Name of the Lab: Communication Lab
Class: 5th sem (ET&T)
Title of experiment: Study of different type of a microphone their sensitivity & directivity

Q1. What do you mean by microphone?
A1. A microphone is an acoustic-to-electric transducer or sensor that converts sound pressure variation into an electrical signal of same frequency and phase and amplitudes in the same proportion as in the original pressure variation.
Q2: Define sensitivity of a microphone?
A2: Sensitivity indicates how well the microphone converts acoustic pressure to output voltage. A high sensitivity microphone creates more voltage and so needs less amplification at the mixer or recording device. The international standard is made in millivolts per Pascal at 1 kHz. A higher value indicates greater sensitivity. Sensitivity, so −60 dB is more sensitive than −70 dB.
Q3: Define directivity of a microphone?
A3: The directivity of a microphone is defined with the help of a polar diagram .The angle for half power point in a polar diagram represents directivity.
Q4: Requirements of a good microphone?
A4: A good microphone should have high sensitivity, high s/n ratio, flat frequency response, natural frequency, and very low distortion.
Q5: What are the different types of microphone?
A5: Condenser microphone, Dynamic microphone, Carbon microphone, piezoelectric microphone etc
Q6: Can you explain condenser microphone briefly?
A6: Condenser microphone: In a condenser microphone, also called a capacitor microphone or electrostatic microphone, the diaphragm acts as one plate of a capacitor, and the vibrations produce changes in the distance between the plates.
Q7: Can you explain Dynamic microphone briefly?
A7: Dynamic microphone: It work via electromagnetic induction A small movable induction coil, positioned in the magnetic field of a permanent magnet, is attached to the diaphragm. When sound enters through the windscreen of the microphone, the sound wave moves the diaphragm. When the diaphragm vibrates, the coil moves in the magnetic field, producing a varying current in the coil through electromagnetic induction.
Q8: Can you explain Carbon microphone briefly?
A8: Carbon microphone: A voltage is applied across the metal plates, causing a small current to flow through the carbon. One of the plates, the diaphragm, vibrates in sympathy with incident sound waves, applying a varying pressure to the carbon. The changing pressure deforms the granules, causing the contact area between each pair of adjacent granules to change, and this causes the electrical resistance of the mass of granules to change. The changes in resistance cause a corresponding change in the current flowing through the microphone, producing the electrical signal.
Q9: Can you explain piezoelectric microphone briefly?
A9: piezoelectric microphone: A crystal microphone or piezo microphone uses the phenomenon of piezoelectricity — the ability of some materials to produce a voltage when subjected to pressure — to convert vibrations into an electrical signal. An example of this is Rochelle salt (potassium sodium titrate), which is a piezoelectric crystal that works as a transducer.
Q10: Application of microphone?
A10: Microphones are used in many applications such as telephones, tape recorders, hearing aids, motion picture production, live and recorded audio engineering, FRS radios, megaphones, in radio and television broadcasting and in computers for recording voice, speech recognition, VoIP, and for non-acoustic purposes such as ultrasonic checking or knock sensors.

Title of experiment: Study various types of loudspeaker & their characteristic & application.

Q1: What do you mean by loudspeaker?
A1: A loudspeaker (or "speaker") is an electroacoustic transducer that converts an electrical signal into sound waves of the same frequency. The speaker moves in accordance with the variations of an electrical signal and causes sound waves to propagate through a medium such as air or water.
Q2: Requirements of a good loudspeaker?
A2: A good loudspeaker should be efficient, not produce noise signal, have flat frequency response, has low distortion, desired directivity.
Q3: What are the different types of loudspeaker?
A3: Horn type loudspeaker (direct radiating speaker), moving coil cone type loudspeaker (indirect radiating speaker).
Q4: Explain Horn type loudspeaker briefly?
A4: Horn loudspeakers use a shaped waveguide in front of or behind the driver to increase the directivity of the loudspeaker and to transform a small diameter, high pressure condition at the driver cone surface to a large diameter, low pressure condition at the mouth of the horn. This increases the sensitivity of the loudspeaker. The size of the throat, mouth, the length of the horn, as well as the area expansion rate along it must be carefully chosen to match the drive to properly provide this transforming function over a range of frequencies.
Q5: What is the principle of loudspeaker?
A5: The most common type of driver uses a lightweight diaphragm, or cone, connected to a rigid basket, or frame, via a flexible suspension that constrains a coil of fine wire to move axially through a cylindrical magnetic gap. When an electrical signal is applied to the voice coil, a magnetic field is created by the electric current in the voice coil, making it a variable electromagnet. The coil and the driver's magnetic system interact, generating a mechanical force that causes the coil (and thus, the attached cone) to move back and forth, thereby reproducing sound under the control of the applied electrical signal coming from the amplifier.
Q6: Which materials use for diaphragm?
A6: The diaphragm is usually manufactured with a cone- or dome-shaped profile. A variety of different materials may be used, but the most common are paper, plastic, and metal. The ideal material would be stiff, to prevent uncontrolled cone motions; light, to minimize starting force requirements and energy storage issues; and well damped, to reduce vibrations continuing after the signal has stopped.
Q7: What is efficiency of loudspeaker?
A7: Loudspeaker efficiency is defined as the sound power output divided by the electrical power input. Most loudspeakers are actually very inefficient transducers; only about 1% of the electrical energy sent by an amplifier to a typical home loudspeaker is converted to acoustic energy. The remainder is converted to heat, mostly in the voice coil and magnet assembly. The efficiency of loudspeaker drivers varies with frequency as well.
Q8: Can you explain enclosure?
A8: To adequately reproduce a wide range of frequencies, most loudspeaker systems employ more than one driver. Including the particular way two or more drivers are combined in an enclosure to make a speaker system. The role of the enclosure is to provide a place to physically mount the drivers, and to prevent sound waves emanating from the back of a driver from interfering destructively with those from the front; these typically cause cancellations (e.g., comb filtering) and significantly alter the level and quality of sound at low frequencies.
Q9: Explain woofer, subwoofer and tweeter?
A9: A woofer is a driver that reproduces low frequencies (16 Hz – 1 kHz). The driver combines with the enclosure design to produce suitable low frequencies. The drivers are named subwoofers for very low frequencies (20-200Hz). A tweeter is a high-frequency (2 – 20 kHz) driver that reproduces the highest frequencies in a speaker system. Many varieties of tweeter design exist, each with differing abilities with regard to frequency response, output fidelity, power handling, maximum output level, etc.
Q10: Why is the efficiency of a horn type speaker higher as compared to PMMC type?
A10: Due to good matching between the mechanical impedance of the horn and the acoustic impedance of the air outside.

Title of experiment: Study of pre - amplifier and its controls.

Q1: What is the pre-emphasis & de- emphasis?
A1: Pre-emphasis of high notes and de- emphasis of low notes before recording take care of noise and saturation. The process is reversed during playback to restore the originality of sound.
Q2: Why we use pre- emphasis?
A2: Pre- emphasis and equalization essentially are needed to improve S/N ratio and to maintain high fidelity.
Q3: What do you mean by preamplifier?
A3: A preamplifier (preamp), or control amplifier, is an electronic amplifier which prepares an electronic signal for further amplification or processing.
Q4: Why we use preamplifier in sound equipment?
A4: A preamplifier, also referred to as a preamp, is a device typically used along with sound equipment to help improve the overall quality of sound. In order to accomplish this, the preamplifier helps prepare the main amplifier, which increases the power and sound of the equipment, for receiving the electronic signal. Through the help of the preamplifier and the main amplifier, the sound is not altered in quality, but it is much louder.
Q5: Explain the principle of preamplifier and what type of control is applied?
A5: In general, the function of a preamplifier is to amplify a low-level signal to line-level. A list of common low-level signal sources would include a pickup, microphone, turntable or other transducer. Equalization and tone control may be applied. In an audio system, the second amplifier is typically a power amplifier (power amp).
Q6: What are the characteristics of preamplifier?
A6: The preamplifier provides voltage gain from 10 millivolts to 1 volt, but no significant current gain. The power amplifier provides the higher current necessary to drive loudspeakers.
Q7: Where preamplifier is mounted?
A7: Preamplifiers may be
1) Incorporated into the housing or chassis of the amplifier they feed
2) In a separate housing
3) Mounted within or near the signal source, such as a turntable, microphone or musical instrument.
Q8: Can you explain microphone preamplifier?
A8: a simple microphone preamplifier circuit which you can use between microphone and stereo amplifier. This circuit amplifier microphone suitable for use with normal home stereo amplifier line/CD/aux/tape inputs. This microphone preamplifier can take both dynamic and electret microphone inputs (preamplifier provides power foe electret microphone elements
Q9: What is meant by "low impedance"?
A9: Impedance is an electrical term that refers to how much a device impedes the flow of current and is measured in ohms. While there is no set standard, low impedance usually refers to a range of between 150 and 800 ohms. Most professional audio microphones are low impedance. This amplifier circuit designed to work with any low to medium impedance source.
Q10: What is tone control?
A10: To off-set the effect of noise present in the signal, provision of bass and treble controls is made. The combined control is called tone control.

Title of experiment: Study of sound mixer.

Q1: What is sound mixer?
A1: A sound mixer is a device which takes two or more audio signals, mixes them together and provides one or more output signals. As well as combining signals, mixers allow you to adjust levels, enhance sound with equalization and effects, create monitor feeds, record various mixes, etc.
Q2: What is the alternative name of sound mixer?
A2: The alternative name of sound mixer is "fader".
Q3: What is the function of sound mixer?
A3: The function of the mixer stage is to effectively isolate different channels from each other before feeding to the main amplifier.
Q4: What are the applications of sound mixer?
A4: Some of the most common uses for sound mixers include:
1) Music studios and live performances: Combining different instruments into a stereo master mix and additional monitoring mixes.
2) Television studios: Combining sound from microphones, tape machines and other sources.
3) Field shoots: Combining multiple microphones into 2 or 4 channels for easier recording.
Q5: Which type of channel use in sound mixer?
A5: Mixers are frequently described by the number of channels they have. For example, a "12-channel mixer" has 12 input channels, i.e. you can plug in 12 separate input sources. You might also see a specification such as "24x4x2" which means 24 input channels, 4 subgroup channels and two output channels.
Q6: What are the channel inputs?
A6: The first point of each channel's pathway is the input socket, where the sound source plugs into the mixer. It is important to note what type of input sockets are available — the most common types are XLR, 6.5mm Jack and RCA. Input sockets are usually located either on the rear panel of the mixer or on the top above each channel.
Q7: What is the input level of sound mixer?
A7: The level of an audio signal refers to the voltage level of the signal. Signals can be divided into three categories: Mic-level (low), line-level (a bit higher) and loudspeaker-level (very high). Microphones produce a mic-level signal, whereas most audio devices such as disc players produce a line-level signal. Loudspeaker-level signals are produced by amplifiers and are only appropriate for plugging into a speaker — never plug a loudspeaker-level signal into anything else
Q8: Which type of controls using in sound mixer?
A8: When a signal enters the mixer, one of the first controls is the input gain.
<u>Phasing:</u> Some mixers have a phase selector which will change the phasing at the input stage.
<u>Phantom Power</u>: Some mixers have the option to provide a small voltage back up the input cable to power a microphone or other device.
<u>Equalization controls:</u> Most mixers have some of sort equalization controls for each channel. Channel equalizers use knobs and can be anything from simple tone controls to multiple parametric controls.
Q9: what are the outputs of sound mixer?
A9: The main output from most mixing devices is a stereo output, using two output sockets which should be fairly obvious and easy to locate. The connectors are usually 3-pin XLRs on larger consoles, but can also be 6.5mm TR (jack) sockets or RCA sockets. Many mixers include a number of additional outputs, for example, Monitor Feed, Headphones, and Communication Channels etc.
Q10: Types of sound mixer?
A10: There are three types of sound mixer. The simplest one does not use per-amplifiers and amplifiers, but uses only gain controls and isolating series resistors. A little more sophisticated one uses common per-amplifier after isolating resistors. The most sophisticated one has separate per-amplifiers, for separate channels and then after the gain control potentiometers and isolating series resistors, there is a common amplifier.

Title of experiment: Study of stereo system & controls.

Q1: Define the word ‘stereophonic’?
A1: The word "stereophonic"—derived from the Greek, stereos = "solid" and phone = "sound" by analogy with the word "stereoscopic".
Q2: Explain stereophonic system?
A2: Stereophonic sound, commonly called stereo, is the reproduction of sound using two or more independent audio channels through a symmetrical configuration of loudspeakers in such a way as to create the impression of sound heard from various directions, as in natural hearing. Stereo recordings are used in FM broadcasting and Digital Audio Broadcasting (DAB) and in several television systems
Q3: What is the meaning of monophonic?
A3: Monophonic, or "mono" sound, where audio is in the form of one channel, often centered in the sound field. And in monophonic system there is no sense of direction is produces.
Q4: Explain two channel stereo systems?
A4: During two-channel stereo recording, two microphones are placed in strategically chosen locations relative to the sound source, with both recording simultaneously. The two recorded channels will be similar, but each will have distinct time-of-arrival and sound-pressure-level information. During playback, the listener's brain uses those subtle differences in timing and sound level to triangulate the positions of the recorded objects.
Q5: What are the elements of stereo system?
A5: A complete stereo system has several elements including speakers, components, sources and the listening room. The acoustic quality of your listening room is the foundation of a good stereo system and plays an important role in the way your system ultimately sounds. Your listening room is as at least as important as choosing the right speakers and components
Q6: In stereo disc recording which method is used?
A6: Stereo recording on a disc is done by the 45 degree angle method. In this method each channel is recorded along a direction at 45 degree to the horizontal plane of the disc surface.
Q7: In stereo tape recording which method is used?
A7: Each channel’s output goes to a separate head and records a separates track on the tape.
Q8: Which phenomenon is used by listener in stereophony?
A8: Direction.
Q9: Lack of sense of direction in the reproduced sound is called?
A9: Spatial distortion.
Q10: What is the function of balance control in a stereo recording system?
A10: When two independent amplifiers give equal o/p for same I/p. If due to different characteristics of ckt devices there may be variation in the o/p. The balance control compensates such variations.

Title of experiment: Study of Hi-Fi system frequency response.

Q1: What do you mean by fidelity?
A1: In audio system it is used to indicate faithful reproduction of sound. In ideal fidelity system have infinite S/N ratio, flat frequency response for whole audio range, distortion should be zero, high dynamic range, ability to give sense of direction.
Q2: What is the meaning of Hi- Fi?
A2: High fidelity (or HiFi or hi-fi) is the reproduction of sound with high degree of similarity to the original sound. Hi-fi aims to achieve minimal or unnoticeable amounts of noise and distortion. The term "hi-fi" can be applied to any reasonable quality home music system.
Q3: Can you explain the characteristics of Hi- Fi system?
A3: No sound system can be so perfect as to give ideal fidelity. The best fidelity system is less perfect but similar to ideal system that is call hi- fi system. Hi- Fi system have S/N ratio > 50 dB, frequency response within 1 dB, non linear distortion < 1%, dynamic range > 80 dB, stereophonic effect.
Q4: In reproduced sound which type of noise are presented?
A4: Noises in the reproduced sound are hum, hiss, cross-talk, self- oscillation, wow and flutter etc. All these noises reduce S/N ratio.
Q5: What are the modern equipments of hi-fi system?
A5: Modern hi-fi equipment can include signal sources such as digital audio tape (DAT), digital audio broadcasting (DAB) or HD Radio tuners. Some modern hi-fi equipment can be digitally connected using fibre optic TOSLINK cables, universal serial bus (USB) ports (including one to play digital audio files), or WiFi support.
Q6: How to improve frequency response of hi-fi system?
A6: Good design, natural frequency should keep below 40 Hz and above 15 kHz as far as possible .Combination of woofer, subwoofer and tweeter provides good frequency response.
Q7: How to reduce noise in Hi-Fi system?
A7: Selection of low noise transistor for pre-amlifires, adequate filtering and decoupling , use of servo-controlled synchronous motors, steady supply voltage, perfect solder joints, use of diamond stylus and vinyl disc, high dynamic range of the system, all contribute towards reduction of noise.
Q8: Wow and flutters are produced in the recording due to?
A8: Variations in tape speed.
Q9: How to reduce the non-linear distortion?
A9: Non-linear distortion can be reduced by properly selecting the operational point, bias and using negative feed back in amplifier.
Q10: How to create pleasing environment?
A10: Environmental conditions are improved by using suitable absorbers and insulators. New environment can be made even better than the original environment.

Title of experiment: Study of record players, and their operation.

Q1: How is audio program recorded on a gramophone?
A1: Current flows through the coli of disc placed in a magnetic field, the force on the coli makes a cutter vibrate on the lacquer a produced groove of the same frequency as audio current. It is like a motor action.
Q2: How are audio program reproduced from a record?
A2: When the needle (called stylus) of a playback cartridge moves radially over a gramophone record, it picks up vibration from the grooves on the record. These vibrations are converted into audio signals by e.m. induction in the same way as a microphone's diaphragm produces audio signals.
Q3: Why is diamond cutter preferred to a steel cutter in preparing gramophone records?
A3: Diamond cutter is more durable than steel.
Q4: What is the function of cartridge in playback system?
A4: The cartridge converts vibrations received from the stylus into electrical signals the same frequency.
Q5: How is smoothness of a groove improved?
A5: A well-polished, hard and properly shaped stylus can produce reasonably smooth grooves. The smoothness is further improved considerably by heating the stylus.
Q6: What is equalization in the recording and playback system?
A6: Equalization is the process of improving signal to noise ratio by modifying the frequency characteristics before recording and neutralization that modification in the playback amplifier.
Q7: What type of working of disc recording unit?
A7: Like a motor.
Q8: What is the difference between coarse-grooves and micro-grooves?
A8: Coarse-grooves are used with 78 rpm discs and micro-grooves with 45 and 33 rpm discs.
Q9: Why are micro-grooves preferred over Coarse-grooves?
A9: By reducing the size of the grooves, play time is increased.
Q10: Which disc is stamped over to make playback discs?
A10: Negative disc (copper disc) disc is stamped over to make playback discs.

Title of experiment: Study common faults in record player and their rectification.

Q1: When power supply output is zero than what type of fault arises in record Player takes place?
A1:No output on playbook, no recording no erase type problem arise to shoot out this problem check main leads, fuse, switch & pump. If Ok checks power supply.
Q2. What type of fault takes place if dirty or worn out head is there?
A2: If dirty or warm out head is there than high frequency response is poor in playback.
Q3.What is the symptom when microphone or its leads or jack is defective?
A3: Erase normal, playback normal but no recording take place.
Q4. If loudspeaker or its jack is not working property than which type of work is not performed by record player?
A4: Erase is normal but no playback is take place when loudspeaker is not working properly.
Q5. If low pitch noise and playback take place which type of fault is present in record player?
A5: Motor speed is not steady.
Q6: What is the difference between corrective maintenance and quality maintenance?
A6: Corrective maintenance just finds out the faulty component or connection and rectifies the fault. Quality maintenance requires that after the fault is removed, specifications of the set must be checked and if there is any discrepancy in the specifications, it must be removed and specifications should be stored to the original value within tolerable limits.
Q7: Why is it necessary to find out cause of a fault before repairing the fault?
A7: Once a faulty component is located, it is necessary to reason out its cause. It may happen that there is some other fault in the set which became the cause of the fault detected.
Q8: What we do first if audio problems occur?
A8: For audio problems, the first step is to clean the heads, doing a manual wet cleaning of the heads is more effective. Use isopropyl alcohol for a cleaning solution. Do not get the alcohol on the rubber parts of the tape transport. After cleaning, run a tape over the heads.
Q9: After cleaning the head if audio problem still have, so what can we do?
A9: If you still have audio problems, check the connections from the tape deck to your receiver. Check to make sure the problem isn't in the receiver by plugging the audio cables into another input on the receiver. Try listening to the deck through a pair of headphones to see if the problem is coming from the deck itself.
Q10: What are intermittent faults?
A10: Sometimes a fault might be intermittent. Dry solder joint, heating, loose connectors, etc. may cause intermittent faults. Such faults are troublesome to locate. Heating by blower may make an intermittent fault permanent and then the fault diagnosis can be done easy.

Title of experiment: Study of cassette recorder and identification of mechanical and electrical parts.

Q1. What is the basic of the principle of reproduction of magnetic recording of sound?
A1: In magnetic recording, sound pressure variations are recorded in the form of elementary magnets whose length and strength depends on the audio signal. The audio signal is passed through a coil wound a core of soft iron. The core called head become the electromagnet the core has a minute gap. When a tap coated with iron oxide pass across the gap the line of force get a path through the iron oxide, which then is transform into permanent elementary magnets. Thus sound is recorded on the tap.
Q2. Write down basic of the principle of reproduction of sound from magnetic tap?
A2: For reproducing, the tap is again made to pass through a similar gap & the varying magnetic field on the tap causes change of the head, causing induced emf which is amplified & converted into sound.
Q3. Why is iron oxide, & not soft iron used on the tap?
A3: Iron oxides become a permanent magnet according to the audio single, while soft iron achiever varying magnetism. Soft iron provides temporary magnetism on the tap which get vanish after the tap came out of the gap.
Q4. What will happen if the speed of tape is (i) too high (ii) too low?
A4: If the speed of tape is too high, programmed of only small duration will be recorded. If the speed is too low, magnetic cycle will occupy small space, reducing resolution between adjacent cycles.
Q5. Why is soft iron used for the head core?
A5: Soft iron produces temporary magnetism which varies with variations in the input audio signal.
Q6. Why is ac biasing required in magnetic tape recording?
A6: To increase output without distortion.
Q7: What is wow, flutter, rumble, hissing noise?
A7: Variation in speed of motor up to 6Hz gives rise to a distortion called wow and Variation of more than 6Hz and up to about 100Hz is heard as flutter. Motor vibrations are produced as rumble sound. Tiny irregularities in tape cause a hissing noise.
Q8: Advantages of magnetic recording over gramophonic recording?
A8: Tape recording has many advantages over disc recording are: there are no vibrating parts, editing and dubbing is easy, recording can be monitored simultaneously , can be used for immediate playback and same tape can be used again and again for recording different programmes.
Q9: What are the basic components used in a magnetic tape recorder?
A9: The basic component of a magnetic tape recorder is: 1. recording head 2. Magnetic head 3. Reproducing head 4. Tape transport mechanism 5. Conditioning devices
Q10: What are the types of recorder?
A10: There are two types of recorder:

(1) Analog recorder (Magnetic tape recording, Gramophone recording)
(2) Digital recorder (Compact disc recording).

Title of experiment: Study of Trouble shooting of tape recorders and cassette recorders.

Q1: What is difference between trouble shooting, maintenance & servicing?

A1: Troubleshooting means to locate the faulty compunction in the equipment,
Maintenance: - Include corrective repairing.
Servicing: - Include preventive maintenance.

Q2: In which section of tape recorder fault can arise?

A2: Recording section
Playback section
Tope transport section.

Q3: How can you understand playback section is not working properly?

A3: If the output is weak, distorted noisy than playback section is not working properly to rectify this problem we will clean the head, check the amplifier for distortion & rectify the fault.

Q4: How can you remedy the rumble noise on playback?

A4: By checking shock absorbers we can eliminate the rumble noise on playback.

Q5. What defects are found in the tape transport section?

A5: When tape does not more at all or more unsteadily or with jerks unsteadily speed produces wow & flutters & body vibration of motor produce rumble noise.

Q6: How head gap of cassette recorder get worn out?

A6: Due to dirt, magnetized, lack of proper pressure ageist the tap in the head tap it gets worn out & causes poor-high frequency response & more noise in playback. If can be reedit by cleaning head gaps periodically by pure cotton socked in alcohol.

Q7: Why tape dose not move properly?

A7: If loose or broken belt, defective motor, pressure roller not pressing properly, all contribute to improper tape move.

Q8: If driven wheel not pressuring on the flywheel, what type of fault is produced?

A8: No rewinding takes place.

Q9: when trouble occurs in cassette deck, what things are to be checked?

A9: Several things can be done to check for different problems on a cassette deck i.e. Heads, Belts, connections etc.

Q10: If sound seems to oscillating, what can we do?

A10: If the tape mechanism doesn't seem to be turning, or if the sound seems to be oscillating, check to see if the belts are worn or snapped; you may have to remove the cover from the tape deck to check this. If the belts are loose or worn, replace them.

.

Title of experiment: Study of complete circuit drawing of PA and their installation.

Q1. What is the full form of PA system?
A1: Public address is the full from of PA system.
Q2. What is the name of the microphone used in PA system?
A2: Moving coin microphone.
Q3. What is the function of mixer in a PA system?
A3: To isolate different channels is the function of mixer in a PA system.
Q4. Why horn type speaker are more suitable then can type in public meeting?
A4: Efficiency of horn type speaker is 40% whereas of cone type is about 5%.
Q5: Why we use PA system?
A5: The intensity of sound decreases with distance. Hence when a 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 fulfils this function is called public address system.
Q6: Application of PA system?
A6: PA system is used in sports meets, public meetings, auditorium, concerts and functions. It is also used to convey information to isolated locations as at railway stations, airports, hospitals, factories, etc.
Q7: What is acoustic feedback?
A7: When sound from a loudspeaker reaches the microphone, it causes a loud howling sound, called acoustic feedback.
Q8: How should loudspeakers be oriented to optimize the output?
A8: Loudspeakers should be so oriented as to direct the sound towards the audience and not towards the walls. Also, any distant structure should not reflect the sound to cause echoes.
Q9: What are the precautions need to be taken while installing PA system?
A9: The following precautions need to be taken while installing PA system?

1) Acoustic feedback should not occur.
2) Amplifier's audio output power should be uniformly divided by using several loudspeakers.
3) Loudspeakers should be properly oriented to the audience.
4) Excessive reverberation should be reduced by appropriate furnishing.

Q10: Intensity required by audience for comfortable hearing is?
A10: Intensity required by audience for comfortable hearing is 0.1mW.

Title of experiment: Study of CD player

Q1. How of recording on compact discs done?
A1: Recording of sound compact discs us done with the help of laser beam of Infrared light. The beam us incident a photo resists material an a rotating dice & form Pits of varying length & fixed depth thus, the regna is recorded in binary form flats And pit making logic 1, & logic 0.
Q2. How do pits & plats allow recovery of base band reproduction in a compact disc?
A2: Light incident and the compact disc us reflected by the pits thus, the reflected light represents the sequence of 0s, 1st sound in digital form is thus reproduced. Which is converted into analog form by a digital to analog converter?
Q.3. How is optical disc protected from dust grease and scratches?
A3; The disc us covered by a transport plastic or lacquer.
Q.4. What are further development and CD?
A4: Video was also recorded a CD which in known as VCD. In place of infrared light laser beam of small wavelength is used to increases the capacity of CD.
Q5. How much wide is the width of soundtrack on a move film?
A5: 2.5mm wide is the width of soundtrack on a move film.
Q6. Where is the laser beam not reflected in video discs?
A6: Fran pit the laser beam is not reflected in video disc.
Q7: How is the reflecting property of an optical disc increased?
A7: The reflecting property of the disc is enhanced by adding a thin layer of aluminium on to the disc.
Q8: How is sound reproduced from film?
A8: A sharply focused narrow beam of light is made to fall on the sound- track of the film. As the film moves, light passing through the film falls on a photocell which converts the varying intensity of light into electrical signals. The output of the photocell will be audio voltage.
Q9: Advantages of compact discs?
A9: 1) As it is covered by transparent plastic, the track and recording are not affected by dust.
2) S/N ratio is high, as high as 90 dB.
3) Wow and flutter does not exist.
4) Frequency response is excellent and covers complete audio range.
5) Size is quite small.
Q10: Disadvantages of compact discs?
A10: At present, the cost of CD is more than the hi-fi conventional disc. Compared to tape, it has another disadvantage is the recording cannot be erased, and hence fresh recording cannot be done on the same disc.

Title of experiment: Familiarization with studio acoustic.

Q1. Distinguish between echo & reverberation?
A1: - When the reflected sound is heard distinctly, it is called echo. When the reflected sound is not heard distinctly, but causes gradual fading of the continuing refection, it is called reverberation.
Q2. Define reverberation time?
A2: - The reverberation time is defined as the time taken for sound energy in a drop to 10-6 times of its initial value or 60 db below its initial value.
Q3. Why is the reverberation time for a sound studio lower than that for a TV studio?
A3: - In a sound studio, impression of action is given by sound alone that is, one can visualize the seen for the dialogues. But dialogues cannot be seeing the picture.
Q4. Why is the typical value of revere aeration period for a TV studio?
A4: - 0.4 see.
Q5. Reverberation is caused by?
A5: - Reverberation is caused by reflection.
Q6. What is the unit of absorption coefficient?
A6: - Absorption coefficient = $\frac{\text{energy absorbed by unit surface area}}{\text{Total energy received by unit surface area}}$

Being a ratio between similar quantities it has no unit.
Q7. What is the formula of reverberation time?
A7: - $T = 55.3 \frac{V}{Ca}$

C = Velocity of sound = 344m/s
V = Volume of room
a = Total absorption.
Q8: Acoustic requirement of the auditoriums?
A8: The important requirement of the auditorium is comfortable listening in every part, uniform distribution throughout, insulation from outside sound.
Q9: Why we prefer parabolic stage?
A9: The stage can be in paraboloid shape to distribute sound energy uniformly and the parabolic reflecting surface sends parallel beams of sound to the hall.
Q10: Which materials are used for insulation?
A10: All good absorbers like heavy curtains; celotex, etc. are also good insulators and stop the unwanted sound.