MODEL ANSWER

WINTER– 17 EXAMINATION

Subject Title: Fundamental Of Communication Subject Code: 17316

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. No.	Sub Q.N.	Answer	Marking Scheme
Q.1	**A**	**Attempt any six.**	**12-Total Marks**
	a)	**Define the term Reverberation.**	**2M**
	Ans:	**Reverberation:-** Due to multiple reflections from walls, ceiling, floor etc. the sound in an enclosure fades away only gradually after the source of sound stops. This continuing echo is called reverberation.	**2M**
	b)	**Define pitch and overtone.**	**2M**

Ans: **Pitch :** — **1M each**

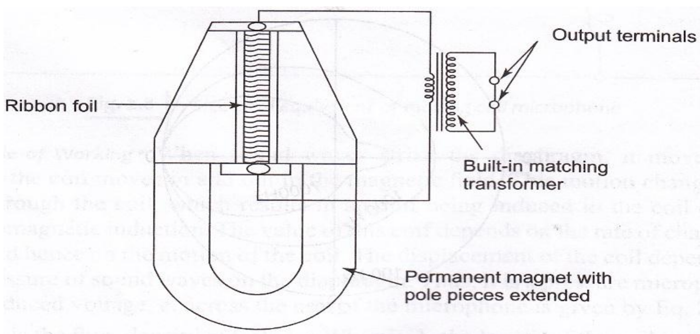
It is a characteristic of sound mainly related to frequency, in pure tones without harmonics, it represents frequency alone, but with harmonics it is related to intensity also in addition to frequency.

Overtone:-

Frequencies other than fundamental frequencies are called overtones.

c) **Draw the neat circuit diagram showing constructional details of ribbon microphone.** — **2M**

Ans: **Ribbon microphone:-** — **Labeled diagram 2 M**



d) **State the characteristics of audio amplifier (any two).** — **2M**

Ans: **Characteristics of audio amplifier:-** — **(Any two) 1M each**

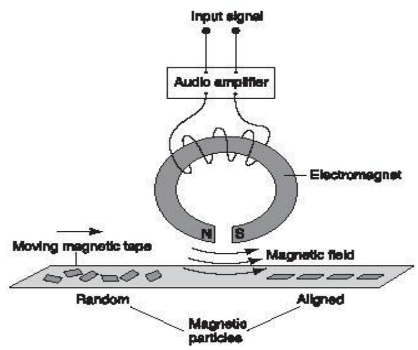
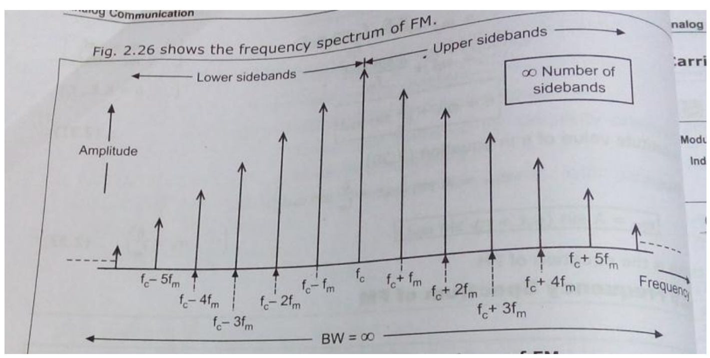
1. Gain
2. Bandwidth
3. Distortion
4. Power output
5. Impedance

e) **What is the difference between parametric and graphic equalizer?(any two)** — **2M**

Ans: — **(Any two) 1M each**

Sr. No.	Parametric equalizer	Graphic equalizer
1.	It provide variable boost or cut up to about 15dB.	Each band has individual slider control which can cut or boost the signal from +15 to -15dB.
2.	The parameters like frequency and bandwidth are varied throughout the audio spectrum (16Hz to 20KHz)	Here, complete audio spectrum is divided into narrow bands.
3.	The frequency response can be adjusted very precisely and selectively	The shape of the response curve is given by joining the slider positions

f) **State the principle of magnetic recording.** — **2M**

<table>
<tr><td></td><td>Ans:</td><td>Principle of magnetic recording:-
Magnetic recording is storage of the sound pressure variations in the form of elementary magnets whose length and strength depend on audio signals. When certain material like iron oxide comes in contact with the magnetic field, get magnetized and retain that magnetism permanently until it is changed.
The sound pressure variations are recorded on the magnetic tape in the form of elementary magnet or varying magnetic field.

</td><td>2M
(Diagram is optional)</td></tr>
<tr><td></td><td>g)</td><td>Draw the frequency spectrum of the FM wave.</td><td>2M</td></tr>
<tr><td></td><td>Ans:</td><td>Frequency spectrum of the FM wave:-

</td><td>2M</td></tr>
<tr><td></td><td>h)</td><td>What is the bandwidth required for FM signal in which modulating frequency is 2kHz and the maximum deviation is 10 kHz (No. Of side band =8).</td><td>2M</td></tr>
<tr><td></td><td>Ans:</td><td>Solution:- Given, fm =2KHz. δmax=10KHz, Number of sidebands

Formula used:-Bandwidth=2(δmax +fm) OR Bandwidth =2(Δf +f_m)

=2(10+2)

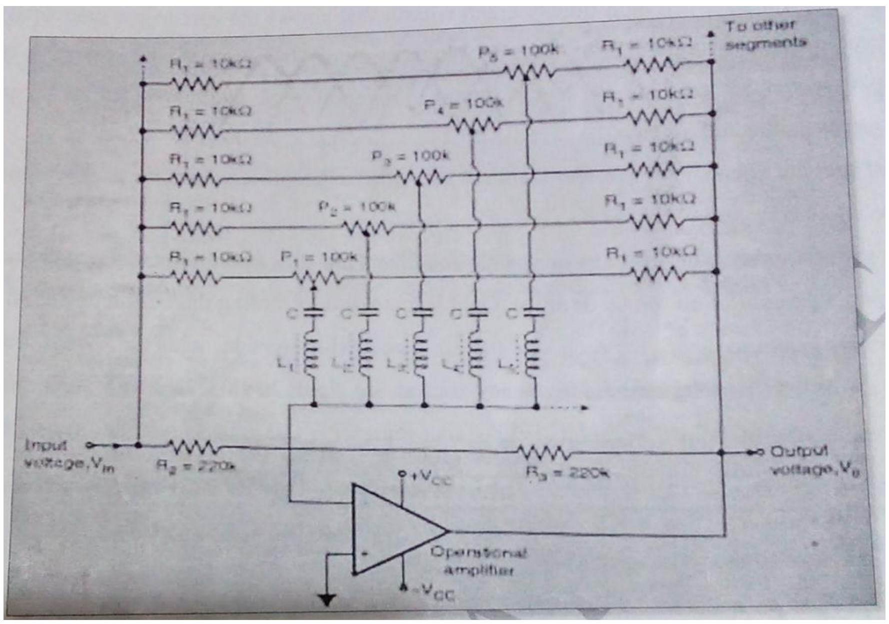
=24 KHz

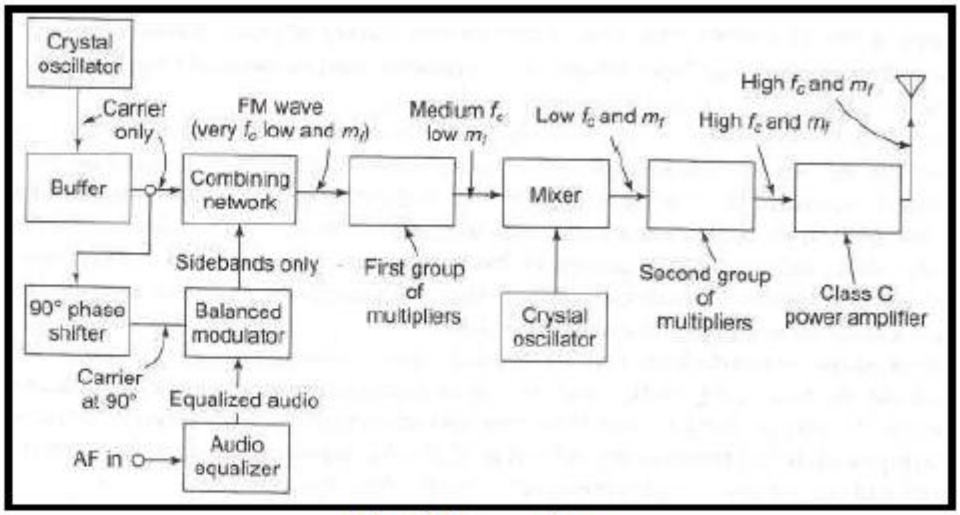
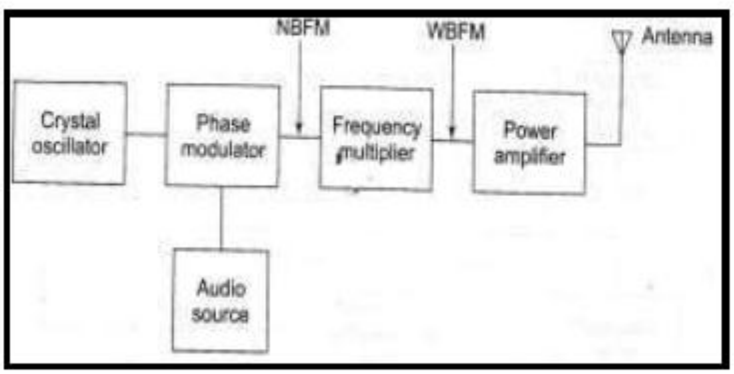
Bandwidth of FM=24 KHz

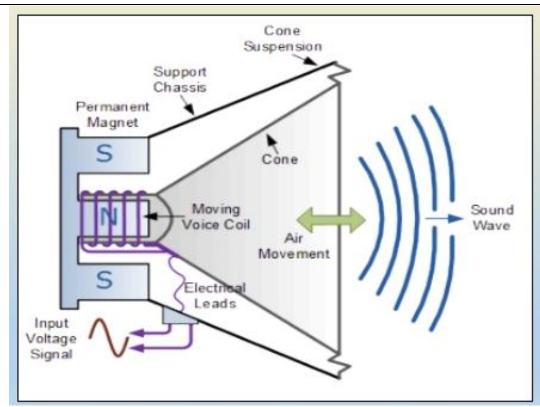
<u>OR</u></td><td>

1M

1M</td></tr>
</table>

	Bandwidth = 2(Number of sidebands * fm) = 2(8*2000) =32KHz	
B)	**Attempt any Two.**	**8-Total Marks**
a)	**Draw the well labeled diagram of graphic equalizer.**	**4M**
Ans:	**5-Point graphic equalizer:**  	**3M diagram** **1M labelling**
b)	**Define amplitude modulation. Explain the need for modulation in communication system.**	**4M**
Ans:	**Amplitude modulation:** It is the technique of modulation in which the instantaneous amplitude of carrier signal varies in accordance with amplitude of modulating signal keeping frequency and phase of carrier signal constant. **Need for modulation:** 1. Reduction in height of antenna. 2. Avoids mixing of signals 3. Increase the range of communication 4. Multiplexing is possible	**2M** **2M (any 4 Points)** .

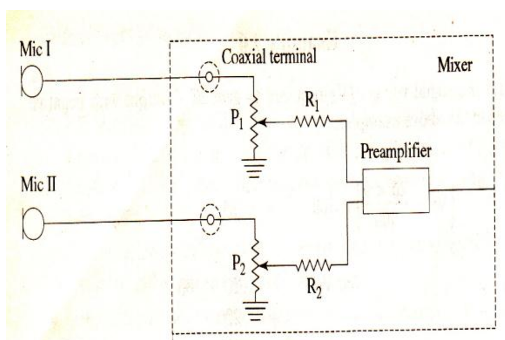
		5. Improves quality of reception	
	c)	**Draw the block diagram of indirect method of generation of frequency modulation.**	**4M**
	Ans:	 **Fig. FM transmitter** **OR** 	**1M label & 3M diagram**
Q 2		**Attempt any four.**	**16-Total Marks**
	a)	**Draw the construction of moving coil cone type loud speaker and give its working principle.**	**4M**
	Ans:	**Construction of moving coil cone type loud speaker:-**	**2M**



Working principle: **2M**

It works on the principle of interaction of magnetic field and current in the same way as an AC motor works. A coil called voice coil is placed in a uniform magnetic field, when the audio signal current passes through the voice coil there is an interaction between magnetic field and current resulting in a force working on the movable coil. This force is proportional to the audio current and hence vibratory motion in the coil.

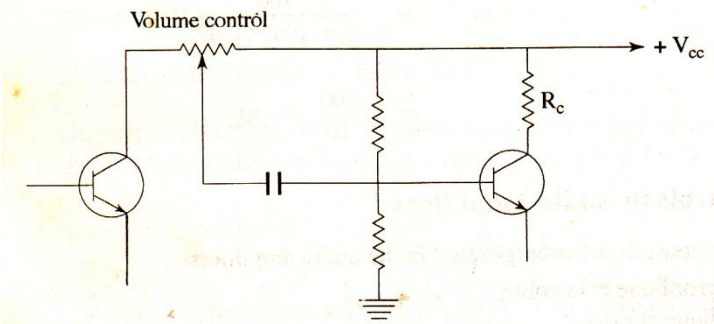
b)	**Draw the circuit diagram of Audio with different controls, stating the function of each.**	**4M**
Ans:	**Microphone gain control:-**	**(Any two circuits)** **2M each**



Its function is to adjust th microphone output depending upon speaker’s style of speaking.

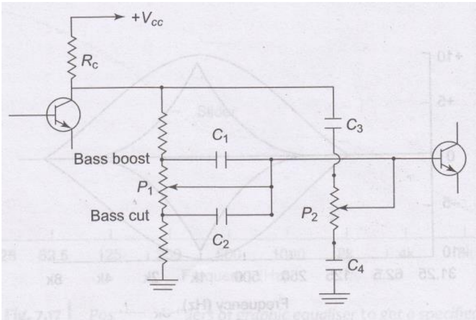
Volume control:-

It is used after the mixer stage to control overall volume of the amplifier output.

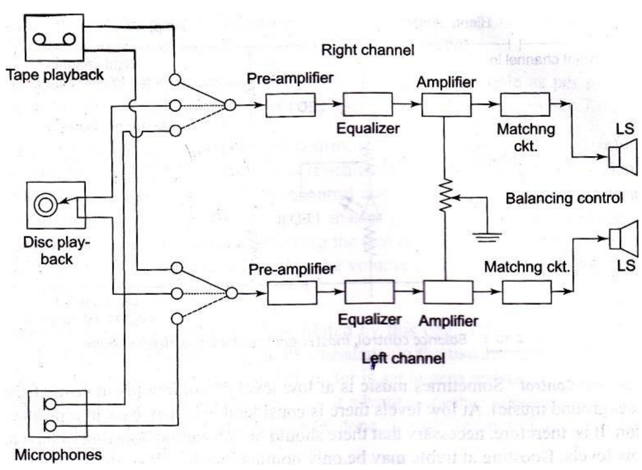


Tone control:-

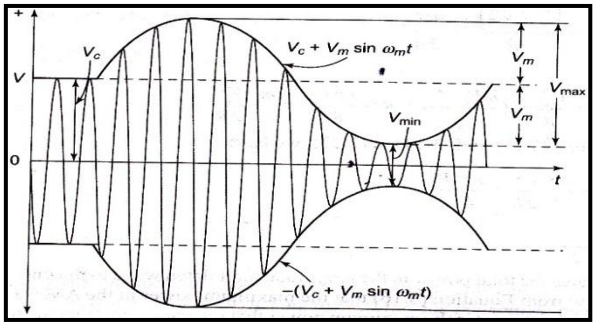
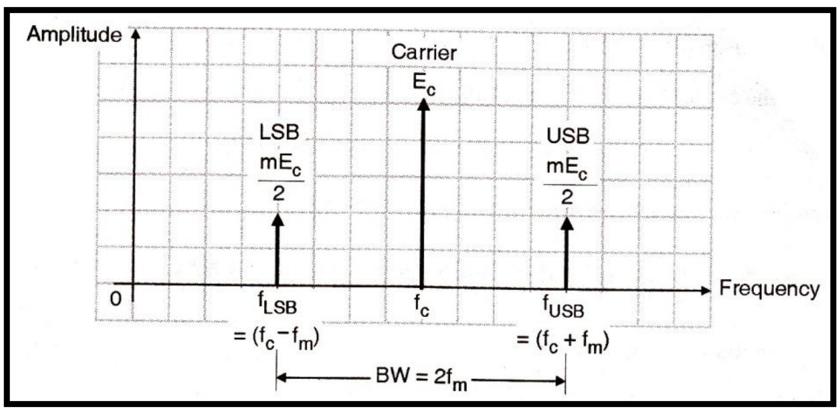
It is the combination of bass and treble control which is used to get flat frequency response over the whole audio range from 16Hz to 20KHz so that people can hear low notes called bass and high notes called treble by adjusting this tone control.



Bass and treble control

c)	**Draw and explain the block diagram of a Hi-Fi system.**		**4M**
Ans:	**Block diagram of a Hi-Fi system:-** 		**2M**

		Explanation: • 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.	**2M**
	d)	**Why pre-emphasis and De-emphasis circuits are used for noise reduction? (Four points).**	**4M**
	Ans:	• In processing electronic audio signals within a frequency band, pre-emphasis refers to a system designed to increase the magnitude of some higher frequencies with respect to the magnitude of other lower frequencies in order to improve the overall signal-to-noise ratio by minimizing the adverse effects of such phenomena as attenuation distortion or saturation of recording media in subsequent parts of the system. • The noise suppression ability of FM decreases with the increase in the frequencies. Thus increasing the relative strength or amplitude of the high frequency components of the message signal before modulation is termed as Pre-emphasis • In the de-emphasis circuit, by reducing the amplitude level of the received high frequency signal by the same amount as the increase in pre-emphasis is termed as De-emphasis. • The pre-emphasis process is done at the transmitter side, while the de-emphasis process is done at the receiver side. • Thus a high frequency modulating signal is emphasized or boosted in amplitude in transmitter before modulation. To compensate for this boost, the high frequencies are attenuated or de-emphasized in the receiver after the demodulation has been performed. Due to pre-emphasis and de-emphasis, the S/N ratio at the output of receiver is maintained constant. • The de-emphasis process ensures that the high frequencies are returned to their original relative level before amplification. • Pre-emphasis circuit is a high pass filter or differentiator which allows high frequencies to pass, whereas de-emphasis circuit is a low pass filter or integrator	**(Any Four points)** **1M each**

		which allows only low frequencies to pass.	
	e)	**Draw the time domain and frequency domain spectrum of AM wave.**	**4M**
	Ans:	**<u>Time domain representation of AM:</u>**  **<u>Frequency domain representation of AM:</u>**  	**2M** **2M**
	f)	**Define:** **1) Frequency deviation** **2) Modulation index** **3) Deviation ratio and** **4) Percentage modulation for FM wave.**	**4M**
	Ans:	• **Frequency deviation:-** The amount by which the carrier frequency varies from its unmodulated value is called frequency deviation. • **Modulation index:-** It is defined as the ratio of frequency deviation(δ) to the modulating frequency ($f_{m)}$	**1M each**

- **Deviation ratio:-**

The modulation index corresponding to maximum deviation and maximum modulating frequency is called deviation ratio.

- **Percentage modulation for FM wave:-**

It is defined as the ratio of the actual frequency deviation produced by the modulating signal to the maximum allowable frequency deviation.

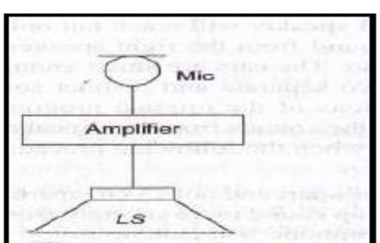
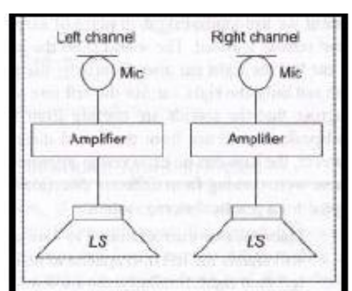
Q 3		Attempt any four.	16-Total Marks
	a)	**Explain the concept of stereophony. What is the difference between monophony and stereophony**	4M

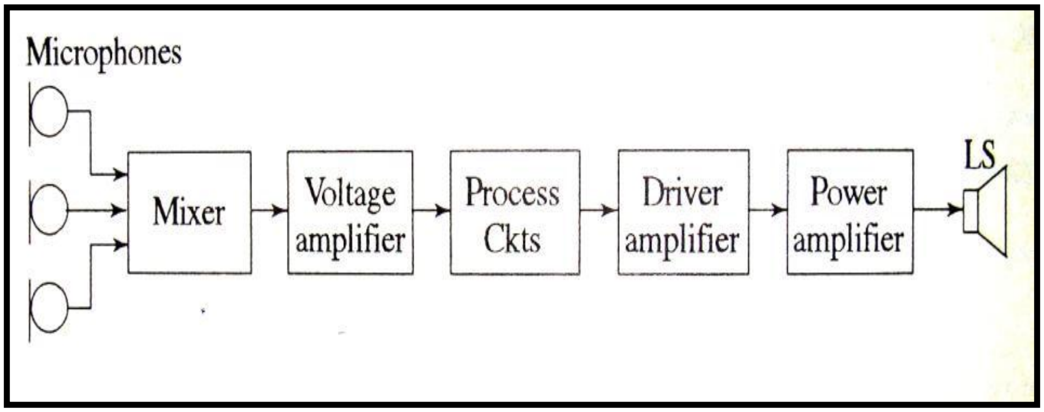
Ans:

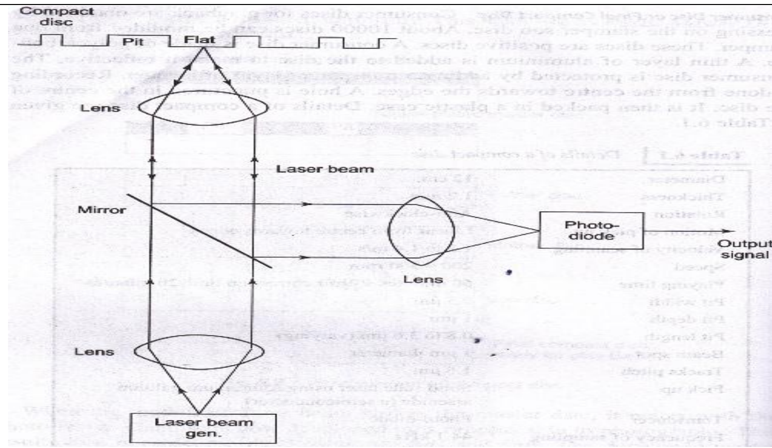
Concept of stereophony: (1M)

Stereophony is a method of sound reproduction that creates an illusion of multi-directional audible perspective. This is usually achieved by using two or more independent audio channels through a configuration of two or more loudspeakers (or stereo headphones) in such a way as to create the impression of sound heard from various directions, as in natural hearing

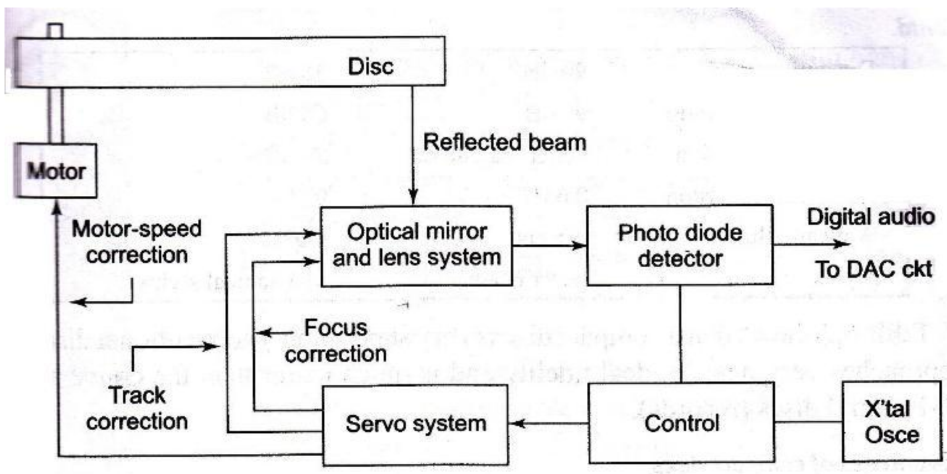
Difference between monophony and stereophony:- (Any 3 points) 1M each

Monophony	Stereophony
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.
 Fig. Basic monophonic system	 Fig. Basic stereophonic system

b)	**With neat block diagram explain the working of public address system.**	**4M**
Ans:	**Block diagram of public address system:-** 	**2M**
	Working:- **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 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.	**2M**
c)	**What is meant by detection in optical sound recording? Describe its operation.**	**4M**
Ans:	**Detection in optical sound recording:-** A layer beam is incident on the compact disc through a half silver mirror. The returning beam is reflected from the aluminum flat surface and represents the logic 1. There is only a little reflection from a pit and it represents logic 0. The binary digits are reproduces when this ON-OFF reflected light falls on a photosensitive diode. The digital output of the diode is analog signal by using digital to analog converter.	**2M**

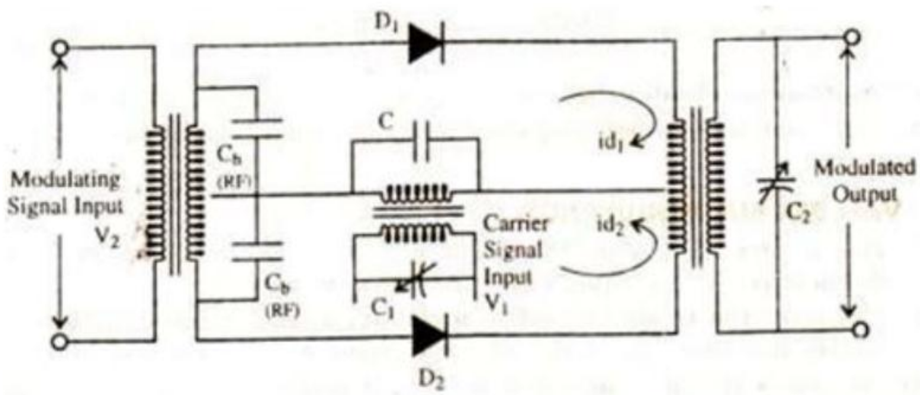
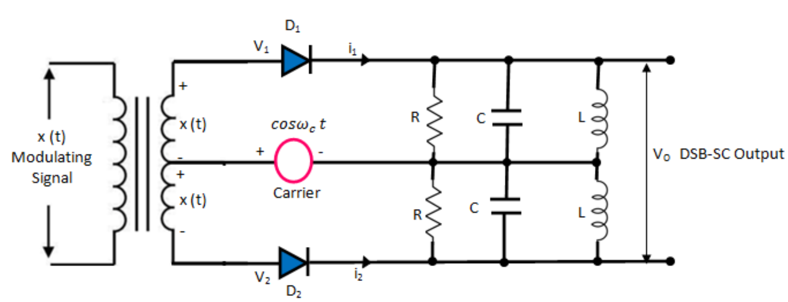


Operation :- 2M



- Detection in optical recording is equivalent to playback process. In this a laser beam produced by a solid state laser of semiconductor aluminum gallium arsenide is made incident on the CD through half silver mirror the mirror allows the beam to pass through itself but does not allow the returning beam to pass.
- The returning beam is reflected from the aluminum flat surface & represents digit 1.there is only little reflection from a pit & it represents 0. Thus the laser beam is the replica of the original laser beam modulated by digits of audio signal.
- Light is not reflected from the pit fully reflected from flat surface. Thus binary digits are reproduces when this ON-OFF reflected light falls on a photodiode.
- The digital output of photodiode is processed & converted into the original signal by using DAC
- Control signals allow any combination of track to be played in any sequence with the help of keyboard.

A clock signal is obtained from the disc itself. It is compared with a crystal oscillator signal. Any discrepancy result in generation of a correction signal which is applied to the servo system.

	d)	**Explain the concept of vestigial side band.**	**4M**
	Ans:	**Concept of vestigial side band:-** • Vestigial sideband is a type of Amplitude modulation in which one side band is completely passed along with trace or tail or vestige of the other side band. • VSB is a compromise between SSB and DSBSC modulation. In SSB, we send only one side band, the Bandwidth required to send SSB wave is w. SSB is not appropriate way of modulation when the message signal contains significant components at extremely low frequencies. • To overcome this VSB is used. • The main advantage of VSB modulation is the reduction in bandwidth.	**4M**
	e)	**Explain the method for generating of DSBSC AM signal using diode balanced modulator.**	**4M**
	Ans:	**Generation of DSBSC AM signal using diode balanced modulator:-**  : Balanced Modulator circuit using diodes **OR**  **Explanation:-** 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	**Diagram 2M** **2M**

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

- A non-linear resistance or non-linear device may be used to produce Amplitude Modulation i.e. one carrier and two sidebands.
- However, a DSB-SC signal contains only two sidebands. Thus, if two non-linear devices such as diodes, transistors etc. are connected in a balanced mode so that they suppress the carriers of each other, then only sidebands are left and a DSB-SC signal is generated .
- Therefore, a balanced modulator may be defined as a circuit in which two non-linear devices are connected in a balanced mode to produce a DSB-SC signal. Fig. shows the balanced modulator using diodes as non-linear device.

The modulating signal x(t) is applied equally with 180^o phase reversal at the inputs of both the diodes through the input center tapped transformer .
The carrier is applied to the center tap of the secondary.

Hence, input voltage to D_1 is given by:

$$v_1 = cos\omega_c t + x(t) \quad \text{........ (1)}$$

And the input voltage to D_2 is given by:

$$v_2 = cos\omega_c t - x(t) \quad \text{.......... (2)}$$

The parallel RLC circuits on the output side form the band pass filters

The diode current i_1 and i_2 are given by:

$$i_1 = av_1 + bv_1{}^2$$

The diode current i_1 and i_2 are given by:

$$i_1 = av_1 + bv_1{}^2$$

$$i_1 = a[x(t) + cos\omega_c t] + b[x(t) + cos\omega_c t]^2$$

$$i_1 = ax(t) + acos\omega_c t + bx^2(t) + 2bx(t)cos\omega_c t + bcos^2\omega_c t \quad \text{\_\_\_\_\_\_\_\_\_\_\_\_(3)}$$

Similarly,

		$i_2 = av_2 + bv_2^{\,2}$ $i_2 = a[x(t) - cos\omega_c t] + b[x(t) - cos\omega_c t]^2$ $i_2 = av_2 + bv_2^{\,2} = ax(t) - acos\omega_c t + bx^2(t) - 2bx(t)cos\omega_c t + bcos^2\omega_c t$ ___________(4) The output voltage is given by: $v_o = i_1 R - i_2 R$ Substituting the expression for i1 and i2 from equations (3) and (4),we get $v_o = R[2\,a\,x(t) + 4\,b\,x(t)\,cos\omega_c\,t]$ OR $v_o = \underbrace{2aRx(t)}_{\text{Modulating Signal}} + \underbrace{4bRx(t)\,cos\omega_c t}_{\text{DSB-SC Signal}}$ Hence, the output voltage contains a modulating signal term and the DSB-SC signal . The modulating signal term is eliminated and the second term is allowed to pass through to the output by the LC band pass filter section. Therefore, final output = 4 b R x(t) cos ωct = K x(t) cos ωct Thus, the diode balanced modulator produces the DSB-SC signal at its output.	
	f)	**Explain the generation of FM using varactor diode.**	**4M**
	Ans:	**Generation of FM using varactor diode:-**	**2M**

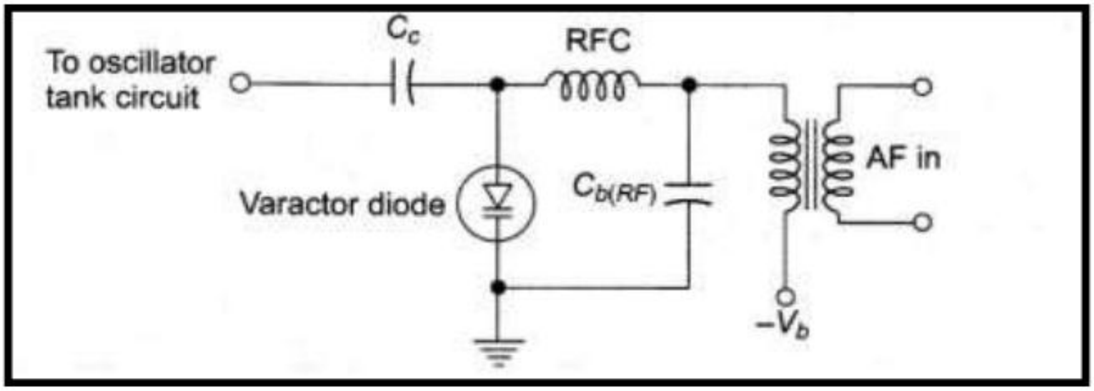
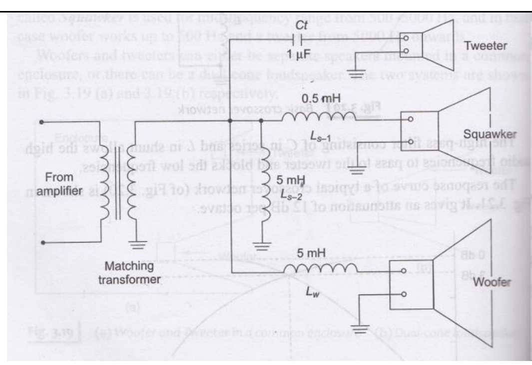


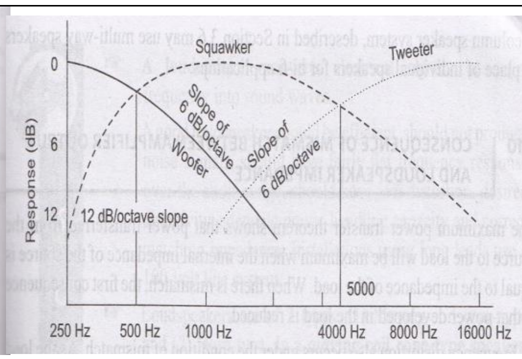
Fig. Generation of FM wave using varactor diode modulator

Explanation:- — 2M

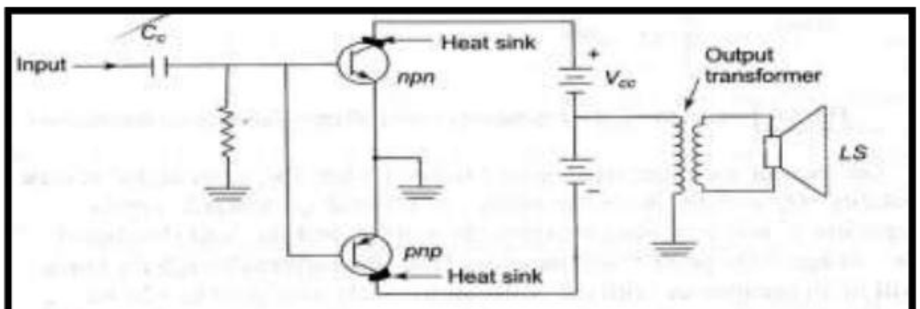
- A varactor diode is a semiconductor diode whose junction capacitance varies linearly with the applied voltage when the diode is reverse-biased.
- It may also be used to produce frequency modulation. Varactor diodes are certainly employed frequently, together with a reactance modulator, to provide automatic frequency correction for an FM transmitter.
- The above figure shows such a modulator. It is seen that the diode has been reverse-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.

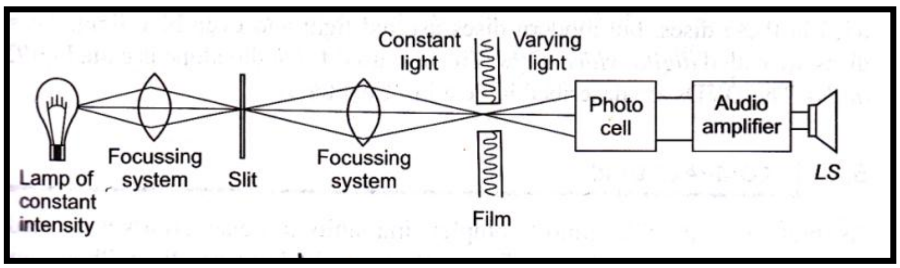
Q. 4	**A)**	**Attempt any four.**	**16-Total Marks**
	a)	**Draw multiway speaker system and describe its working.**	**4M**
	Ans:		**Diagram 2M**

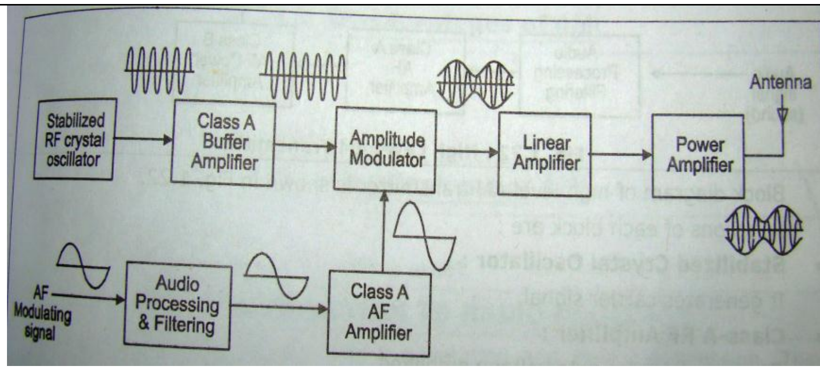




Three way cross over network **Response curve (*Optional to Draw*)**

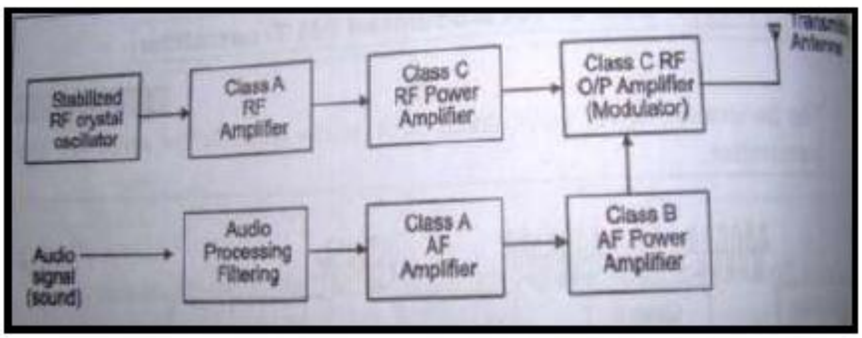
		Explanation: • The multi-way speaker system is used to get flat frequency response for the entire range of audio frequency it is essential to have a cross over network. The cross over network divides the incoming signal into separate frequency ranges for each spectrum. • In absence of cross over network, 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. Lw of 5mH in series with woofer prevents high and mid frequencies reaching to woofer. • Ls1 and Ls2 allows only mid frequency range to reach to squawker.	**2M**
	b)	**Draw circuit diagram and explain the working of complementary symmetry push-pull amplifier.**	**4M**
	Ans:	**Circuit diagram of complementary symmetry push-pull amplifier:-**  Fig. Complementary symmetry push-pull amplifier circuit with output transformer	**2M**
		Explanation: • 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.	**2M**

	c)	**State the need and application of public address system.**	**4M**
	Ans:	**Need of public address system:-** The intensity of sound decreases with distance therefore when large gathering is to be addressed, sound needs to be amplified so that people at a distance from the stage may receive good intensity of sound for comfortable listening. The system which fulfills this function is called PA system. **Application of public address system:-** • It is used in sports meeting, public meeting auditorium, concerts, functions etc • It is also used to convey information to isolated location like railway station, airport, hospitals, factories, schools etc.	**2M** **2M (any two application)**
	d)	**Explain the principle of reproduction of sound from a recorded film.**	**4M**
	Ans:	**Diagram :-**  <u>**Explanation:**</u> • The principle of reproduction is illustrated in above figure. • 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, fall 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.	**2M** **2M**
	e)	**Draw the block diagram of AM transmitter and state function of each block.**	**4M**
	Ans:	**Block diagram of Low level AM transmitter**	**2M**



OR

Block diagram of High level AM transmitter:-



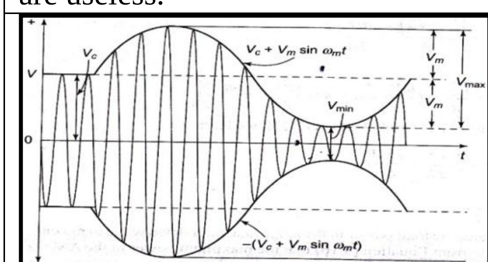
- **Stabilized RF crystal oscillator:** RF oscillator generates the carrier signal. RF oscillator is stabilized to maintain carrier frequency deviation in limit
- **Buffer amplifier :** Carrier signal is amplified
- **Audio processing and filtering:** Sound information converted in to electrical signal.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

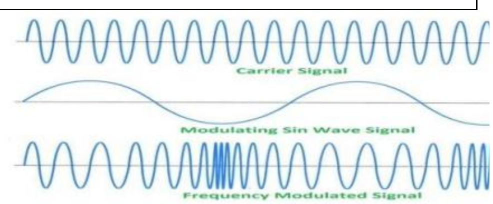
2M

f)	**Differentiate FM from AM (four points).**	**4M**

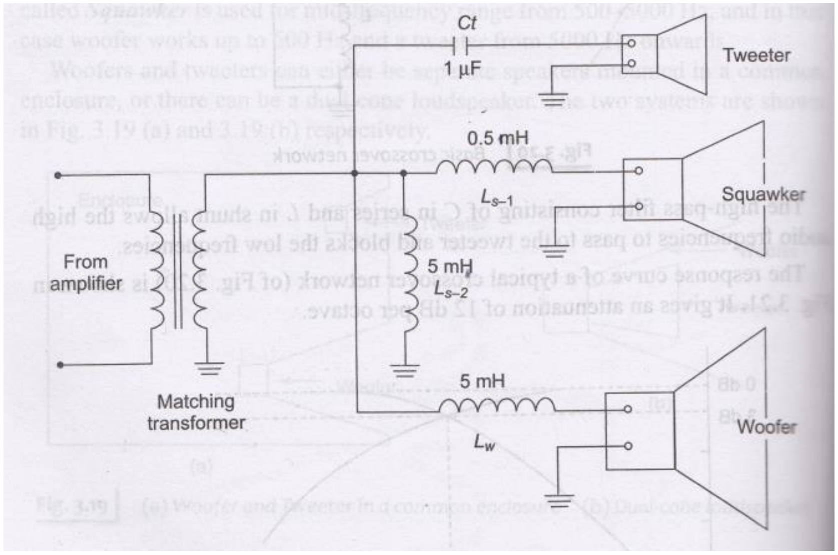
Ans: **(Any four points) 1M each**

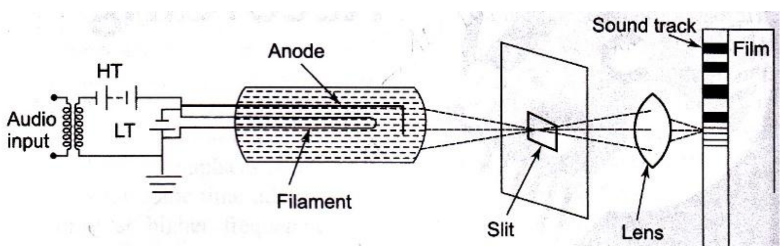
AM	FM
AM signal have low noise immunity	FM is higher noise immunity compared to AM.
AM modifies the amplitude of the carrier frequency	FM modifies the frequency of the carrier
AM is much more simpler compared to FM	FM is much more complex compared to AM
ground wave & sky wave propagation is used therefore large area is covered than FM	space wave is used for propagation do radius of transmission is limited to line of sight.
AM is more prone to signal distortion And degradation	FM signal doesn't degrade as easily as AM
Applications: Radio & TV broadcasting,	Application : Radio & TV broadcasting, police wireless, point to point communication
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)
Carrier power & one sideband power are useless.	All the transmitted power are useful.

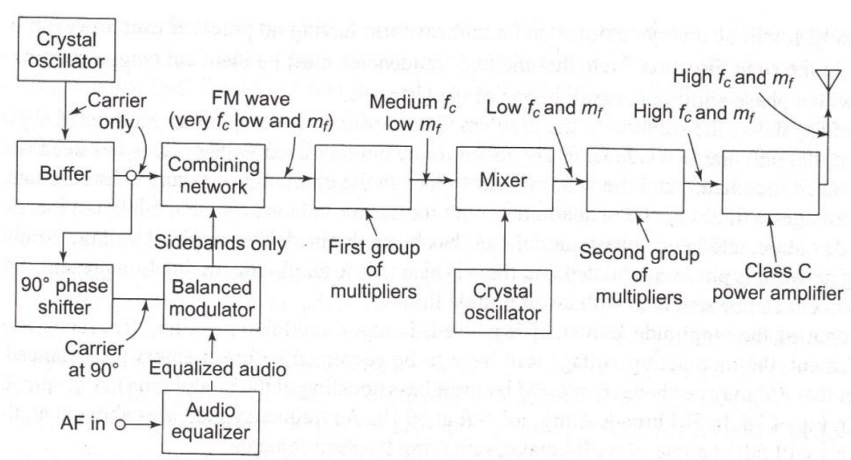


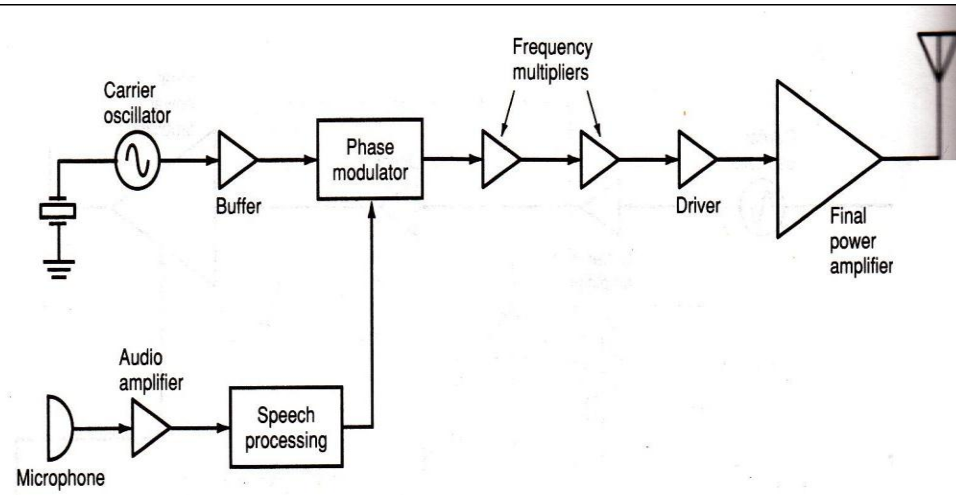


Q.5		**Attempt any four.**	**16-Total Marks**
	a)	**Why cross over network is necessary? Describe the operation of 3 way cross over network.**	**4M**
	Ans:	**Need of cross over network:-** When multi-way speaker system is used to get flat frequency response for the entire range of audio frequency it 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.	**1M**

		Operation of 3 way cross over network:-  **Explanation:-** • Ct of 1μf in series with tweeter prevents low 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	**2M** **1M**
	b)	**What are the causes affecting fidelity? Give their remedies.**	**4M**
	Ans:	**<u>Causes affecting fidelity:</u>** • High signal to noise ratio(S/N ratio) • Flat frequency response • Low nonlinear distortion • Large dynamic range • Creating sense of direction. **<u>Remedies:</u>-** 1. S/N ratio can be improved by using preamplifier of low noise figures proper shielding, grounding, Decoupling & filtering circuits, stabilized power supply, microphones 2. By using coupling capacitor and shunt capacitor in audio amplifier circuits 3. 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. 4. Dynamic range can be increased by using solid-state amplifier; dynamic microphones & L.S. which are capable of withstanding the large change in loudness. **5.** Creating sense of direction can be improved by using high fidelity system.	**(Any four-2M)** **(Any four - 2M)**
	c)	**State the four specification of public address system.**	**4M**

	Ans:	**Specifications of public address system:-** 1. Acoustic feedback 2. Distribution of sound intensity 3. Reverberation 4. Orientation of loudspeakers 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	**Any 4 points** **1M each**
	d)	**Draw and describe optical recording of sound on film is done by variable density method.**	**4M**
	Ans:	**<u>Variable density method:</u>**  **<u>Explanation:</u>** • 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.	**2M** **2M**
	e)	**A 500 watt carrier is modulated to depth of 80% calculate.**	**4M**

		1) Total power in AM wave **2) Power in sidebands.**	
	Ans:	**Solution:** **Given:** $P_c = 500$ watt $m = 0.8$ **(i) Total Power:** $P_t = \left(1 + \frac{m^2}{2}\right) P_c$ $= \left(1 + \frac{(0.8)^2}{2}\right) \times 500$ $\boxed{P_t = 660 \text{ Watt}}$ **(ii) Power in sidebands:** $P_{USB} = P_{LSB} = \frac{m^2}{4} \times P_c$ $= \frac{(0.8)^2}{4} \times 500$ $= 80$ Watt $\boxed{P_{USB} = P_{LSB} = 80 \text{ Watt}}$	**2M** **2M**
	f)	**Draw the block diagram of FM transmitter and explain its operation.**	**4M**
	Ans:	**Block diagram of FM transmitter:-**  **OR**	**2M**



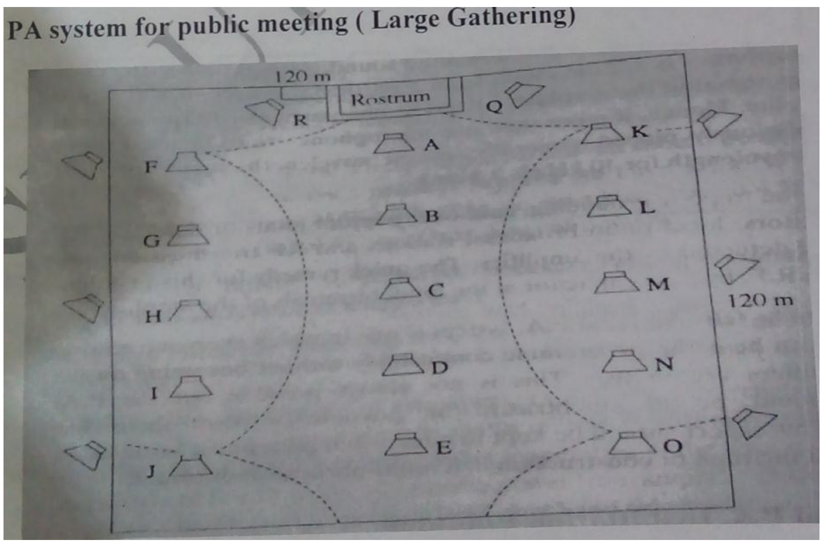
2M

Explanation:

The indirect method of frequency modulation generation is used.

- A stable crystal oscillator is used to generate the carrier signal and a buffer amplifier is used to isolate it from the remainder of the circuitry
- The carrier signal is then applied to a phase modulator.
- The voice input is then amplified and processed to limit the frequency range & prevent over deviation. The modulator output is desired FM signal.
- Most FM transmitter are used in the VHF and UHF range and crystal are not available to generate those frequencies directly as result, the carrier is usually generated at frequency considerably lower than the final output frequency.
- To achieve the desired output frequency one or more frequency multipliers stage are used.
- A frequency multiplier is class C amplifier whose output frequency is some integer multiple of the input frequency by a factor 2, 3, 4 & so on. Because of class C amplifier provides a modest amount of power amplification.
- The frequency multiplier not only increases the carrier frequency to the desired output frequency but is also multiplies the frequency deviation produced by the modulator.
- After the frequency multipliers, a class C driver amplifier is used to increase the power level sufficiently to operate the final power amplifier.

Q.6		**Attempt any four.**	**16-Total Marks**
	a)	**State the characteristics of human ear response to the Audio frequency.**	**4M**
	Ans:	**Characteristics of human ear response:-** 1. The ear is most sensitive from 3 KHz to 4 KHz for all ages.	**(Any 4 points)**

		2. Sensitivity of the ear decreases with age for high frequencies. The ear has good sensitivity from 500 Hz to 10000 Hz for youngsters, but from 500 Hz to 5000 Hz only for old people. Children's ears are sensitive for up to 20000Hz. 3. Sensitivity of the ear for all ages decreases as frequencies decrease below 500 Hz for low and medium volumes of sound. 4. Ears can judge the direction of the source of sound because of the phase difference between sounds reaching two ears simultaneously.	**1M each**
	b)	**With neat sketch, explain installation of PA system for public meeting.**	**4M**
	Ans:	**Installation of PA system for public meeting:-**  **Explanation:-** **1.** The L.S A, B, C, D & E in the centre line will give the sense of direction to most of the audience and can be mounted on poles. 2. L.S. F, G, H and I on one side and K, L, M and N on the other side will give full coverage to meeting ground both sides of the central area. 3. To cover the remote semicircular side and corner areas, L.S. J and Q are used. These will throw sound power towards corners. 4. The L.S Q and R will cover the left and right sides respectively near the rostrum. 5. There may be some Loudspeakers S, T, V, X, Y, Z to give coverage to audience standing outside the meeting park. These may be slightly inclined as shown in figure. 6. Microphones should be of cardioids type and the loudspeakers may be of the horn type.	**Diagram 2M** **2M**
	c)	**State the reasons due to which noise is reduced in Dolby system as compared to other audio system.**	**4M**

Ans: Dolby noise reduction is a form of dynamic pre-emphasis employed during recording, plus a form of dynamic de-emphasis used during playback, that work in tandem to improve the signal-to-noise ratio. While Dolby A operates across the whole spectrum, the other systems specifically emphasize the audible frequency range where background tape hiss, an artifact of the recording process that is similar to white noise, is most noticeable (usually above 1 kHz, or two octaves.

4M

The Dolby pre-emphasis boosts the recorded level of the quieter audio signal at these higher frequencies during recording, effectively compressing the dynamic range of that portion of the signal, so that quieter sounds above 1 kHz receive a proportionally greater boost.

Diagram is optional

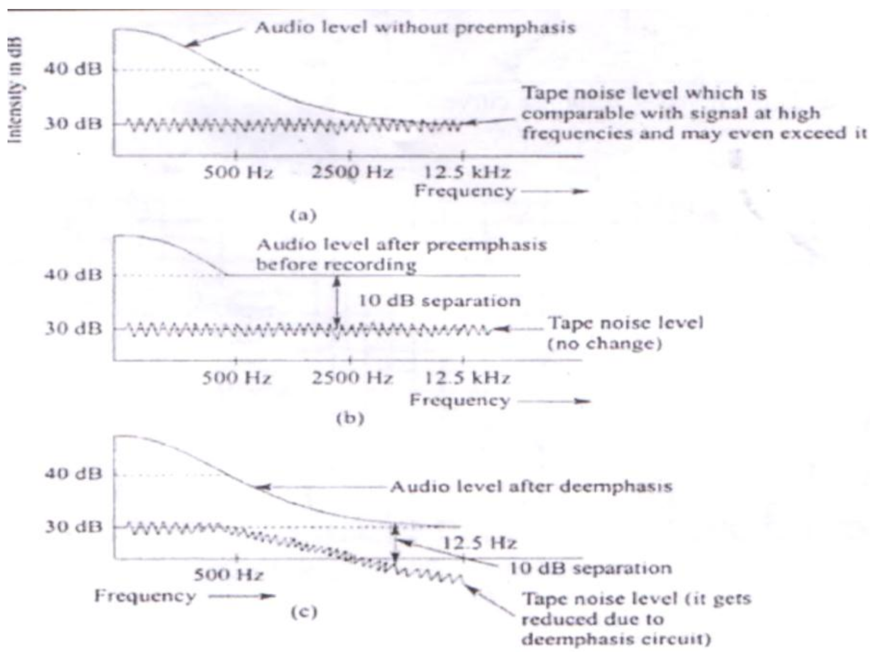
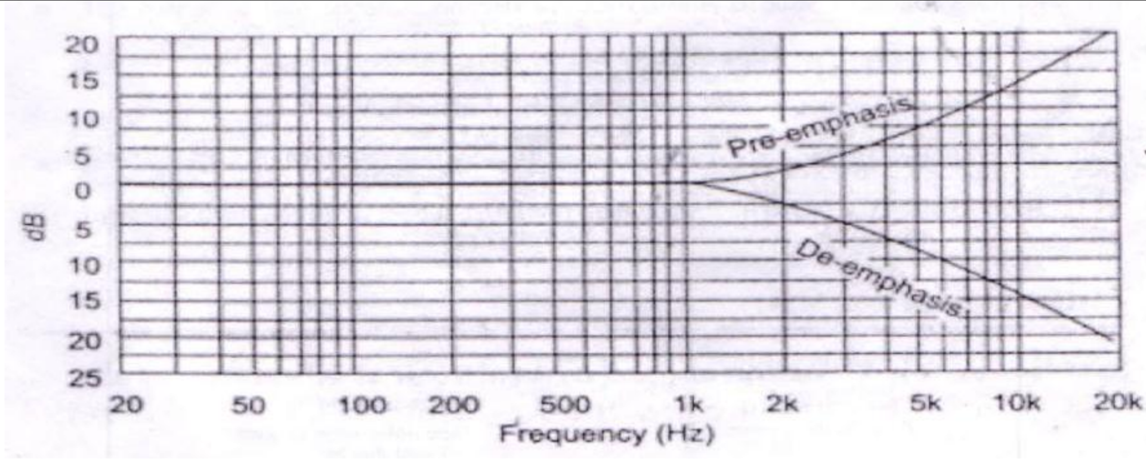


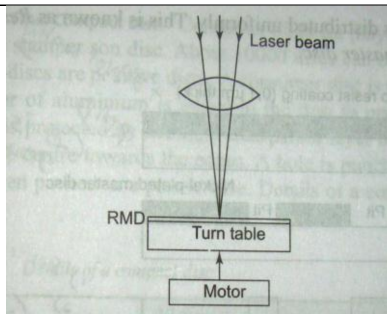
Fig. Reduction of noise by 10 dB in Dolby system
(a) Position without pre-emphasis
(b) Position after pre-emphasis
(c) Position after de-emphasis

OR

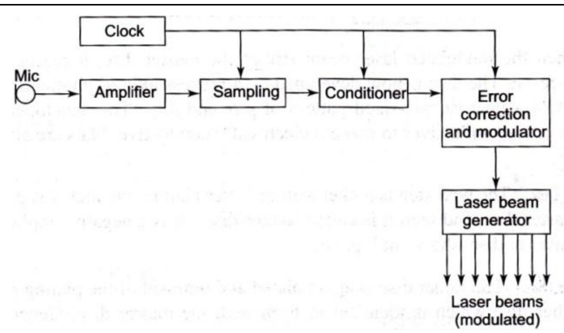


d) **Draw neat block diagram and explain optical recording process in CD's.** **4M**

Ans: **2M**



<u>OR</u>



<u>Recording on CD:</u> **2M**

- 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) **Define modulation index of an AM wave and give the mathematical representation of AM wave.** **4M**

Ans: **Modulation Index**: It in AM is defined as the ratio of amplitude of modulating signal to the amplitude of carrier signal. **2M**

		$$m = \frac{V_m}{V_c}$$ **Mathematical expression for amplitude modulated wave** **$V_{AM} = V_c \sin \omega_c t + mV_c/2 \cos (\omega_c - \omega_m)t - mV_c/2 \cos (\omega_c + \omega_m)t$**	**2M**
	f)	**Define phase modulation and its modulation index.**	**4M**
	Ans:	**<u>Phase modulation:</u>** The phase shift of the carrier signal is varied in proportional with the amplitude of the modulating signal. The amplitude of the carrier remains constant. **<u>Modulated index:</u>** The modulating index is defines as: Mp = δp is expressed in radiance where δp is maximum frequency deviation.	**2M** **2M**