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 judgement 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.

Q.1 a) Answer any SIX of the following : 12 M

i) Define Pitch and Timbre

Answer:

PITCH : Pitch is a characteristic of sound mainly related to frequency. 01 M

Pure tone pitch is determined by frequency alone.

But in speech & music the pitch of sound depends on frequency as well as on intensity.

TIMBRE : 01 M

The proportion of tones & overtones in a sound form the special characteristics by which a particular sound can be recognized. When we hear the sound of a relative or a friend, even if the person is not visible. This quality of sound is called timbre & is related to the proportion in which overtones are present in the sound.

ii) List any four controls of Audio Amplifier.

Answer : **(Each control ½ M)**

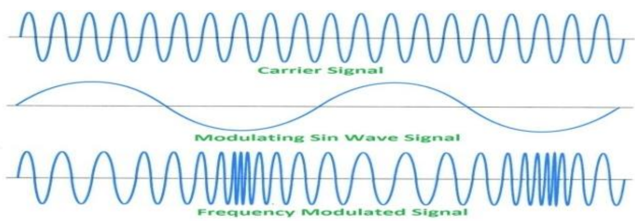
For controls of Audio amplifiers are

1. microphone gain control
2. volume control
3. Bass control
4. Treble control

iii) Define frequency modulation and draw FM wave.

Answer : (Definition 01 M , Diagram 01 M)

The modulation process in which the frequency of the carrier signal changes according to instantaneous value of modulating signal keeping amplitude & phase constant.



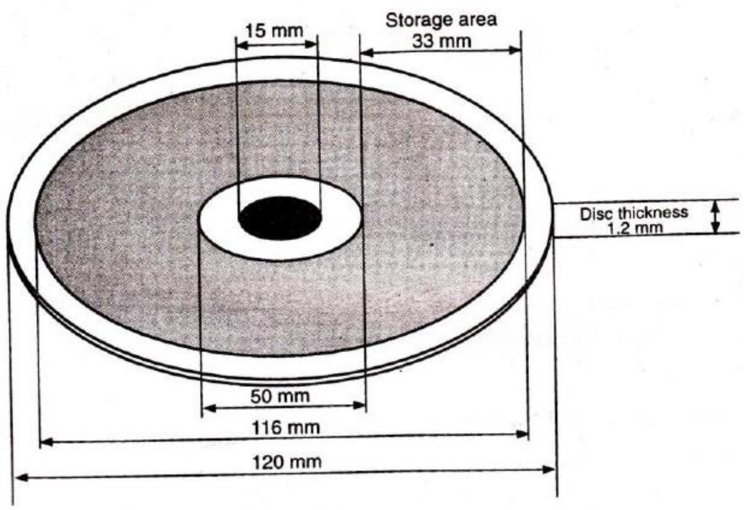
iv) State the principle of magnetic recordings.

Answer : 02 M

Magnetic recording is storage of the sound pressure variations in the form of elementary magnets. Magnetic recording is based on the principle that certain materials (like iron oxide) when brought in a magnetic field, get magnetized and retain that magnetism permanently until altered.

v) Draw and label the structure of CD.

Answer : **(Diagram 01 M and labeling 01 M)**



Construction of a Compact Disc (CD)

vi) Give at least one application of Tie clip microphone and shotguns type microphone.

Answer :

Application of Tie clip microphone: 01 M

1 In small PA systems as in clubs.

Application of Shot gun type microphone (Any One) 01 M

a) Used in field recording of wildlife.
b) Outdoor TV interviews in noisy environment

vii) Define bass and Treble

Answer :

Audio frequency amplifiers are designed to give flat frequency response over the whole audio range from 16 Hz to 20 KHz

Depth of sound is given by low notes called as bass.

Sharpness of sound is given by high notes called as treble.

viii) Give the application of Hi-Fi Amplifier.

Answer : **02 M**

The main application of Hi-fi amplifier is in Hi-fi system and PA system.

b) Answer any TWO of the following. **08 M**

i) Derive the mathematical expression for Amplitude modulated Wave.

Answer : **04 M**

To describe amplitude modulation mathematically, consider the carrier wave given as a sinusoidal wave , with a frequency of fc and amplitude Vc. This can be written as

$$v_c = V_c \sin 2\pi f_c t$$

Without modulation this sine wave will convey no information.

Reason: We can calculate its value at anytime from previously known values.

What modulation does is to modify the constant value with the signal, which carries the information. This results in the amplitude of the modulated carrier varying in proportion to the amplitude of the information signal.

Let the information signal be given by

$$v_s = V_s \sin 2\pi f_s t$$

Let $\omega_s = 2\pi f_s$ and $\omega_c = 2\pi f_c$

then $v_c = V_c \sin \omega_c t$

and $v_s = V_s \sin \omega_s t$

The amplitude of the resulting modulation is the sum of the amplitude of the carrier and the signal.

$$A = V_c + v_s$$

Substituting for v_s

$$A = V_c + V_s \sin \omega_s t$$

$$A = V_c(1 + m \sin \omega_s t)$$

where $m = \frac{V_s}{V_c}$ this is index of modulation

The resulting AM wave will thus be

$$\begin{aligned} v &= A \sin \omega_c t \\ &= V_c(1 + m \sin \omega_s t)\sin \omega_c t \end{aligned}$$

$$v = V_c \sin \omega_c t + m V_c \sin \omega_s t \sin \omega_c t$$

$$v = V_c \sin \omega_c t + \frac{m}{2} V_c \cos(\omega_c - \omega_s)t$$

$$- \frac{m}{2} V_c \cos(\omega_c + \omega_s)t$$

$$v = V_c \sin 2\pi f_c t + \frac{m}{2} V_c \cos 2\pi(f_c - f_s)t$$

$$- \frac{m}{2} V_c \cos 2\pi(f_c + f_s)t$$

The result is AM equation.

ii) Differentiate between AM and FM on the basis of sidebands, modulation index, Noise and Transmitted power.

Answer : **(Each point 01 M)**

Parameter	**AM**	**FM**
1.Sidebands	2	Infinite (more than 2)
2. Modulation Index	m<1	m>1 till 250
3. Noise	AM signal has low noise immunity	FM is higher noise immunity compared to AM.
4. transmitted power	Carrier power & one sideband power are useless.	All the transmitted power are useful.

iii) Give reasons why optical recording is better that magnetic recording system.

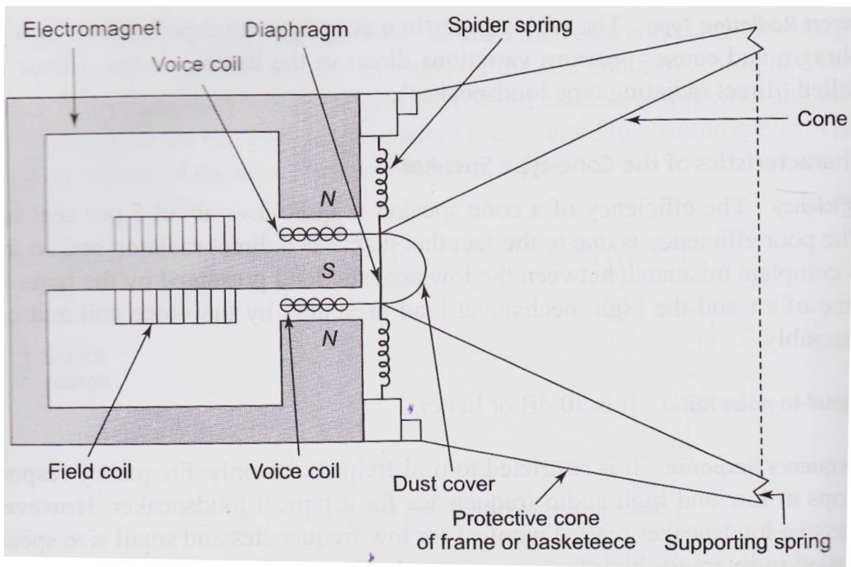
Ans :- Any Four Points **04 M**

1. No Direct contact with film / disc. So no wear and tear.
2. Less maintenance required.
3. Long life.
4. Memory storage is more
5. Less chances of data loss
6. Signal to noise ratio is high
7. High dynamic range (around 90 dB)
8. High channel separation (around 80 dB)
9. Wow does not exist
10. Flutter does not exist
11. Overall distortion is low
12. Excellent frequency response covering complete audio range from 20 Hz – 20 kHz within only 0.5 dB
13. Small size

Q.2 Answer any FOUR of the following. **16 M**

a) Explain the working principle of Electrodynamic loudspeaker with its schematic diagram.

Answer : Diagram of Electrodynamic loudspeakers **2 M**

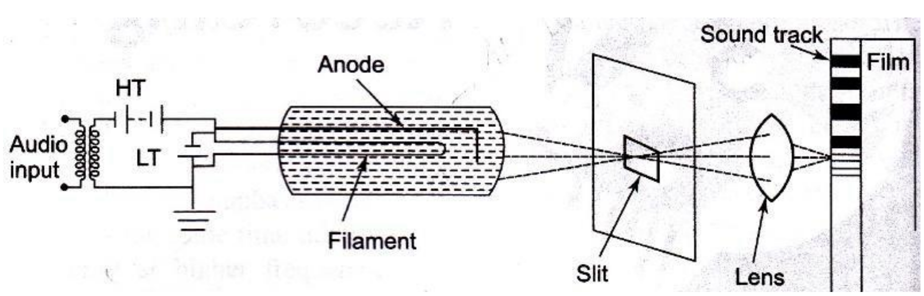


Explanation : **2 M**

- Loudspeakers of more than 25 watt and upto a few hundred watts are of the electrodynamic type.
- The strong and steady magnetic field is produced by a large field coil wrapped around a core.
- The voice coil is wound on the fibre or aluminium. It is placed in a annular gap. The audio signal from the amplifier output transformer is applied to the voice coil. This signal causes a varying magnetic field. The resultant interaction between the two magnetic fields (one due to electromagnet and other due to audio current) produces mechanical vibrations in the coil assembly, which corresponds to the audio signal. The vibration of the coil is transmitted to the attached cone which creates sound waves in the air, in the listener's area and hence radiates sound energy directly.

b) Explain variable density method of optical recording of sound.

Answer : Variable density method: **2 M**



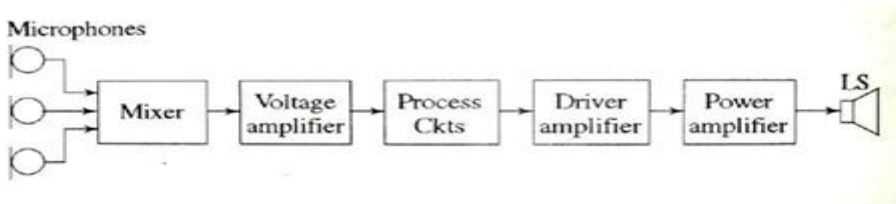
Explanation: **2 M**

- 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.

c) Draw block diagram of PA system and give its functioning.

Answer :

Ans: Public Address system- **01 M**



Explanation :- **03 M**

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 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. Function of preamplifier & amplifiers to amplify weak signals.

3. Voltage amplifiers- Amplifies the output of mixer stage.

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

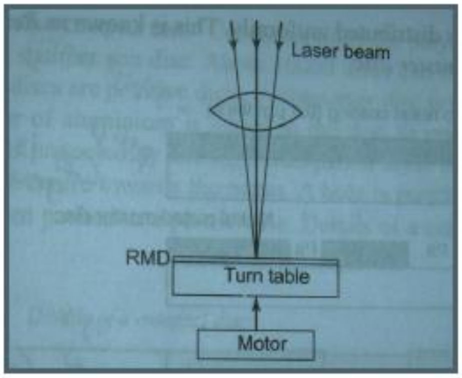
5. Driver amplifier - It gives voltage amplification to the signal to such an extent that when feed to power amplifier (next stages) the into internal resistance of that stage is reduces. Thus drives the power amplifier to give more power.

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.

d) Explain how optical recording has done on compact disc.

Answer: **(Diagram: 2 mark, Explanation: 2 mark)**



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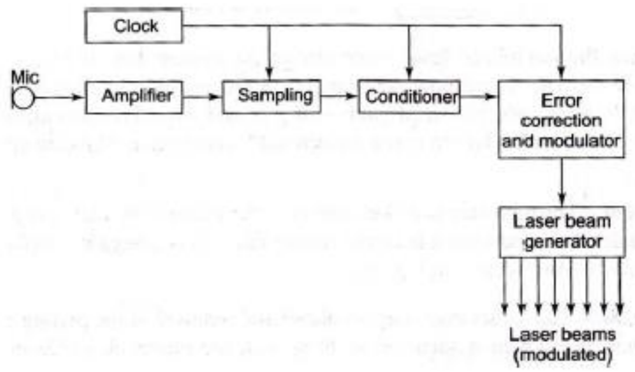


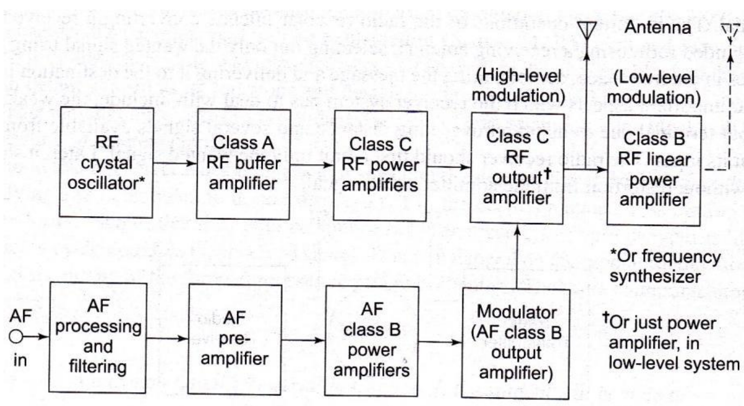
Recording on CD: **2 M**

- This is done with the help of laser beams, made ON and OFF by digitized audio signals
- These beams fall on a photo resist material on a rotating disc and caused pits of varying width & fixed depth & thus records signals in binary form, flats & pits making 1s & 0s respectively.
- Recording is done on Resist Master Disc (RMD) with help of a powerful laser beam as shown as fig. The laser beam is modulated by digitized audio signals. The audio signal is sampled at rate of 44.1 KHz. the quantum level pertains to 16 bits.

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

Answer:

Block diagram of high level & low level AM transmitter: -



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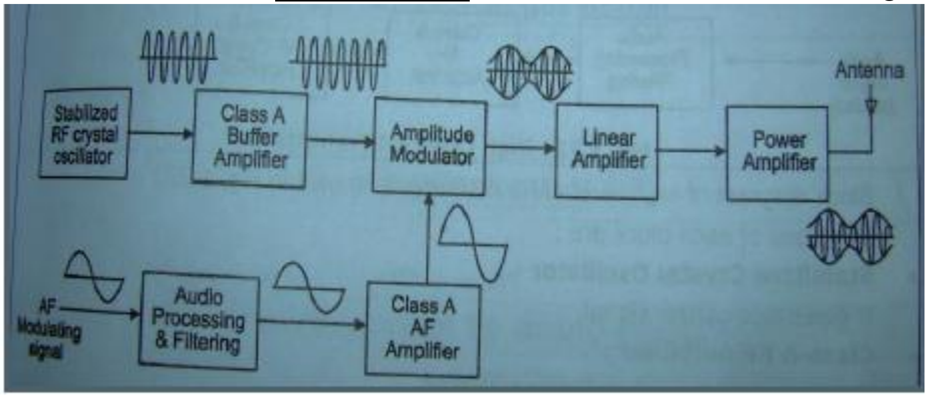
Block diagram of low level AM transmitter:

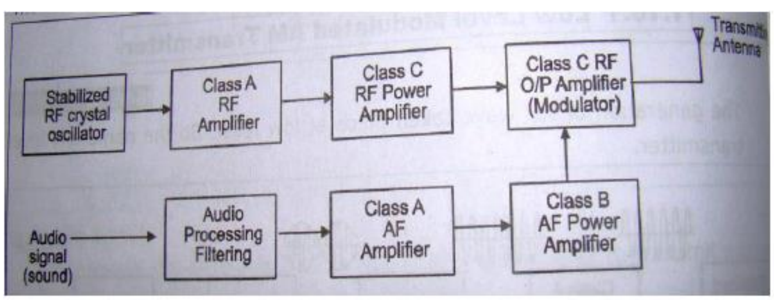


Functions of each block are :(Low level transmitter) **2M**

- 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
- Transmitting antenna : AM wave of high power is transmitted in free space

OR

Block diagram of high level AM transmitter:- **2M**



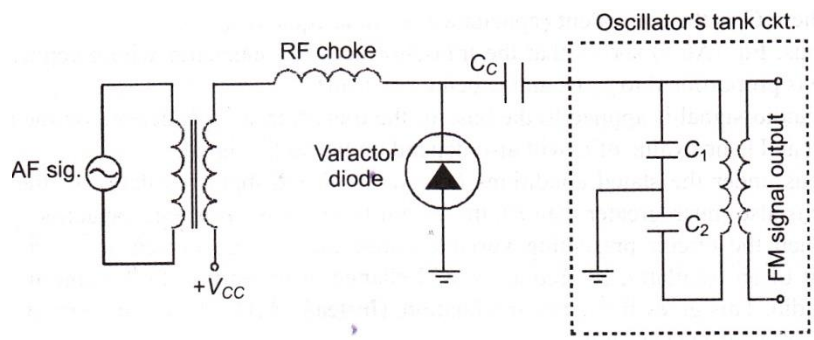
Functions of each block are : (High level Transmitter) **02 M**

- 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.

f) Explain generation of FM using varactor diode modulator.

Answer : Circuit Diagram **2M**



Explanation : 2 M

Varactor diode uses transition capacitance of a diode junction. A fixed DC voltage + V_{cc} is connected across the diode to make its capacitance such that the oscillator circuit connected to it oscillates at the RF carrier frequency. The Oscillator consists of tank circuit which uses C1, C2 and L.

The audio signal is impressed through a transformer so that it changes the value of reverse bias voltage across the varactor diode.

This variable voltage is made available across the tank circuit of the oscillator.

Capacitor C_C couples the varactor diode to the oscillator circuit. RFC is radio frequency choke which prevents the RF Signal going to +V_{cc} supply and to the audio section.

Q3) Answer any FOUR of the following: **16 M**

a) Derive the mathematical expression for power in AM. 04 M

Ans: An AM wave consist of carrier & two sidebands. The two sidebands depends on modulation index `m'. The total power in AM wave is a function of the value of modulation index `m'.
01 M

The total power in an AM wave :

Pt=[carrier power]+[power in USB]+[power in LSB]

$$Pt = \frac{E_{carr}^2}{R} + \frac{E_{USB}^2}{R} + \frac{E_{LSB}^2}{R}$$

When E_{carr}, E_{USB}, & E_{LSB} are the rms values of carrier & R is the characteristics resistance of antenna in which total power is dissipated

Carrier power (Pc)

The carrier power is given by 01 M

$$Pc = \frac{E_{carr}^2}{R} = \left[\frac{Ec/\sqrt{2}}{R}\right]^2 = \frac{Ec^2}{2R}$$

Where Ec = peak carrier amplitude

Power in the sidebands : 02 M

The power in two sidebands is given as

$$P_{USB} = P_{LSB} = \frac{E_{SB}^2}{R}$$

Where E_{SB} = R_{MS} value of sideband magnitude.

Peak amplitude of each sideband in $\frac{mEc}{2}$

$$P_{USB} = P_{LSB} = \frac{\left[\frac{mEc}{2}\sqrt{2}\right]^2}{R} = \frac{m^2Ec^2}{8R}$$

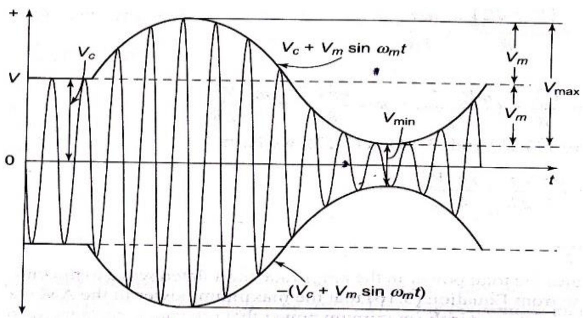
$$P_{USB} = P_{LSB} = \frac{m^2}{4} \text{ x } \frac{Ec^2}{2R}$$

$$P_{USB} = P_{LSB} = \frac{m^2}{4} \text{ x } Pc$$

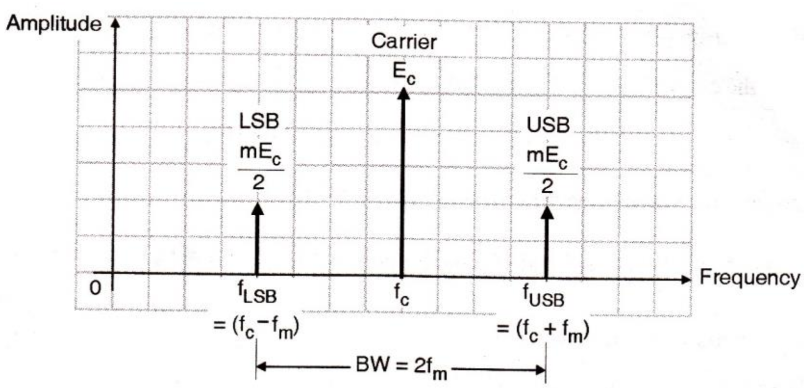
b. Draw time domain and frequency domain spectrum in AM **04 M**

Answer:

Time domain – **2M**

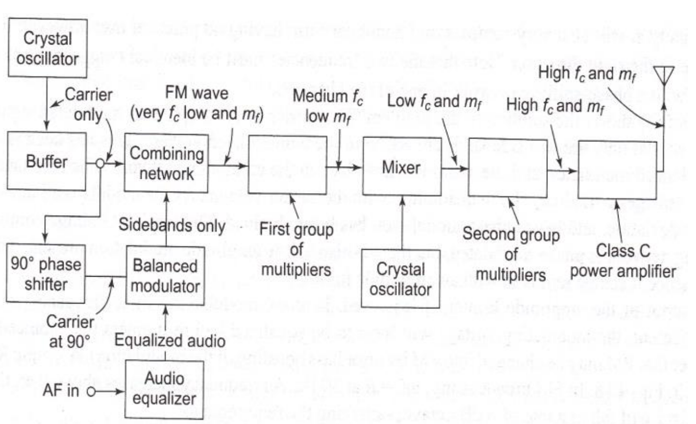


Frequency domain- **2M**



c. Draw block diagram of FM transmitter and give its operation of working. **04 M**

Answer:- Block diagram:- **2M**



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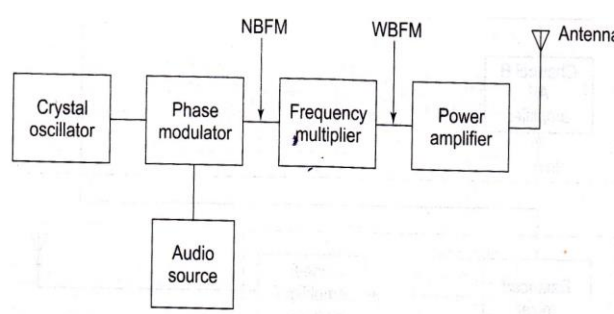


Operation:- **2M**

- 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

d. A 800 watt carrier amplitude modulated to the depth of 70 % . Calculate the total power in the modulated wave and power in sidebands. **04 M**

Answer :$P_c = 800$ 02 M

M=70% = 0.7

Total power

$$P_t = P_c\left[1 + \frac{M^2}{2}\right]$$

$$P_t = 800\left[1 + \frac{0.7^2}{2}\right]$$

$$= 800\left[1 + \frac{0.7^2}{2}\right]$$

= 99600

P_{SB} = P_t - P_c 02 M

= 996-800

= 196 W

$$P_{USB} = P_{LSB} = \frac{m^2\ x\ Pc}{4}$$

$$= \frac{0.7^2 \ x\ 800}{4}$$

= 98 W

e. Explain the use of pre-emphasis and de-emphasis technique as noise reduction tecchnique.04 M

Answer :

Emphasizing low intensity sound before recording is called pre-emphasis the process of de-emphasizing the

Playback circuit to bring originality is called

equalization.

Pre-emphasis:- **02 M**

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.

D**e-emphasis**:- **02 M**

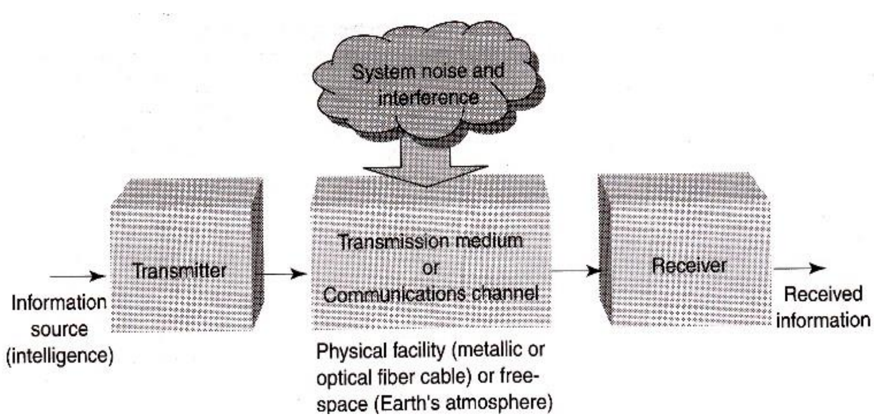
At the receiver, it is essential that the reproduced sound possess the same proportions of

intensities for low & high notes as were present in the original sound, De-emphasis will bring

back the originality.

f. Draw and explain block diagram of communication system **04 M**

Answer :- Block Diagram: **[Block diag. 2M, Explanation 2M]**



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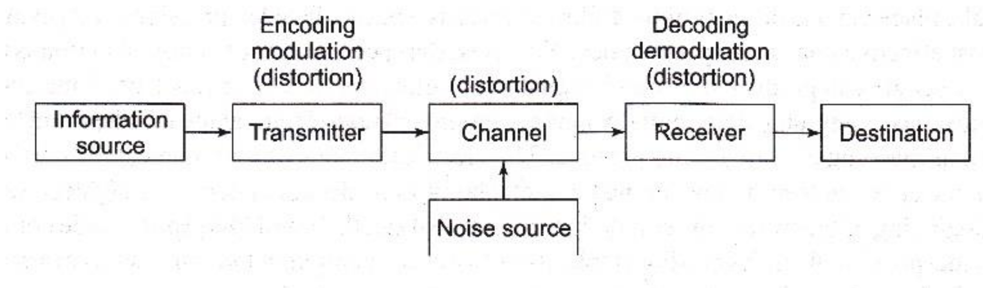


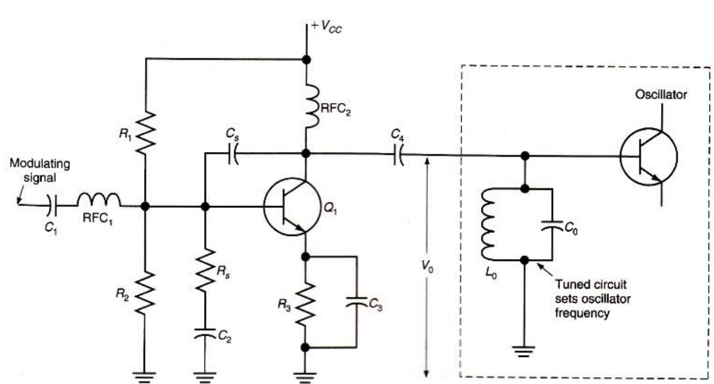
Explanation:

- Figure shows the simplified block diagram of an electronic communication system that includes a transmitter, a transmission medium, a receiver and system noise.
- A transmitter is collection of one or more electronic devices or circuits that converts the original source information to a form more suitable for transmission over a particular transmission medium.
- The transmission medium or communication channel provides a means of transporting signals between transmitter and receiver and can be used simple as pair of copper wires or as complex as microwaves, satellites or optical fiber communication system.
- System noise is any unwanted electrical signals that interfere with information signal.
- A receiver is collection of electronic devices and circuits that accepts the transmitted signals from the transmission medium and then converts those signals back to their original form.

Q 4 Answer any Four of the following. 16M

a) Give working Principle of reactance modulator with the help of circuit diagram.

Answer : C**ircuit diagram: -**
2M



Principle: - 2M

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.

Working:

The modulating signal is applied to the modulator circuit through C and RFC. The RFC helps keep the RF signal from the oscillator out of the audio circuit from which the modulating signal will vary the base voltage and current of Q will also vary in proportional.

As the collector current amplitude varies the phase-shift angle changes with respect to the oscillator voltage, which is interpreted by the oscillator as a change in the capacitance. So as the modulating signal changes the effective capacitance of the circuit varies and the oscillator frequency is varied accordingly the circuit produces direct frequency modulation.

b) List Different types of analog modulation technique and specify why modulation is needed.

Answer: Different types of Analog modulation 2M

Modulation

- **Amplitude Modulation**
  - **DSBSC** **(Double sideband Full carrier)**
  - **DSBSC** **(Double sideband suppressed carrier)**
  - **SSBSC** **(Single sideband supressed carrier)**
  - **VSB** **(Vestigial side band)**
- **Angle Modulation**
  - **FM** **(Frequency modulation)**
  - **PM** **(Phase modulation)**

Need of modulation **2M**

- Low frequency signals cannot be transmitted for long distance that's why there is a need of
- Modulating the information signal in short, it improves the signal strength.
- To reduced antenna heights, noise & distortion
- To narrow banding the signal, to reduce the complexity of the equipment
- To increase the bandwidth of the signal & to multiplexed more number of the signal

c) Define any four specification of PA system

Answer : Public Address system- **4M**

1) Power O/P : 75W
2) I/P channels : 3*mic 0.6mv/ 4.7kΩ
   1x aux 100mv/330kΩ
3) Frequency response. 50Hz -15khz , ± 3db
4) S/N ratio : 55db
5) Tone control
   Bass: ± 10db at 100Hz
   Treble: ± 10db at 10khz
6) O/P : line 400mv / 3.57kΩ
7) Speaker O/P = 4Ω , 8Ω ,16 Ω and 100 V
8) Digital player : MP3 player with usb reader.
9) Power supply Ac: 220-240V,50/60Hz
   Dc: 12-14V car battery
10) Dimensions : W305 x H12xD 305mm
11) Weight: 7.0 kg

Note

- Students can write down any 4 specifications
- Above specification are for Ahuja P.A system student can write specifications of any other P.A system.

d) Draw 3-way cross over network and give its working

Answer:

Working **2M**

When a multi way loudspeaker system is used to get flat frequency response for the entire range of audio frequencies, it is essential to have a crossover network to divide the incoming signal in to separate frequency ranges for each speaker.

In the absence of crossover networks, 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.

Crossover networks make use of the fact that the capacitive reactance decreases with increase in frequency

[$X \alpha 1/fC$)], and the inductive reactance increases with increase in frequency ($X \alpha fL$).

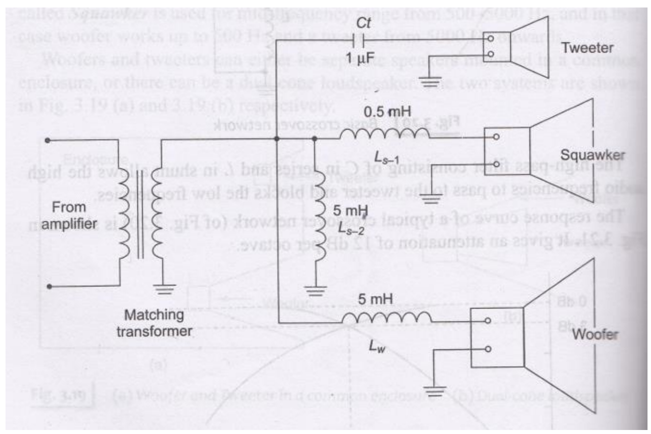
Circuit diagram and frequency response of three-way cross over network give the values of L and C Where, R1 is the impedance of a loudspeaker in ohms and fc is the crossover frequency in Hz, L is The inductance and C, the capacitance of LC circuits.

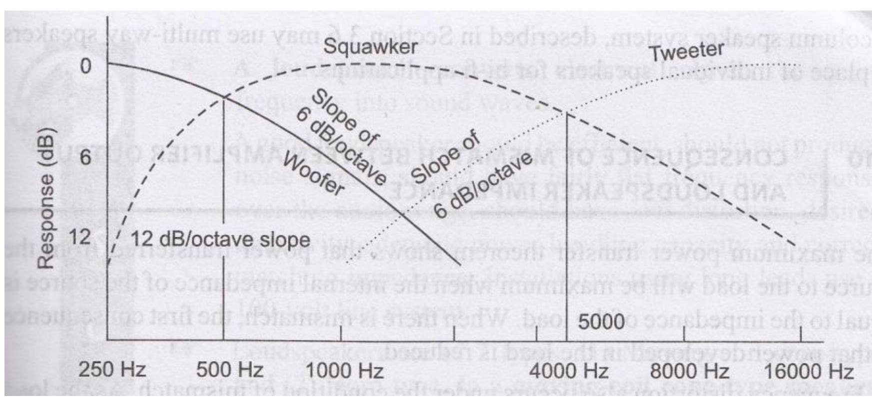
A commercial three-way divider network is shown in Fig. In this circuit the capacitor C1 of 1µF in series with the tweeter prevents low and mid-frequencies from reaching the tweeter. Similarly, the inductance Lw of 5 mH in series with the woofer prevents high frequencies from reaching the woofer. Inductances Ls1 and Ls2 of 0.5 mH and 5 mH, respectively in the squawker circuit allow only mid-frequencies and prevent too low and too high frequencies from reaching the squawker.

A typical divider curve for a three-way network of Fig. is shown in Fig.

A single element in filtering gives attenuation of 6 dB per octave and double element in filtering gives attenuation of 6 dB per octave and double elements give 12 dB octave.

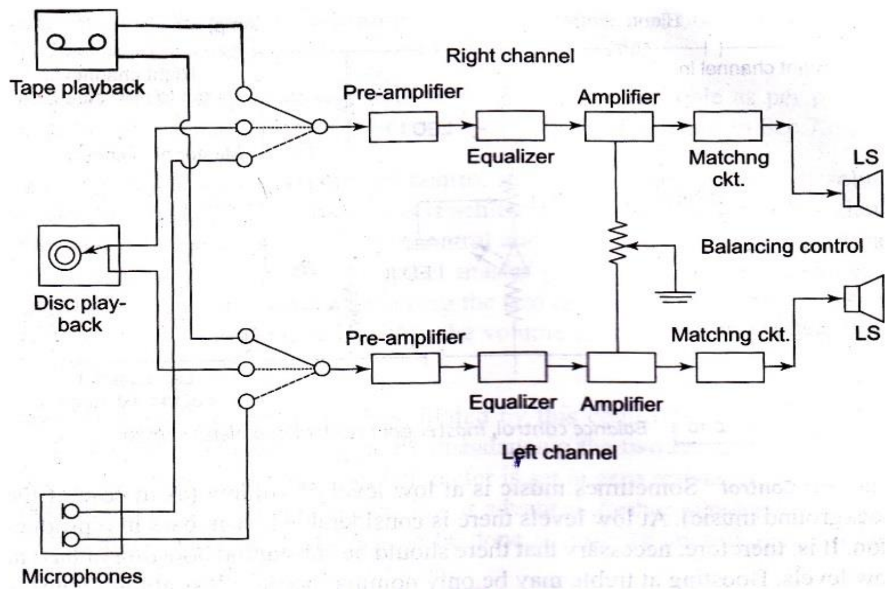
Diagram **(2M)**





e) Draw the block diagram of Hi Fi amplifier and give function **2M**

Answer:



Explanation: - **2M**

- Fidelity means faithfulness. In audio system it is used to indicate faithful reproduction of sound. Figure shows block diagram of Hi- Fi system
- High fidelity sound can be obtained from the recorded stereo tape or in live system from the microphone or from record player.
- The stereo signal is fed to two independent amplification channels through a tape-mic switch . The amplifier consists of a low noise high gain preamplifier, equalizer, well designed amplifier giving flat frequency response & little distortion by using negative feedback circuit & then the matching transformer.
- A balancing circuit is incorporated to balance out any imbalance in the characteristics of identical circuits.
- The secondary of the matching transformer of each channel is connected to the respective loudspeaker column.
- For hi fi the L.S columns consisting of woofer, squawker & tweeter are used.

f) Explain how compact disk is prepared with relevant schematic diagrams

Answer:

Preparation of compact Discs consist of following important stages **2M**

1) Preparation of resist master disc-: in this stage a master disc made up of optically ground glass disc is used. The glass is polished and spotlessly clean. It is coated with photoresist compound. The coating is 0.12am thick and is distributed uniformly when modulated laser beam is allowed to strike

this disc, it reacts with the photoresist. The disc is then developed by a process similar to photography ie microscopic size pits and flats are created an the disc. The developed disc is coated with silver to make it electrically conductive

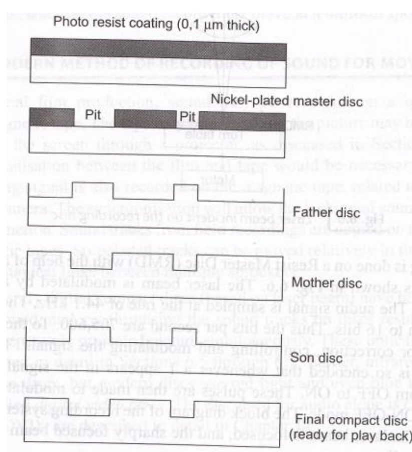
2) Preparation of father disc: The master disc is then plated with nickel. After plating the nickel is peeled off the master disc and then it is called father disc. It is a negative replica of master disc.

3) Preparation of 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 10 mother disc are obtained from a single master disc.

4) Formation of son disc or stamper: the mother discs are plated and when the plating is removed, it gives son disc or stamper. It is identical with the father disc. Several sons are obtained from single mother. It is also called as negative nickel plated stamper.

5) Preparation of final compact disc-: consumer discs are obtained by pressing on the stamper son disc. About 10000 discs can be modulated form one stamper. It is made up of polycarbonate. In order to make it reflective, a thin layer of aluminum is added. A transparent layer of lacquer is also added for protection of disc.

Diagram: **2M**



Q 5 Answer any Four of the following. **16M**

a) Differentiate between direct and indirect methods of FM generation. **4M**

Answer:

Direct method	Indirect method
1) In direct FM 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.

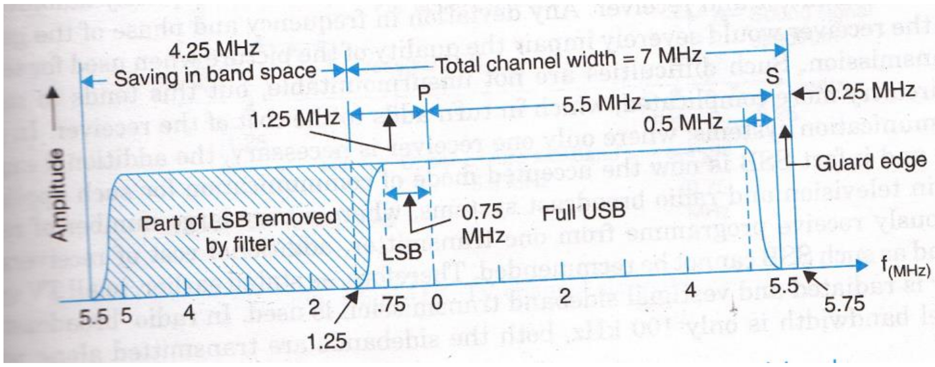
b) What is VSB? Give its application. Draw the VSB in spectrum AM.

Answer: 1M

- During transmission of AM signal, many times instead of transmitting complete frequency sideband contain same information, this do not create any problem.
- But during transmission a part of LBS also get transmitted along with usb (vestige). This is because it is very difficult to desimafilter with very share cut off.
- This type of transmission is called as vestigial sideband transmission

Application -: In T.V transmission system for transmitting picture information 1M

VSB Spectrum diagram 2M



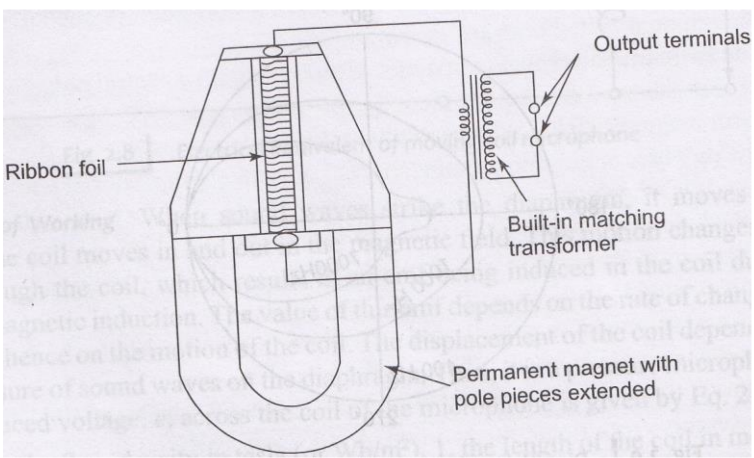
c) Explain construction and working principle of ribbon microphone

Answer: **Explanation:**

Construction: - **(2M)**

- The main parts of ribbon microphone are permanent magnet, ribbon conductor
- The permanent magnet is specially designed horse shoe magnet with extended pole pieces. It provides a strong magnetic field.
- The ribbon is a light aluminum foil. it is corrugated at right angle to its length to provide greater surface area.
- The main feature is the lightness of the ribbon which is about 0.2 mg in weight less than 1 micron and about 20mm long and 3 mm wide.
- It is suspended in the magnetic field of the permanent magnet.
- The whole unit is enclosed in circular or rectangular baffle.

Diagram.



Principle of working: - **(2M)**

- When the ribbon conductor placed in the magnetic field, is made to move at right angles to the magnetic field by the force of sound pressure, there is a change of magnetic flux through the ribbon conductor.
- Due to this change , an e. m. f is induced across the ribbon this e. m. f is proportional to the rate of change of magnetic flux which in turn proportional to the force of sound waves striking the ribbon.
- It is also called as pressure gradient or velocity microphone.

Note: It is not necessary to draw diagram. But if Student draws diagram and NOT write down construction, 1M can be given.

d) Compare complementary symmetry push-pull amplifier with symmetry push-pull amplifier.

Answer: **(each point 1M)**

Sr. no	**Complementary symmetry push-pull amplifier**	**Symmetry push-pull amplifier**
1	Circuit consists of two transistors, one PNP & other NPN.	The circuit consist of two similar transistors(both, PNP or both NPN)
2	I/P is fed directly to bases of the transistors	I/P is applied through a transformer
3	Circuits do no use transformer that it is cheap.	Use of transformer makes circuit costly
4	Circuit is small in size & light in weight	Circuit is bulky & heavy
5	O/P is free from noise.	Noise is present in the O/P
6	O/P is free from distortion	O/P is distorted particularly at high frequencies
7	No such resonance effect is present	It gives resonance effect at particular frequently due to inductance transformer coil and its self-capacitance
8	Circuit requires two power supplies.	Circuit requires only one power supply.

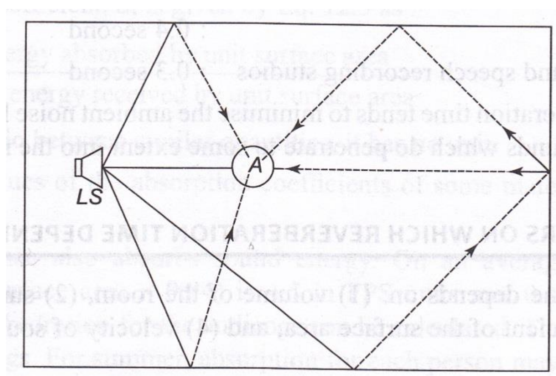
Note; student can write any 4 points.

e) What is reverberation? Explain the necessity of reverberation

Answer: **Reverberation:** **3M**

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.

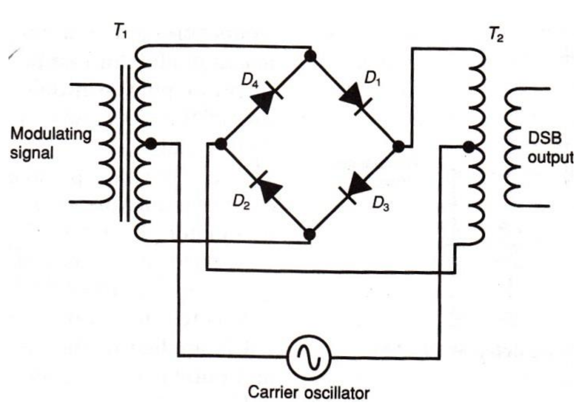
Reverberation to some extends is pleasing & should be incorporated in the design of rooms.

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 generation of DSBSC AM signals using balanced modulator with circuit diagram.

Answer: **Diagram** **2M**



Explanation: **2M**

A balance modulator generates a DSB signal. The input to the balance modulator is the carrier and a modulating signal. The output of a balance modulator is the upper and lower side bands with suppresses the carrier.

It consists of an input transformer T_1 an output transformer T_2 and four diode connected in bridge circuit. The carrier signal is applied to the center tops of the input and output transformer. The modulating signal is applied to the input transformer. The modulating signal is applied to the input transformer $T_{1.}$ The output appears across the secondary of the output transformer $T_{1.}$

Assume that the modulating input is zero when the polarity to the carrier is shown in fig. diode D_1 and D_2 are forward biased. At this time D_3 and D_4 are reverse biased and act like open circuit as you see current divides equally in the upper and lower portions of the primary winding of $T_{2.}$ The current in the upper part of the winding produces a magnetic field that is equal and opposite to the magnetic field produced by the current in the lower half of the secondary. Therefore these magnetic fields cancel each other out and no output is induced in the secondary. Thus the carrier is effectively suppressed.

Q. 6 Answer any Four of the following **16 M**

a. Give reason why multiway speaker system is needed for good sound quality. **04 M**

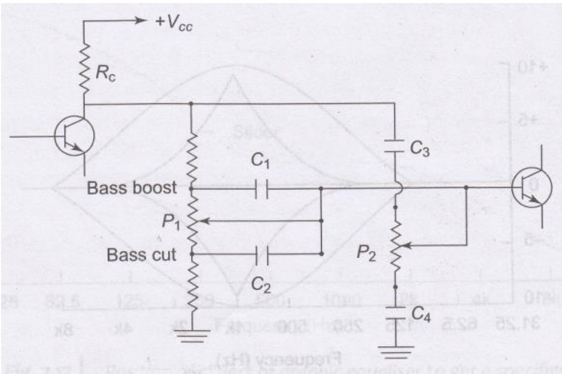
Answer :

- A single loudspeaker cannot have flat response for the whole audio frequently range from 16 Hz to 2000Hz.
- Low frequencies are weakened by the back sound waves of reverse phase in open speaker
- The loss at high frequencies is due to mass or inductive effects of the diaphragm.
- Thus a single speaker cannot produce both the good solid bass and the smooth crisp treble.
- To solve the problem audio frequency spectrum is divided into two or three parts. Speakers are designed to cover a small frequency range. Hence multiway speaker is required for good sound quality.

b. Draw circuit diagram of tone control and explain how bass and treble controlled in sound system.

Answer : **(Diagram 02 M Explanation 02 M)**

The potentiometer in series with capacitor forms the treble control. When slider P2 is at the lower end, maximum signal develops. As slider is moved upwards, less resistance of the potentiometer comes in series and there is more cut in the high frequency signals. Cut is maximum when slider is at the top end, short circuiting the potentiometer completely. This position is called treble cut. The other position where treble cut is minimum is called treble boost.

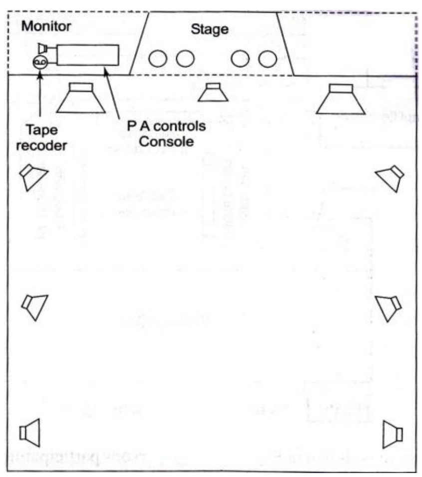


Bass and treble control

Bass would be cut if capacitive reactance in series of signal increases. Lower the capacitance greater will be reactance. When slider P1 of potentiometer R is at upper end, the capacitor C1 is shorted and signal directly goes to next stage, bypassing capacitor C1 and hence bass has minimum attenuation. It is called bass boosts when slider is at the lower end, capacitor C1 in parallel with whole resistance 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. Give the PA system layout and planning for an auditorium. **04 M**

Answer: - PA system installation for an auditorium. **[Diagram 2M, Explanation 2M]**



Explanation:

- 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.

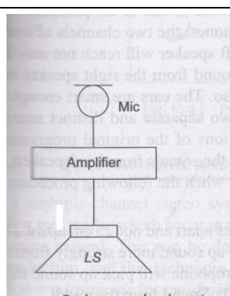
d. What is stereophony ? Give the difference between monophony and stereophony ststem with the help of block diagram.

Answer : Definition of stereophony 02 M

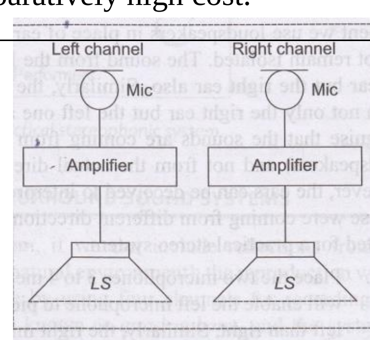
The word stereophony is derived from two Greek words stereos and phone meaning solid and sound respectively. Thus it means solid sound or three dimensional sounds. In a programme different sources of sound are placed at different positions on the stage. When such a programme is amplified and reproduced, the originality of sound would be restored if the reproduced sound appears to come from different directions simulating the original programme. This three dimensional reproduction is called stereo

Difference between Monophony amplifier and stereo phony amplifier (Any two points) 02 M

Monophony amplifier	**stereo phony amplifier**
1. Only one amplifier is used. Single amplifier stage is known as mono amplifier (as shown in fig.)	1. At least two independent amplifiers are used. These part of amplifiers is called as stereo signal (as shown in fig.)
2. No naturalness	2. Provides naturalness of sound signal.
3. Listener cannot judge the direction of sound	3. Listener can judge the direction of souond
4. Low cost	4. Comparatively high cost.



Basic monophonic system



Basic stereophonic system

e : Compare power amplifier and voltage amplifier. **04 M**

Answer :(any four points 1M for each)

Parameter	Voltage amplifier	Power amplifier
1. Operation region	Linear	Linear as well as non linear
2. Input signal	Small	Large
3. Signal distortion	No	Yes
4. Harmonies in output	Not present	Present
5. Power transistor	Not required	Required
6. Power	Low	High
7. Heat links	Not used	Essential.
8. Output impedance	High	low

f : What is the bandwidth required for FM signal in which modulating frequency is 3 KHz and maximum deviation is 15 KHz.(No. of sidebands 8)

Answer :

According to carson's Rule

$B.W. = 2\ [Fd(\max) + fm(max)]$ 01 M

$= 2\ [15KHz + 3KHz]$

= 36KHz 03 M

OR

Band width = 2 * fm * N 01 M

Where N = 8 side bands

BW = 2 x 3 x 8

= 48 KHz 03 M