R} + \frac{m_a^2 V_c^2}{8R}$$

$$P_t = \frac{V_c^2}{2R}\left[1 + \frac{m_a^2}{4} + \frac{m_a^2}{4}\right]$$

$$P_t = P_c + \frac{m_a^2}{4} P_c + \frac{m_a^2}{4} P_c$$

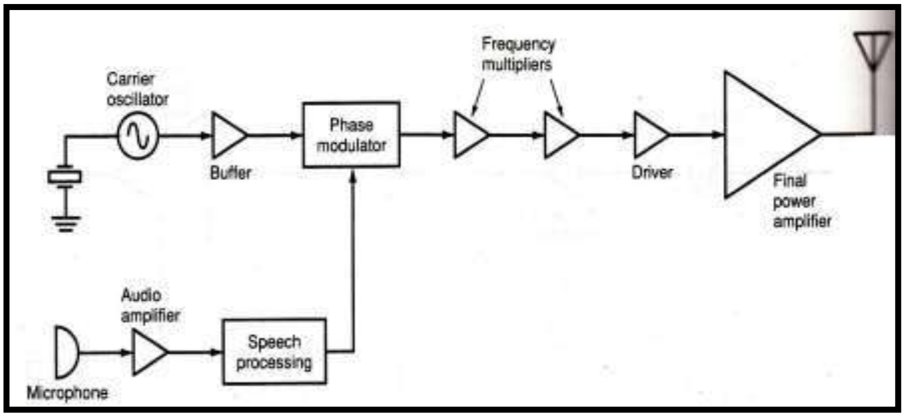
$$P_t = P_c\left(1 + \frac{2m_a^2}{4}\right)$$

$$\boxed{P_t = P_c\left(1 + \frac{m_a^2}{2}\right)}$$ 01M

c) Draw neat diagram of Armstrong frequency modulation technique.

Ans: (4 marks for any correct diagram given below)

Diagram: **4M**



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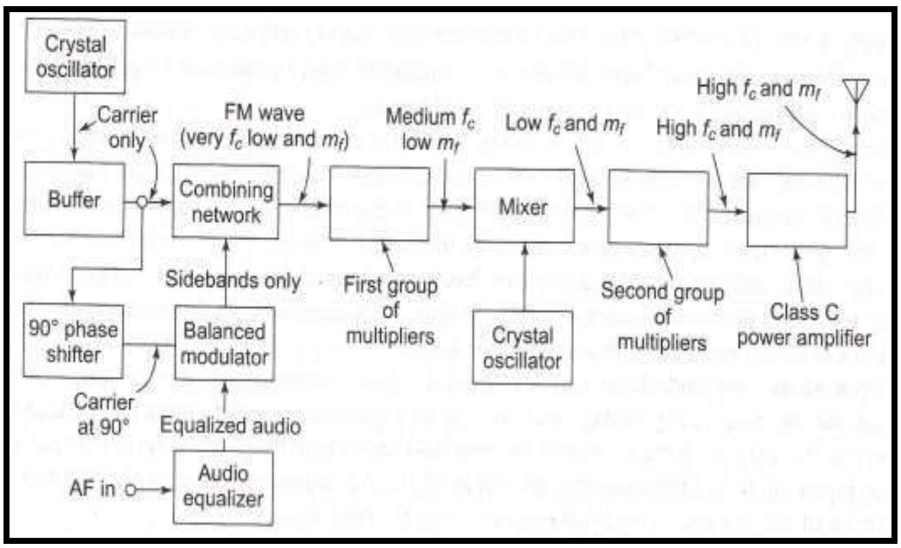


Fig. Armstrong frequency

d) Define amplitude modulation and modulation index.

Ans: (Each definition 2 marks)

Definition of AM: **2M**

Amplitude modulation is a process of varying amplitude of modulating signal in accordance with instantaneous values of carrier signal keeping phase and frequency constant.

Modulation index: **2M**

It is defined as ratio of modulating signal voltage to carrier signal voltage.

Mathematically,

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

e) Explain Dolby-A system of noise reduction.

Ans: (Diagram-2 Marks, Explanation-2 Marks)

Diagram: **2M**

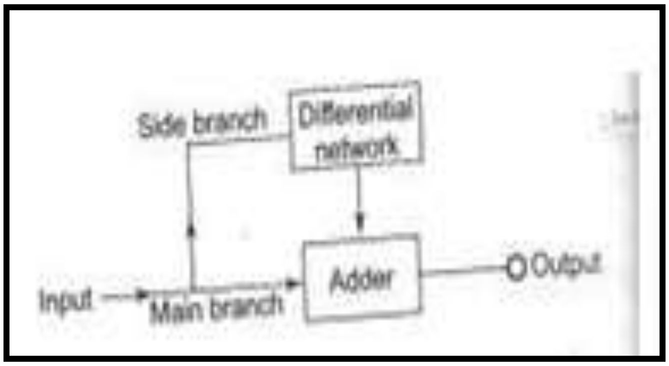


Fig. Coding of signal in Dolby method

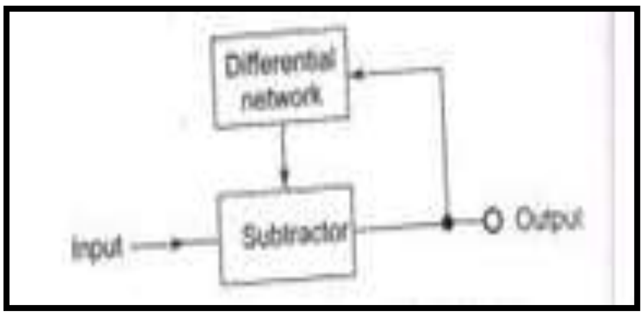


Fig. Decoding of Dolby signal

Explanation: **2M**

- Boosting is done in 4 bands:
  1. Below 80Hz
  2. 80Hz to 2999Hz
  3. 3000 Hz and above
  4. 9000 Hz and above
- Each band is processed separately by using low-pass, band-pass and high-pass filters and limiters.
- The 16 Hz to 80Hz signal goes to a low pass filter which causes improvement in signal to noise ratio with respect to hum and rumble.
- The 80Hz to 2999Hz signal goes to a band pass filter which deals with the mid band noise.
- Most of the sound energy for music is concentrated in this band. The 3000Hz and 9000Hz high pass filters improve signal to noise ratio with respect to hiss and modulation noise.
- The output of the four separate units is added. All this is done in a side branch, and this branch is known as the differential network.
- The output of the differential network goes to the adder of the main branch as shown in fig.(a)
- In playback, the differential network separates out the boosted signals in the side branch and subtracts them from the input signal as shown in fig. (b)

f) Draw neat diagram of FM transmitter and explain any two block of it.

Ans: (Diagram-2 Marks, Explanation of any two blocks 1 Mark each)

Diagram: **2M**

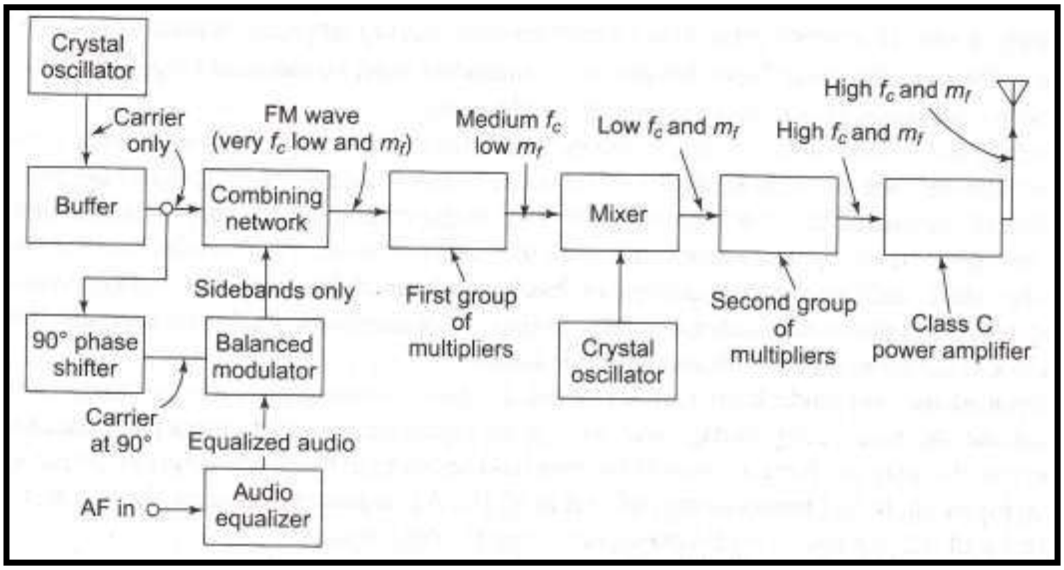
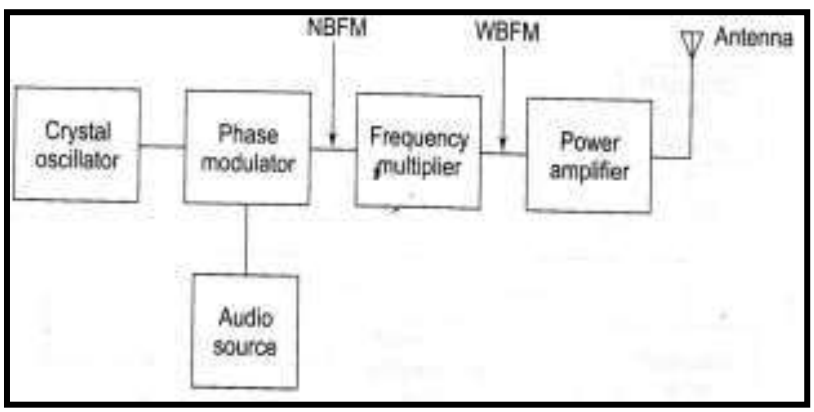


Fig. FM transmitter

OR



Explanation: **2M**

- The crystal oscillator generates the carrier at low frequency typically at 1 MHz. This is applied to the combining network and a 90 degree phase shifter.
- The modulating signal is passed through an audio equalizer to boost the low modulating frequencies.
- The modulating signal is then applied to a balance modulator.
- The balance modulator produces two sidebands such that their resultant is 90 degree phase shifted with respect to the un-modulated carrier.
- The un-modulated carrier and 90 degree shifted sidebands are added in the combining network .The output of combining network is equivalent to FM wave. This FM wave has low carrier frequency Fc and low value of the modulating index mf .

- The carrier frequency and the modulation index are then raised by passing the FM wave through the first group of multipliers. The carrier frequency is then raised by using a mixer and then the Fc and mf both are raised to required high values using the second group of multipliers.
- The FM signal with high Fc and high mf is then passed through a class C power amplifier to raise the power level of the FM signal.

Q.4 Attempt any FOUR 16M

(a) Compare AM with FM (any four points).

Ans: Comparison: (1 Mark for each point)

AM	FM
1. AM signal have low noise immunity	1. FM is higher noise immunity compared to AM.
2. AM modifies the amplitude of the carrier frequency	2.FM modifies the frequency of the carrier
3.AM is much more simpler compared to FM	3.FM is much more complex compared to AM
4. ground wave & sky wave propagation is used therefore large area is covered than FM	4. space wave is used for propagation do radius of transmission is limited to line of sight.
5.AM is more prone to signal distortion And degradation	5.FM signal doesn't degrade as easily as AM
6. applications: Radio & TV broadcasting,	6. application : Radio & TV broadcasting, police wireless, point to point communication
7. Bandwidth Required for AM is Twice the highest modulating frequency (less as compared to FM)	7. Bandwidth is Twice the sum of the modulating frequency and the frequency deviation. (20 times More as compared to AM)
8. Carrier power & one sideband power are useless.	8. All the transmitted power are useful.

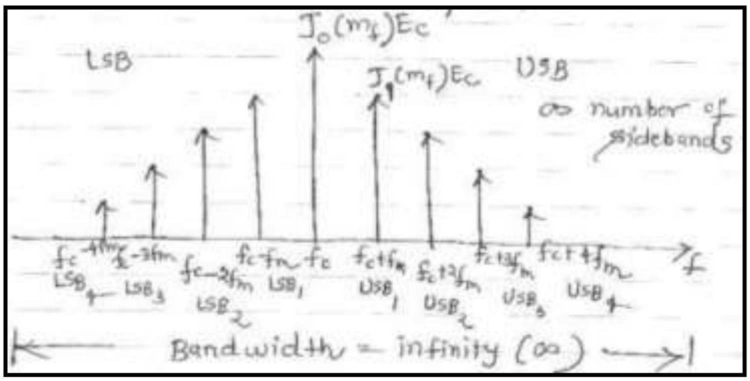
NOTE: Waveforms to be considered

(b) Draw time domain spectrum and frequency domain spectrum.

Ans: (2 marks time domain spectrum, 2 marks for frequency domain)

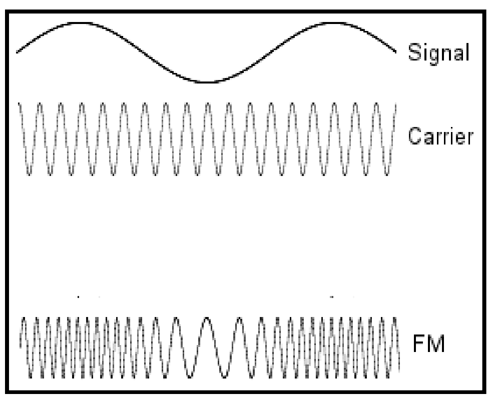
(Marks should be given if the students draw frequency spectrum for AM or FM)

Frequency domain representation of FM: **1M**

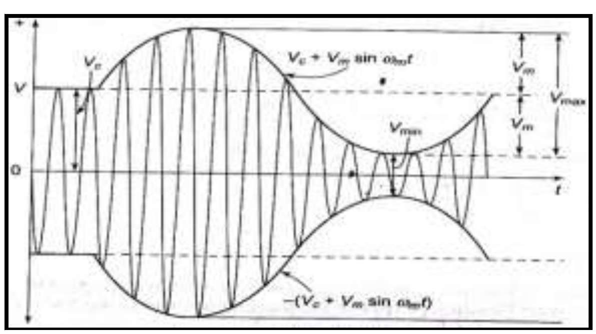


OR

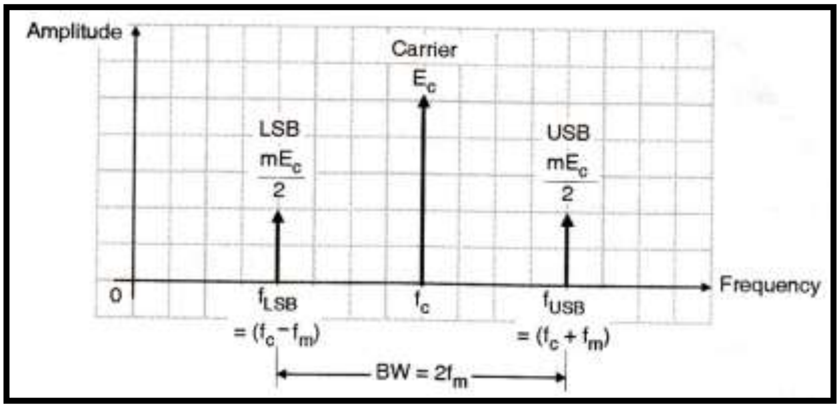
Time domain representation of FM: **1M**



Time domain representation of AM: **1M**



Frequency domain representation of FM: **1M**



c) **Draw neat sketch and explain installation of PA system in an auditorium.**

Ans: (Diagram-2 Marks, Explanation-2 Marks)

Diagram: **2M**

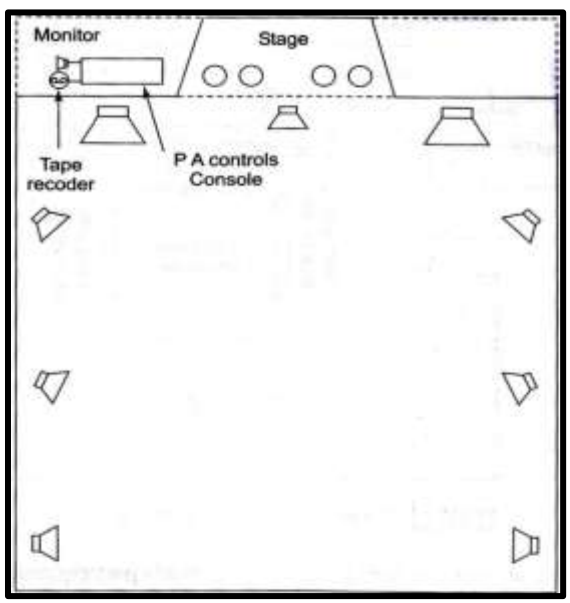


Fig. PA system in an auditorium

Explanation: **2M**

- An auditorium may be used for wide range of activities like public meeting, conferences, cultural programme etc. Hence the loudspeakers system should have a wide dynamic range and good frequency coverage from 20 Hz to 16 KHz.
- Columns of loudspeakers having good bass and treble response should be mounted facing towards the front on the either side of the stage.
- If the hall is wide, a small column may also be mounted in the center of the front line. Another pair of small columns slightly inclined may be placed at about one third and two third ways down the hall from the front. The fourth pair placed last need not be inclined.

- A separate versatile mixer unit is desirable. It may have tape and disc input and several microphone inputs
- The amplifiers should be 50 to 100 watt
- Microphone should be of moving coil type.
- If the hall has a line of pillars small loudspeakers may be mounted on the pillars facing the audience.
- The back wall should be furnished with good sound absorbing material.
- The height of the side speakers should be such that their axes point towards ear of the audience and not towards the flooring or ceiling.

d) Draw neat sketch showing constructional details of dynamic microphone and list its four characteristics.

Ans: (2 Marks for diagram, any four characteristics-1/2 Marks each)

Constructional Diagram: **2M**

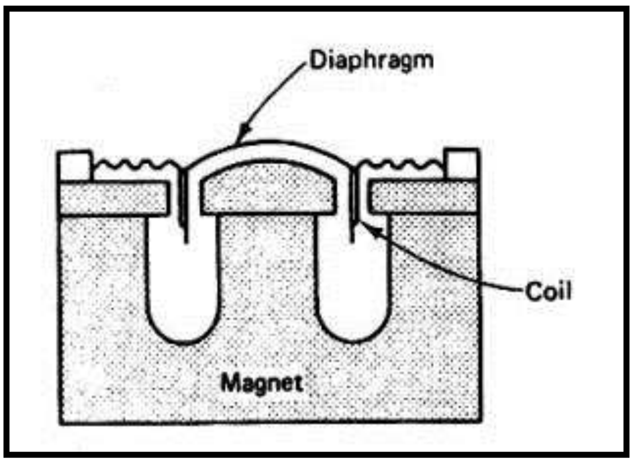
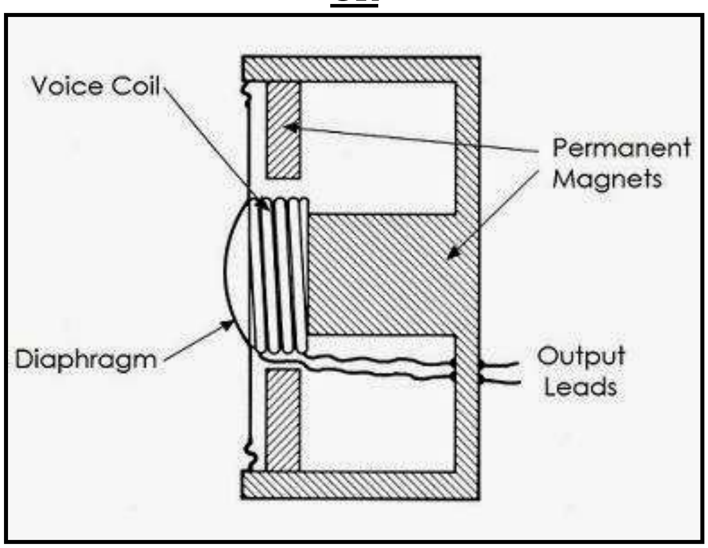


Fig. Dynamic microphone

OR



Characteristics: **2M**

- Sensitivity: 30 microvolts
- Signal to noise ratio-30dB
- Frequency response: 60Hz to 8000 Hz
- Distortion: Less than 5%
- Directivity: omnidirectional
- Output impedance: 25 ohms

e) Draw circuit diagram of class B push pull amplifier and explain its working.

Ans: (Diagram-2 Marks, Working-2 Marks)

Circuit Diagram: **2M**

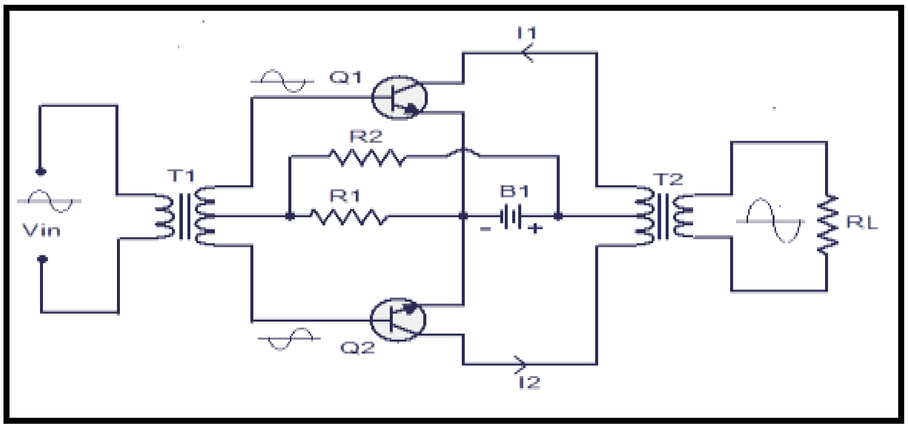


Fig. class B push pull amplifier

Working: **2M**

- Class B amplifiers are biased in a way that they give output for half the cycle only. Their efficiency is high (78%) but they produce severe distortion as only half the cycle is reproduced.
- It used two similar NPN transistors. Input signals from the transformer T1 are fed to the base of two transistors Q1 and Q2 and are opposite in phase to each other.
- Direct current to the two collectors flows through the primary of the transformer T2 in opposite direction as shown by arrows.
- This saves the transformer from becoming saturated by dc, and hence eliminates non-linear distortion which would have been produced due to saturation of the core.
- The audio current will flow through the output transformer in the same direction from both collectors, due to the two outputs being in the opposite phase with respect to each other.
- When the collector of Q1 is a positive going audio, that Q2 will be a negative going audio and hence the audio currents in the primary will add up.
- Thus the audio will develop fully without any saturation of the core by dc.

f) Explain principle of magnetic recording.

Ans: (Diagram-2 Marks, Principle-2 Marks)

Diagram: **2M**

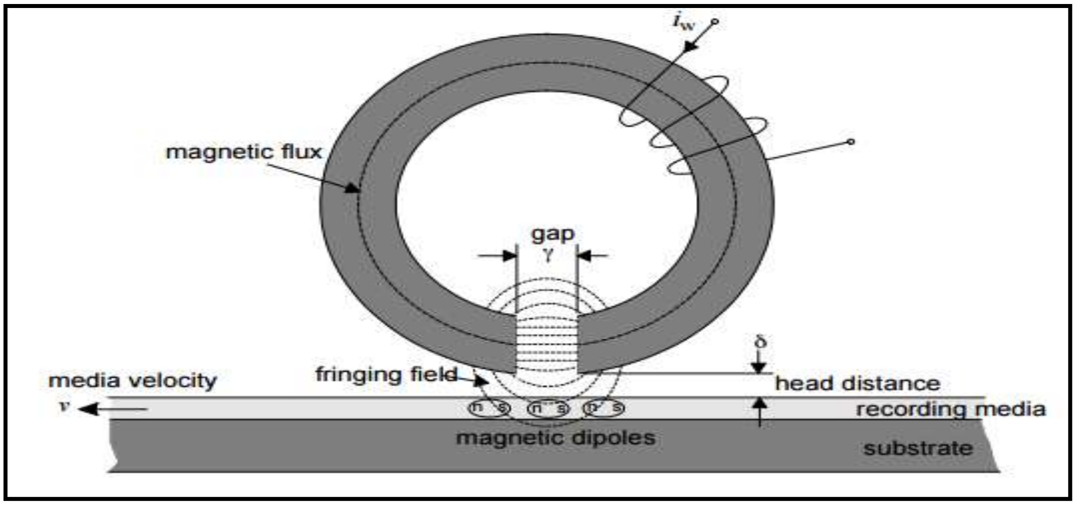


Fig. Magnetic recording

Principle: **2M**

- When certain material like iron oxide comes in contact with magnetic field, get magnetized and retain that magnetism permanently until altered.
- Sound pressure variations are converted into electrical variations by a microphone. The audio output of the microphone is amplified and fed to the coil of an electromagnet.
- The electromagnet has a minute gap through which magnetic lines of force cannot pass easily due to high reluctance of air.
- When a tape with a coating of a magnetic material is made to pass across the gap, the lines of force get an easy path through the iron oxide which is formed into elementary magnets.
- The magnetic strength of electromagnet, and hence through the gap covered by the iron oxide of the tape depends on the audio current.
- Thus the coating of iron oxide on the tape is magnetized in accordance with the audio current and hence, in accordance with the sound pressure variations.
- The magnetism in the iron oxide can be retained for long time. This means that sound has been recorded in the form of varying magnetic field.

Q5) Attempt any FOUR: 16M

a) Derive the formula for instantaneous value of FM voltage and modulation index.

Ans: (Equation of instantaneous frequency 1M,
Diagram 1M,
Equation of instantaneous value of FM voltage 1M,
Modulation index 1M)

Mathematical representation of FM:

Instantaneous frequency f of the frequency modulated wave is given by

$$f = f_c + k_f V_m \sin \omega_m t \qquad \textbf{(eq.1)}$$

where f_c is un modulated (or average) carrier frequency, k_f is proportionality constant expressed in Hz/volt and $V_m \sin \omega_m t$ is instantaneous modulating voltage.

The maximum deviation for this signal will occur when the sine term has its maximum value, ±1. Under these conditions, the instantaneous frequency will be

$$f = f_c \pm k_f V_m, \qquad \textbf{(eq.2)}$$

So that the maximum deviation δ_f will be given by

$$\delta_f = k_f V_m \qquad \textbf{(eq.3)}$$

The instantaneous amplitude of the FM signal will be given by a formula of the form

$$V_{FM} = V_c \sin [f(\omega_c, \omega_m)] = V_c \sin\Theta \qquad \textbf{(eq.4)}$$

Where $f(\omega_c, \omega_m)$ is some function of the carrier and modulating frequencies. This function represents an angle and will be called Θ for convenience. The problem now is to determine the instantaneous value (i.e. formula) for this angle.

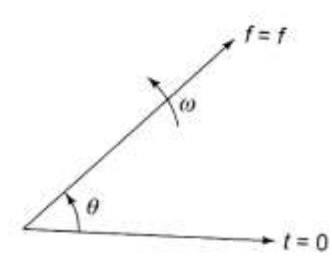


Fig Frequency modulated vectors.

Figure shows , Θ is the angle traced by the vector V_c in time t. If V_c were rotating with a constant angular velocity, for example, p, this angle Θ would be given by pt (in radians). In this instance, the angular velocity is anything but constant. It is governed by the formula for ω obtained from equation (1) that is,

$$\omega = \omega_c + 2\pi k$$

In order to find Θ, ω must be integrated with respect to time. Thus

$$\Theta=\int \omega\, dt= \int(\omega_c + 2\pi k_f V_m \sin \omega_m t)\, dt$$

$$\Theta=\int \omega_c t+ \frac{2\pi\ k_f\ V_m \sin \omega_m t)}{\omega_m}$$

$$\Theta=\int \omega_c t+\frac{\delta_f}{f_m} \cos\omega_m t$$

$$\Theta=\int \omega_c t+\frac{\delta_f}{f_m} \cos\omega_m t \qquad \textbf{(eq.5)}$$

These deviation utilized in turn, the fact that ω_c is constant, the formula $\int \cos nx dx = \sin nx/n$ and equation 3 . Equation 5 may now be substituted into equation 4 to give the instantaneous value of the FM voltage; therefore

$$V_{FM}= V_c \sin (\omega_c t+ \omega_c t+ \frac{\delta_f}{f_m} \cos\omega_m t) \qquad \textbf{(eq.6)}$$

The modulation index for FM, m_f is defined as

$$m_f =\frac{(maximum) frequency\ deviation}{modulating\ frequency} = \frac{\delta_f}{f_m} \qquad \textbf{(eq.7)}$$

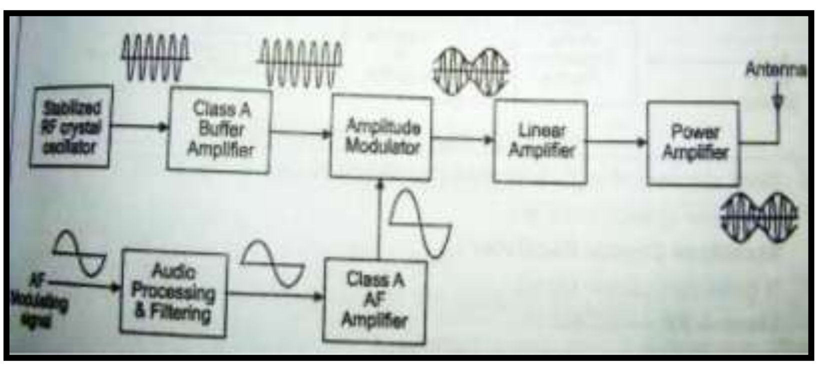
Substituting equation 7 into 6, we obtain

$$V_{FM}= V_c \sin(\omega_c t + m_f \cos\omega_m t) \qquad \textbf{(eq.8)}$$

b) Draw block diagram of AM transmitter and state the function of each block.

Ans: (Diagram-2Marks, Functions-2Marks)

<u>Block diagram of AM transmitter</u>:- **2M**



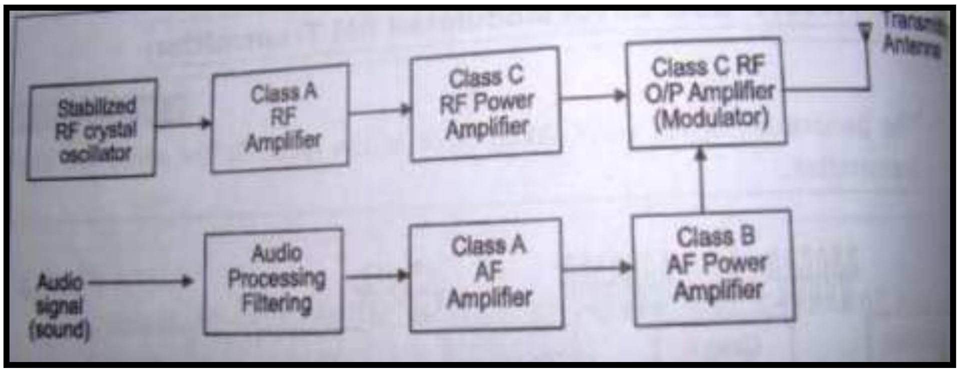
<u>Functions of each block</u> : **2M**

- **Stabilized RF crystal oscillator:**
  RF oscillator generates the carrier signal. RF oscillator is stabilized to maintain carrier frequency deviation in limit
- **Buffer amplifier:**
  Carrier signal is amplified
- **Audio processing and filtering:**
  Sound information converted in to electrical signal. It is processed and filtered
- **Class –A AF amplifier:**

It amplifies the modulating signal

- **Amplitude modulator:**
  Modulating and carrier signal applied to modulator to generate AM wave
- **Linear amplifier:**
  It avoids wave form distortion if any
- **Power amplifier:**
  Power of AM wave is amplified or increased

OR



Functions of each block : **2M**

- **Stabilized RF crystal oscillator:**
  RF oscillator generates the carrier signal. RF oscillator is stabilized to maintain carrier frequency deviation in limit
- **Class –A RF amplifier:**
  It amplifies carrier signal.
- **Class C RF power amplifier:**
  It increases power level of carrier
- **Class –A amplifier:**
  Modulating signal amplified after audio processing and filtering
- **Class- B power amplifier:**
  Amplified modulating signal amplified at adequate power level
- **Class –C RF output amplifier:**
  Carrier signal and modulating signals are applied here so it gives AM wave as output.

c) Explain the concept of tie clip microphone and state its application.

Ans: (Concept-2Marks, Application-2Marks)

Concept: **2M**

- It is an electret tiny microphone which can be clipped on to a tie, lapel or any other convenient part of clothing.
- An external amplifier made on a tiny clip of silicon is used inside the microphone.
- Even with a tiny amplifier and its cell, it is very light.

Application: (any two) **2M**

(i) Tie clip microphone is used for lecturers

(ii) It is used as radio (wireless) microphones in sports meets.

(iii)It is used in small P.A system for clubs and small halls.

(iv)It is used in sound level meters.

d) Draw neat circuit diagram and explain operation of complementary symmetry push-pull amplifier.

Ans: (Diagram-2Marks, Explanation-2Marks)

Diagram: **2M**

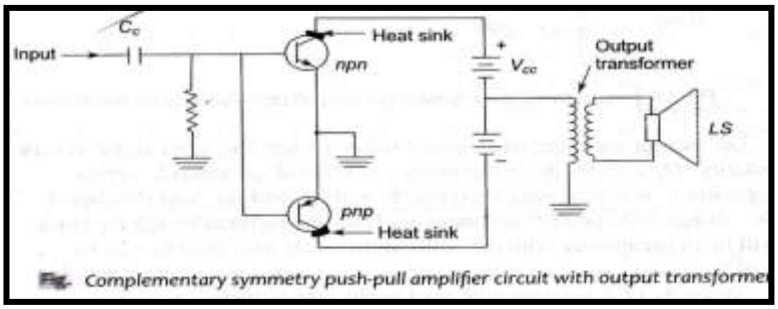


Fig. Complementary symmetry push-pull amplifier circuit with output transformer

Explanation: **2M**

- The circuit for a complementary symmetry push pull amplifier is shown in figure.
- It requires the same polarity at the input of two transistors.
- The circuit uses two transistors, one of npn type and the other of pnp type.
- Input signals to the two transistors are in the same phase. (Inter-Stage transformer for input is not required.)
- The npn collector gets positive dc voltage and the pnp collector, negative dc voltage.
- Direct current, through the primary of the transformer will be in the opposite directions. The audio currents from the two transistors will add in the primary and then will give all the advantages of push-pull configuration.

e) Define reverberation. State its necessity.

Ans: (Definition-3Marks, Necessity-1Marks)

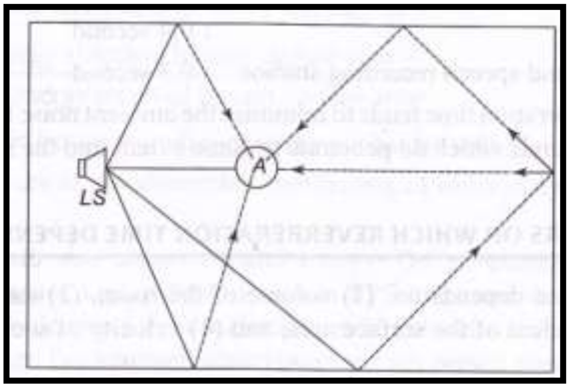
Definition: **3M**

Reverberation:

- As the auditoriums and studios and even living rooms a person receives sound directly from the source as well as sound reflected from the walls, celling, floor etc. the reflected sound is heard as a distance echo if the time gap between the original wave and reflected wave is more than oms.
- Reflection over shorter distance shall simply prolong the sound due to multiple reflection in hall as shown in fig. in which loud speaker is the source of the sound and A, the listener who receive the

direct sound as well as the reflected sound. The sound persists even after the source of sound has stopped sounding.

- It fades away only gradually. The gradual fading of the continuing echo is called reverberation



- Reverberation time is defined as the time taken for sound energy in room to drop to 10^{-6} time of its initial value.
- Reverberation to some extends is pleasing & should be incorporated in the design of rooms.

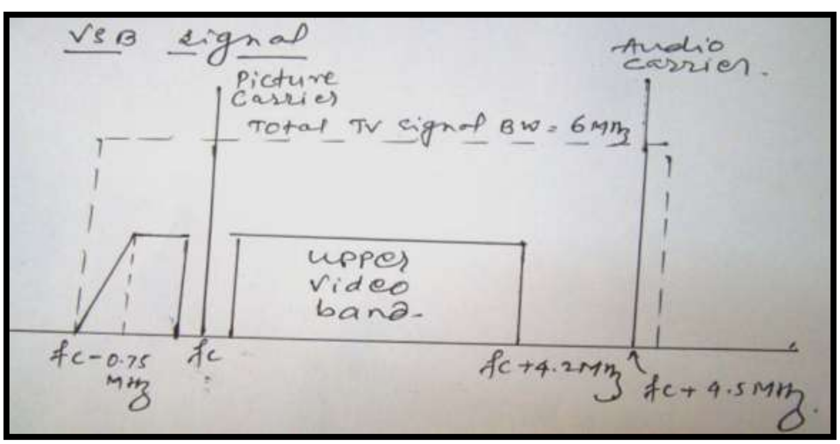
Necessity: **1M**

- To give the natural or pleasing sound because all natural sound in a hall or auditorium includes a proportion of continuing echoes variation in this proportion give sound a quality of liveliness or richness. Hence reverberation is necessary.

f) Explain the concept of vestigial sideband.

Ans: (**Diagram-2marks, Explanation-2marks**)

Diagram:- **2M**



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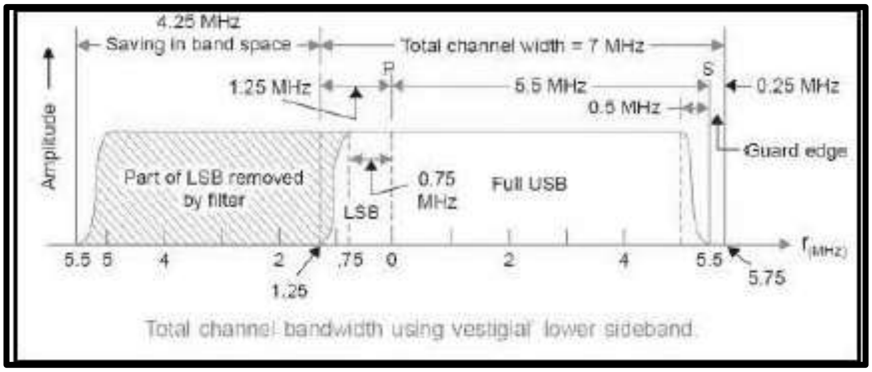


Total channel bandwidth using vestigial lower sideband

Explanation: - 2M

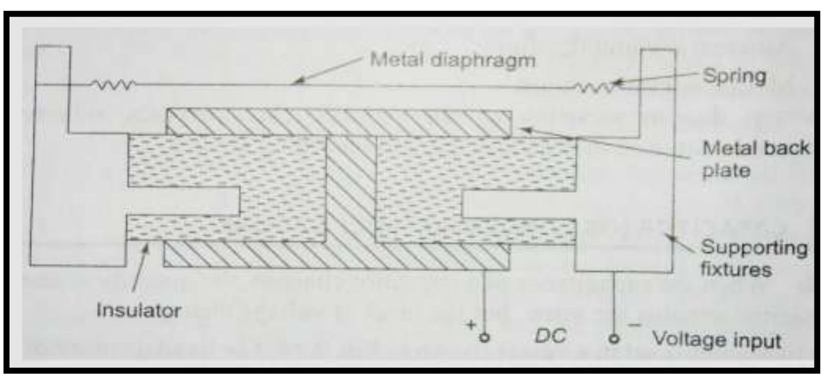
- In the TV broadcasting the audio carrier in frequency modulated and video information in amplitude modulated.
- The picture carrier is transmitted, but one sideband is partially A portion of the lower side band of the TV signal suppressed leaving only a small vestige of lower sideband. Such an arrangement is known as vestigial sideband signal.
- Video signal above 0.75 MHz (750 KHz) are suppressed in the lower sideband while all video frequency are transmitted in the upper vestigial sideband suppressed.

Q6) Attempt any FOUR of the following: 16M

a) Draw neat sketch showing construction of condenser microphone and explain its operation.

Ans: (Diagram-2Marks, Explanation-2Marks)

Diagram:- 2M



Explanation:- 2M

- When sound waves strikes the diaphragm, it moves. During compression, it moves towards the fixed back plate and increases capacitance.
- During rarefaction, it moves away from the back plate and therefore decreases the capacitance.
- The change in capacitance changes the DC voltage across the capacitor plates. as the distance between the plates changes, its capacitance changes as per the below equation

$$C = kA/d$$

Where,

"k"is a dielectric constant of the medium between the plates.

"A" is area of the cross section of the plates.

"d "is the distance between the plates.

b) Explain BASS control circuit with the help of neat circuit diagram.

Ans: (Diagram-2Marks, Explanation-2Marks)

Diagram: **2M**

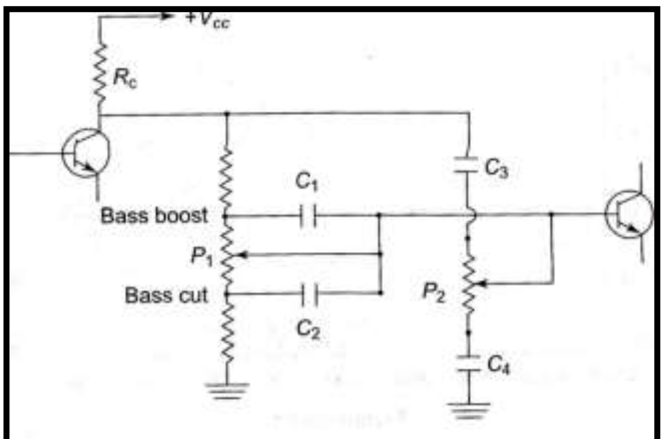


Fig Bass and treble control

Explanation: **2M**

- Bass would be cut if capacitive reactance in series of signal increases.
- Lower the capacitance, greater will be the reactance ($X_c = ½ \pi f C$).
- Hence for cut in base, the value of series capacitance is reduced as illustrated in figure.
- When the slider of the potentiometer R is at the upper end, the capacitor C_1 is shorted and the signal directly goes to the next stage, bypassing the capacitor C_1 and hence, bass has the minimum attenuation.
- It is called bass-boost. When R of the potentiometer comes in series with the signal.
- In this position, bass will have maximum attenuation. This position is called bass cut.

c) What precautions will you take while installing PA system (Explain any four).

Ans: Precautions:(any four 1M each)

One should consider the following points while installing a PA system**.**

- **Acoustic Feedback:**
  Sound from the loudspeaker should not reach feedback will result in loud crawling sound**.**
- **Distribution of sound intensity:**
  1. Loudness of the sound is contained in low notes, and the intelligibility in high notes.

2. High notes suffer greater attenuation with distance than lower notes.
3. A good P.A. system should take care of this fact, and sound should be uniformly distributed amongst the audience i.e. instead of 1 or 2 powerful loudspeaker near the stage above, audio power should be divided between several loudspeakers to spread it up to the farthest point so that each one specified area.
4. The no. and wattage of L.S.S is should be sufficient to handle the maximum Power of amplifier.

- **Reverberation:**
  1. In a reverberating medium, the intelligibility is poor. This is due to overlapping of successive sound waves. P.A. system should throw additional power in those areas where the direct sound gets submerged in the echoes.
  2. The problem of reverberating halls can be solved by locating several small points.
  3. L.s.s at various points of the auditorium rather than using a single high power circuit.
- **Orientation of Loudspeaker:**

  To make maximum best use of available power of the P.A system L.S should be oriented direct towards the audience not towards the walls.
- **Ambient Noise:**

  Place like sports events place, where ambient noise is high P.A system should boot the high. Frequencies to restore the intelligibility.
- **Dynamic Range Limitation :**

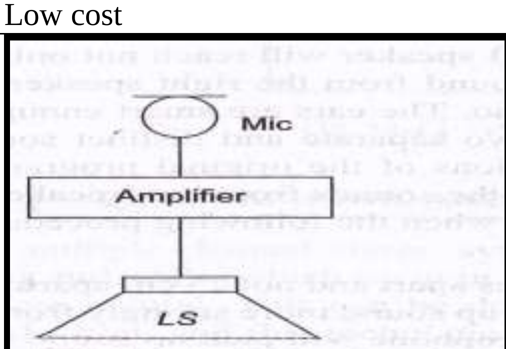
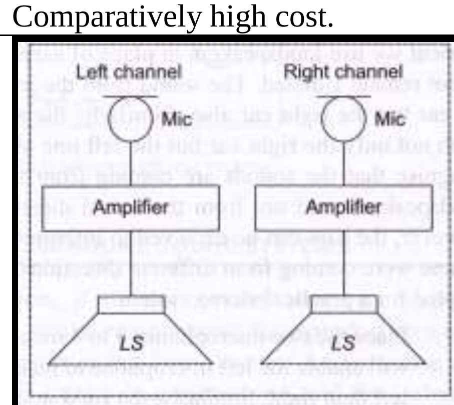
  A good P.A system should have level meter which keeps the o/p level constant. When i/p level exceeds a certain predetermined value.
- **Selection of microphone:**
  - Microphones preferably car diode type, so that they do not pick up reverberation sound & sound from L.S.
  - Sense of direction of the source of sound. The L.S should be so placed that the sound appears to be coming from the direction of the sound.
- **Phase delay:**
  - Sound from nearest loud speaker may be heard along with the sound from the other loud speaker with time difference. Delayed sound impairs the intelligibility delay to 45ms or more. Hence loud speaker should not located beyond 16 meter apart.
  - Matching of total loud speaker impedance with the output impedance of the amplifier is necessary for maximum transfer of energy. So series parallel combination of loud speaker should be such as to ensure maximum power transfer.
- **Grounding:**

  Chasses and shields of equipment and coaxial cable should be earthed properly.
- **Amplifier power:**
  - P.A system gives out amplified sound so that it is comfortably sound audible to audience at distance from few watts to watts, depends on the need.
  - The total output of power of the amplifier required should be calculated on the basis of sound intensity equal to 80dB over the threshold of hearing should be available to the audience at the farthest point.

- **Choice of Loud speaker:**
  Loud speaker should be able to withstand the output power to the amplifier.
- **Close Ring connection of loud speaker:**
  For better reliability, loud speaker leads should form a closed ring. If the lead is broken at any point, it will not make any loudspeaker in active**.**
- **Placement of microphones:**
  Placement of microphone should be made in such a way that they give total coverage of the all source of program sound and at the same time not respond to unwanted sound.
- **RF pick up :**
  Due to poor grounding, or cold or dry solder joints or defective RF by capacitors local radio station and RF transmission are pick up and detected by the amplifier. For this connect RF by pass Capacitors at the output of amplifier.
- **Presence not to be felt:**
  An ideal P.A system is one in which everyone among the audience can hear the programme comfortably without becoming aware that amplifiers are in use.

d) Differentiate between monophonic and stereophonic system (any 4 points).

Ans: <u>Comparison:</u>- (any four points) **4M**

Monophonic system	**Stereophonic system**
1. Only one amplifier is used. Single amplifier stage is known as mono amplifier	1. At least two independent amplifiers are used. These part of amplifiers is called as stereo signal
2. No naturalness	2. Provides naturalness of sound signal.
3. Listener cannot judge the direction of sound	3. Listener can judge the direction of sound
4. Low cost	4. Comparatively high cost.
5.  Fig. Basic monophonic system	5.  Fig. Basic stereophonic system

e) List various causes affecting fidelity of system. What are their remedies?

Ans: (Causes-2marks, Remedies- 2Marks)

Causes affecting fidelity: (Any four 2M)

- High signal to noise ratio.(s/n ratio\)
- Flat frequency response
- Low nonlinear distortion
- Large dynamic range
- Creating sense of direction.

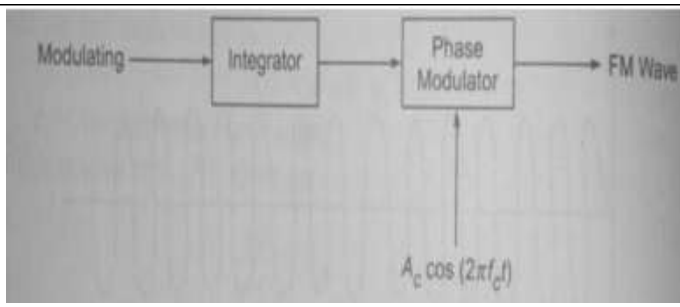
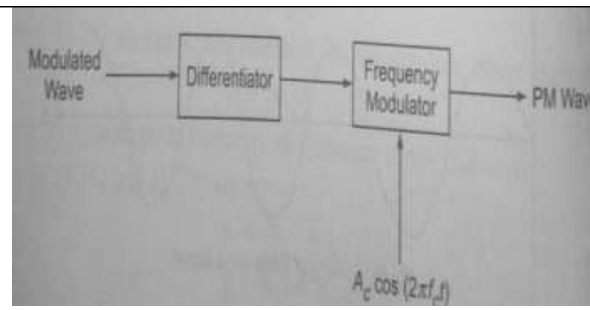
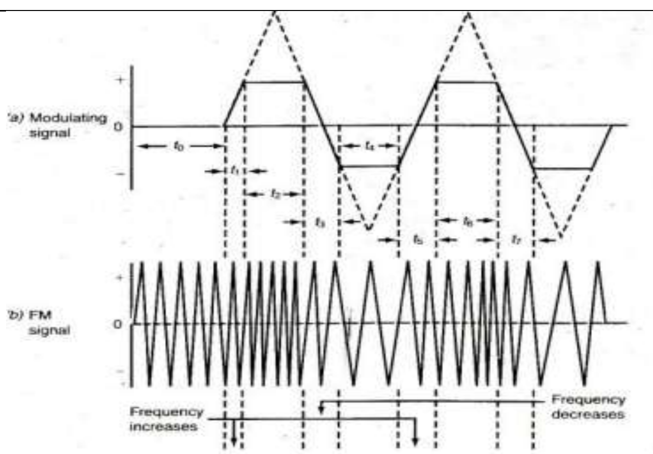
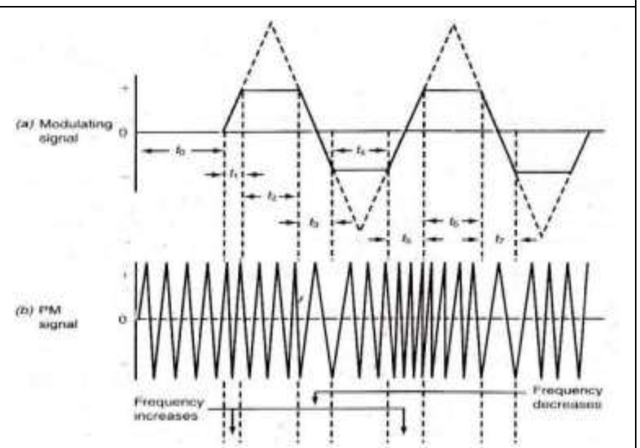
Remedies: (Any four 2M)

- S/N ratio can be improved by using preamplifier of low noise figures proper shielding, grounding, decoupling & filtering circuits, stabilized power supply, microphones
- By using coupling capacitor and shunt capacitor in audio amplifier circuits.
- Nonlinear distortion can be reduced by using negative feedback in amplifier, designing bias circuit to keep Q point in the middle of linear portion of the characteristics curve.
- Dynamic range can be increased by using solid-state amplifier, dynamic microphones & L.S. which are capable of withstanding the large change in loudness.
- Creating sense of direction can be improved by using high fidelity system.

•

f) Explain the difference between frequency and phase modulation. (any 4 points)

Ans: Comparison: (Any Four Points) **(4M)**

SR NO	FM	PM
1.	Frequency of the carrier is varied according to the modulating signal.	Phase angle of the carrier is varied according to the modulating signal.
2.	FM can be generated by integrating PM and then using the resulting signal to phase modulate the carrier.	PM signal can be generated by first differentiating modulating signal and then using the resulting signal as the input to a frequency modulator to modulate the carrier.
3.		
4.	In FM wave the maximum deviation occurs at the peak positive and negative amplitude of the modulating signal.	In PM the maximum frequency deviation modulator occurs during the „time" that the modulating signal is changing at its most rapid rate.
5.		
6.	The fm modulation index will increase as modulating frequency is reduced and vice versa	The pm modulating index will remain constant as modulating frequency is change