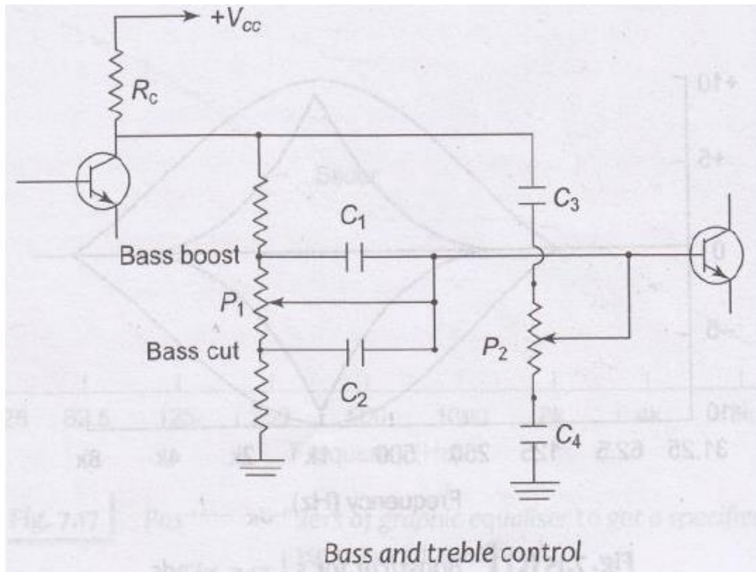
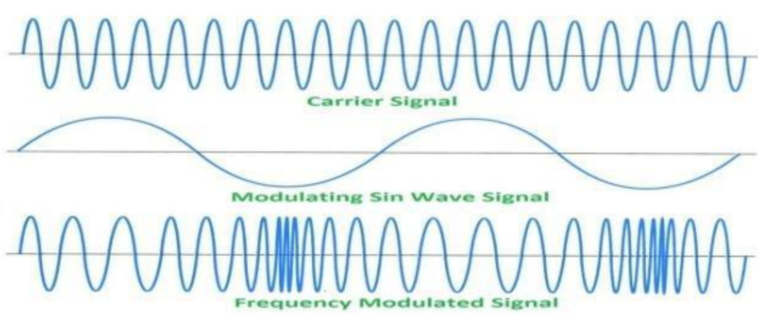
MODEL ANSWER
SUMMER– 18 EXAMINATION

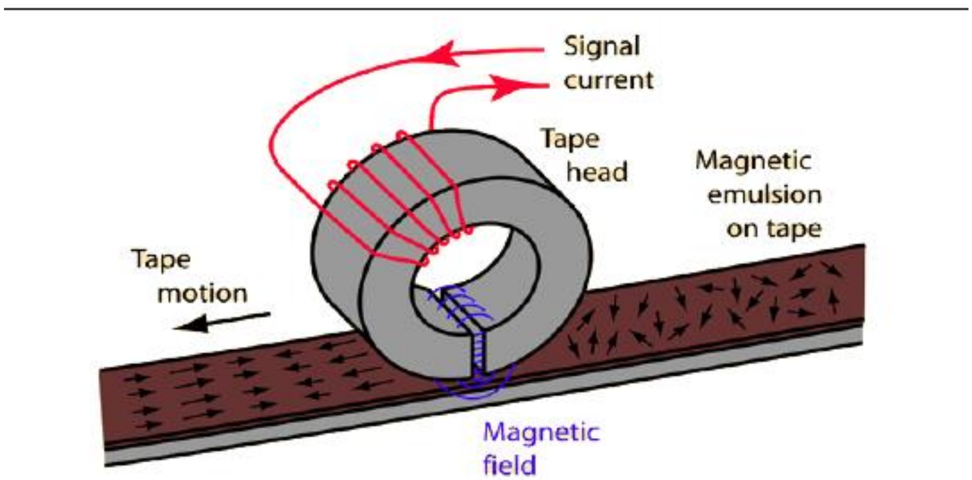
Subject Title: Fundamentals of communications **Subject Code:-** 17316

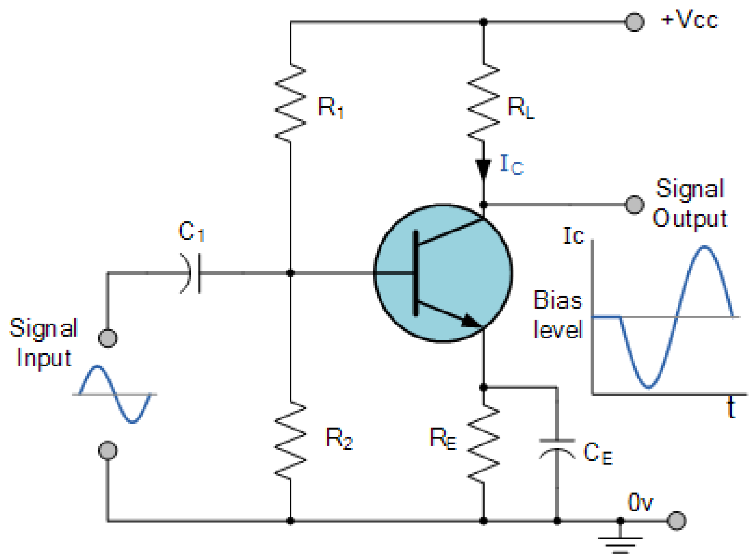
Important Instructions to examiners:

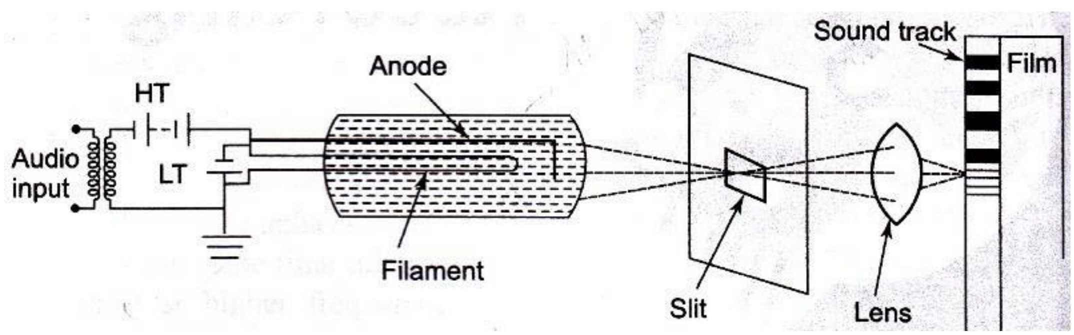
1) The answers should be examined by key words and not as word-to-word as given in the model answer scheme.
2) The model answer and the answer written by candidate may vary but the examiner may try to assess the understanding level of the candidate.
3) The language errors such as grammatical, spelling 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.

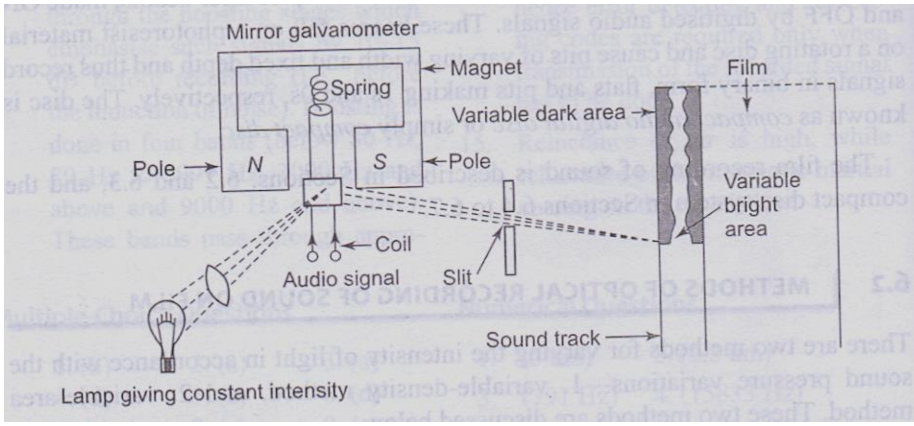
Q. No.	Sub Q.N.	Answer	Marking Scheme
Q.1	**A)**	**Attempt any six :**	**12 Marks**
	a)	**Define overtone and timbre.**	**2 Marks**
	Ans:	**TIMBRE :** The proportion of tones & overtones in a sound form the special characteristics by which a particular sound can be recognized. When we hear the sound 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. **OVERTONE:** An overtone is any frequency greater than the fundamental frequency of a sound. The fundamental and the overtones together are called Harmonics	**1 Mark each Definition**
	b)	**List any four characteristics of loudspeaker.**	**2 Marks**
	Ans:	**Characteristics of loudspeaker.** 1) Efficiency 2) Noise 3) Frequency response 4) Distortion 5) Directivity	**Any four ½ Marks each Characteristics**

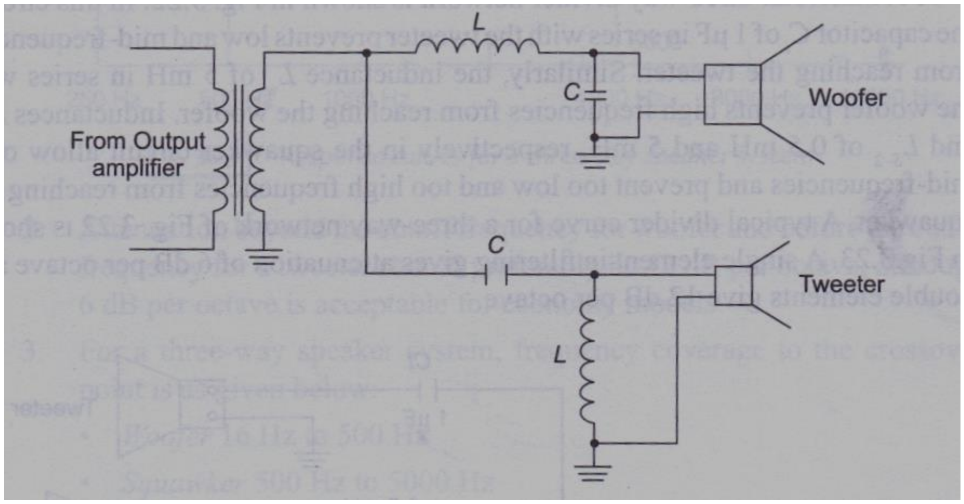
		6) Power 7) Impedence	
	c)	**Draw neat circuit diagram of Bass and treble control.**	**2 Marks**
	Ans:		**Diagram** **2 Mark**
	d)	**Define frequency modulation and modulation index of FM .**	**2 Marks**
	Ans:	**Frequency modulation:** The modulation process in which the frequency of the carrier signal changes according to instantaneous value of modulating signal keeping amplitude & phase constant.  **Modulation Index for FM:** The modulation index for FM, *mf* is defined as max. frequency deviation to the max modulation frequency. $m_f=\delta/fm$	**1 Mark each Defination**

	e)	**State the principle of magnetic recording.**	**2 Marks**
	Ans:	**Diagram:**  **Principle:** 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.	**Principle 2 Mark**
	f)	**List any four advantages of CD's.**	**2 Marks**
	Ans:	**Advantages of CD:** 1. Signal to noise ratio is high 2. Compact disc is immune to the surface contamination 3. Dynamic range is high 4. Channel separation is high 5. Wow does not exist 6. Flutter does not exist 7. Total distortion is low **8.** Frequency response is excellent & covers complete audio range	**½ marks for each point** **(Any four points)**
	g)	**List the different controls of Audio amplifier.**	**2 Marks**
	Ans:	For controls of Audio amplifiers are 1. microphone gain control 2. volume control 3. Bass control 4. Treble control	**½ marks for each point**

	h)	**Draw a neat labeled circuit diagram of single stage power amplifier.**	**2 Marks**
	Ans:	**Circuit diagram:**  	**Diagram 2 Mark**
	B)	**Attempt any two :**	**8 Marks**
	a)	**A 500 watt carrier is modulated to depth of 80%.** **Calculate :** **i) Total power in AM wave** **ii) Power in sidebands.**	**4 Marks**
	Ans:	**Given:** Pc=500 W, ma=0.8 **Formula**: $Pt= Pc(1+ma^2/2)$ i) Total power in AM Wave $Pt= Pc(1+ma^2/2)$ $=500(1+0.8^2/2)$ =660 watt ii) power in side band: $P_{USB} = P_{LSB} = \frac{m^2 \, x \, Pc}{4}$ $= (0.8)^2 * 500/4$ **(01M)** = 80 Watt	**i) Formula 1 Mark** **Total power calculation 1 Mark** **ii) Formula 1 Mark** **side band power calculation 1 Mark**
	b)	**In FM, if the maximum deviation is 75 KHZ and max. modulating frequency is 10 KHZ. Calculate modulation index of FM.**	**4 Marks**

	Ans:	**Given:** Max deviation δm= 75 KHZ max. modulating frequency fm= 10 KHZ Modulation Index (mf)= δm/fm = 75/10 **= 7.5**	**Formula 2 Mark** **Calculation 2Mark**
	c)	**Describe optical recording of sound on film with neat diagram.**	**4 Marks**
	Ans:	**Different methods of optical recording of sound on film:-** • Variable density method OR • Variable area method **Diagram:** **Variable density method:**  **Explanation:** • 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	. **Diagram 2 Mark** **Explanation 2 Mark**

		on a moving photographic film where the image is recorded in the form of bars of varying density and distance on the film. **OR** **Variable area method:** **Diagram:**  **Explanation:** • Light of constant intensity falls on a slit. The area of the slit opened for this light varies in accordance with the variant of the sound pressure .Hence , the light falls on the variable area on the soundtrack edge of the film .Thus , the area which is bright to light varies. The area of the slit is made variables with the help of a mirror or galvanometer as illustrated in fig.6.2. • The audio signals are amplified and reach the coil of a mirror galvanometer. The current-carrying coil is placed in a magnetic field and hence , deflects in accordance with amplitude of the audio signals . A mirror is attached to the coil assembly. The mirror also deflects. Light from a lamp, duty focused by a lens systems, is made to fall on the mirror the light reflected from the mirror goes to a narrow slit, when the mirror deflects , the slit area exposed to the light changes, i.e, the slit is partially illuminated . The extent to which the slit area is illuminated depends on the extent of the deflection of mirror and hence and hence on the strength of audio current. The light from the variable area of the slit falls on the soundtrack edge of the film and is recorded in the form of a photograph of a variables area.	
Q 2		**Attempt any four :**	**16 Marks**
	a)	**Draw and explain two way crossover network.**	**4 Marks**
	Ans:	**Diagram:**	



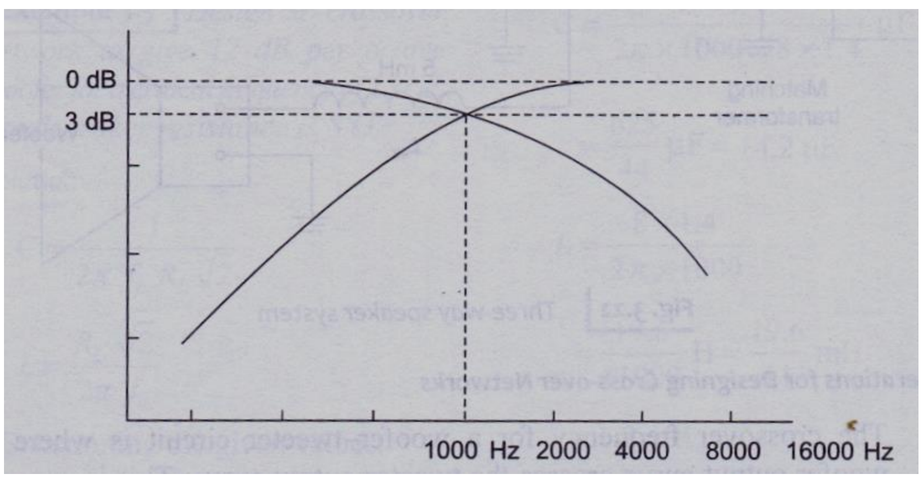


Diagram 2 Mark

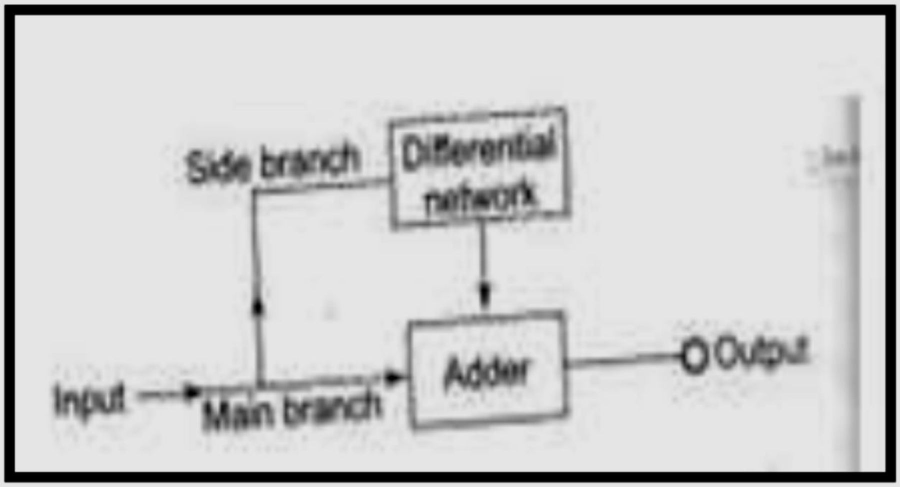
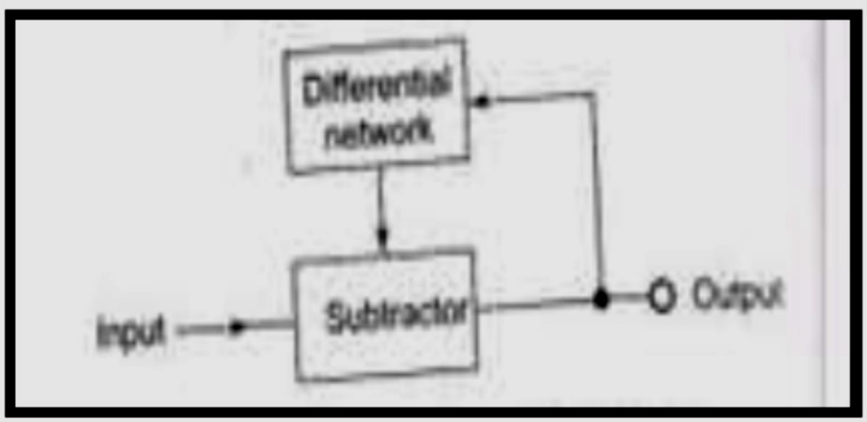
Explanation 2 Mark

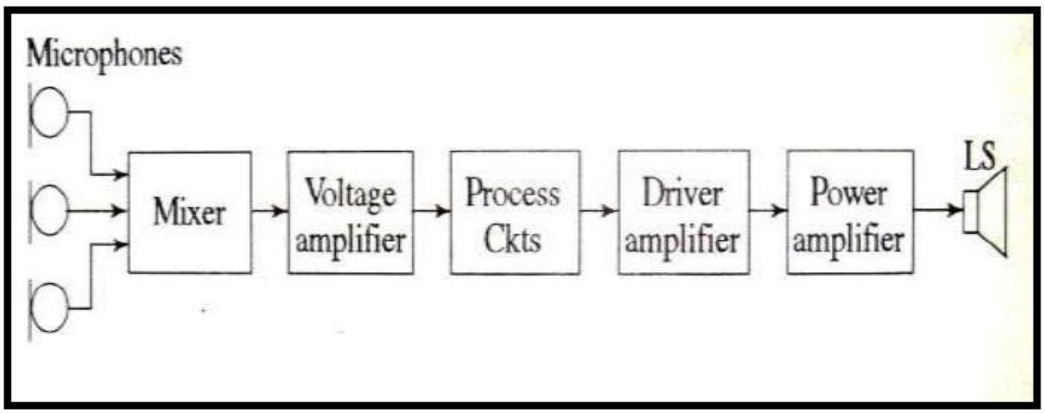
Explanation:

Crossover networks make use of the fact that the capacitive reactance decreases with increase in frequency $[X_c=1/(2\pi fL)]$, and the inductive reactance increases with increase in frequency$X_1=2\pi fL$). A basic crossover network is illustrated in fig.3.20.The circuit consists of a low-pass LC filter across the woofer and a high-pass LC filter across the tweeter. The low-pass filter permits only low audio frequencies (16Hz to 1000Hz) to go the woofer . the series reactance of L and shunt reactance of C for high audio frequencies prevents these frequencies from going to the woofer.

The high-pass filter consisting of C in series and L in shunt allows the high audio frequencies to pass to the tweeter and blocks the low frequencies.

The response curve of a typical crossover network(of fig.3.20) is shown in fig.

	b)	**Describe Dolby—A method of noise reduction.**	**4 Marks**
	Ans:	**Diagram:**  **Fig. Coding of signal in Dolby method**  **Fig. Decoding of Dolby signal** **Explanation:** Boosting is done in 4 bands: 1. Below 80Hz 2. 80Hz to 2999Hz 3. 3000 Hz and above 4. 9000 Hz and above Each band is processed separately by using low-pass, band-pass and high-pass filters and limiters.	**Diagram 2 Mark** **Explanation 2 Mark**

		The 16 Hz to 80Hz signal goes to a low pass filter which causes improvement in signal to noise ratio with respect to hum and rumble. The 80Hz to 2999Hz signal goes to a band pass filter which deals with the mid band noise. Most of the sound energy for music is concentrated in this band. The 3000Hz and 9000Hz high pass filters improve signal to noise ratio with respect to hiss and modulation noise. The output of the four separate units is added. All this is done in a side branch, and this branch is known as the differential network. The output of the differential network goes to the adder of the main branch as shown in fig.(a) In playback, the differential network separates out the boosted signals in the side branch and subtracts them from the input signal as shown in fig.	
	c)	**Draw and explain block diagram of PA system.**	**4 Marks**
	Ans:	**Block Diagram:**  **Fig. Block diagram PA system** **Function:-** 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.	**Diagram 2 Mark** **Explanation 2 Mark**

		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** 1) Simplest – no amplifiers only gain controls (faders) and isolating services resistors. 2) Little sophisticated- common amplifiers after isolating resistors. 3) 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.	
	d)	**Draw neat sketch and explain step by step procedure of preparation of CD's on large scale.**	**4 Marks**
	Ans:	**Diagram:**	

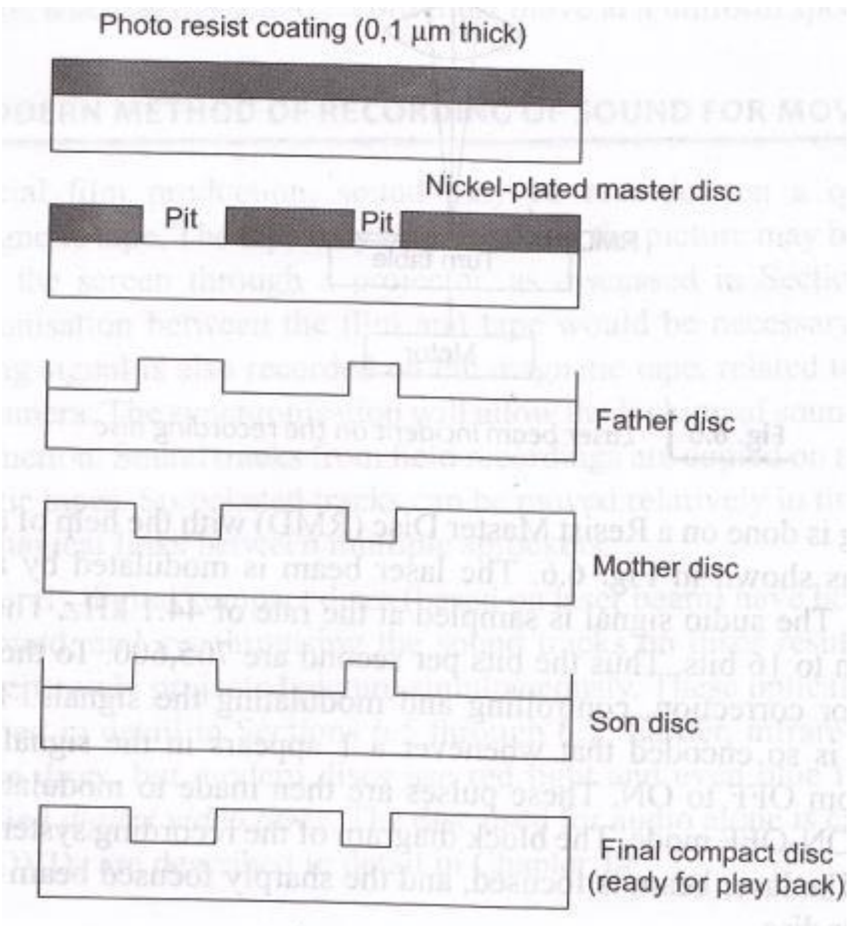


Diagram 2 Mark

Explanation 2 Mark

Preparation of compact Discs consist of following important stages

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.

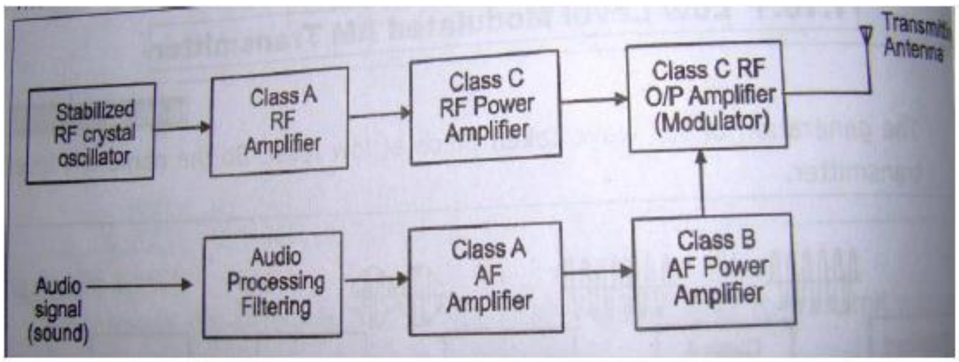
e) **Compare AM and FM (any 8 points).** **4 Marks**

Ans:

1/2 marks for each point (Consider any 08 points)

AM	FM
1. AM signal have low noise immunity	1. FM is higher noise immunity compared toAM.
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 Anddegradation	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.

f) **Draw the block diagram of high level AM transmitter and state the function of each block.** **4 Marks**

<table>
<tr><td></td><td>Ans:</td><td>High level AM transmitter:
Block diagram:

Functions of each block are : (High level Transmitter)

 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.

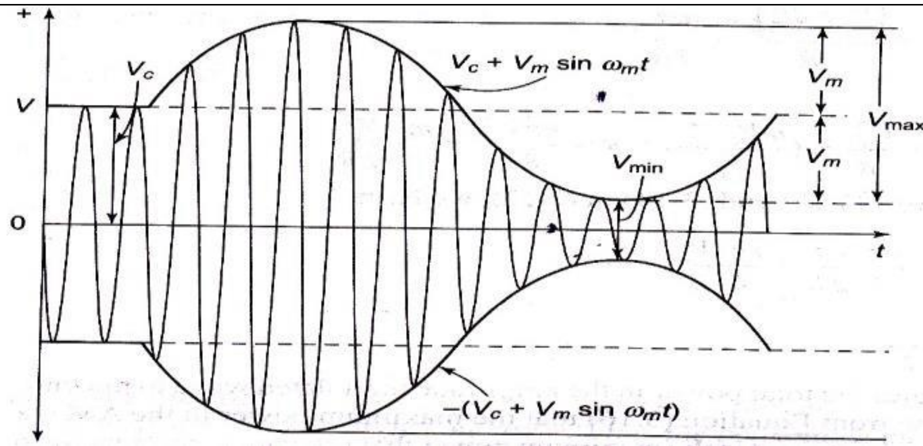
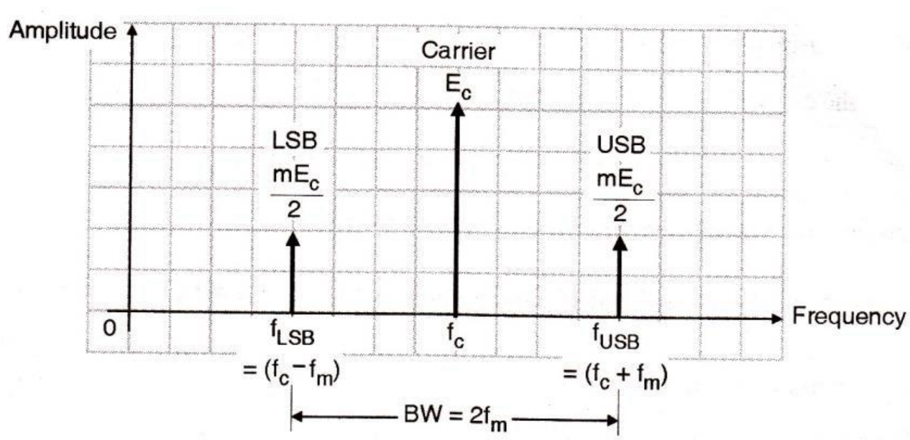
 Class A amplifier : modulating signal amplified after audio processing and filtering.

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.</td><td>Diagram 2 Mark

Explanation 2 Mark</td></tr>
<tr><td>Q. 3</td><td></td><td>Attempt any four :</td><td>16 Marks</td></tr>
<tr><td></td><td>a)</td><td>Derive the mathematical equation for total power in AM.</td><td>4 Marks</td></tr>
</table>

	Ans:	$$P_{total} = \frac{(V_{carrier})^2}{R} + \frac{(V_{LSB})^2}{R} + \frac{(V_{USB})^2}{R}$$ 01M The above expression is represented in terms of Peak values, but for the power rms values are considered. So $$V_{c(rms)} = \frac{V_c}{\sqrt{2}} \text{ using concept } V_{rms} = \frac{V_{max}}{\sqrt{2}}$$ $$V_{LSB(rms)} = +\frac{m_a V_c}{2\sqrt{2}} \quad \text{where } V_{LSB} = \frac{m_a V_c}{2}$$ ... Derived in the sideband expressions $$V_{USB(rms)} = -\frac{m_a V_c}{2\sqrt{2}} \quad \text{where } V_{USB} = -\frac{m_a V_c}{2}$$ ... Derived in the side band expressions $$P_c = \frac{(V_{carrier})^2}{R} = \frac{(V_c/\sqrt{2})^2}{R} = \frac{V_c^2}{2R}$$ 01M $$P_{total} = \left(\frac{V_c}{\sqrt{2}}\right)^2 \frac{1}{R} + \left(\frac{m_a V_c}{2\sqrt{2}}\right)^2 \frac{1}{R} + \left(-\frac{m_a V_c}{2\sqrt{2}}\right)^2 \frac{1}{R}$$ 01M $$P_t = \frac{V_c^2}{2R} + \frac{m_a^2 V_c^2}{8R} + \frac{m_a^2 V_c^2}{8R}$$ $$P_t = \frac{V_c^2}{2R}\left[1 + \frac{m_a^2}{4} + \frac{m_a^2}{4}\right]$$ $$P_t = P_c + \frac{m_a^2}{4} P_c + \frac{m_a^2}{4} P_c$$ $$P_t = P_c\left(1 + \frac{2m_a^2}{4}\right)$$ $$\boxed{P_t = P_c\left(1 + \frac{m_a^2}{2}\right)}$$ 01M	**1 Mark** **1 Mark** **1 Mark** **1 Mark**
	b)	**Draw AM wave in frequency and time domain.**	**4 Marks**
	Ans:	**Time domain**	**Time domain diagram 2 Mark**

		 **Frequency domain** 	**Frequency domain diagram 2Mark**
	c)	**Explain with neat diagram, the generation of FM wave using reactance modulator.**	**4 Marks**
	Ans:	**Diagram:**	**Diagram 2 Mark** **Explanation 2 Mark**

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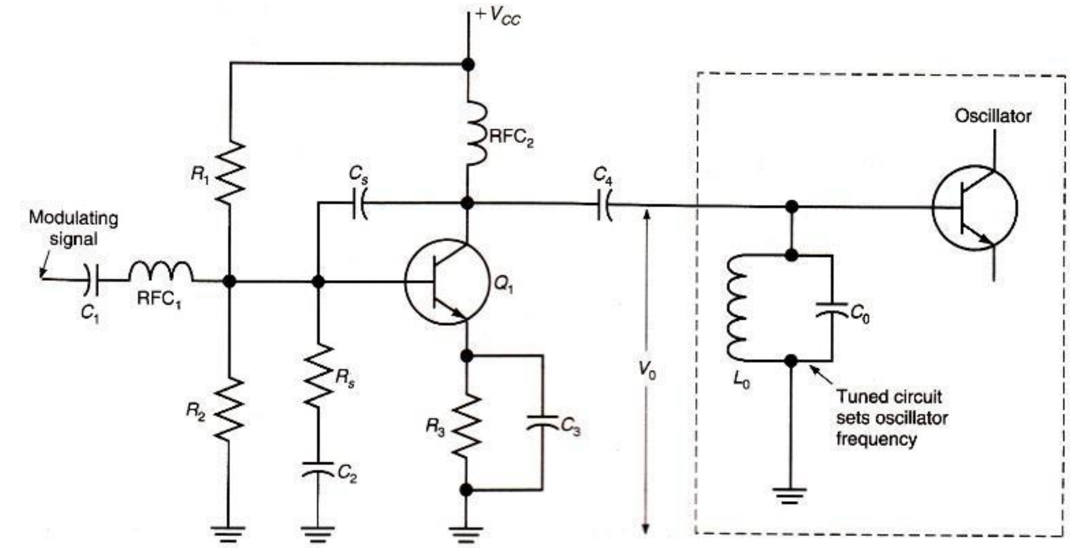

Principle: -
In reactance modulator a transistor is operated as a variable reactance and it is connected across the tuned circuit of an oscillator.
As the instantaneous value of modulating voltage changes, the reactance offered by the transistor will change proportionally. This will change the frequency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

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<td>d)</td>
<td>Explain block diagram of communication system.</td>
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<td>Ans:</td>
<td>Diagram:</td>
<td>Diagram
2 Mark

Explanation
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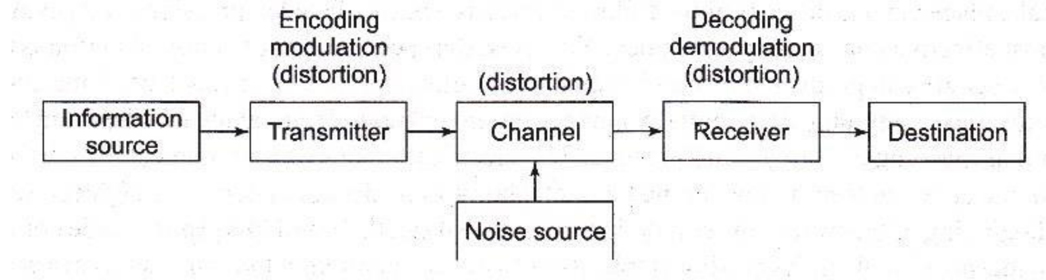
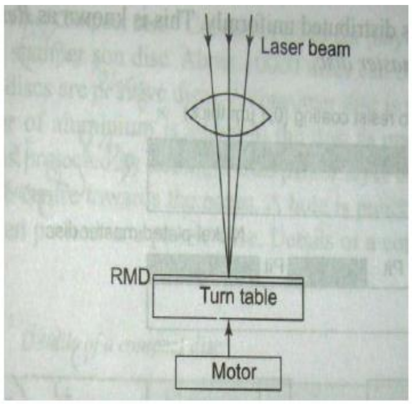


Explanation:

- 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.

	e)	**Draw and explain the block diagram and operation of optical recording on CD.**	**4 Marks**
	Ans:	**Diagram:**	**Diagram 2 Mark** **Explanation 2 Mark**



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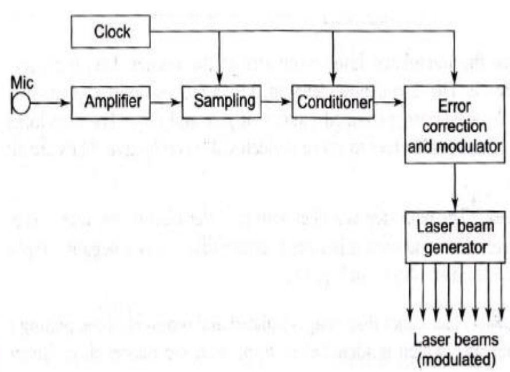


		Explanation: **Recording on CD:** This is done with the help of laser beams, made ON and OFF by digitized audio signals These beams fall on a photo resist material on a rotating disc and caused pits of varying width & fixed depth & thus records signals in binary form, flats & pits making 1s & 0s respectively. 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.	
	f)	**Explain the neat block diagram of Armstrong ?frequency modulator system.**	**4 Marks**
	Ans:	**Block diagram:**	**Diagram** **2 Mark** **Explanation** **2 Mark**

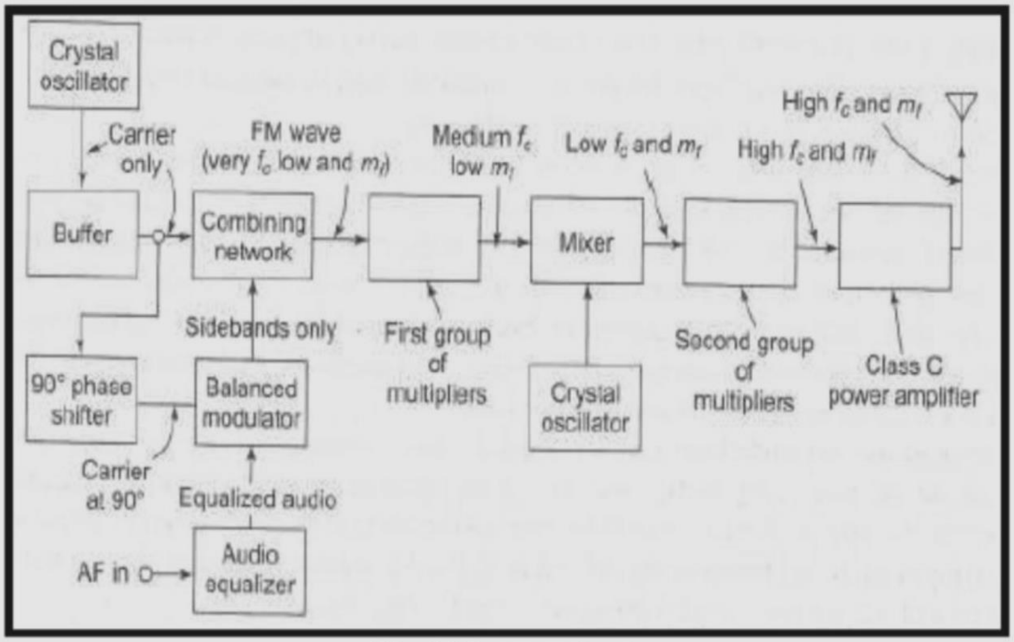
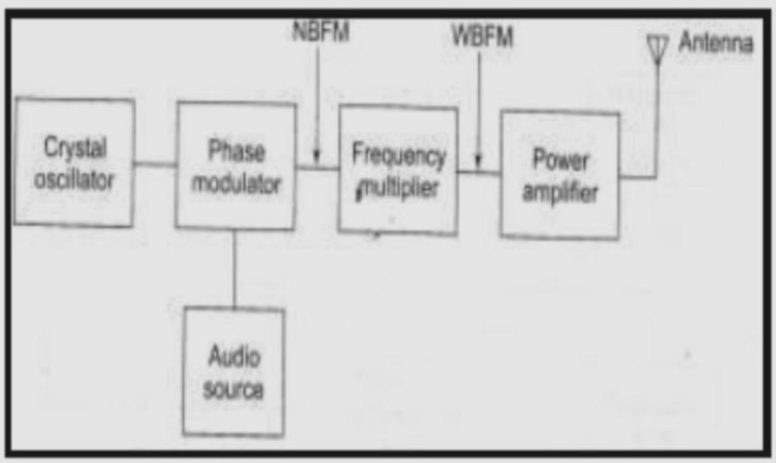


Fig. FM transmitter

OR



Explanation:-

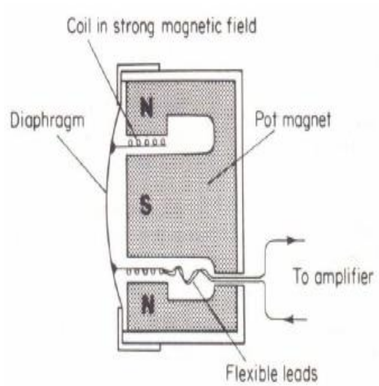
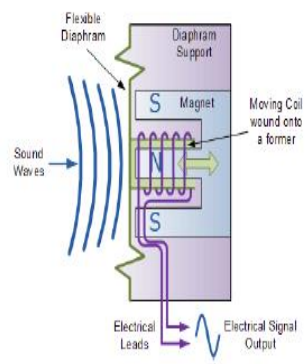
 The crystal oscillator generates the carrier at low frequency typically at 1 MHz .This is applied to the combining network and a 90 degree phase shifter.

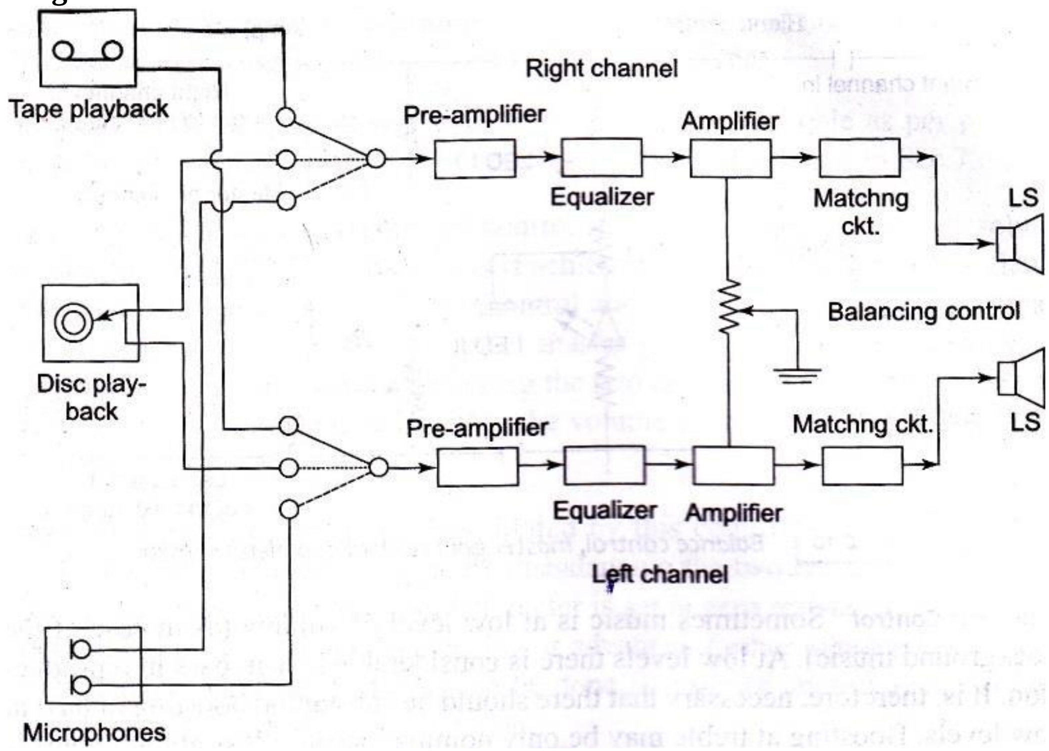
 The modulating signal is passed through an audio equalizer to boost the low modulating frequencies.

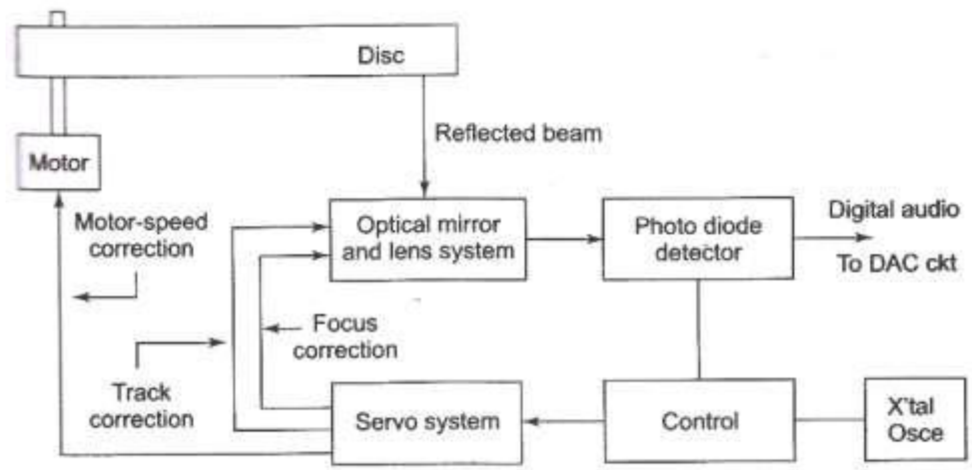
 The modulating signal is then applied to a balance modulator.

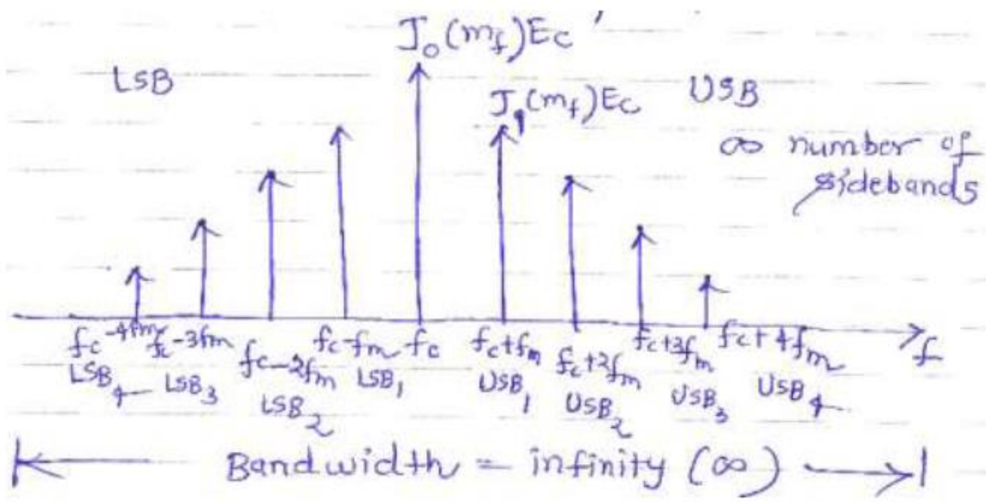
 The balance modulator produces two sidebands such that their resultant is 90 degree phase shifted with respect to the un-modulated carrier.

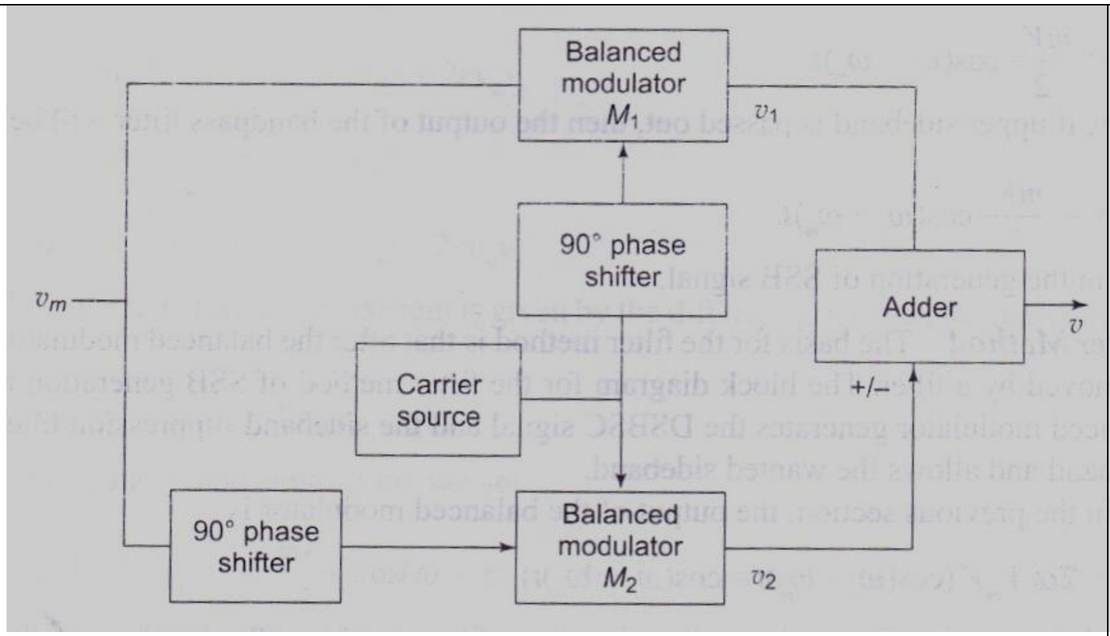
		☐ The un-modulated carrier and 90 degree shifted sidebands are added in the combining network. The output of combining network is equivalent to FM wave. This FM wave has low carrier frequency Fc and low value of the modulating index mf . ☐ The carrier frequency and the modulation index are then raised by passing the FM wave through the first group of multipliers. The carrier frequency is then raised by using a mixer and then the Fc and mf both are raised to required high values using the second group of multipliers. ☐ The FM signal with high Fc and high mf is then passed through a class C power amplifier to raise the power level of the FM signal.	
Q. 4		**Attempt any four :**	**16 Marks**
	a)	**Define phase modulation and modulation index of PM.**	**4 Marks**
	Ans:	**Phase modulation:** 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. **Modulated index:** The modulating index is defines as Mp = δp is expressed in radiance. where δp is maximum frequency deviation.	**Each Defination 2 Mark**
	b)	**Define modulation and state the need of modulation.**	**4 Marks**
	Ans:	**Modulation:** • **Modulation** is a process of mixing a signal with a sinusoid to produce a new signal. • Its process by which modulating signal is superimposing on carrier signal to from modulated signal. • The process by which any parameter of carrier signal (ie. Amplitude, frequency or phase) change with respect to modulating signal. **Need of modulation: -** 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	**Defination 1 Mark** **Need 3 Mark**

	c)	**State the need of PA system. State any four applications of PA system.**	**4 Marks**
	Ans:	**Need of PA system:-** The intensity of sound decrease with distance. Hence 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. **Application of PA system:-** 1) Sports meets 2) Public meetings 3)Auditoriums 4) Concerts & function. 5)To convey information to isolated locations as at railway station, airports, hospitals, factories etc.	**Need 2 Mark** **Application 2 Mark**
	d)	**Draw and describe the working principle of moving coil microphone.**	**4 Marks**
	Ans:	**Diagram:**  **OR**  **Working principle:** Moving coil type microphone uses electromagnetic induction to convert the sound waves into an electrical signal. It has a very small coil of thin wire suspended within the magnetic field of a permanent magnet. As the sound wave hits the flexible diaphragm, the diaphragm moves back and forth in response to the sound pressure acting upon it causing the attached coil of wire to move within the magnetic field of the magnet. The movement of the coil within the magnetic field causes a voltage to be induced in the coil as defined by Faraday’s law of Electromagnetic Induction. The resultant output voltage signal from the coil is proportional to the pressure of the sound wave acting upon the diaphragm so the louder or stronger the sound wave	**Diagram 2 Mark** **Explanation 2 Mark**

		the larger the output signal will be, making this type of microphone design pressure sensitive.	
	e)	**Draw the block diagram of Hi-Fi system and explain the function of each block.**	**4 Marks**
	Ans:	**Diagram:**  **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.	**Diagram 2 Mark** **Explanation 2 Mark**

		□ 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 the block diagram of detection circuit used in CD player.**	**4 Marks**
	Ans:	**Block diagram:**  **Explanation:** □□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.	**Diagram 2 Mark** **Explanation 2 Mark**

		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. 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.	
Q.5		**Attempt any four :**	**16 Marks**
	a)	**Explain the effect of modulation index on bandwidth of FM with neat sketch.**	**4 Marks**
	Ans:	**Diagram:**  **Explanation:** FM has an infinite number of sidebands, as well as the carrier. They are separated from the carrier by fm, 2fm, 3fm,...., and thus have a recurrence frequency of fm. The J coefficients eventually decrease in value as It increases, but not in any simple manner. As seen in Fig. 4.4, the value fluctuates on either side of zero. gradually diminishing. Since each J coefficient represents the amplitude of a particular pair of sidebands, these also eventually decrease, but only past a certain value n. The modulation index determines how many sideband components have significant amplitudes	**Diagram 2 Mark** **Explanation 2 Mark**
	b)	**Explain with neat sketch, the generation of SSB-AM wave using phase shift method.**	**4 Marks**
	Ans:	**Diagram:**	**Diagram**



2 Mark

Explanation:

The phase shift method avoids filters and some of their inherent disadvantages, and instead makes use of two balanced modulators and two phase shifting networks, as shown in Fig. 3.19. One of the balanced modulators, M1, receives the 90° phase shifted carrier and in phase message signal, whereas the other, M2, is fed with the 90° phase shifted message and in phase carrier signal. Both the modulators produce the two sidebands. One of the sidebands, namely, the upper sideband will be in phase in
both the modulators, whereas, the lower sideband will be out of phase. Thus by suitable polarity for M2 output end adding with M1 output results in suppressing one of the sidebands.

Explanation 2 Mark

c) **Draw neat diagram of Ribbon microphone. State its two applications.** **4 Marks**

Ans: **Diagram:**

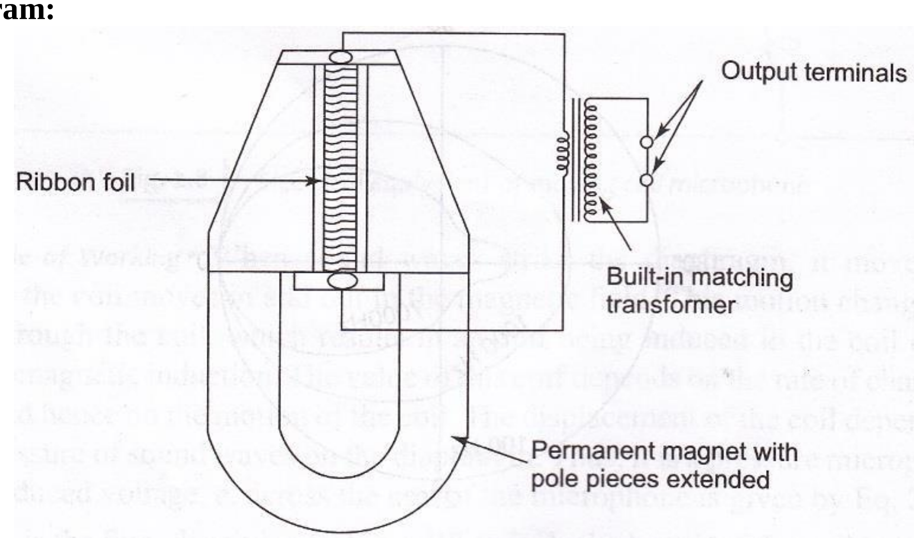
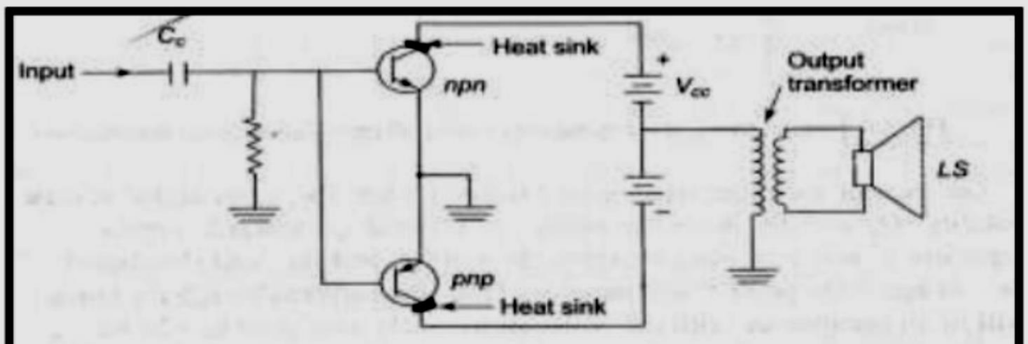
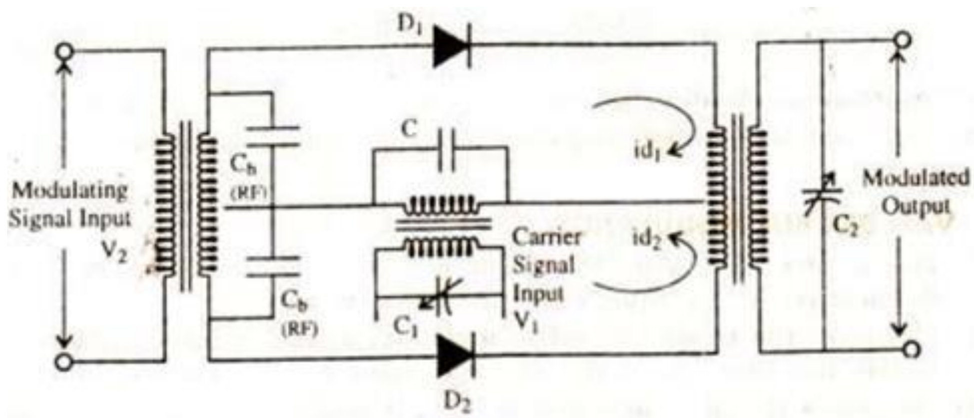
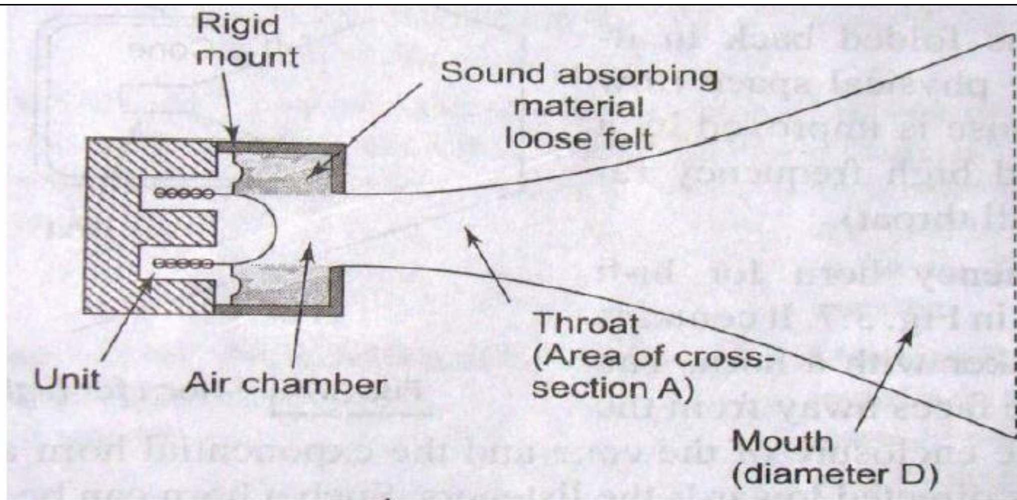


Diagram 2 Mark

Application 2 Mark

		Applications: 1) Used in dramas. 2) Use with public address systems	
	d)	**Draw and explain the working of complementary symmetry push-pull amplifier.**	**4 Marks**
	Ans:	**Diagram:**  Fig. Complementary symmetry push-pull amplifier circuit with output transforme **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.	**Diagram 2 Mark** **Explanation 2 Mark**
	e)	**State the 1 application of each specified microphones :** **i) Lavalier microphone** **ii) Tie-clip microphone** **iii) Shotguns type microphone** **iv) Wireless microphone.**	**4 Marks**
	Ans:	**i) Lavalier microphone** Application: Broadcast presenting and lectures. **ii) Tie-clip microphone** Application: For delivering the lectures	**Each application 1 Mark**

		iii) Shotguns type microphone **Application:** Recording of wildlife, outdoor TV interview in noisy environment **iv) Wireless microphone.** Application: Useful in sports and oath taking ceremony	
	f)	**Draw and explain generation of DSBSC AM signal using diode balanced modulator.**	**4 Marks**
	Ans:	**Diagram:**  : Balanced Modulator circuit using diodes **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 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	**Diagram 2 Mark** **Explanation 2 Mark**
Q.6		**Attempt any four :**	**16 Marks**
	a)	**Describe construction and working principle of horn type loudspeaker.**	**4 Marks**
	Ans:	**Construction:**	**Construction 2 Mark**



Working Principle 2 Mark

Working Principle:-

- A horn type loudspeaker uses a moving coil placed in a magnetic field similar to paper cone type, but instead of radiating acoustic power direct in open space of the listener's area, the power is first delivered to the air trapped in a fixed non vibrating tapered or flared horn and from there to the air in the listener's area.
- Thus it radiates sound power to the air in the space not direct from the diaphragm but indirectly through the horn.
- This is the reason why the horn type loudspeaker is called indirect radiating loudspeaker. The horn does acoustically what the cone does mechanically.

b) **Draw and explain operation of stereo controls used in Hi-Fi system.** **4 Marks**

Ans: **Diagram:**

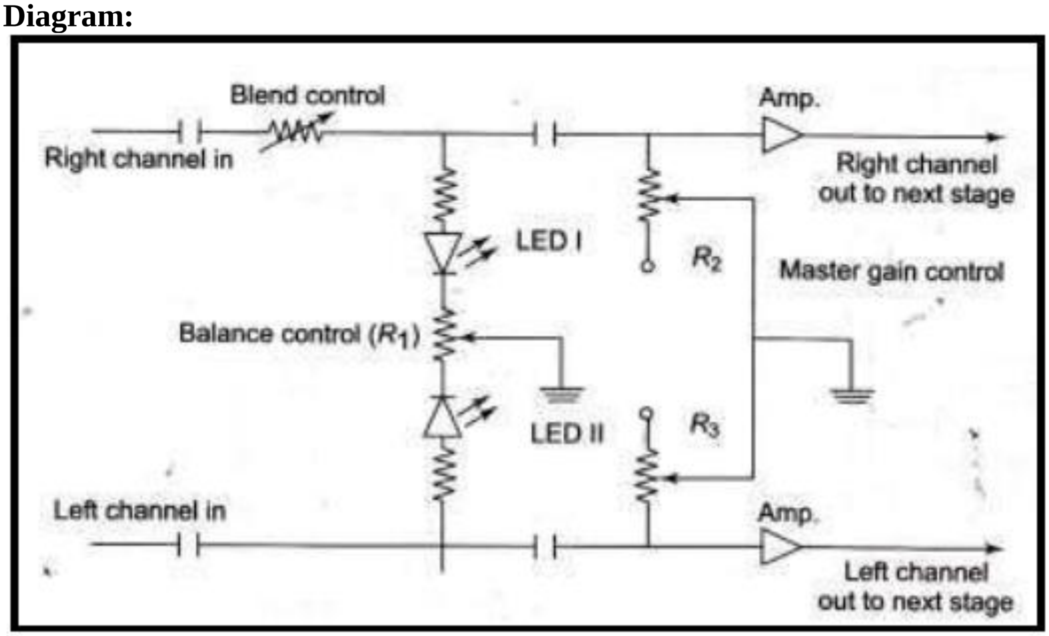
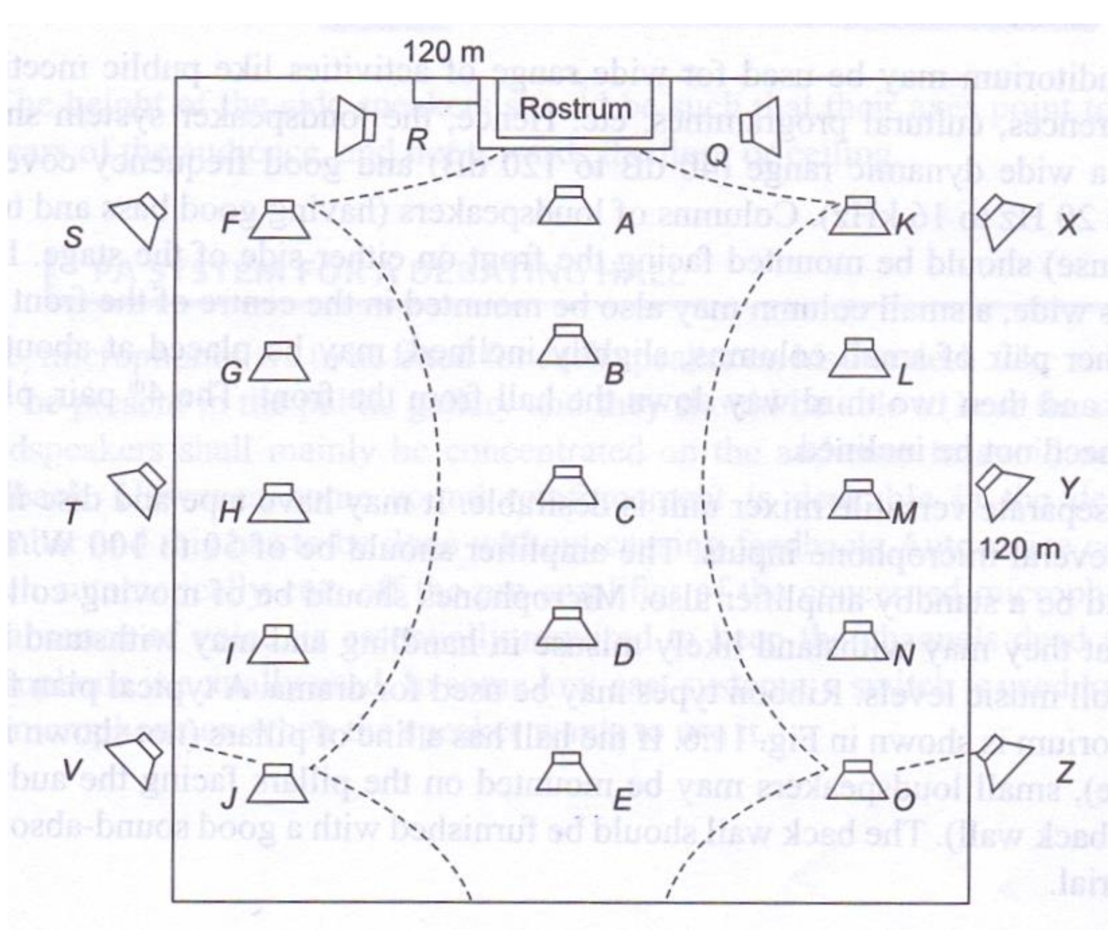
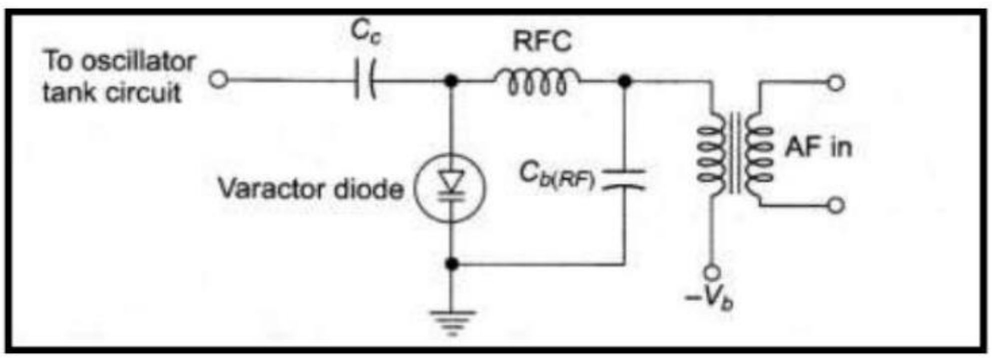


Diagram 2 Mark

		Explanation: 1. **Balance control :** Two amplifier of a stereo system , although independent of each other , are built as matched pair to give equal output for the same input. 2. **Master Gain Control :** A Master gain control is used for adjusting overall volume without disturbing the balance . This is achieved by using dual concentric shafts, the inner shaft adjusts the balance control and the outer shaft, the overall gain or volume of the amplifier. 3. **Blend Control :** The stereo effect is diluted by this control when it is too much left-right effect. Diluting is done by disbalancing the two channels.	**Explanation 2 Mark**
	c)	**Explain how will you install PA system for public meeting.**	**4 Marks**
	Ans:	**Diagram:**  **Explanation:** 1. The loudspeakers A, B, C, D and E in the centre line will give the sense of direction to most of the audience and can be mounted on poles.	**Diagram 2 Mark**

		2. Loudspeakers, 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 on both sides of the central area. 3. To cover the remote semicircular side and comer areas, loudspeakers J, and O are used. These will throw sound power towards corners. 4. The loudspeakers 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 the figure. 6. Microphones should be of cardioid type and the loudspeakers may be of horn type. 7. The output audio power of the amplifier may be calculated by using the formula given in Eq.1l.l. 8. It is preferable to use HOT standby amplifiers with batteries.	**Explanation 2 Mark**
	d)	**Compare monophony and stereophony (any 4 points).**	**4 Marks**
	Ans:	(see table below)	**For each point 1 Mark**
	e)	**Write the causes which affect the fidelity. How it can be minimized ?**	**4 Marks**
	Ans:	**Causes affecting fidelity:** i. High signal to noise ratio.(s/n ratio)	**Causes 2 Mark Remedies 2**

Monophony amplifier	Stereophony amplifier
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.

		ii. Flat frequency response iii. Low nonlinear distortion iv. Large dynamic range v. Creating sense of direction. **Remedies:** i. S/N ratio can be improved by using preamplifier of low noise figures proper shielding, grounding, decoupling & filtering circuits, stabilized power supply, microphones ii. By using coupling capacitor and shunt capacitor in audio amplifier circuits. iii. 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. iv. Dynamic range can be increased by using solid-state amplifier, dynamic microphones & L.S.which are capable of withstanding the large change in loudness. Creating sense of direction can be improved by using high fidelity system.	**Mark**
	f)	**Explain with neat sketch, the generation of FM wave using varactor diode modulator.**	**4 Marks**
	Ans:	**Diagram:**  **Explanation:** • A varactor diode is a semiconductor diode whose junction capacitance varies linearly with the applied voltage when the diode is reverse-biased. • It may also be used to produce frequency modulation. Varactor diodes are certainly employed frequently, together with a reactance modulator, to provide automatic frequency correction for an FM transmitter. • The circuit of fig shows such a modulator. It is seen that the diode has been back- biased to provide the junction capacitance effect, and since this bias is varied by the modulating voltage which is in series with it, the junction capacitance will also vary, causing the oscillator frequency to change accordingly.	**Diagram 2 Mark** **Explanation 2 Mark**