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 of the following: **12M**

i) List four selection criteria of microphones

Ans:-

Selection criteria of microphones: [Any four points] **½ M each**

Microphone possess following characteristics

- Sensitivity
- S/N ratio
- Frequency response
- Distortion
- Directivity
- o/p impedance

ii) State different characteristics of audio amplifier.

Ans:-

Characteristics of Audio amplifier:- [Any four] **½ M each**

- Gain
- Bandwidth
- Distortion
- Power output
- Impedance

iii) Define: Frequency Modulation and its Modulation Index

Ans:-

Frequency Modulation:- **01M**

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

Modulation Index:- **01M**

The modulation index for FM, m_f is defined as

$$m_f = \frac{(maximum)\ frequency\ deviation}{modulating\ frequency} = \frac{\delta_f}{f_m}$$

iv) List advantages of CD

Ans:-

Advantages of CD:- [Any four] **½ M each**

- As the CD surface is covered by transparent plastic or transparent lacquer, the tracks & recording remains safe and are not affected by dust, grease & scratches. It is immune to surface contaminations.
- High S/N ratio (about 90 dB)
- High dynamic range (around 90 dB)
- High channel separation (around 80 dB)
- Wow does not exist
- Flutter does not exist
- Overall distortion is low
- Excellent frequency response covering complete audio range from 20 Hz – 20 kHz within only ±0.5 dB
- Small size

v) State different methods of optical recording of sound on a film.

Ans:-

Different methods of optical recording of sound on film:- **01M each**

- Variable density method
- Variable area method

vi) Define AM.

Ans:-

Amplitude Modulation(AM):- **02M**

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

vii) List different tone control.

Ans:-

Different Tone control:- **01M each**

- Bass Control
- Treble Control
- Volume control
-

viii) List any three voltage and power amplifiers

Ans:-

Voltage Amplifiers:- **01M**

- Transformer coupled
- RC coupled
- Direct Coupled

Power Amplifier (any three):- **01M**

- Class A
- Class B
- Class C
- Class AB

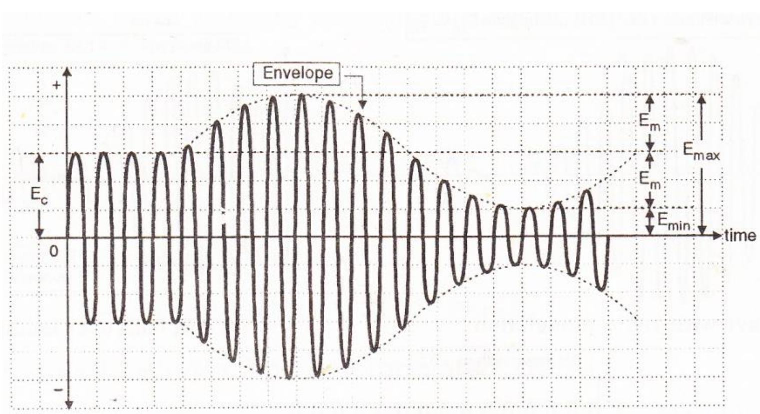
Q1)

b) Attempt any two:- **08M**

i) Derive the mathematical expression for modulation index in AM.

Ans:-

Waveform:- **01M**



AM wave for calculation of m

- To calculate modulation index "m" which is:

$m = \frac{E_m}{E_c}$, we must express E_m and E_c in terms of E_{max} and E_{min}

- Referring to fig. we can write,

$$E_m = \frac{E_{max} - E_{min}}{2} \quad \text{.......................................}(1)$$ **01M**

And $E_c = E_{max} - E_m$ ....(2)

- Substitute the value of E_m from equation(1) into equation (2) to get,

$$E_c = E_{max} - \left[\frac{E_{max} - E_{min}}{2}\right] = E_{max} - \frac{E_{max}}{2} + \frac{E_{min}}{2}$$

Therefore

$$E_c = \left[\frac{E_{max}+E_{min}}{2}\right] \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(3)$$ **01M**

- But $m = \frac{E_m}{E_c}$
- So substituting the value of E_m and E_c from the equation (1) and (3) we get,

$$m = \frac{(E_{max}-E_{min})/2}{(E_{max}+E_{min})/2}$$

Therefore

$$m = \left[\frac{E_{max}-E_{min}}{E_{max}+E_{min}}\right] \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(4)$$ **01M**

- This is the required expression

[<u>NOTE</u>:- Students can use other notation instead of "E", "V" can be considered]

ii) What is the bandwidth required for FM signal in which modulating frequency is 3KHz and the maximum deviation is 15KHz.

Ans:-

Given:-

Modulating frequency (f_m) = 3 KHz. **½ M**
Maximum deviation (δ) = 15 KHz. **½ M**

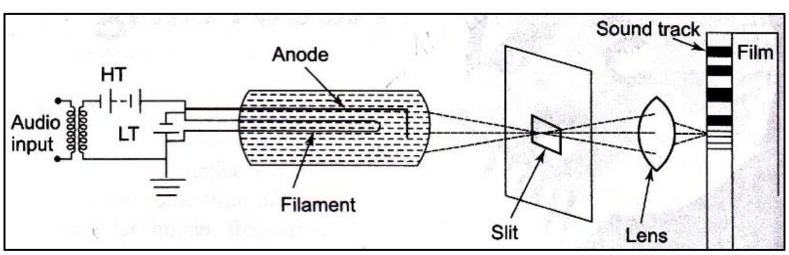
Bandwidth $= 2[\delta + f_m]$ **01M**
$= 2[15 + 3]$
$= 2[18] = 36$ KHz. **02M**

iii) Explain variable density method of optical recording of sound

Ans:-

<u>Variable density method:</u>

<u>Diagram:-</u> **02M**



<u>Explanation:</u> **02M**

- In this method, sound is picked up by a microphone and converted into electrical signals which are amplified by audio amplifier & is fed to the anode of a special type of vacuum tube, called an AEO lamp. This lamp consists of a little quantity of helium gas.
- High DC voltage (HT) is applied to the anode in series with the audio voltage
- The filament of the lamp is connected to the low DC voltage (LT)

The intensity of light coming from lamp varies in accordance with the audio signal. This varying light passes through a slit and a focusing lens. The focused light falls on a moving photographic film where the image is recorded in the form of bars of varying density and distance on the film.

Q2) Attempt any four: 16M

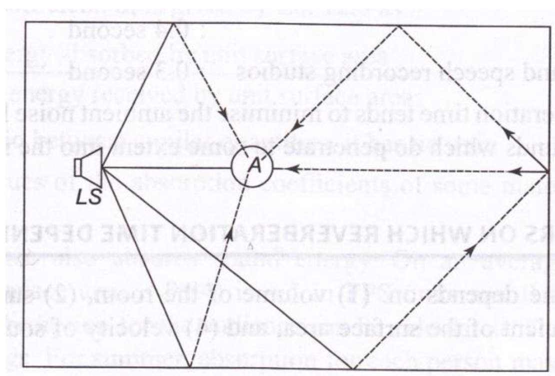
i) Explain concept of reverberation and its necessity.

Ans:-

Reverberation: **03M**

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: **01M**

- 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.

ii) Define and explain the terms:
Pre-emphasis and de-emphasis.

Ans:-

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 :- **02M**

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 :- **02M**

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.

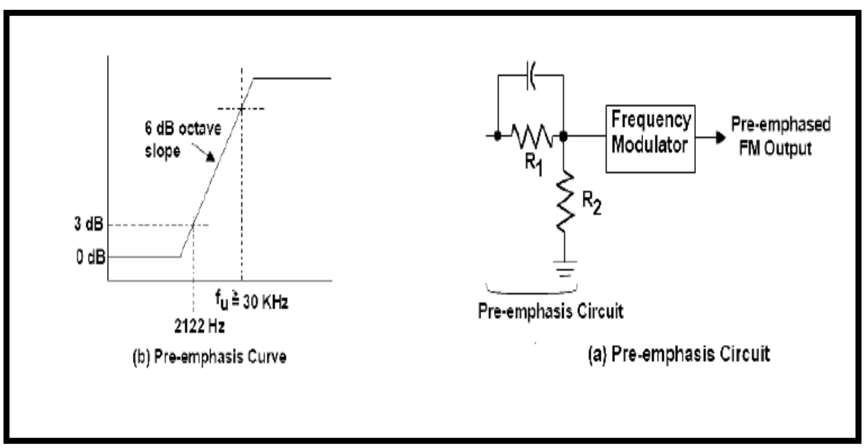
OR

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.



- 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.

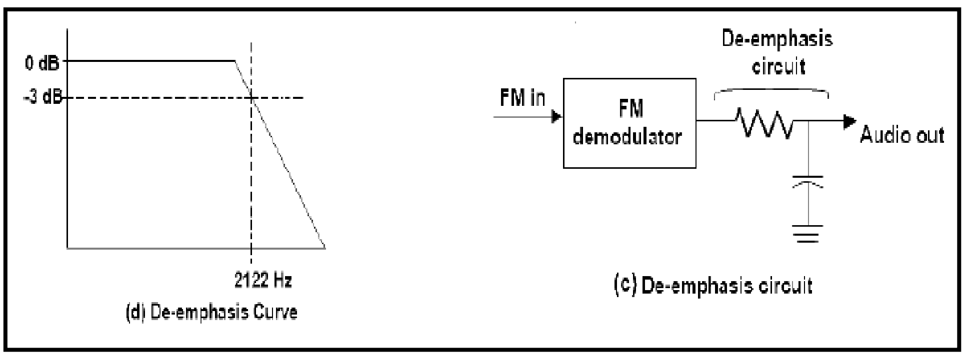


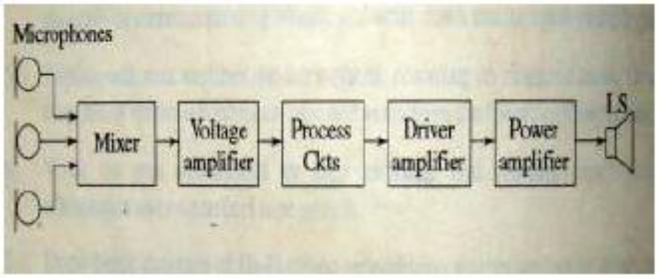
Fig. De-emphasis Circuit

- 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.

iii) Draw neat block diagram of PA system and give function of each block.

Ans:-

Block Diagram:- **02M**



Function:- **02M**

The intensity of sound decreases with distances. Hence when a large gathering is to be addressed, sound needs to be amplified so that people at a distance from the rostrum or stage may receive good intensity of sound for comfortable listening. The system which fulfills the above requirement is called public address system or P.A system.

The in used in sports meet public meeting auditorium, concerts, functions etc. also used to convey information to isolated location like, railway station airport, hospitals, factories, schools etc. In an electro acoustic system in which sound in first converted into electrical signals by a microphone

1) **Microphone –**
It picks-up sound wave and convert them to equivalent electrical signal called audio signals. Generally 2 or more microphones are used and in addition, an auxiliary input for tape/record player CD player.

2) **Mixer-**
The out of microphones in fed to mixer stage. The function of the mixer stage in 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.

3) **Voltage amplifiers**-
Amplifiers the output of mixer stage.

4) Processing circuit-

These circuits have master-gain control (volume control) and tone control circuit.

5) Driver amplifier –

It gives voltage amplification to the signal to such an extent that when feed to power amplifier (next stages) the into internal resistance of that stage is reduces. Thus drivers the power amplifier to give more power.

6) Power amplifier –

It gives desired power amplification to the signal generally push pull amplifier is used, so that harmonics are eliminated from the output and transformer core us bit saturated, The output of the power amplifier is connected to the loudspeaker through a matching transformer to match the low impedance of the L.s for max transfer of power

7) Loudspeaker-

Converts electrical signal into pressure variation resulting in sound.

iv) Give reasons why optical recording is better than magnetic recording system.

Ans:- (Any four points) **01M each**

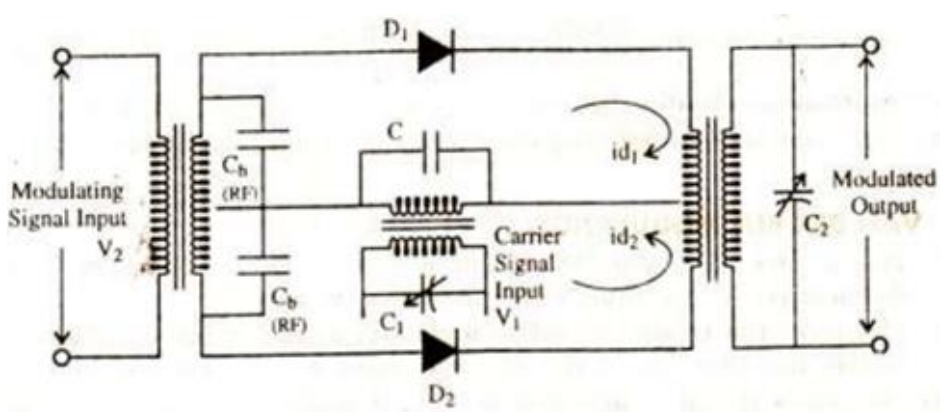
Optical recording is better than magnetic recording as:-

1. Optical recording is based on digital signal and have all the advantage of digital system over analog system.
2. Frequency response is flat from 20Hz. To 20KHz. Than magnetic recording
3. Wow and flitter distortion are absent because of digital nature of data.
4. Optical recording has error correction capability. If there are errors in the data they are corrected
5. Signal to noise ratio is high.
6. Channel separation and dynamic range are improved in optical recording.
7. The data density of CD is very high as compared to magnetic tapes

v) Describe how DSBSC AM signal is generated by diode balance modulator with neat diagram.

Ans:-

Diagram: - **02M**



: Balanced Modulator circuit using diodes

Explanation: **02M**

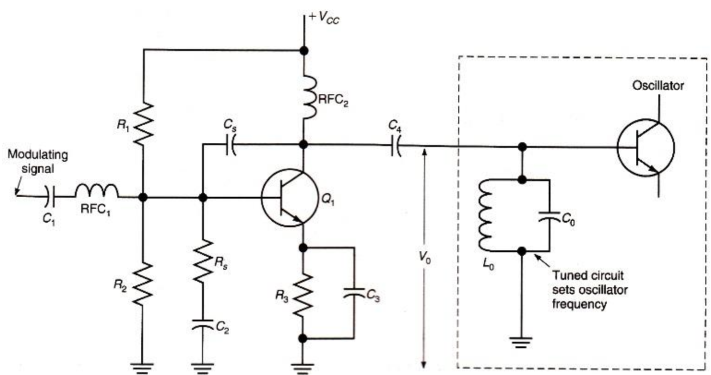
This is a circuit can be used for generating the two side bands with the suppression of carrier. The balanced modulator is constructed using components which are of non-linear behavior can be analyzed by certain mathematical equations,

1) i = bv where b = conductance
2) if the circuit operates in amplifier form then equation is i = a + bv where a = dc component
3) If the circuit is constructed using certain non-linear devices then equation modifies to i = a + bv + cv2 where c = non-linear constant may be positive or negative

vi) Draw neat diagram of reactance modulator and write its operating principle.

Ans:-

Diagram:- **02M**



Principle:- **02M**

In reactance modulator a transistor is operated as a variable reactance and it is connected across the tuned circuit of an oscillator. As the instantaneous value of modulating voltage changes, the reactance offered by the transistor will change proportionally. This will change the frequency of oscillator to produce FM wave.

Q3. Attempt any FOUR: **16M**

i) If 60 KW of carrier power is radiated by broadcast AM transmitter, then what will be the power radiated at 75% modulation.

Ans:-

Solution

Given,

Carrier Power(P_c)=60 KW **½ M**

Modulation Index(m)=0.75 **½ M**

Total radiated power

$P_t=P_c[1+\frac{m^2}{2}]$ **01M**

$=60[1+\frac{(0.75)^2}{2}]$

=76.87 KW **02M**

Total radiated power=76.87KW.

ii) Define Modulation. Why is it needed?

Modulation:-

It is the process of superimposing low frequency information signal to a high frequency carrier signal. **01M**

Need of modulation: – **03M**

a)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

b) 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!

c) Therefore, we cannot transmit low frequency signals directly. As per equation (1), if we increase frequency of electrical signal

d) 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.

iii) What is BW required for an FM signal in which the modulating frequency is 1 kHz and maximum deviation is 7 kHz.

Ans:-

Solution:-

Given,

F_m=1KHz. **½ M**

δ_{max}=7KHz **½ M**

Band Width=2(δ_{max}+f_m) **01M**
=2(7+1)
=16 KHz
Band Width of FM=16 KHz **02M**

iv) Calculate the power in side bands if 800 W carrier is amplitude modulated to the depth of 70%.

Ans:-

Solution:

Given,
Pc=800 watt. **½ M**
M=0.7 **½ M**
$P_{USB}=P_{LSB}=\frac{m2}{4}*Pc$ **01M**
$=\frac{(0.7)^2}{4}*800$
=98 watt. **02M**
$P_{LSB}=P_{USB}$=98 watt.

v) Explain Dolby-B system of noise reduction.
Ans:-
Diagram: **02M**

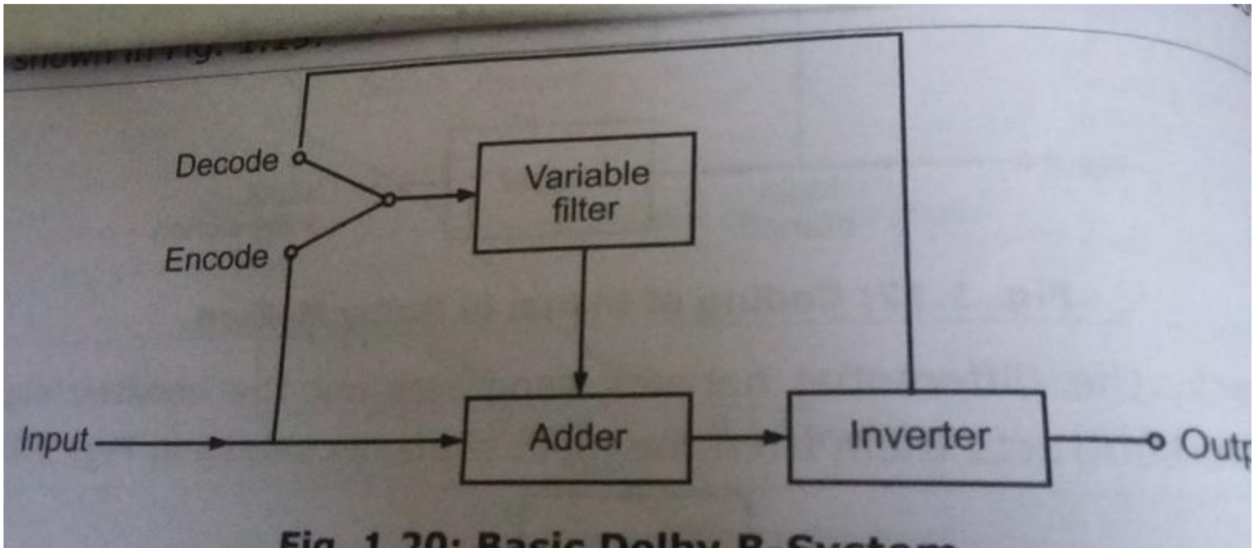


Fig. 1.20: Basic Dolby B-System

Explanation : **02M**

Dolby B was developed after Dolby A and presented in 1968, as a single sliding band system providing about 9 dB noise reduction (A-weighted), primarily for cassettes. It was much simpler than Dolby A and therefore much less expensive to implement in consumer products. Dolby B recordings are acceptable when played back on equipment that does not possess a Dolby B decoder, such as most inexpensive cassette players. However, Dolby B provides less effective noise reduction than Dolby A, generally by a factor of more than 3 dB.

vi) Draw and explain block diagram of communication system.

Ans:-

Diagram: **02M**

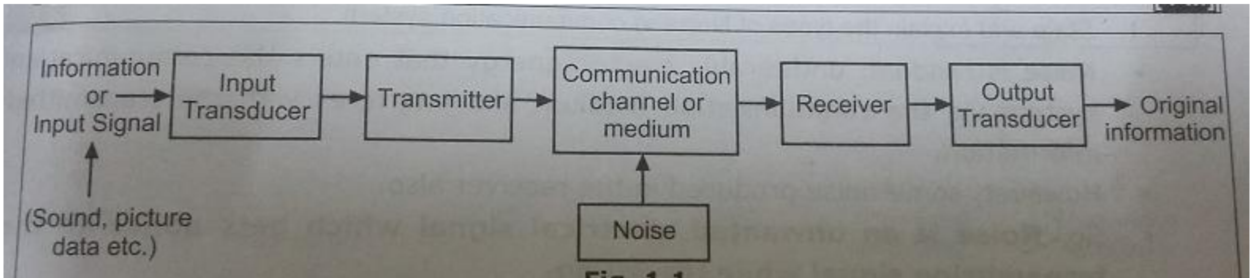


Fig. 1.1

• Fig. 1.1 shows the basic elements of communication system.

Explanation : **02M**

Input Signal:
The information or Input signal can be in the form of sound, picture or data coming from the computer.

Input Transducer:
It converts the original information into equivalent electrical signal.

Transmitter:
It converts electrical equivalent of information into suitable form and also increases power level of the signal to cover long range.

Channel:
It is the medium used for transmission of electromagnetic signals from one place to another. It may be wire or fiber optic or free space.

Noise:
It is unwanted signal which gets added in the transmitting signal while travelling.

Receiver:
Received signal is amplified, demodulated and converted back to suitable form.

Output Transducer:
It converts electrical signal into original information signal.

Q4 Attempt any four **16M**

i) Explain Armstrong frequency modulator system with its operation

Ans: -

Diagram:- **02M**

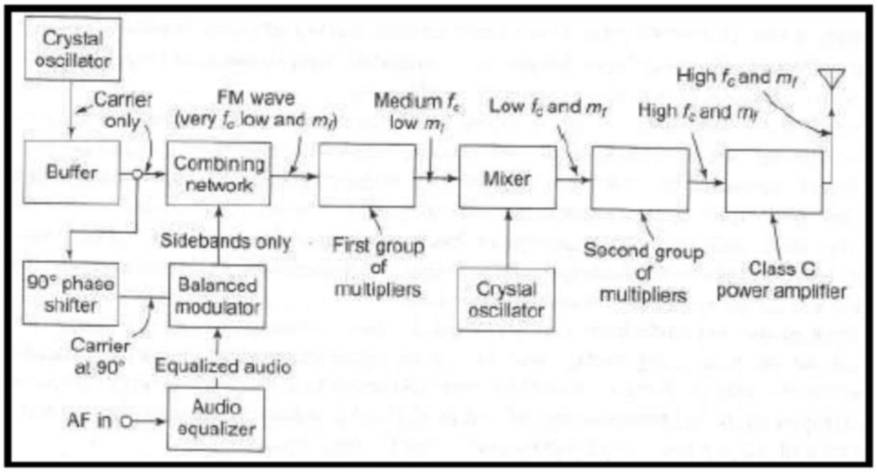
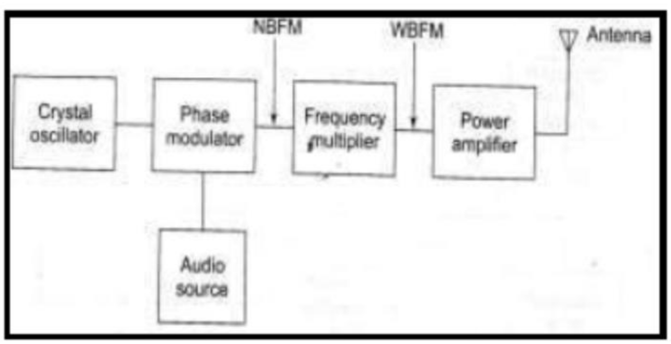


Fig. FM transmitter
OR



Explanation:- **02M**

- 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.

ii) List different types of analog modulation techniques and give reason why it is needed.

Ans-

<u>Types of analog modulation techniques:</u> - **02M**

a) <u>Continuous wave modulation</u>

1. Amplitude modulation
   a. High level modulation
   b. Low level modulation
2. Frequency Modulation
   a. Narrow Band
   b. Wide Band
3. Phase modulation

<u>Reason for modulation:</u> - **02M**

1. It is impractical to propagate information signals over standard transmission media so that it is necessary to modulate the source information onto a higher frequency analog signal called carrier.
2. It is extremely difficult to radiate low frequency signals from an antenna in the form of EM energy.
3. To reduce the height of antenna.
4. To avoid mixing of signals.
5. To increase the range of communication

iii) List any four specifications PA system.

Ans:-

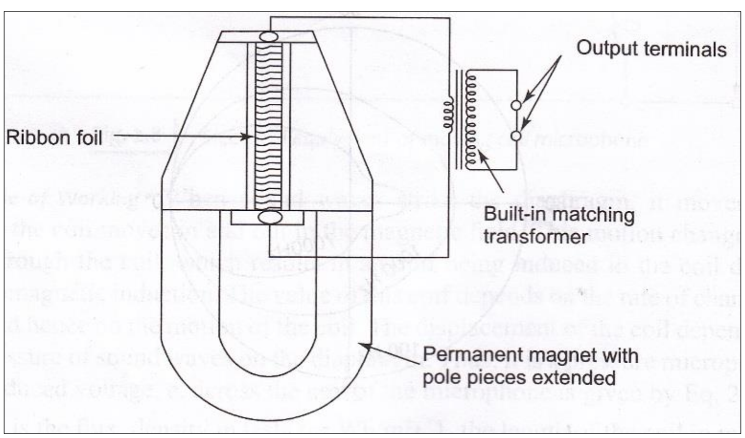
Specification of PA system: - (Any four) **01M each**

1. Acoustic feedback
2. Distribution of sound intensity
3. Reverberation
4. Orientation of loud speaker
5. Ambient noise
6. Dynamic range limitation
7. Selection of microphone
8. Sense of direction of the source sound
9. Phase delay
10. Matching
11. Grounding

iv) Give constructional details of ribbon microphone with neat diagram.

Ans:-

Diagram: - **02M**



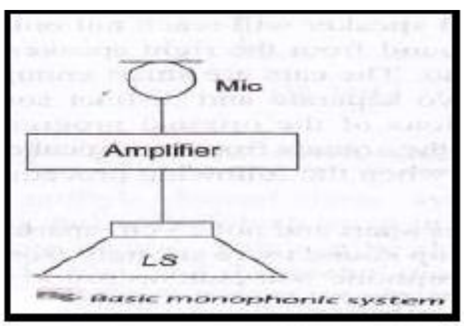
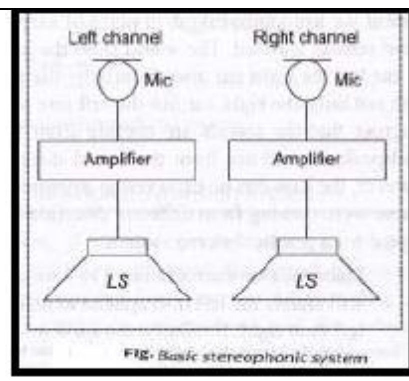
Constructional details: - **02M**

- The main parts of a ribbon microphone are : permanent magnet and ribbon conductor
- The permanent magnet is a specially designed horse-shoe magnet with extended pole pieces. It provides strong magnetic field.

- The ribbon is a light aluminum foil. It is corrugated at right angles to its length to provide greater r surface area.
- The main feature is the lightness of the ribbon which is only about 0.2 mg in weight, a few microns thick and about 3mm wide.
- It is suspended in the magnetic field of permanent magnet and stiffness of suspension is small.
- The whole unit is enclosed in a circular or rectangular baffle.

v) Compare monophony and stereophony.

Ans: - (any four points) **01M each**

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

vi) Explain manufacturing process of CD with relevant diagram.

Ans: -

Diagram: - **02M**

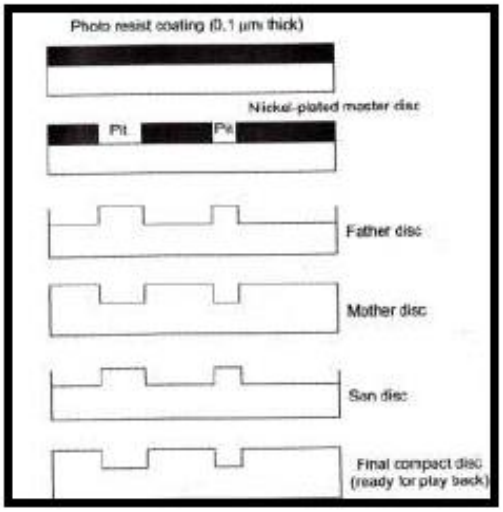


Fig. Preparation of CD

Explanation: - **02M**

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. It 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 molded 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 center of the disc. It is then packed in a plastic case.

Q5 Attempt any four **16M**

i) Differentiate between direct and indirect methods of FM generation.

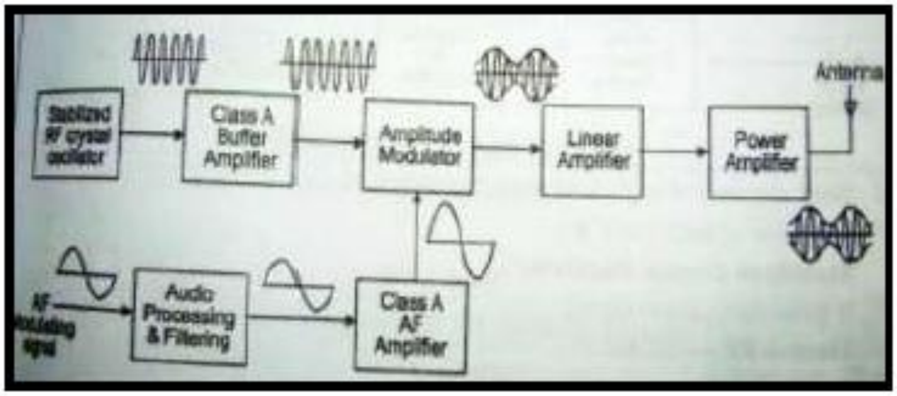
Ans:- **01M each**

Direct method	Indirect method
1. In direct f_m generation the instantaneous frequency of the carrier is changes directly in proportional with the message signal.	In indirect method use of phase modulation to obtain frequency modulation.
2. Oscillator frequency is not stable.	Oscillator frequency is stable.
3. Simplicity of the modulator and their low cost	Complex of the modulator and their high cost
4. Eg. Reactance modulation, varactor diode modulation	Eg. Armstrong frequency modulator.

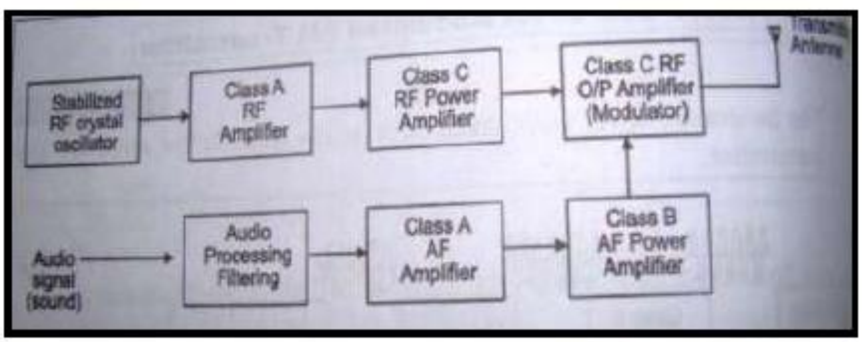
ii) Draw block diagram of AM transmitter. Write function of antenna.

Ans:-

Diagram: - **03M**



OR



Function of Antenna:- **01M**

- It is used for transmission and reception of information in terms of EM waves.

iii) Explain the concept of tie clip microphone and state its applications.

Ans:-

Concept: - **02M**

- 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)** **02M**

- Tie clip microphone is used for lecturers
- It is used as radio (wireless) microphones in sports meets.
- It is used in small P.A system for clubs and small halls.
- It is used in sound level meters.

iv) Compare audio and power amplifier.

Ans: - (any four point) **01M each**

Audio amplifier	Power amplifier
1. Transistor used in audio amplifier has a large value of current gain (β) is nearly 100 as compare to that of power amplifier.	1. Transistor used in power amplifier has current gain between 20& 50.
2. The input voltage to base of transistor in audio amplifier is low (a Few mV)	2. The input voltage to base of transistor in power amplifier is high (2-4 V)
3. Input resistance is quite low as compare to output resistance.	3. Input resistance is very large as compared to output resistance.
4. The physical size of the transistor used in voltage amplifier is used in voltage small known as low or medium power transistor.	4. Power amplifier uses larger size transistors.
5. Audio amplifier uses RC coupling for inter-stage connection.	5. Power amplifier uses transformer coupling

v) Give any four points of comparison between woofer and tweeter.

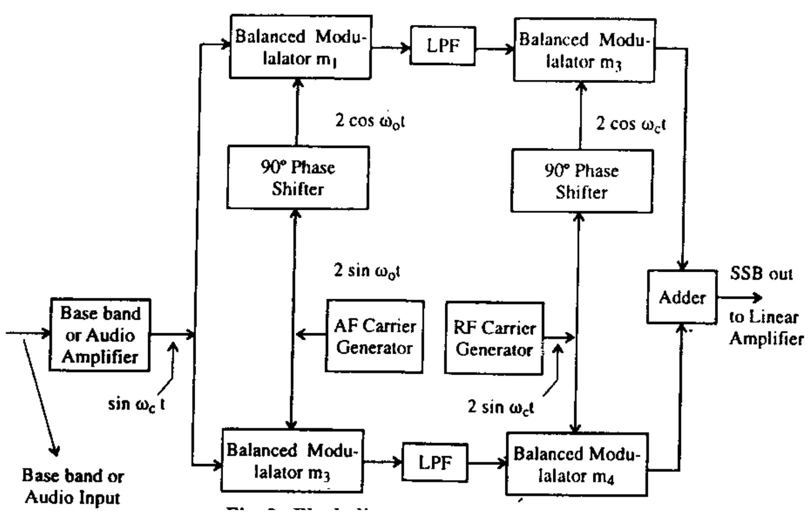
Ans:- **(any four point)** **01M each**

Parameter	Woofer	Tweeter	Squawker
Size and physical structure	Large	Small	Medium
Frequency range	Low	High	Medium
Audio frequency range	50-500 Hz	5KHz-15KHz	500Hz-5KHz
Placement of speaker	Bottom	Top	Middle
Weight	More	Light	Medium
Attenuation	High frequency attenuated	Low frequency attenuated	Lower than 500Hz and greater than 5KHz

vi) With neat diagram, describe generation of SSB AM signal using third method.

Ans: -

Diagram **02M**



Explanation:- **02M**

- It was developed by weaver to retain the advantage of the phase shift method, such as its ability to generate SSB at any frequency and use low audio frequencies.
- It is very complex and not often used method commercially.

- The later part of the block diagram is identical to phase shift method, but the way in which the appropriate voltages are fed to the last two balanced modulators at points C & F has been changed that is instead of phase shifting the whole range of audio frequencies, this method combines them with an AF carrier F_o which is the fixed frequency in the middle of audio band.
- A phase shift is then applied to this fixed frequency only.
- The resulting voltage at the output of the balanced modulators M_1 and M_2 are applied to low pass filters whose cut off frequency is designed to be f_0 to ensure that the input to the last stage of the balanced modulators i.e. M_3 and M_4 results in proper side band suppression
- If a lower sideband signal if required at the final output the phase of the carrier voltage being fed to M_1 should be changed by 180^0

Q6. Attempt any FOUR: 16M

i) Give reason why multi-way speaker system is needed for good sound quality.

Ans:- 04M

- It is essential to have a multi-way speaker to divide the incoming signal into separate frequency range for each speaker.
- In the absence of multi-way speaker the speakers will suffer overheating and the output will be distorted when full power at frequencies outside their range is fed to them. The overall efficiency will be much reduced in the absence of crossover networks.

ii) Explain operation of BASS control circuit with help of neat circuit Diagram.

Ans:-

<u>Diagram-</u> **02M**

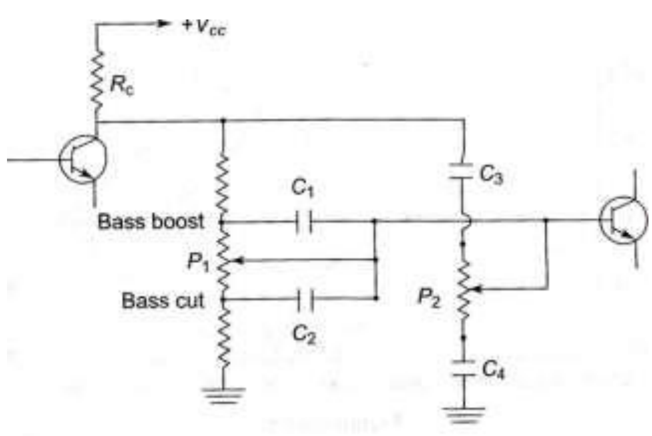


Fig Bass and treble control

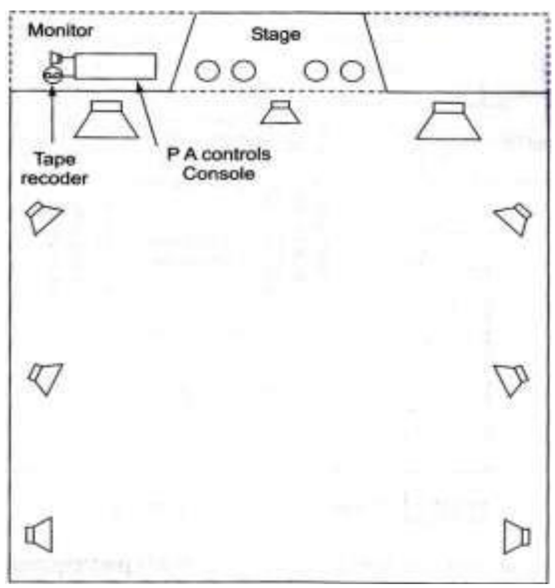
Explanation: **02M**

- Bass would be cut if capacitive reactance in series of signal increases.
- Lower the capacitance, greater will be the reactance ($Xc = ½ \pi f C$).
- 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 C1 is shorted and the Signal directly goes to the next stage, bypassing the capacitor C1 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.

iii) Give the PA system layout and planning for an auditorium.

Ans:

Diagram- **02M**



Explanation: **02M**

- An auditorium may be used for wide range of activities like public meeting, conferences, cultural programmer 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.

iv) Draw block diagram of Hi-Fi system. List any 2 applications of it.
Ans:-

Diagram **03M**

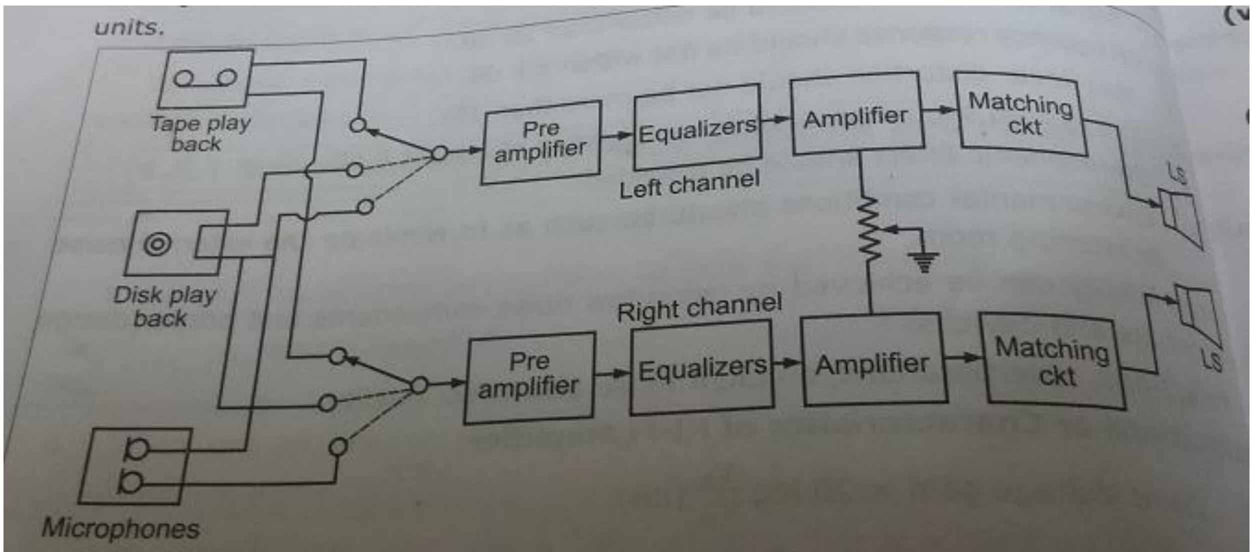


Fig. 1.7: Block Diagram of Hi-Fi Amplifier System

Applications : **01M**

1.Digital Media Players.
2. digital audio players

v) List various causes affecting fidelity of system.
Ans:- **01M each**
Causes affecting fidelity: (Any four)

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

vi) Compare AM and FM (any 4 points).

Ans:- **01M each**

<u>Comparison</u>:

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.

<u>(Note:- Wave form to be considered)</u>