Government Girls' Polytechnic, Bilaspur

Name of the Lab: **Communication Lab**

Practical: **Audio Engineering & Acoustic Lab**

Class :**V Semester (ET&T)**

Teachers Assessment: 10 End Semester Examination: 30

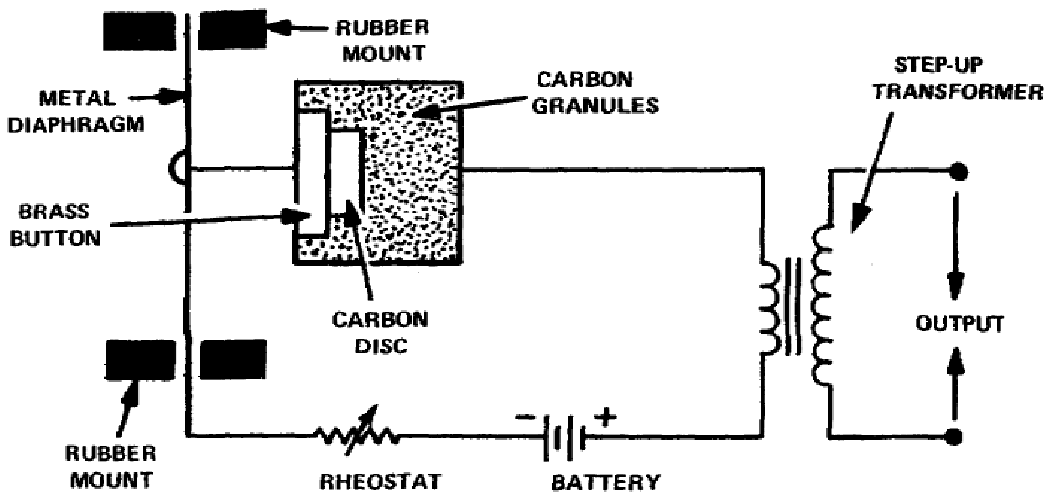
EXPERIMENT No- 1

OBJECTIVE: - Study of different type of a Microphone their sensitivity & directivity.

THEORY: - A microphone (called a mic or mike) is an acoustic-to-electric transducer or sensor that converts sound into an electrical signal. In 1876, Emile Berliner invented the first microphone used as a telephone voice transmitter. Microphones are used in many applications such as telephones, tape recorders, karaoke systems, 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. Most microphones today use electromagnetic induction (dynamic microphone), capacitance change (condenser microphone, pictured right), piezoelectric generation, or light modulation to produce an electrical voltage signal from mechanical vibration.

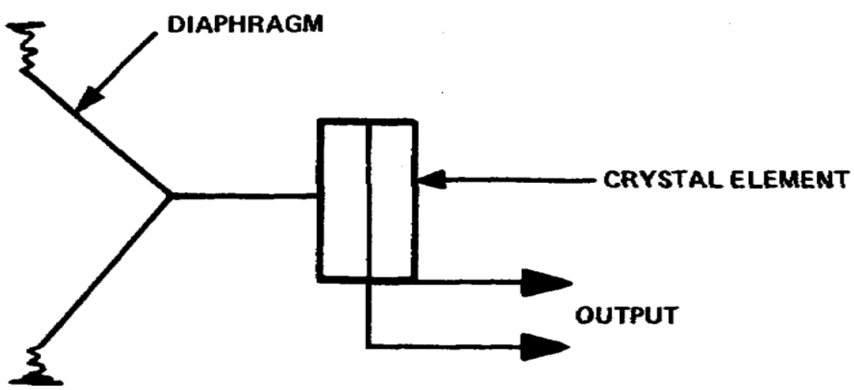
TYPES OF MICROPHONE :- To further narrow a microphone's use for specific job applications, study the specific characteristics of each type. These specific characteristics must be considered to best select a microphone to perform a particular job. The five types of microphones presented here begin with the least effective and progress to the most efficient.

1) CARBON MICROPHONE:- This microphone is activated by carbon granules, and is held in a container (a brass cup) attached to a metallic diaphragm. Sound waves striking the diaphragm cause a change in contact resistance among the granules. This change in contact causes a current from a battery, connected in series with the carbon button (brass cup), and the input of a transformer to vary in amplitude. The result is a current waveform similar to an acoustic waveform striking the diaphragm.
After leaving the output of the transformer, the sound becomes amplified and reproduced. This principle of operation causes a high internal noise making the carbon microphone limited in its usage. All telephone receivers use carbon microphones because telephone transmissions require only intelligibility and voice clarity.
a. Frequency Response: 200-2000 Hz with much distortion present.
b. Sensitivity: Low or poor, must be close to sound source.
c. Impedance: Low.
d. Internal Noise: Very high due to the friction of the carbon granules.
e. Other characteristics: Inexpensive and rugged. Primarily used for communications (telephone).



Carbon microphone

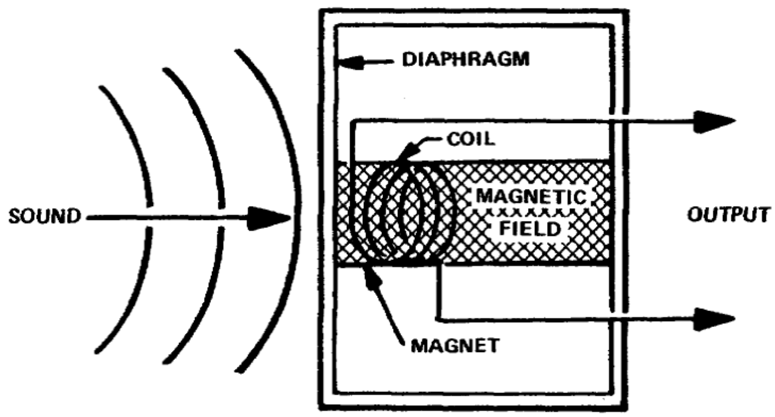
<u>2) CRYSTAL MICROPHONE</u>:- The crystal microphone uses one or more Rochelle salts placed in such a way that when a sound wave strikes them they bend or twist. This action produces an electrical current called the piezoelectric effect (pressure electricity).
a. When exposing the crystal to a mechanical stress such as a sound wave striking a diaphrag (bimorph) or the crystal itself (sound cell), a minute current develops which is directly proportional to the mechanical pressure.
b. The Bimorph and Sound Cell (trade names) make up the two most popular types of designs for crystal microphones.



Bimorph microphone

c. In the Bimorph, sound waves strike the surface of the diaphragm, creating pressure. The crystals consist of two slabs, separated by a thin piece of foil, which are connected to one side of the external circuit. The outer surfaces of the crystal slabs covered with foil connect to the other side of the external circuit.
(1) Frequency Response: 80 - 6590 Hz
(2) Directivity: Nondirectional
(3) Impedance: High
(4) Other Characteristics: Pressure-operated, inexpensive and small in size

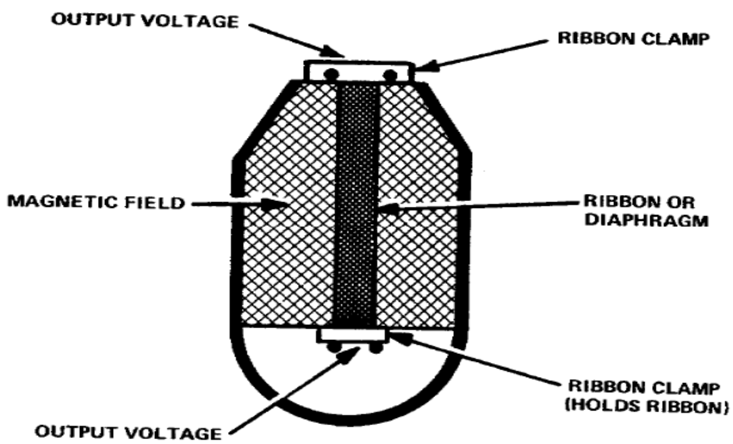
3) DYNAMIC MICROPHONES:-. Known as the most ruggedly constructed of all broadcast microphones, the dynamic microphone may be found in sports, remote broadcasts, and studio recording situations. Because “wind pickup” noise does not affect the microphone, it is used outdoors.
(1) The dynamic microphone favors high frequencies over low. In the studio, this aids the speaker with a deep or bass voice to achieve a higher pitch. Due to its sensitivity to higher frequencies, it accentuates sibilance in a person's voice. (Some people produce a hissing sound while pronouncing s's and ch's. This hissing sound is sibilance.)
(2) Dynamic Microphone Characteristics. This microphone operates on the moving coil generator principle (fig 2-27). It uses a diaphragm and coil which move the field of a permanent magnet. Sound waves striking the diaphragm cause the coil to be moved. This movement in a magnetic field generates an output voltage.



Dynamic microphone

(a) Frequency Response: 20 - 15,000 Hz which is nearly all of the audio spectrum.
(b) Directivity: Omnidirectional
(c) Sensitivity: Very high
(d) Impedance: Low
(e) Other Characteristics: Pressure-operated, rugged and sturdy.

4) VELOCITY MICROPHONE:- Because of its pickup element, manufacturers and studio broadcoasters also call the velocity microphone a ribbon microphone. Do not use the velocity microphone outdoors or in a studio recording situation where it will have to be moved a great deal. Strong winds or rough handling could cause damage to the ribbon.
(1) The velocity microphone favors low over high frequency sounds and consequently may be used to deepen a voice which is too high in pitch. The closer a performer is to this microphone the deeper his voice will sound.
(2) When used in its normal mounted position, the velocity microphone possesses two live and two dead sides. Its pickup pattern is bidirectional. A performer may talk into either of the live sides of the microphone, or two performers may use both live sides alternately or simultaneously.

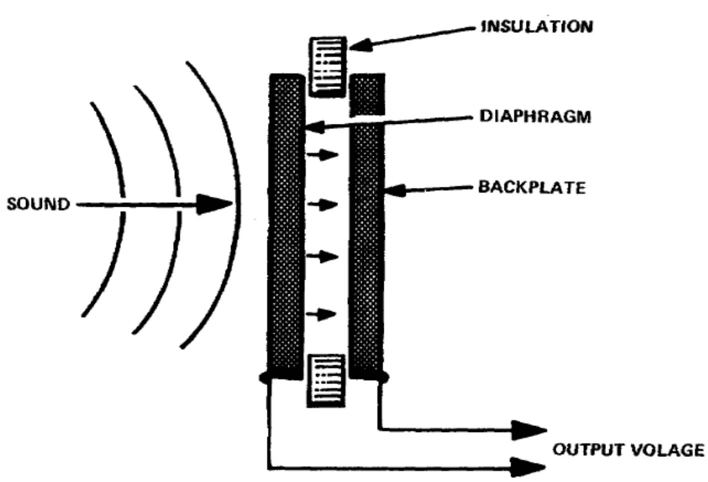


Velocity (ribbon) microphone

Principles of operation. A metallic ribbon is suspended between the poles of a permanent magnet which vibrates freely in a magnetic field. This ribbon, constructed of a thin aluminum strip, serves as the diaphragm. Movement of this ribbon by a sound wave causes the magnetic lines of force to be cut crosswise, creating a corresponding voltage between the two ends of the ribbon.

(a) Frequency Response: 20 - 15,000 Hz
(b) Directivity: Basically bidirectional
(c) Sensitivity: Excellent
(d) Impedance: Low
(e) Other Characteristics: Since the ribbon is easily damaged, never use it outdoors

<u>5) CAPACITOR (CONDENSOR) MICROPHONE</u>:- These microphones get extensive use in recording studios because of the broad frequency range, low distortion, little internal noise, and excellent sensitivity. But, due to their principle of operation, they require a preamplifier as an integral part of the housing, plus a power supply for the preamplifier. They also require an output transformer that converts the extremely high impedance of the microphone capacitor head to a low impedance for unlimited cable length. Their quality and principle of operation combine to make them the most expensive professional microphones.



Capacitor (condensor) microphone

Capacitor Microphone Characteristics. The capacitor microphone operates on the storage of an electrical charge. The head of the microphone contains two plates. One is a stretched diaphragm, the other is heavy backplate. The backplate is insulated from the diaphragm and spaced parallel to the rear surface of the diaphragm.

(2) As sound waves enter the microphone, the pressure causes a change in the spacing of two plates thereby varying the internal capacitance. This produces a signal voltage proportional to the pressure component of the wave. It then gets amplified and passed through the output transformer so conversion from high impedance to low impedance may occur.

(a) Frequency Response: Widest of all microphones; nearly covers the complete audio spectrum

(b) Directivity: Earlier models all omnidirectional, may now be obtained with omni-, bi-, and unidirectional characteristics

(c) Sensitivity: Exceptional

(d) Impedance: High impedance. The output transformer converts the high impedance of the microphone capacitor head to a low impedance

(e) Other Characteristics: Has very faithful signal reproduction. The disadvantages of this microphone are its fragility, and cost. Its extreme sensitivity may prove a disadvantage, depending upon the recording requirements.

RESULT:- Study of microphone has been completed.

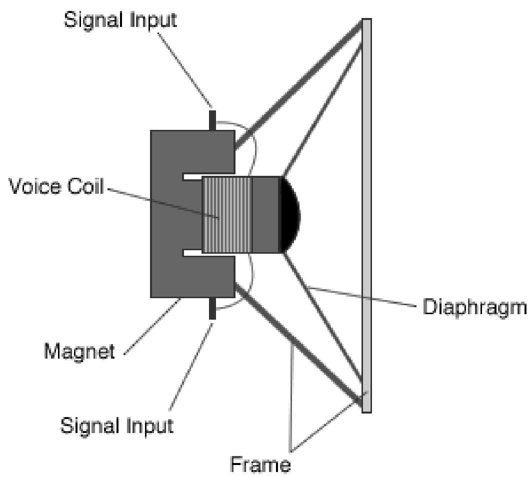
EXPERIMENT No- 2

OBJECTIVE: - Study various types of Loudspeaker & their characteristic & application

THEORY:- A loudspeaker (or "speaker") is an electroacoustic transducer that converts an electrical signal into sound. 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.Loudspeakers (and other electroacoustic transducers) are the most variable elements in a modern audio system and are usually responsible for most distortion and audible differences when comparing sound systems.The term "loudspeaker" may refer to individual transducers (known as "drivers") or to complete speaker systems consisting of an enclosure including one or more drivers. To adequately reproduce a wide range of frequencies, most loudspeaker systems employ more than one driver, particularly for higher sound pressure level or maximum accuracy. Individual drivers are used to reproduce different frequency ranges. The drivers are named subwoofers (for very low frequencies); woofers (low frequencies); mid-range speakers (middle frequencies); tweeters (high frequencies); and sometimes supertweeters, optimized for the highest audible frequencies. The terms for different speaker drivers differ, depending on the application. In two-way systems there is no mid-range driver, so the task of reproducing the mid-range sounds falls upon the woofer and tweeter. Home stereos use the designation "tweeter" for the high frequency driver, while professional concert systems may designate them as "HF" or "highs". When multiple drivers are used in a system, a "filter network", called a crossover, separates the incoming signal into different frequency ranges and routes them to the appropriate driver. A loudspeaker system with n separate frequency bands is described as "n-way speakers": a two-way system will have a woofer and a tweeter; a three-way system employs a woofer, a mid-range, and a tweeter.

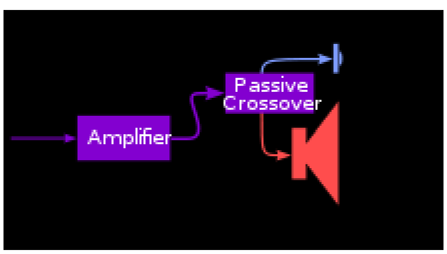
TYPES OF LOUDSPEAKER:

1) HORN LOUDSPEAKERS: Horn loudspeakers are the oldest form of loudspeaker system. The use of horns as voice-amplifying megaphones dates at least to the 17th century, and horns were used in mechanical gramophones as early as 1857. 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 and focuses the sound over a narrower area. 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 (every horn performs poorly outside its acoustic limits, at both high and low frequencies). The length and cross-sectional mouth area required to create a bass or sub-bass horn require a horn many feet long. 'Folded' horns can reduce the total size, but compel designers to make compromises and accept increased complication such as cost and construction. Some horn designs not only fold the low frequency horn, but use the walls in a room corner as an extension of the horn mouth. In the late 1940s, horns whose mouths took up much of a room wall were not unknown amongst hi-fi fans. Room sized installations became much less acceptable when two or more were required.

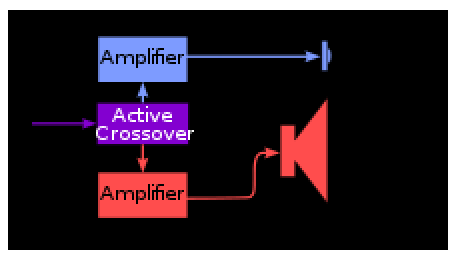


2) PIEZOELECTRIC SPEAKERS: Piezoelectric speakers are frequently used as beepers in watches and other electronic devices, and are sometimes used as tweeters in less-expensive speaker systems, such as computer speakers and portable radios. Piezoelectric speakers have several advantages over conventional loudspeakers: they are resistant to overloads which would normally destroy most high frequency drivers, and they can be used without a crossover due to their electrical properties. There are also disadvantages: some amplifiers can oscillate when driving capacitive loads like most piezoelectrics, which results in distortion or damage to the amplifier. Additionally, their frequency response, in most cases, is inferior to that of other technologies. This is why they are generally used in single frequency (beeper) or non-critical applications.Piezoelectric speakers can have extended high frequency output, and this is useful in some specialized circumstances; for instance, sonar applications in which piezoelectric variants are used as both output devices (generating underwater sound) and as input devices (acting as the sensing components of underwater microphones). They have advantages in these applications, not the least of which is simple and solid state construction which resists the effects of seawater better than, say, a ribbon based device would.

LOUDSPEAKER SYSTEM DESIGN:-



A passive crossover.



Bi-amped.

Used in multi-driver speaker systems, the crossover is a subsystem that separates the input signal into different frequency ranges suited to each driver. The drivers receive only the power in their usable frequency range (the range they were designed for), thereby reducing distortion in the drivers and interference between them.Crossovers can be passive or active. A passive crossover is an electronic circuit that uses a combination of one or more resistors, inductors, or non-polar capacitors. These parts are formed into carefully designed networks and are most often placed between the power amplifier and the loudspeaker drivers to divide the amplifier's signal into the necessary frequency bands before being delivered to the individual drivers. Passive crossover circuits need no external power beyond the audio signal itself, but do cause overall signal loss and a significant reduction in damping factor between the voice coil and the crossover. An active crossover is an electronic filter circuit that divides the signal into individual frequency bands before power amplification, thus requiring at least one power amplifier for each bandpass. Passive filtering may also be used in this way before power amplification, but it is an uncommon solution, due to inflexibility compared to active filtering. Any technique that uses crossover filtering followed by amplification is commonly known as bi-amping, tri-amping, quad-amping, and so on, depending on the minimum number of amplifier channels. Some loudspeaker designs use a combination of passive and active crossover filtering, such as a passive crossover between the mid- and high-frequency drivers and an active crossover between the low-frequency driver and the combined mid- and high frequencies.

ENCLOSURES:-

Most loudspeaker systems consist of drivers mounted in an enclosure, or cabinet. 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.

RESULT:- . Study of loudspeaker has been completed.

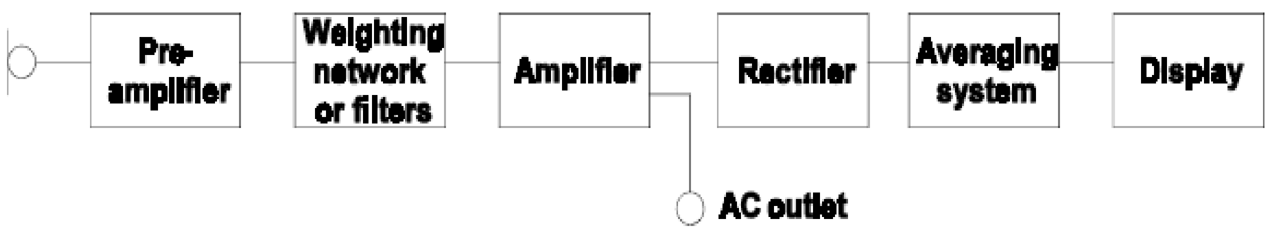
EXPERIMENT No- 3

OBJECTIVE: - Study of Pre-amplifier and its controls.

THEORY:- Many types of measuring systems can be used for the measurement of sound depending on the purpose of the study, the characteristics of sound and the extent of information that is desired about the sound. The various elements in a measuring system are:

a. the transducer; that is, the microphone;
b. the electronic amplifier and calibrated attenuator for gain control;
c. the frequency weighting or analyzing possibilities;
d. the data storage facilities;
e. the display.

Not all elements are used in every measuring system. The microphone can, for instance, be connected to a sound level meter or directly to a magnetic tape recorder for data storage and future measurement or reference.

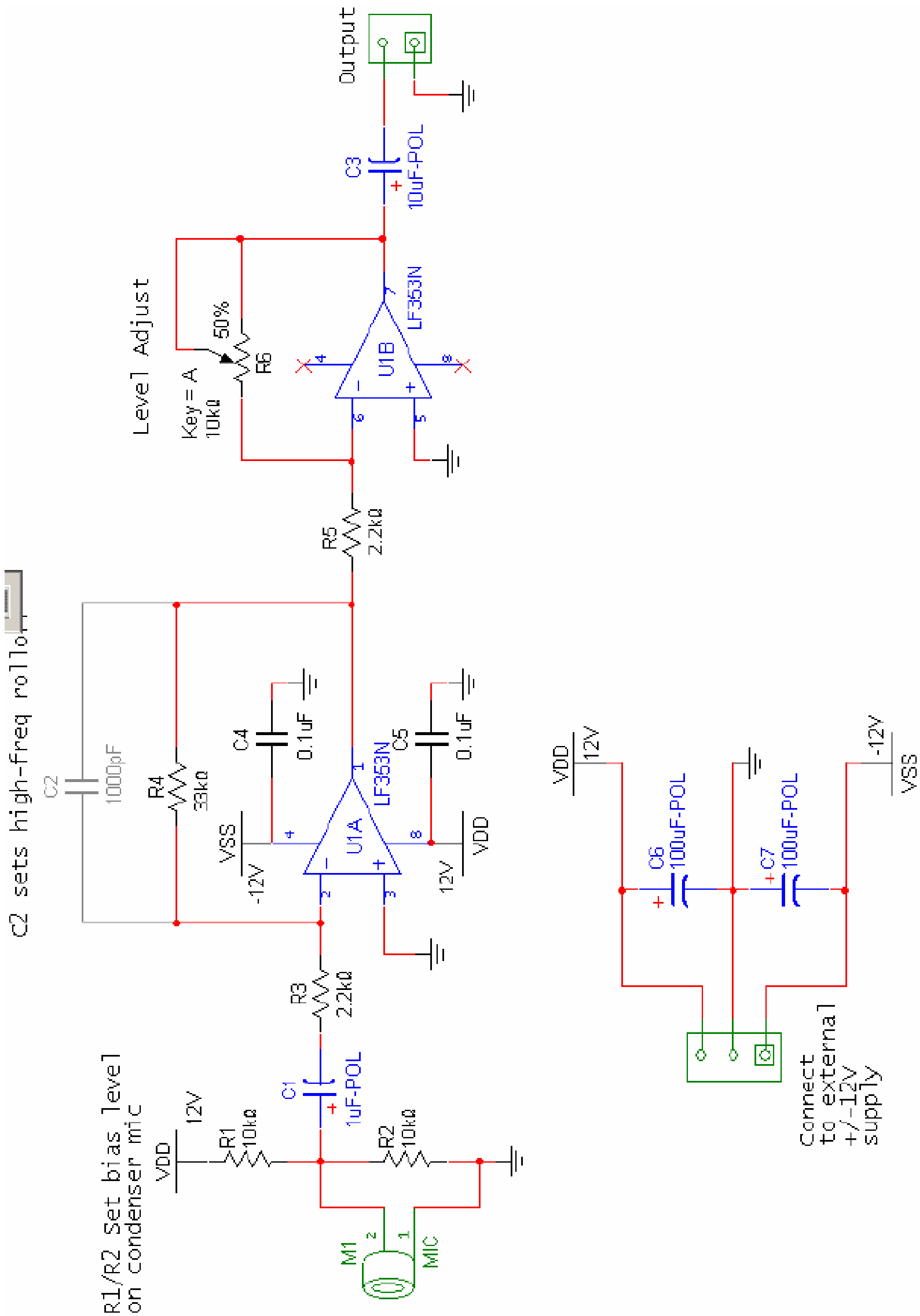


Sound level meter block diagram

The two main characteristics are:

1. The frequency response: that is, the deviation between the measured value and the true value as a function of the frequency. As the ear is capable of hearing sounds between 20 Hz and 20 kHz, the frequency response of the sound level meter should be good, with variations smaller than 1 dB, over that range.
2. The dynamic range: that is, the range in dB over which the measured value is proportional to the true value, at a given frequency (usually 1000 Hz). This range is limited at low levels by the electrical background noise of the instrument and at high levels by the signal distortion caused by overloading the microphone or amplifiers

CIRCUIT DIAGRAM: -



Schematic for the microphone preamplifier.

OBSERVATION :- **Microphone preamp circuit**

Low frequency 3dB cutoff: Predicted: __________ Hz Measured: __________ Hz
High-frequency cutoff (w/C2): Predicted: _________ kHz Measured: __________ kHz
Measured Mid-band Gain: ____________
Typical output signal amplitude for a
whistling tone in mind-band at 6” away from mic: ____________ V
Approximate output noise amplitude: ____________ V

RESULT: - Study of preamplifier has been completed.

PRECAUTIONS: - 1) All the connection should be tight.

2) Ammeter is always connected in series in the circuit while voltmeter is parallel to the conductor.
3) The electrical current should not flow the circuit for long time, Otherwise its temperature will increase and the result will be affected.
4) It should be care that the values of the components of the circuit is does not exceed to their ratings (maximum value).
5) Before the circuit connection it should be check out working condition of all the Component.

EXPERIMENT No- 4

OBJECTIVE: - Study of Sound mixer.

THEORY: In professional audio, a mixing console, or audio mixer, also called a sound board, mixing desk, or mixer is an electronic device for combining (also called "mixing"), routing, and changing the level, timbre and/or dynamics of audio signals. A mixer can mix analog or digital signals, depending on the type of mixer. The modified signals (voltages or digital samples) are summed to produce the combined output signals.Mixing consoles are used in many applications, including recording studios, public address systems, sound reinforcement systems, broadcasting, television, and film post-production. An example of a simple application would be to enable the signals that originated from two separate microphones (each being used by vocalists singing a duet, perhaps) to be heard through one set of speakers simultaneously. When used for live performances, the signal produced by the mixer will usually be sent directly to an amplifier, unless that particular mixer is "powered" or it is being connected to powered speakers.

AUDIO MIXING

1. Definition of Audio Mixing. Audio mixing is defined as the technical and aesthetic blending of sound from two or more sound sources. Each sound source is called a channel. In this definition, there are two key words, “technical” and “aesthetic”. The term technical refers to the actual physical operation and manipulation of equipment and how well it is operated. Aesthetic is used to describe the listening portion of the mixing process. How well does it sound? Is it pleasing to the ear?

2. Three Areas of Audio Mixing. The three areas of audio mixing are really an extension of the definition of audio mixing. The three areas are: sound levels, balance, and operating techniques. The first two terms, sound levels and balance, have to do more with aesthetics while the last term, operating techniques, deals with the technical aspects.

3. Sound Levels. A sound level is the amount of sound loudness heard in the medium. Volume controls on the loudspeakers control sound. Signal levels are found within the equipment in the form of electrical signals and come from mics, turntables, or recorders, etc. Signal levels are controlled by attenuators or pots on the audio console; 100 percent modulation or 0 db is read on the VU meter. Sound levels are

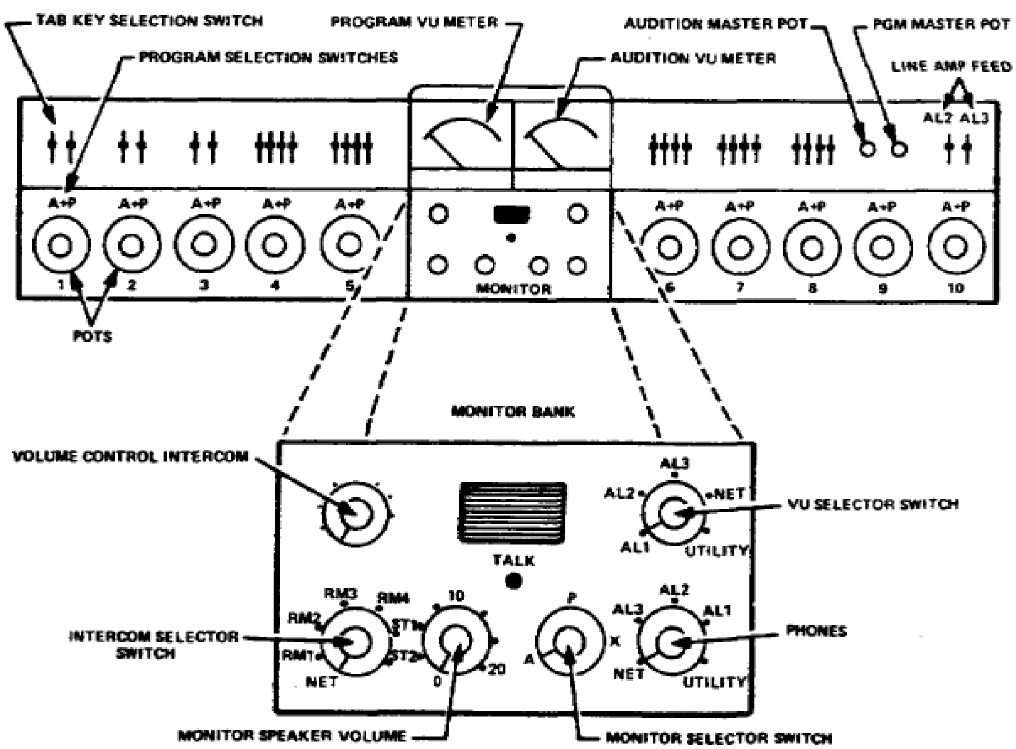
STRUCTURE

A typical analog mixing board has three sections:

- Channel inputs
- Master controls
- Audio level metering

The channel inputs are replicated monaural or stereo input channels with pre-amp controls, channel fader and pan, sub-group assignment, equalization and auxiliary mixing bus level controls. The master control section has sub-group faders, master faders, master auxiliary mixing bus level controls and auxiliary return level controls. In addition it may have solo monitoring controls, a stage talk-back microphone control, muting controls and an output matrix mixer. On smaller mixers the inputs are on the left of the mixing board and the master controls are on the right. In larger mixers, the master controls are in the center with inputs on both sides. The audio level meters may be above the input and master sections or they may be integrated into the input and master sections themselves

AUDIO CONSOLES:-



Audio console, Gates Diplomat

1. Audio consoles basically consist of the same functions and associated equipment. However, the following equipment and associated equipment may not be included on all audio consoles.

2. Subassemblies. The major subassemblies for audio consoles usually consist of a power supply, a program amplifier, a monitor amplifier, cue amplifiers, a pre-amplifier, and a relay system. Each of these subassemblies have a special purpose in the functioning of the audio console.
a. Power Supply. The power supply is the section of the audio console which contains the master power ON-OFF switch and fuses for the console. Its primary purpose is to furnish all the power to the console's components.
b. Program Amplifier. The program amplifier uses a standard 600 ohm impedance and its output level is controlled by the master attenuator. Its primary purpose is to amplify console output.
c. Monitor Amplifier. The monitor amplifier's output level is controlled by the monitor attenuator. Its primary purpose is to amplify monitoring signals, audition signals, and talkback facilities.
d. Cue Amplifiers. The cue amplifier can be used for talkback. Its primary purpose is to amplify signals from inputs for cueing before programming.
e. Pre-amplifiers. The pre-amplifier is used with signals from microphones or phono cartridges. Its primary purpose is to amplify signals from a low level input.
f. Relay System. The relay system can sometimes be DC operated and operates the warning lights. Its primary purpose is to prevent feedback by muting loudspeakers in the presence of live microphones.

3. Controls. The controls for an audio console usually consist of master pots, channel pots, program selector switches, and VU meters.
a. Master Pots. The master pots are used to regulate the overall output of the board. Once the master pots are set up using a 1 KHz tone, they remain as they are until the entire board is neutralized.
b. Channel Pots. The channel pots are used to regulate the output of the individual channels. Channel pots must be adjusted to their individual sources.
c. Program Selector Switches. Program selector switches usually consist of keys or interlocking pushbuttons which are used for rapid selection of inputs to the console. Program selector switches are used to program sound source, monitor a sound source for auditioning or to neutralize a sound source.
d. VU Meters. VU meters are used to measure the average of the complex waves which come from the console. There are usually two meters.
e. Remote Channel. These pots connect the remote lines to TB (talk-back), CUE (monitor output), or MIX (over the air) (Pot 8).
f. Network Channel. Pot 9 is the network channel and can be used as input for a 600 ohm network input such as an off-the-air tuner, or cart tape machine.
g. Nemo Channel. The nemo channel as referred to here; is not used as a remote input. The use of the term here refers simply to a high level channel which has been provided for auxiliary use.
h. Line Amp Feed Switches. The line amp feed switches are the tab key selector switches located above. Line amp feed, key AL3, may be amplified for simultaneous feed from both channels.
i. VU Meters. There are two VU meters located in the center of the audio console. The meter on the left is the program VU meter while the one on the right is used for addition.
j. VU Selector Switch. The VU selector switch allows for monitoring of AL1, AL3, and network on the audition VU meter. AL2 and utility are not used.
k. Monitor Bank. The monitor bank is the area directly below the two VU meters. It consists of six smaller dials, a talk button and a small speaker.
l. Volume Control-Intercom. This dial controls the volume for the intercom/cueing speaker located in the top center of the monitor bank.
m. Intercom Selector Switch. The intercom selector switch determines the booth or remote that is run through the intercom facilities.
n. Monitor Selector Volume. This switch controls the volume for the monitor speaker found in the studio.
o. Monitor Selector Switch. This provides for monitoring of either the audition or program channel or external source over the monitor speaker.
p. Phones Switch. The phones switch allows you to monitor any circuit over the headphones. Only lines AL1 and AL3 are used.

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CONCLUSION:- It's small, sturdy and simple to use, w a perfect, small scale all-round mixer ith enough features to make it a perfect, small scale all-round mixer.In a live situation, we were impressed with its low noise and tough casing, and at home it's a great little unit for putting together sound recordings and

podcasts (although the lack of a USB port is a pain).
Best of all, with a little shopping around on the Internet, the Mackie DFX-6 can be found for as little as £99 which makes it a truly outrageous bargain. Highly recommended.
Our verdict
Ease of use: 87%
Build: 90%
Value for money: 88%
Overall: 88%

SPECIFICATIONS:-

Six-channel on-stage mixer with EMAC digital effects
12-segment stereo LED metering
5-band stereo assignable graphic EQ with bypass switch
EMAC 32-bit digital stereo effects with footswitch bypass, and Level Set LED
Aux 1/monitor send with level control
Aux 2/effects send with level control
Effects-to-monitor control
60mm main mix faders
Two 60mm stereo Aux Return faders with mute switches
Break switch for intermissions
Balanced XLR and 1/4? stereo main outputs
RCA CD/Tape inputs and Tape outputs
Vocal Eliminator circuit for CD/Tape
Master +48 V phantom power switch with LED indicator
Headphone output with level control
Power LED indicator

RESULT:- Study of sound mixer has been completed.

EXPERIMENT No- 5

OBJECTIVE: - Study of Stereo system & controls.

THEORY:- The word "stereophonic"—derived from the Greek, stereos = "solid" and phone = "sound"—was coined by Western Electric, by analogy with the word "stereoscopic". In popular usage, stereo usually means two-channel sound recording and sound reproduction using data for more than one speaker simultaneously. In technical usage, stereo or stereophony means sound recording and sound reproduction that uses stereographic projection to encode the relative positions of objects and events recorded. A stereo system can include any number of channels, such as the surround sound 5.1- and 6.1-channel systems used on high-end film and television productions. However, in common use it refers to systems with only two channels. The electronic device for playing back stereo sound is often referred to as a "stereo".

The term Stereophonic, commonly called stereo, sound refers to any method of sound reproduction in which an attempt is made to create an illusion of directionality and audible perspective. This is usually achieved by using two or more independent audio channels through a configuration of two or more loudspeakers in such a way as to create the impression of sound heard from various directions, as in natural hearing. Thus the term "stereophonic" applies to so-called "quadraphonic" and "surround-sound" systems as well as the more common 2-channel, 2-speaker systems. It is often contrasted with monophonic, or "mono" sound, where audio is in the form of one channel, often centered in the sound field (analogous to a visual field). Stereo sound is now common in entertainment systems such as broadcast radio and TV, recorded music and the cinema. Stereo sound systems can be divided into two forms: The first is "true" or "natural" stereo in which a live sound is captured, with any natural reverberation or ambience present, by an array of microphones. The signal is then reproduced over multiple loudspeakers to recreate, as close as possible, the live sound.

These days "true" stereo is mainly confined to recordings or broadcast of live, acoustic music, particularly classical music. Almost all pop records and movie soundtracks are of the "artificial" variety. In technical usage, true stereo means sound recording and sound reproduction that uses stereographic projection to encode the relative positions of objects and events recorded. 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. Stereo recordings often cannot be played on monaural systems without a significant loss of fidelity. Since each microphone records each wavefront at a slightly different time, the wavefronts are out of phase; as a result, constructive and destructive interference can occur if both tracks are played back on the same speaker. This phenomenon is known as phase cancellation.

A complete stereo system has several elements including speakers, components, sources and the listening room. Whether you're a stereo novice or an experienced listener, this overview covers the essential parts of a good stereo and how to get the best sound from your system.

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. Optimizing speaker placement, listening position and purchasing room acoustic treatments is the best way to get the most performance from your system. Click on the links below for more information and guidelines about speaker placement, room acoustic treatments and listening position.

Stereo Speakers

Stereo speakers determine the overall sound quality of your stereo system more than any other component. Speakers come in all sizes, types, shapes and prices so you have a lot of choices. There is no 'best' speaker, only the one that is right for you and your needs. Sound is a very personal decision and you should listen to several models before purchasing speakers. Learn more about selecting speakers in the following articles.

- Speaker Basics
- How to Choose Speakers
- How to Properly Connect Speakers

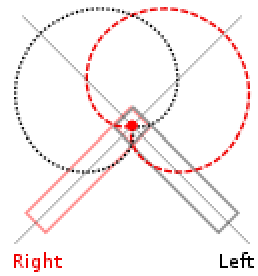
Stereo Components & Product Reviews
Stereo components are available in a wide variety of types and prices from separate components, stereo receivers, integrated amplifiers, or as a pre-packaged system. The stereo components best for you depends on your budget, listening preferences and how often you listen to music. You get a lot for your money with stereo components and even a modest stereo system can provide years of music enjoyment. The following articles and product reviews will help you make the best buying decisions.

- Introduction to Stereo Components
- Buying a Pre-Packaged Stereo System
- Product Reviews

Stereo Source Components
A source component is first in the audio reproduction chain and is just as important as a receiver or speakers. Source components can be analog or digital. As an example, a digital source component can be a CD or DVD player, and an analog source component could be a tape player or phonograph. Learn more about different source components in this section.

RECORDING METHODS:-

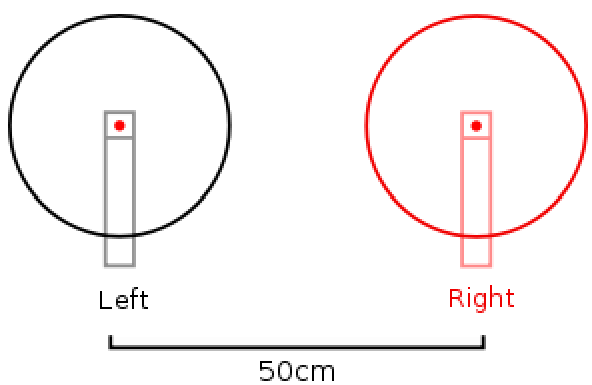
1) X-Y TECHNIQUE: INTENSITY STEREOPHONY:



X-Y stereo microphone placement

Here, two directional microphones are at the same place, typically pointing at an angle between 90° and 135° to each other (see also "The Stereophonic Zoom" by Michael Williams). The stereo effect is achieved through differences in sound pressure level between two microphones. A difference in levels of 18 dB (16 to 20 dB) is needed for hearing the direction of a loudspeaker. Due to the lack of differences in time-of-arrival/phase ambiguities, the sonic characteristic of X-Y recordings has less sense of space and depth when compared to recordings employing an A-B setup. When two figure-eight microphones are used, facing ±45° with respect to the sound source, the X-Y setup is called a Blumlein Pair. The sonic image produced is realistic, almost "holographic". (See also acoustic intensity).

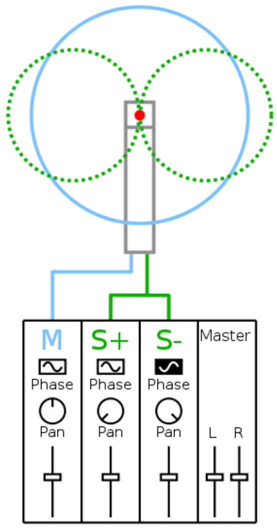
2) A-B TECHNIQUE: TIME-OF-ARRIVAL STEREOPHONY:



A-B stereo microphone placement

This uses two parallel omnidirectional microphones some distance apart, capturing time-of-arrival stereo information as well as some level (amplitude) difference information—especially if employed in close proximity to the sound source(s). At a distance of about 60 cm (0.6 m), the time delay (time-of-arrival difference) for a signal reaching the first microphone and then the other one from the side is approximately 1.5 msec (1 to 2 msec). If you increase the distance between the microphones, you effectively decrease the pickup angle. At a 70 cm distance, it is approximately equivalent to the pickup angle of the near-coincident ORTF setup.

3) M/S technique: Mid/Side stereophony:



Mid-Side stereo microphone technique

This coincident technique employs a bidirectional microphone facing sideways and another microphone (generally a variety of cardioid, although Alan Blumlein described the usage of an omnidirectional transducer in his original patent) at an angle of 90°, facing the sound source. The left and right channels are produced through a simple matrix: Left = Mid + Side; Right = Mid − Side (the polarity-reversed side signal). This configuration produces a completely mono-compatible signal and, if the Mid and Side signals are recorded (rather than the matrixed Left and Right), the stereo width can be manipulated after the recording has taken place. This makes it especially useful for film-based projects.

Binaural recording:

Engineers make a technical distinction between "binaural" and "stereophonic" recording. Of these, binaural recording is analogous to stereoscopic photography. In binaural recording, a pair of microphones is put inside a model of a human head that includes external ears and ear canals; each microphone is where the eardrum would be. The recording is then played back through headphones, so that each channel is presented independently, without mixing or crosstalk. Thus, each of the listener's eardrums is driven with a replica of the auditory signal it would have experienced at the recording location. The result is an accurate duplication of the auditory spatiality that would have been experienced by the listener had he or she been in the same place as the model head. Because of the inconvenience of wearing headphones, true binaural recordings have remained laboratory and audiophile curiosities. However "loudspeaker-binaural" listening is possible with Ambiophonics

Playback:

Stereophonic sound attempts to create an illusion of location for various sound sources (voices, instruments, etc.) within the original recording. The recording engineer's goal is usually to create a stereo "image" with localization information. When a stereophonic recording is heard through loudspeaker systems (rather than headphones), each ear, of course, hears sound from both speakers.

RESULT:- Study of Stereo system has been completed.

EXPERIMENT No- 6

OBJECTIVE: - Study of Hi-Fi system frequency response.

THEORY:- High fidelity—or hi-fi—reproduction is a term used by home stereo listeners and home audio enthusiasts (audiophiles) to refer to high-quality reproduction of sound [1] or images. Ideally, high-fidelity equipment has minimal amounts of noise and distortion and an accurate frequency response as set out in 1973 by the German Deutsches Institut für Normung (DIN) standard DIN 45500. This standard was well intentioned, but only mildly successful in defining 'high-fidelity'. DIN 45 500 approval provided audio equipment buyers with reassurance that their equipment was capable of good quality reproduction. In theory, only stereo equipment that met the standard could bear the words 'hi-fi', but in practice, the term was widely misapplied to audio products that did not remotely approach the DIN basis specifications.

HISTORY:- The 1920s saw the introduction of electronic amplification, microphones, and the application of quantitative engineering principles to the reproduction of sound. Much of the pioneering work was done at Bell Laboratories and commercialized by Western Electric. Acoustically-recorded disc records with capriciously peaky frequency response were replaced with electrically recorded ones. The Victor Orthophonic phonograph, although entirely acoustic, was created by engineers who applied waveguide technology to the design of the interior folded horn to produce a smooth frequency response which complemented and equalled that of the electrically recorded Victor Orthophonic records.

Meanwhile, the rise of radio meant increased popularity for loudspeakers and tube amplifiers, so there was a period of time during which radio receivers commonly used loudspeakers and electronic amplifiers to produce sound, while phonographs were still commonly purely mechanical and acoustic. Later, electronic phonographs became available, as stand-alone units or designed to play through consumer's radios. The now ubiquitous RCA connector was first introduced by the Radio Corporation of America for this purpose.

The development of Sound film in the 1930s led motion picture companies to develop amplification and loudspeaker systems to fill movie theaters with good quality sound at a reasonable volume. To achieve this result, they employed loudspeakers with separate sections for low and high frequencies ("woofers" and "tweeters"), connected via an audio crossover network, and more carefully engineered enclosures. This development exposed the public to better fidelity than home equipment was capable of at the time. Some movie stars purchased movie theater sound equipment for use in their homes but the cost and size put them out of reach for anyone of modest means.

ASCERTAINING HIGH FIDELITY:

In a double-blind experiment, neither the individuals nor the researchers know who belongs to the control group and the experimental group. Only after all the data has been recorded (and in some cases, analyzed) do the researchers learn which individuals are which. A commonly-used variant of this test is the ABX test. This involves comparing two known audio sources (A and B) with either one of these when it has been randomly selected (X). There is no way to prove that a certain lossy methodology is transparent. [2] To scientifically prove that a lossy method is not transparent, double-blind tests may be useful. Modularity Integrated, midi, or lifestyle systems, also known as music centres or minisystems, contain one or more sources such as a CD player, a tuner, or a cassette deck together with a preamplifier and a power amplifier in one box (where Midi has no connection with MIDI technology in electronic instruments). Such products are generally disparaged by audiophiles, although some high-end manufacturers do produce integrated systems. The traditional hi-fi enthusiast, however, will build a system from separates, often with each item from a different manufacturer specialising in a particular component. This provides the most flexibility for piece-by-piece upgrades.

For slightly less flexibility in upgrades, a preamplifier and a power amplifier in one box is called an integrated amplifier; with a tuner, it is a receiver. A monophonic power amplifier, which is called a monoblock, is often used for powering a subwoofer. Other modules in the system may include components like cartridges, tonearms, turntables, Digital Media Players, digital audio players, DVD players that play a wide variety of discs including CDs, CD recorders, MiniDisc recorders, hi-fi videocassette recorders (VCRs) and reel-to-reel tape recorders. Signal modification equipment can include equalizers and signal processors.

This modularity allows the enthusiast to spend as little or as much as he wants on a component that suits his specific needs. In a system built from separates, sometimes a failure on one component still allows partial use of the rest of the system. A repair of an integrated system, though, means complete lack of use of the system.

Another advantage of modularity is the ability to spend one's money on only a few core components at first and then later add additional components to one's system. Because of all these advantages to the modular way of building a high-fidelity system instead of buying an integrated system, audiophiles almost always assemble their system from separates. Some of the disadvantages of this approach are increased cost, complexity, and space required for the components.

Modern equipment

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.

<u>RESULT</u>: - Study of HI-Fi system has been completed.

EXPERIMENT No- 7

OBJECTIVE: - Study of Record players, changers and their operation.

THEORY:- A record changer or autochanger is a device that plays multiple gramophone records in sequence without user intervention. Record changers first appeared in the late 1920s, and were common until the 1980s. Record player or phonograph, device for reproducing sound that has been recorded as a spiral, undulating groove on a disk. This disk is known as a phonograph record, or simply a record (see sound recording sound recording, process of converting the acoustic energy of sound into some form in which it can be permanently stored and reproduced at any time.

In using a record player, a record is placed on the player's motor-driven turntable, which rotates the record at a constant speed. A tone arm, containing a pickup at one end, is placed on the record. The tone arm touches the groove of the record with its stylus, or needle. As the record revolves, the variations in its groove cause the stylus to vibrate. The stylus is part of the pickup, a device that also contains a transducer transducer, device that accepts an input of energy in one form and produces an output of energy in some other form, with a known, fixed relationship between the input and output.

These signals are then increased in size by an amplifier amplifier, device that accepts a varying input signal and produces an output signal that varies in the same way as the input but has a larger amplitude. After leaving the amplifier, they are passed to a loudspeaker loudspeaker or speaker, device used to convert electrical energy into sound. It consists essentially of a thin flexible sheet called a diaphragm that is made to vibrate by an electric signal from an amplifier .

Although sound waves had been recorded in the middle of the 19th cent., the first machine to reproduce recorded sound, the phonograph, was built by Thomas A. Edison in 1877. Edison's records were made of tinfoil, upon which a groove of unvarying lateral direction but varying depth was cut; later this method became known as "hill-and-dale" recording. In 1887, Emile Berliner invented the disk record (patented 1896), which has grooves of unvarying depth but of varying lateral direction. His method, called lateral recording, superseded the earlier method. Berliner also invented the matrix record, from which unlimited duplicate recordings could be pressed. Early turntables were operated by a spring-driven motor that required rewinding for each record played; later the use of an electric motor made rewinding unnecessary.

Instrument for reproducing sounds. A phonograph record stores a copy of sound waves as a series of undulations in a wavy groove inscribed on its rotating surface by the recording stylus. When the record is played back, another stylus (needle) responds to the undulations, and its motions are then reconverted into sound. Its invention is generally credited to Thomas Alva Edison (1877). Stereophonic systems, with two separate channels of information in a single groove, became a commercial reality in 1958. All modern phonograph systems had certain components in common: a turntable that rotated the record; a stylus that tracked a groove in the record; a pickup that converted the mechanical movements of the stylus into electrical impulses; an amplifier that intensified these electrical impulses; and a loudspeaker that converted the amplified signals back into sound. Phonographs and records were the chief means of reproducing recorded sound at home until the 1980s, when they were largely replaced by recorded cassettes and compact discs.

Record player or phonograph, device for reproducing sound that has been recorded as a spiral, undulating groove on a disk. This disk is known as a phonograph record, or simply a record (see sound recording). In using a record player, a record is placed on the player's motor-driven turntable, which rotates the record at a constant speed. A tone arm, containing a pickup at one end, is placed on the record. The tone arm touches the groove of the record with its stylus, or needle. As the record revolves, the variations in its groove cause the stylus to vibrate. The stylus is part of the pickup, a device that also contains a transducer to convert these mechanical vibrations into corresponding electrical signals. These signals are then increased in size by an amplifier . After leaving the amplifier, they are passed to a loudspeaker that converts them into sound.

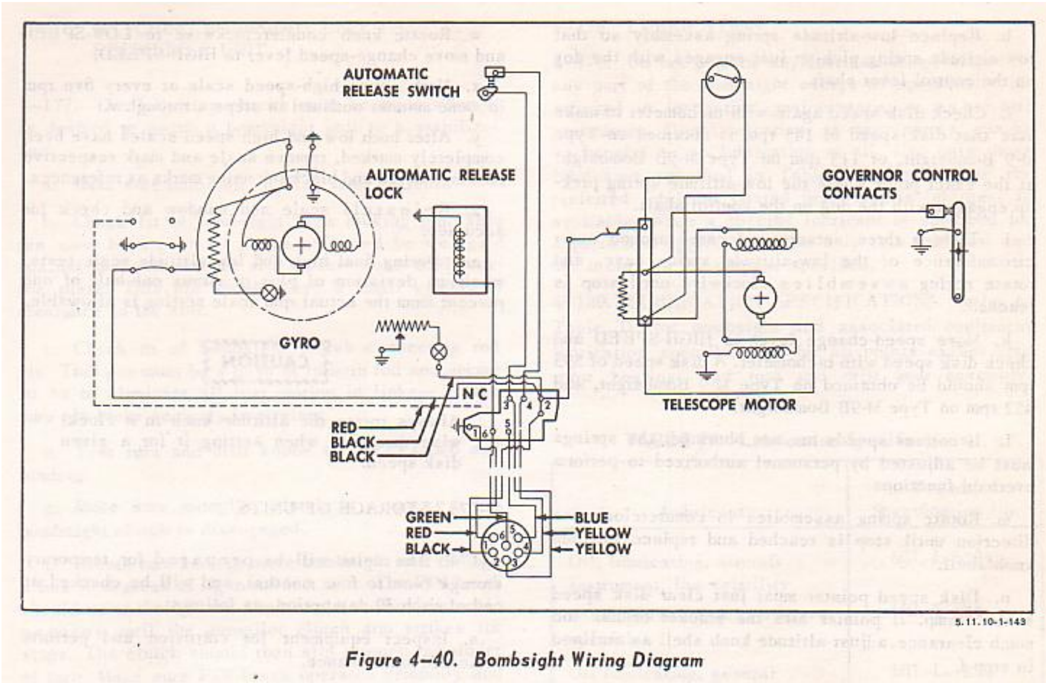


Figure 4–40. Bombsight Wiring Diagram

OPERATION:-

The purely mechanical mechanisms of record changers were often very complex. Changers typically had an extended central spindle that the records were stacked on, and an extra arm designed to hold the stack steady. Some units had feelers that could detect the size of each record (standard sizes 7", 10", or 12") and position the tone arm accordingly. Some, including the changer pictured, used a variable size sensor which allowed sizes other than the three standard sizes to be played.[11] (Note that the pictured Dual 1003 has four sizes loaded, and records sizes can be mixed in any order.) The more basic models required the record diameter to be set manually, and hence did not allow records of different sizes to be stacked together. The following devices were the most popular (with examples):

Three size sensors:

- Size selector knob - no size intermix (BSR 1968 to 1973)
- Size and speed selector knob - no size intermix, some types can't be played automatically
- Rising feelers in or alongside turntable - no size intermix, but automatic sensing of size (PE after 1970)
- Falling record sensor - random intermix - sizes mixed in any order (BSR before 1968)
- Rising and falling record sensors - Intermix 10" and 12" records, 7" played separately (V-M 1950 to 1970)
- Unplayed stack sensor - Arranged intermix - large records before small (Webster Chicago 1950 to 1953)
- Unplayed stack arm tip sensor - Arranged intermix (Collaro/Magnavox after 1967)

INSTRUCTIONS:-

1) Locate the power cord and plug it into a standard household outlet. Open/remove the top of the player to access the turntable and operate the record player.

2) Place a 33 or a 78 vinyl record on the turntable so the center hole is over the post and the record is flat. To play a 45, which has a larger center hole, you'll need an adapter. If there's no pull-up adapter on the record player turntable, you'll need to get one. This adapter is a plastic piece that snaps into the hole on the record. It has a small hole in the middle which fits over the turntable post.

3) Locate the power button on the record player. It's often part of the volume control. Turn the button on to operate the record player. Then locate the speed adjuster, which is usually beside it. Select the proper speed for the record you're playing.

4) Carefully lift up the tone arm from its rest and move it over to the record on the turntable. Place the needle underneath on the outside edge of the vinyl record. Set it down gently so it doesn't scratch the record or damage the needle.

5) Adjust the volume button and the tone button, if applicable. Once the record is finished, carefully lift the tone arm up and back to its original position on its rest. Shut the power off and wait for the turntable to stop. Carefully lift the vinyl record up and off the turntable.

6) Replace the record with another selection. If the next record is also a 45, carefully push the adapter out of the center, then snap it into the center of the next selection.

RESULT: - Study of Record players, changers and their operation has been completed.

EXPERIMENT No- 8

OBJECTIVE: - Study common faults in Record player and their rectification.

THEORY:- Turntables, the rotating part of a phonograph or gramophone, play vinyl records with the aid of a stylus, or needle, attached to a tone arm. Since the advent of compact discs and downloads, phonographs and turntables remain in use mainly by vinyl aficionados and by nightclub DJs who occasionally need to troubleshoot their turntables

TROUBLESHOOTING OF RECORD PLAYER:-

I firmly believe that the best way to listen to music is with vinyl. I don't consider myself an audiophile, or do I groan when friends play CDs or MP3s, but there's something about the large, physical artifact of the vinyl and the simple beauty of a good turntable that I like. Nevertheless, turntables have certain problems that other audio players are exempt from. Here's a look at four of the most common turntable problems, and easy instructions on how to fix them.

1. Dirty stylus: - your turntable suddenly begins to sound, well, not so beautiful, the most likely culprit other than a damaged record is a damaged needle. Always be sure to clean the stylus of your turntable, if possible once daily; it doesn't take too long, just put some isopropyl alcohol on a cotton ball and gently run it from back to front on the stylus. There's also stylus cleaning tools that you can buy. With some simple cleaning, you won't get a distorted or overdriven type of sound, and your stylus will last much longer.

2. Belt issues: - your turntable suddenly stops turning, the belt has probably broken or slipped. New belts can be bought online, but be sure to either look for the exact model of your turntable or measure the length of the belt carefully with string. If you've completely lost the belt, be sure to check the bottom of the platter of the turntable before buying a replacement--I felt like a puts once when I failed to do this, and spent $20 on a belt that I already had.

3. Extremely quiet sound: The most common cause of extremely quiet--or loud--sound occurs when a turntable is plugged into the wrong type of input on a receiver. Older turntables especially, need to be plugged into a "phono" jack, which provides additional power for their fairly weak signals. Otherwise, you'll hear a bad sound, or you'll barely be able to hear anything. The good news is that most stereo receivers have multiple ins to handle both types of turntable outputs.

4. Constant needle skipping:-If your needle skips, no matter how small the vibration, you'll probably need to balance your turntable better. Most turntables have adjustable feet, and with a balance you can achieve much greater stability in your turntable. Keep it on a high, even platform, not on the ground, and away from excessive vibration, to the best of your ability. Remember, records aren't CDs; you have to treat them very carefully, or you'll damage them.

INSTRUCTIONS:-

1) Consult the manual for your turntable to troubleshoot issues. It will have illustrations to help you locate the cause of the problem. Manuals contain troubleshooting or frequently asked questions lists for common problems.

2) Clear off a table or desk where you can fix your turntable. Remove all other items from the desktop and clean the surface.

3) Preserve the tone arm. When examining or fixing your turntable, don't flip them over and put weight on the tone arm to troubleshoot the problem. You'll compromise the tone arm or damage it, and then you'll have to repair it. Remember to clip the tone arm down with string before flipping the turntable over.

4) Examine your turntable for dust. Wipe off your turntable on a regular basis to alleviate excess dirt and dust that seeps into wires and the stylus.

5) Neutralize hum by securing the ground wire. Always attach wires and chords firmly to prevent interruption when playing records.

6) Clean the belt if you have problems with playing speed. Don't use harsh solvents to clean your turntable or loosen stiff parts. The solution will affect other parts of the turntable that should remain dry.

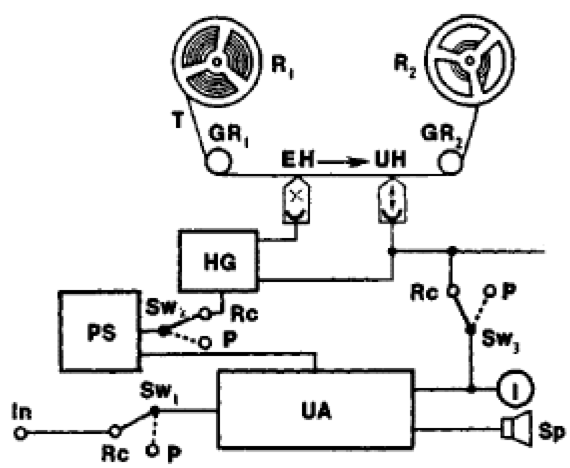
7) Move levers, knobs, the tone arm or other parts gently. Forcing parts to adjust them will only cause more damage

RESULT: - Study of common faults in Record player and their rectification has been completed.

EXPERIMENT No- 9

OBJECTIVE: - Study of Cassette recorder and its complete circuit draw.

THEORY:- **Tape Recorder :**It is an electronic and tape transport mechanism in which a magnetic tape is recorded music or speech The tape recorder consists of following controls.
A tape recording machine is required to perform three important functions viz. recording, playback and erasing. recorder system.
In the block diagram, the same amplifier is used for recording and playback and is called the record/playback amplifier. During the recording process a ganged switch connects the input of the amplifier to the microphone and the output is connected to the record head so that the amplified audio signals are impressed on the moving tape. In the playback position, the magnetic signals picked up or detected from the tape by the same head are applied to the input of the amplifier for reproduction by the loudspeaker. One and the same head performing the combined functions of recording and playback is known as the record playback head. The tape is wound on the supply reel and is moved at constant speed and under constant tension in front of the record / playback head during recording and playback operations. The tape is finally collected by the take-up reel.
An erase head located to the left of the record/playback head wipes off any information from the tape before the tape reaches the record head for a fresh recording. The erase head is energized by the HF oscillator which also applies an AC bias to the record head during the recording process for improving the quality of recording. The erase h egad is not energized ruing the playback process to prevent the recorded matter from being erased. A tape transport system consisting of a motor, drive assembly, belts and pulleys enables the tape to move at constant speed during record and playback process and also to move faster during the fast-forward and rewind processes.



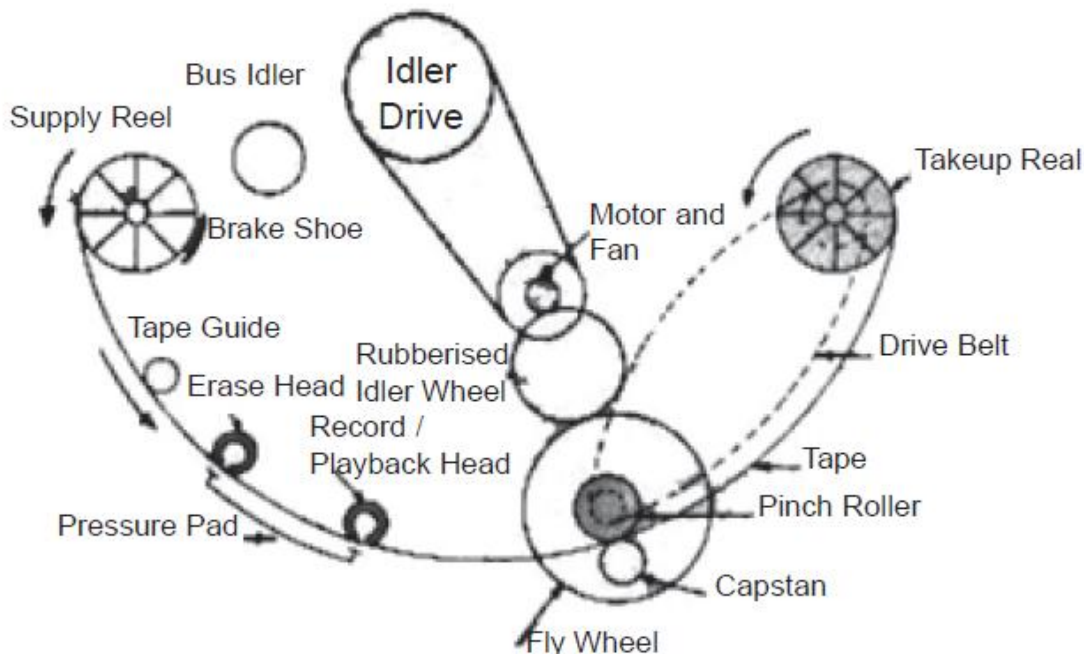
Block Diagram of Tap Recorder

TAPE TRANSPORT MECHANISM:-

1. ON/OFF switch
2. Volume control
3. Play press button
4. Rewinding
5. Fast Forward button
6. Recording
7. Eject / Stop
8. Pause.

All above are called operating controls of a tape recorder. The tape transport mechanism and motor recording / play back heads, cassette and electronic circuit, which operates in recording / playback mode of operation.

Two-in-one: In this system a radio and tape recorder is separated by a lever switch. In the circuit tape recorder operation is same. Radio operation as follows A transistor/IC version receiver is like any other normal receiver in which valves have been replaced by transistors. With the use of transistors, the size and weight of all other components required for the circuit is also considerably reduced. Thus, reasonably prices pocket transistor receivers are not available in all convenient shapes and sizes. Except for some receivers meant ot receive only local stations, all other modern transistor receivers are of the super heterodyne type. A transistor super heterodyne receiver makes use of the stages which function in the same way as the stages of a valve type receiver. However, there are certain differences in the circuits employed in the two cases because of some basic differences between the properties of transistors and vacuum tubes. In order to bring out clearly the points of difference and similarity between the circuits used in the two types of receivers, the circuit diagram of a standard transistor receiver.



TAPE TRANSPORT MECHANISM

FUNCTIONS OF THE TAPE TRANSPORT ARE:-

(1) Reproducing. With the recorder/reproducer turned on and the tape properly threaded, selecting the reproduce mode causes the control circuitry to release the main-reel brakes and starts the reel motors, the capstan motor, and the capstan idler. As a result, the recorded tape is pulled past the heads at a constant speed. The signal sensed by the record or reproduce head is equalized and amplified.

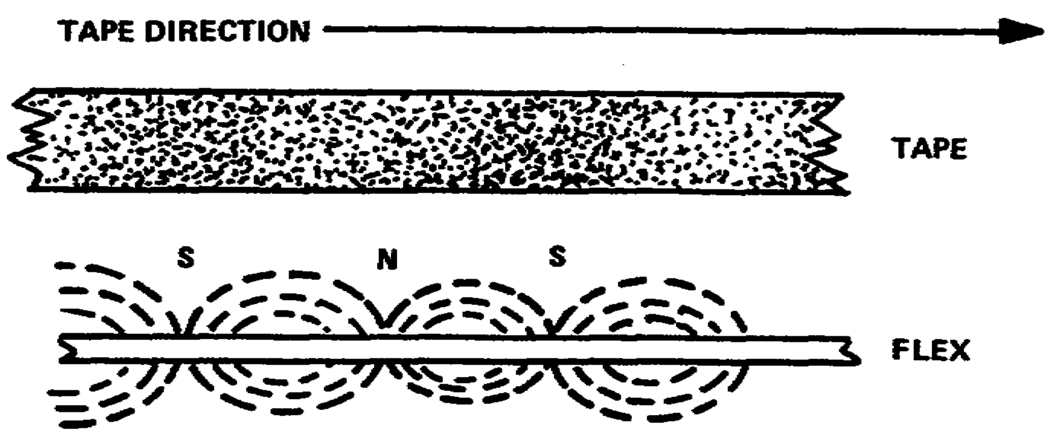
(2) Recording. In the record mode, the tape is moved as in the play mode. During recording, an erase signal from an internal oscillator is fed to the erase head, which clears any previously recorded signals from the tape before it reaches the record head. Information to be recorded is amplified, mixed with a bias signal, and applied to the record head. The information is recorded on the tape as it is pulled past the record head.

(3) Fast Forward/Rewind. With the recorder/reproducer turned on and the tape properly threaded, selecting the fast forward mode locally or remotely causes the control circuitry to release the main-reel brakes and apply full power to the take-up reel motor and hold back power to the supply reel motor. The control circuitry also causes the tape to be lifted away from the heads. Tape is then rapidly wound onto the take-up reel. The rewind mode is similar to the fast forward mode, except that full power is applied to the supply reel motor, and the tape is wound rapidly onto the supply reel.

(4) Edit. Three edit modes are selectable at the front panel of the recorder/reproducer: (1) stop/edit, (2) fast wind/edit, and (3) play/edit. Selecting the stop/edit mode sets only the edit brakes of the tape reel motors, thus facilitating manual cueing and threading of the tape. Selecting the play/edit mode causes the tape to be pulled past the heads and spilled off the right side of the transport. This mode is typically used when unwanted tape is to be cut off. The fast wind/edit mode brings the tape into contact with the heads while the tape is being moved in the fast forward or rewind modes, making the recorded portions audible for high-speed search.

HEAD ASSEMBLY:-Professional tape recorders employ three separate heads (fig 3-31). In their order of head placement they are the erase head, the record head, and the reproduce head. The first to be discussed will be the erase head.

(1) Previously recorded signals are erased from a magnetic tape by passing a high frequency current through the erase head (fig 3-32). The tape passes over the erase head before it arrives at the record head. To erase in the record mode, a high frequency signal supplied by the recorder's erase oscillator, feeds current to the erase head. The current causes a strong alternating magnetic field to be developed. This magnetic field returns a previously recorded tape to its original state before being rerecorded.



(2) As audiotape moves across the record head, molecules on the tape orient themselves with the direction of a magnetic field. The magnetic field is directly proportional to the audio signal being fed to the record head. This audio signal reaches the tape from the record head with unnecessary distortion or quality loss by a carrier signal commonly referred to as the bias.

USE OF A TAPE RECORDER:-

The criteria for the selection of a tape recorder are:

1) The frequency response at the different speeds. Usually the limits are directly proportional to the speed;
2) The range of speeds;
3) The dynamic range;
4) The cross channel attenuation;
5) The presence of band pass filters enabling the elimination of low frequency noise;

6) The quality of the indicating device and of the input potentiometers, preferably graduated in dB;
7) The possibility of controlling the output signal;
8) The protection against dust;
9) The protection against vibration susceptibility which increases the internal noise level;

ADVANTAGES AND DISADVANTAGES:-

Advantages: - Magnetic recording tape provides the audio industry with several positive features:

(1) Easy to use: Bulky recording equipment is not needed. Recorders no larger than the palm of ones hand permit recording at locations where other methods cannot.

(2) Information storage: Several tracks of information can be recorded on magnetic tape and be easily retrieved.

(3) Immediate playback: Recorded information can be played back immediately after it has been recorded without special processing.

(4)Physical makeup: Magnetic tape can be cut, spliced, and easily repaired if broken.

Disadvantages: - There are a few drawbacks of magnetic recording tape:

(1) Effects of external magnetism: Magnetic recording tape is greatly affected by external magnetic fields from motors and electromagnets which can cause partial or even total erasure of the recorded information.

(2) Temperature and humidity variations: Magnetic recording tape expands and contracts with extreme variations in temperature. The plastic backing becomes brittle when cold, and stretchable when hot. Moisture retained within the iron oxide particles causes some of the particles to fall off, resulting in tape head clogging.

(3) Dirt, dust, and oil: Dust or dirt settling on the tape can cause loss of frequency response or loss of the entire signal. Introduction of body oils from the fingers through frequent handling can also cause loss of frequency response.

RESULT: - Study of Cassette recorder has been completed.

EXPERIMENT No- 10

OBJECTIVE: - Trouble-shooting of cassette recorders.

EQUIPMENT REQUIRMENT: - Tape recorder kit, Two-in-one kit and multimeter, servicing tools and instruments

THEORY:- While cassette decks and recorders have dwindled in popularity, many people still have old models that may be starting to see some problems, either with the sound or with parts, such as the rubber wheels, starting to wear out. Several things can be done to check for different problems on a cassette deck. If your cassette tape player or recorder is not working properly, there are some troubleshooting steps you can try before purchasing a replacement or paying to have it repaired.

Please skip to the heading which refers to the problem your tape player is having, then follow the troubleshooting steps below it :-

1) Poor sound quality:-

1. Try replacing the tape player's batteries or connecting it to an AC power source, if it isn't already using one.
2. Use a cleaning cassette in the tape player. Some slightly more expensive cleaners include a de-magnetizing feature which removes unwanted magnetism from the mechanism.
3. The cassette you are using may have been recorded in a recorder with low batteries or another problem; try troubleshooting it with a different tape.

2) Won't play a tape:-

1. Make sure the "PAUSE" and "MUTE" buttons are not pushed in, if it has these. Confirm that any power or speaker cords are properly attached.
2. Many cassette players have an "AC-DC" switch on the front or side; ensure it is in the proper position ("AC" for power cord, "DC" for batteries or cigarette lighter power).
3. Remove and re-insert the cassette. If this doesn't work, try lightly pressing on the cassette door while the "PLAY" button is pushed down.
4. If you can see the cassette moving, but there is no sound, try troubleshooting it by using a different external speaker or headphones.
5. Consider replacing the power adapter/supply, if it has one and there is absolutely no response from the player (no indicator lights, radio, cassette movement, etc.)

3) Plays, won't record:-

1. Check to see if the cassette's plastic write-protection tab(s) were removed; if so, it cannot be recorded to unless you put tape over their holes.
2. If you are using a microphone, look for a mic. Volume control on the cassette recorder. It might be turned down too low. Try using a different microphone, there's a slight chance it could have failed.

4) May need repair:-

If these troubleshooting steps do not make your cassette player or recorder work properly, it might be necessary to replace the motor or rubber belts inside it, and/or make other repairs.

If your cassette tape player is acting up, there are troubleshooting steps you can take to remedy the issue:-

1) Clean and Demagnetize Heads: - For audio problems, the first step is to clean the demagnetize the heads. While cleaning tapes work for light cleaning, doing a manual wet cleaning of the heads is more effective. Use isopropyl alcohol for a cleaning solution, and lint free swabs to apply the alcohol, gently cleaning the heads. Do not get the alcohol on the rubber parts of the tape transport. After cleaning, run a demagnetizing tape over the heads.

2) Check Belts and Heads: - 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. Eventually, the tape heads may wear out. Check the heads themselves to see if they show excessive wear or appear scratched. If they are, have the heads replaced.

3) Check Connections: - 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.

4) Clean the Inside of the Deck: - Because cassette decks have lots of moving parts, dust and small particulates can be a pervasive issue, especially if your cassette deck hasn't been used in awhile. To clean your cassette deck, take a few quarter-sized cotton balls and dip them in rubbing alcohol. Then open the tape-loading deck and rub all of the surfaces with the cotton swab. Continue swabbing the inside until the cotton swab comes out clean, and make sure to get in between all the moving parts.

Operator's maintenance of a turntable involves the following steps:-

1) Disconnect and inspect the AC power cord for serviceability.
2) Clean all exterior surfaces.
3) Check drive belts (if belt driven) for cracks and proper tension. Replace belt if broken, cracked, or stretched, using associated tools.
4) Clean the drive system using denatured alcohol and cotton swabs.
5) Check and lubricate the disc center guide/spindle using a lubricant.
6) Visually inspect the tone arm for damage. Replace if damaged.
7) Remove and inspect the cartridge. Replace if necessary.
8) Remove and inspect the stylus for damage if bent, crimped, or broken.
9) Check for correct speed control using a strobe disc.
10) Check input and output connectors.

RESULT: - Study of troubleshooting of cassette recorder has been completed.

EXPERIMENT No- 11

OBJECTIVE: - Study of PA system and its installation.

THEORY:- There are many different types of public address system available on the market for various requirements. The term PA or public address means different things to different people, for example a PA rig is a common term for a high power amplification system used for performances by artists and bands. Public address and sound reinforcement are terms used for amplification systems used for other purposes. From earliest history, man has felt the need to communicate with others, both singly and in groups. It has been group communication however, that has presented the difficulties as it has been limited by the limited number of people that can be conveniently assembled within earshot of the speaker.

Whenever a very large crowd had to be addressed in earlier times, the only way to do it was with relay speakers, who stood within earshot of the speaker and repeated what he had said at suitable intervals. Others within hearing distance transmitted the speech even further until it reached the edge of the crowd. These speakers were usually arranged in concentric circles, with the original speaker at the centre. There were frequent pauses and delays, and errors would be passed on and added to. Today amplification systems enable gatherings of all sizes to be addressed with ease under varied circumstances. There is increasing demand for such facilities, from clubs and small local halls, to large sports staid, from single speaker meetings to multi speaker conferences, for stage plays and musical events, for indoor assemblies and outdoor. The main requirements of any public address or PA system where speech is being amplified are that the programmed material must be heard comfortably by all the audience or public present, and that the speech is easily intelligible. Naturalness is a desired quality and in instances where speech reinforcement is required, if everyone present can hear clearly without being aware that amplification is in use then the installation can be claimed as successful.

Sound Equipment and PA Systems Consultancy:-

We have built up our own **PA System**s over time, seeking improvements in audio quality, functionality, portability, set-up and take-down simplicity and flexibility, without making the cost prohibitive for the end-user.

We know a lot about what we would get if our budget was bigger, and what we could dispense with if it was smaller.

If you want to build your own system, or want one built for you, we can help with:

Advice:-

This is free. Contact us, tell us what you want to do, & we can give you some general ideas. We may be able to help you directly, or may refer you to PA Equipment suppliers or other contractors whose area of expertise may be more appropriate to your needs. Working in the industry, we know who supplies what, and where to find them. If you then decide you would like our further help, we will tell you what we can do for you, and what it will cost you. We will not ask you to pay us anything until you have agreed it with us.

Design:-

We can design a **PA system** within any budget constraints you give us, and either directs you to suppliers who can provide the various parts of it, or obtain all of them for you. Any charge for this is negotiable (there is no charge for anything we do until we tell you and you agree to it).

Creation:-

We can build a **PA system** for you, and deliver it if it is a touring system, or install it if it is a fixed installation. The first time you use any touring PA system we have built we can set it up for you, and show you how to use it. You will have to pay for your equipment (and our work), but only what you have agreed to pay at the advice or design stage

Public address system amplifies low level sound to a higher level so that it can be heard by large number of people gathered and at a considerable distance. A basic public address system has input source is a microphone, which pick ups low level sound like a human speech. Examples of some other input sources are a cassette player for playback of recorded music, or compact disc player.

AMPLIFIER:- It is an electronic circuit in which sound low level signal to higher level signal and improves the signal gain. The gain depends upon the number amplifiers are used in casteded stages.

This determines the AF power levels. This amplifier consists of 3 or 4 microphone auxiliary circuits at input level with controlling switches for each microphone. Each microphone has got volume controllers to control audio signal levels. Special electronic circuit provides and excellent sound quality. Cut and Boost type Bass and Treble controls, master control, 4 input volume controls and a special feature of adjusting tape sensitivity through tape Hi/Lo switch have been provided. An LED ARRAY gives a visual indication of the output level. Line output in addition to speaker outputs of 4,8,16 ohm and 100W is provided.

FRONT PANEL CONTROLS OF AN AMPLIFIER

SPECIFICATIONS:-

Player Section:

Track system: 2 Track Monaneal
Tape speed: 4.75 Cm/Sec
FF/Revind Time: 190 seconds with C.60 tape
Tape speed accuracy: within +2% - 1%
Wow and flutter: 0.35% Nominal

Amplifier section:

Output power: 100W (RMS) max
75W RMS at 10% THD
70W RMS at 5% THD (Rated)
Frequency response: 55Hz - 15000 Hz 3dB
S/N Ratio: > 55dB
Input channels: 4 X Mic 0.6 mV (Lo - Z) 4.7 K
1 X Aux 100mV / 470 K
Speaker outputs: 4,8,16 and 100W
Line output: 1000mV / 3.5 K
Tone controls: Bass - 10dB + 7 dB at 100Hz
Treble - 10dB + 7 dB at 10 KHz

Microphone Connection:

4 independent microphone inputs have been provided through jacks on the front panel. Mic 4 inputs may be used by keeping the Mic / Tape - Lo Tape Hi selector switch at MIC position. Microphone inputs are unbalanced Low impedance usable with microphones of impedance 200 - 600. In case microphones with both HIGH and LOW impedance are used, be some that switch on the microphone is at low position. Microphone with upto 30 meter long cable can be used always use good quality shielded cables. For good intelligibility of sound, the microphone should be placed atleast 20 Cms away from the speaker’s mouth. Keeping the microphone too close to the mouth will result in distortion of sound and may even cause damage to the diaphragm of the microphone.

Speaker Connections:

Speaker impedance taps of 4, 8 and 16 have been provided for direct connection of speakers to the amplifier when the distance between the amplifier and speakers is less than 50 meters. Use 23/26 or thicker cable for connections. Be sure that the total impedance of the speakers is equal to or more than the impedance specified on the terminal strip.

Phasing of Loudspeaker/ units:
When two or more speakers / units are installed in the same area and are facing in the same direction, it is essential that their cones / diaphragms move in unison. Otherwise, the sound output of one speaker will cancel the sound output of other. To avoid any mistake, the terminals of all driver units are marked L1 & L2. For correct wiring refer the sketches.

RESULT: - PAS is assembled in college ground observed sound distribution through the ground in public meeting.

EXPERIMENT No- 12

OBJECTIVE: - Study of CD player.

THEORY: - **CD player:** A recorded compact disc (CD) is played in a CD/VCD player, detection of pits and present on the CD is also done by a laser beam of low power are focused on the rotating CD. If the rays reach a flat they will come back after reflection from the flat, but if fail on the PIT then they will get absorbed there and will not come back. By processing to reflected rays circuits of CD/VCD player get information about the combinations of PITS and FLATS. On these digital signals thousands of circuits of CD/ VCD player do various type of processing and finally generate analog audio / video signals. Similar to those signals which were given at the input of the video card. By giving audio signals to the loudspeaker and video signals to video section of a T.V, sound /picture similar original can be produced.

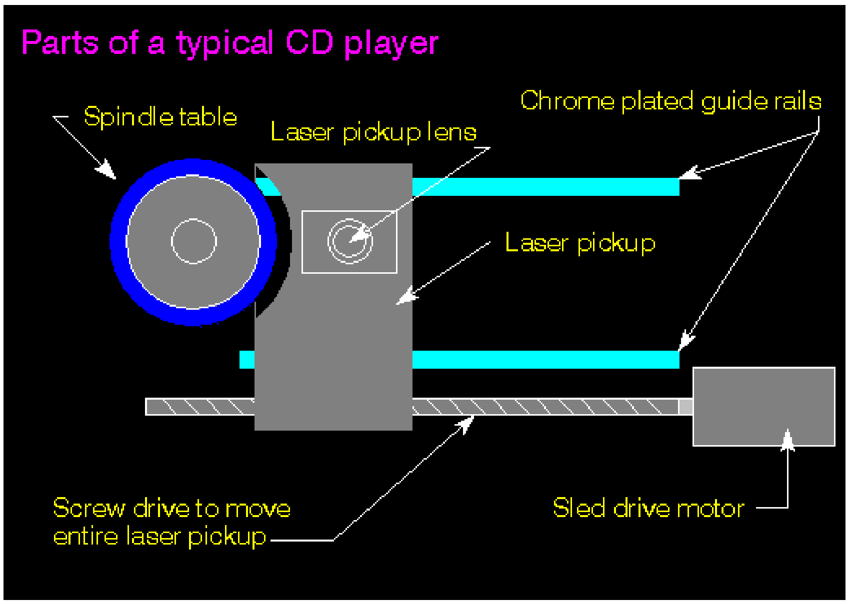
1. CD player speed during recording / play back speed are 2.5 cm/second. A C.D. consists of a outer diameter of 120mm. The aluminum coating present on the CD is 116mm diameter inner diameter of 50mm, a CD consists of a hole with diameter is 15mm. The recording is done on thousands of spiral tracks present on this 33mm wide ring. The CD thickness about 1.2mm. on CD can store 800MB of audio/video programmed. With a duration of 1 hour.
2. Thousands of tracks are present on the 33mm wide ring. A gap is maintained between two nearly tracks on these tracks of a laser beam is focused.
3. A high power laser (Light Amplification by stimulated emission of radiation) used during CD writing. A low power laser beam is used for playback.
4. Laser rays are nothing but a beam of light.
5. CD can be cleaned with soap water.
6. On CD when scratches are formed, play is not possible.
7. A digital signals are two voltage levels ie. 0V, 3V.
8. CD recording is done on the inner most track and the last part of the programmed is recorded on the outer tracks.
9. On a CD in between audio / video signals sync, timing signals are present.

A compact disc, also popularly known simply as a CD, is an optical storage medium with digital data recorded on its surface. A compact disc player is a device that reads the recorded data by means of an optical beam and accurately reproduces the original information (music, pictures, or data). Because the player reads the information by optical means, there is no physical wear and tear on the disc. The basic technology used in all compact disc players is essentially the same, whether the player is designed for audio, video, or computer applications. This article will focus on players designed for audio (specifically, home audio) applications.

Raw Materials: A compact disc player is a very sophisticated piece of electronic equipment. The simple exterior contains complex interior mechanisms to read and process audio signals into very clear and crisp music. The various components include a housing cabinet, an optical pick-up assembly, and printed circuit boards (PCBs), which contain microchips that direct the electronic processes of the system. The cabinet that houses the maze of components is usually made of light, reinforced aluminum. The laser is a small glass tube filled with gas and a small power supply to generate a laser beam, while the photodiode—a semiconducting part that the light that is reflected from the compact disc into an electrical signal—is generally made of silicon or germanium. The lenses and mirrors in the optical pick-up are made of highly polished glass or plastic. This assembly is housed in its own plastic enclosure. The majority of the electronic components—resistors, transistors, and capacitors—are contained on microchips attached to PCBs. The base material of these components is usually silicon. The hardware that connects the various subassemblies together consists of a variety of metal and plastic nuts, screws, washers, pulleys, motors, gears, belts, and cables.

Design: Conceptually, the design of a CD player resembles that of a phonograph (record) player. Like a record, the compact disc is rotated on a turntable, and the audio is read by a pick-up device. However, unlike a record player, the motor does not rotate the turntable at a constant speed but adjusts it in accordance with the distance of the pick-up from the center of the turntable. Furthermore, the pick-up device in a CD player is not a mechanical stylus (a needle) but an optical laser beam that does not come into physical contact with the compact disc. This laser focuses its beam on the disc track that contains the lands and pits, and the CD player's detector (the photodiode) senses the difference between the light reflected from the lands and that reflected by the pits. The photodiode turns this reflected light into an electrical signal. Relayed to the electronic circuit board, this signal is then converted back to sound.

The diagram below shows the physical position of a few parts of a cd player

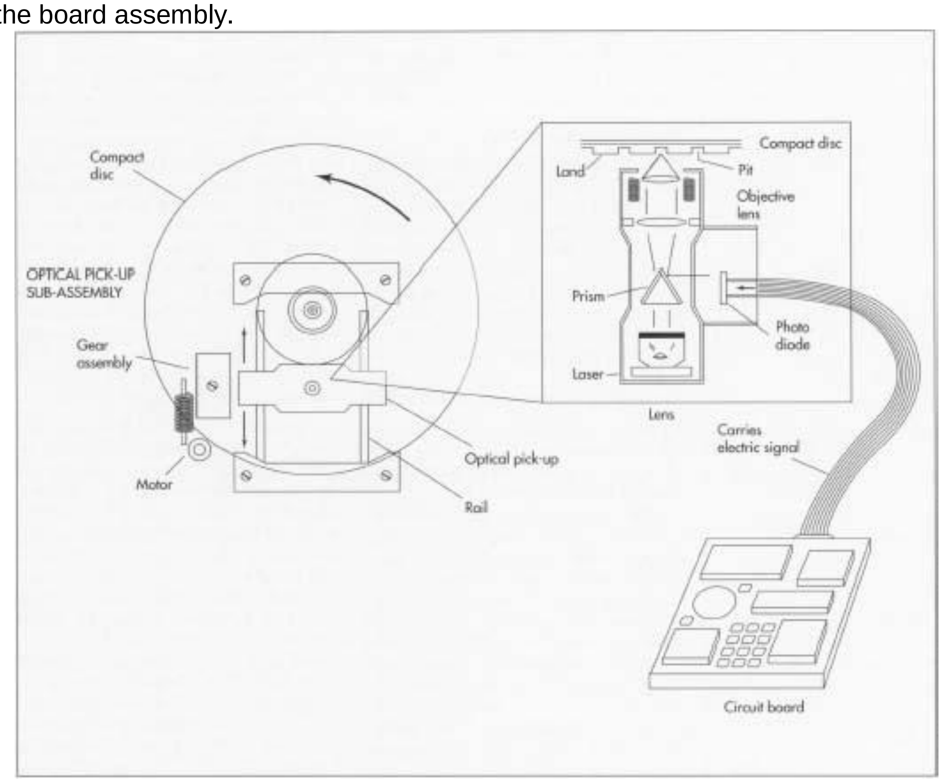


There are basically three subassemblies in a compact disc player: the disc drive mechanism assembly; the optical pick-up assembly; and the electronic circuit board assembly, which coordinates the other systems inside the player and which includes the servo mechanism and data decoding circuitry. By sending signals to the servo mechanism, the circuit board adjusts the motor speed, focusing, and tracking of the optical pick-up; manages the flow of data to the decoding circuitry; and provides display information in response to the various buttons on the control panel.

The disc drive mechanism consists of a spindle that holds the CD and a motor that rotates it. The motor, called the spindle motor, is mounted underneath the plastic disc loading tray or turntable. A separate motor mounted on the chassis (the base or frame of the CD player) moves the loading tray in and out of the player; this is done by means of a gear that is attached to the motor and that also operates a larger gear to raise and lower a clamp for holding the disc in place.

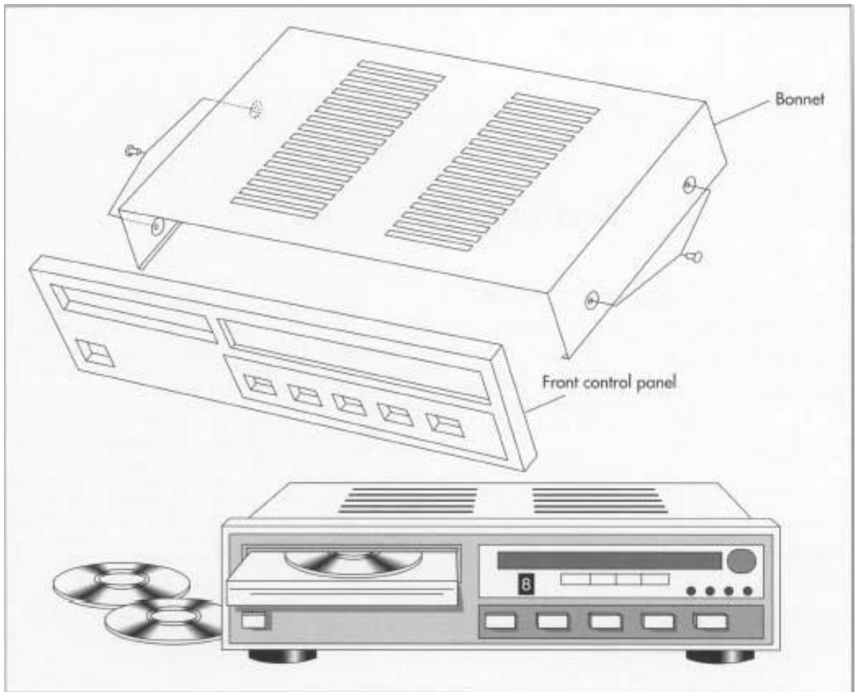
The optical pick-up consists of a laser, a photodiode, and various lenses and mirrors. The entire subassembly slides back and forth on rails and is controlled by the servo mechanism that receives directing signals from the circuit board. The optical pick-up is usually located underneath the clamp that positions the disc, while the motor that moves the assembly is mounted on the chassis close to the rails. The mechanism works by directing a laser beam through lenses and mirrors onto the underside of the compact disc. The lenses and mirrors keep the beam properly focused. If the beam hits a pit on the disc, no light is reflected and the photodiode remains disengaged. If the beam hits a land, light is reflected back through the lenses and mirrors onto the photodiode, which then generates an electrical signal. This signal is transferred to the electronic circuit board assembly, where it is converted by the data decoding system into audio signals for playback.

The electronic circuit board assembly consists of printed circuit boards that contain the circuitry for the servo mechanism, which operates the optical pick-up system, data decoding, and control system. There are many integrated circuits chips, microprocessors, and large scale integrated components on the board assembly.



A key assembly in a compact disc player is the optical pick-up assembly. It is situated on rails so that it can move back and forth underneath the compact disc. It works by directing a laser beam at the CD; if the laser hits a land, the reflected light then travels to the photodiode, which generates an electrical signal. In turn, the signal moves to the CD player's circuit board, which converts the signal into music.

The Future: The CD system technology has come a long way in the last few years, and new applications for compact disc systems are being discovered every day. The market has already seen the introduction of CD-ROMs, CD-Videos and CD-Interactive. The latest product to attract consumer attention is Kodak's Photo-CD, which can display photographs on television and computer screens. These pictures can be edited or cropped by the user, just like clip art images.



The housing for a CD player includes a top cover or "bonnet' and a front control panel. The compact disc rests on a loading tray that slides in and out of the player. These functions will allow the user to display information on the remote control unit itself, such as song titles, artist names, and the actual lyrics of the songs. Compact discs capable of both recording and playback, like a cassette tape, are also in the works. The CD's vast storage capabilities also lend itself to many broad-based multimedia applications and it is quite possible that compact discs will become the common medium of data exchange for all audio, video, and computer applications.

RESULT: - Study of CD player has been completed.

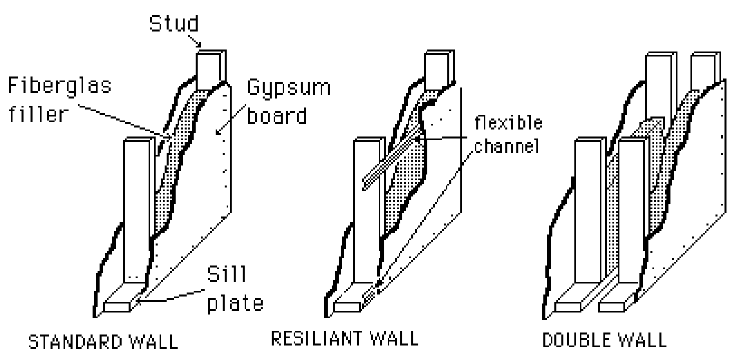
EXPERIMENT No- 13

OBJECTIVE: - Familiarization with studio acoustic.

THEORY: - **ACOUSTIC TREATMENT FOR HOME STUDIOS**:-

Soundproofing: - Commercial recording studios cost hundreds of thousands of dollars to build because they must allow absolutely no sound to enter from a usually noisy urban environment. Double and triple walls, isolated concrete slabs, custom steel doors are all standard but high priced items used in their construction. A studio's sound is its number one asset and most owners will go to any lengths to get it right. The use of microphones is infrequent enough that it can be scheduled for predictably quiet times, and close mic techniques, don't pick up much noise. Given a reasonably quiet, solidly built house to start with, a decent home studio can be created with modest expense and effort.

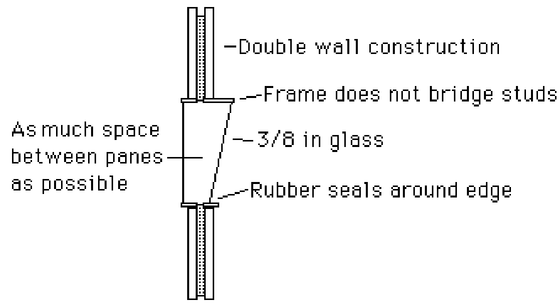
New Construction: - The most effective soundproofing must be designed into a house when it is first built. A typical residential wall is made of a frame of 2x4 wood studs covered with 5/8" thick gypsum board. Properly built (no holes!) this will provide about 35 dB of isolation. Fiberglas filler, R-7 or better, will increase this by 5 to 8 dB and decrease wall resonance. Doubling the thickness of gypsum gives another 3 to 6 dB of overall isolation, but its most important effect is lowering the resonant frequency, hopefully below the audio range. There are two common strategies for reducing coupling between the two sides of the wall. One is to make the gypsum to stud connection springy, either by using metal studs or by hanging the gypboard on resiliant metal bars. The most effective trick is to use separate studs for each face of the wall so there is no direct connection. This eats up a lot of space, but can give a transmission loss of over 60 dB. This is actually better performance than simple cinder block or poured concrete construction!



Construction Details

These same principles can be applied to floors and ceilings. A heavy false ceiling hung on springs can match the performance of a double wall-- If there is a room below the studio, it should get a double ceiling too.

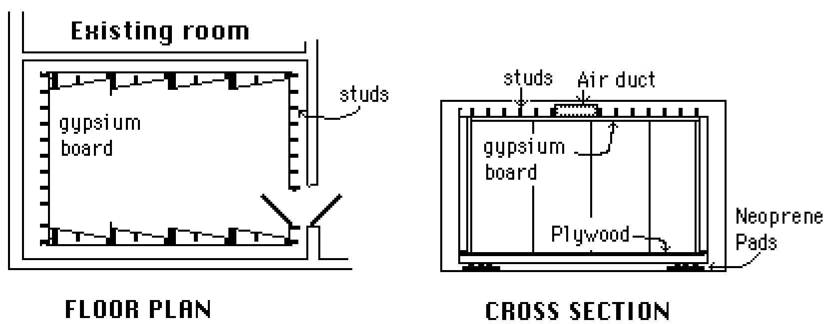
Interior Windows: - The window between control room and studio used to be a traditional feature of a recording facility. The home studio doesn't really need one, because you can get a decent video camera and a large monitor for less than what a good window costs to build. If you want a window, figure 2 shows what has to be done



The traditional window.

The effectiveness of these kinds of construction depends a great deal on the craftsmanship of the builder. There must be no loose studs, and the sill plates must really hug the floor. The gypboard must be well fitted and all potential cracks must be caulked. (Caulk is soft and will not crack when the building settles.) Do not put holes in sound walls for outlets or pipes-- use surface mount electrical fittings and caulk around any wires that pierce the gypboard.

The Ultimate Solution: Double Untruly isolated spaces are created by building a separate room within the room. Both the external room and the internal room have to be tight and heavy and there must be no solid connection between the two, not even the floor. You can buy prefabricated isolation rooms (at a hefty cost), or you can build one using construction techniques similar to that of the house. Something like this should really be designed by an architect to fit your situation, but here is a typical plan to give you the idea.



Plans for a room within a room.

The inner room is built on a platform of 2X4s covered with two layers of 3/4 inch plywood. The platform is supported by neoprene pads that line up with the floor joists. There must be no other connection between the room and the house. The walls and ceiling are built on the platform using 2X4 studs and double gypboard on the inside only. The space between the walls should be at least one inch (wider if practical) and lined with fiberglass. The air duct should be very long and lined with sound absorption material. Get the heaviest solid door and frame you can find, and add gaskets as described above.

These steps can result in a very quiet space, but they get progressively more expensive-- the real question is when is it quiet enough? The easy test is to make a recording of the space. No sound, just a tape of the mic levels at their usual setting with nothing going on. Now turn up the gain and play it back. If you can't hear any difference between the unrecorded and recorded portions of the tape you have reached your goal.

The only way to get an objective measurement of sound levels is to use an SPL meter. (There are some inexpensive models by Gold Line/loft or Radio Shack.) As measured by the "C" scale on these meters you will find the following numbers appropriate for these uses.

- Good restaurant 35-45
- Quiet office 30-40

- Hospital room 25-35
- Church 20-30
- Concert hall 15-25
- Recording studio 10-20

A decent home studio should measure in the 20s. Assuming all noise sources are outside the room, you can calculate the amount of transmission loss the walls have to provide by measuring the sound level with the door open. Close the door and you can figure what you already have. If the level does not change when you close the door, you know where to start!

ROOM TREATMENT:-

Has this ever happened to you? You are playing your latest masterpiece at a party at a friend's place, and when the best song comes on you want to hide under the couch-- the bass is boom, the highs screech, and along with the backup vocals you can definitely hear Gilligan's Island. If you find this experience familiar, you are probably the victim of BAD ACOUSTICS.

You won't be surprised to hear that the shape and furnishings of a room can affect the way things sound-- we have all experienced extreme cases such as large echoed bathrooms and overstuffed restaurants. These effects can easily happen in a subtle way in your studio, causing inaccuracies in the sound from the monitors. When you record or mix you adjust the music till it is right in your control room, but when you play the tape in a neutral environment the sound is overcompensated and strange. There are expensive instruments available to measure the quality of sound in a space, but the best ones are on the sides of your head. You can compare rooms by listening to familiar recordings. (It doesn't have to be on CD-- you can tell a lot from the quality of hiss on a tape.) In a good room, the bass is balanced and clear, cymbals "shine" without being harsh, you can understand words without effort. A mono signal appears to come from a spot exactly between the speakers, and that spot does not jump around with changes of pitch. Now listen to the quiet-- can you hear a refrigerator, a TV, traffic on the street? Clap your hands-- you should hear a slight broadening of the sound, but little reverberation and certainly no pitches or echoes.

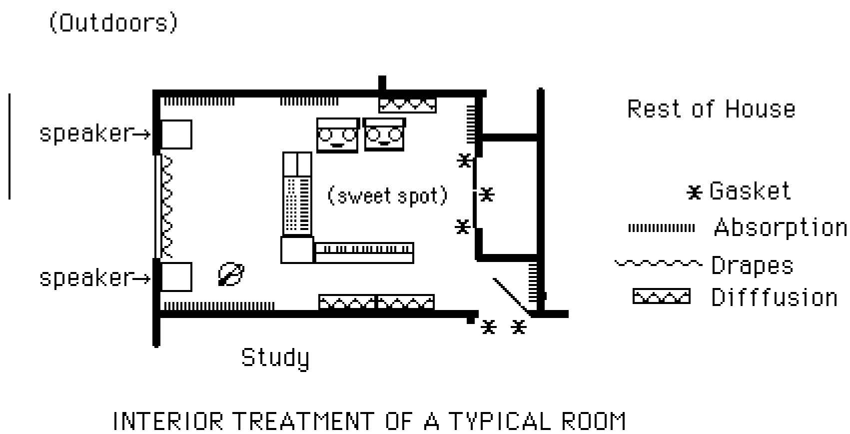
These simple tests should tell you about any severe problems the room may have. Subtle ones will show up in the music produced in the room, as described above. You may be surprised to find that the control of the sound of a room is not really very complicated and can usually be accomplished with inexpensive materials.

The amount of reverberation desired in a room depends on the activity going on. Musicians like fairly long reverberation times; between one and two seconds. This allows them to hear themselves play and enhances the harmonic effects of the music. (In larger rooms even more reverb is desirable because it helps fill the hall with sound.) For listening to speech or music played through loudspeakers this amount of reverb is too much-- values around a second are more comfortable, and for critical listening to speakers the RT60 should be close to a half second. Reverberation time is determined by the volume of the room. It can be reduced by replacing some of the hard, reflective parts of the the walls with soft, absorptive sections. Every material has some absorptive qualities. This is described by its coefficient of absorption, a number between 0 and 1, with 0 being totally reflective and 1 being an open window. For instance the COE of brick is 0.04, whereas that for heavy drapes is around 0.6. The effective absorption of a surface is simply the COE times the area of the surface in square feet. These numbers can be used to compare materials and to predict the results of treatment. The absorption ability of most materials is frequency dependent, which can cause problems as described later.

Interference: - You may be familiar with phase interference from recording work with multiple microphones. If a sound arrives at a single point via two paths at slightly different times, certain frequencies will be reinforced and others will be weakened. You can easily hear this by putting your ear close to a wall: the quality of sound will change because the reflections off the wall interfere with the direct sound. The effect is at its worst when the distance the reflected sound travels is only slightly longer than the direct distance. Phase interference is attacked by careful consideration of the placement of speakers and the listener. In general avoid locating either so that there are short reflective paths off of walls, ceiling, or equipment. The worst problems occur when a speaker winds up in a corner. If this is unavoidable, figure out where the reflections occur, and make that part of the wall or ceiling absorptive.

A SAMPLE DESIGN:-

As an example of how to apply these principles, let us look at an ordinary room in a typical house. (All right, it's my wife's studio in my house!) This room is rectangular, about 11' by 13' with an eight foot ceiling. There is a large closet at the back of the room and a window at the front looking onto a suburban street. The closet helps isolation because it provides something of a double wall between the studio and the living room.



A treated room.

There was a plush carpet over a thick pad on the floor, but no other absorptive material in the room to start with. The clap test in the empty room suggested a moderately long, primarily high frequency reverberation and produced the characteristic "chirp" of a severe standing wave problem.

Some Isolation: - After adding gaskets to the doors, isolation from the rest of the house is adequate as long as recording is limited to quiet times. (We checked this out before we moved in!) Noise from the street is an occasional problem which was helped a little by drapes on the window. An additional drape across the doorway made only a slight improvement in isolation and was really in the way, so we gave it up.

Positioning the Equipment:- After some experimentation, we decided to locate the speakers each side of the window. Since speakers tend to move gypsum as well as air, outside walls are always your first choice if you are concerned with sound control. Incidentally, these are obviously not near field speakers. Near field monitors should not be against a wall, but most large systems depend on a wall backing for extended bass response. The speakers were hung about 6 ft from the floor. This is a bit on the high side, but was necessary to allow the placement of a writing table underneath them.

Wall Treatment: - At this point we were down to two problems: the rising frequency response of the reverberation and the standing wave. We attacked both problems at the same time with some carefully placed absorptive panels. These were made of R-19 fiberglass and measured 2 ft by 6 ft. (They do not need to extend down to the floor because the furniture scatters sound at that level.) Most of this absorption wound up on the walls near the speakers-- this cleaned up the last of the short delay reflections and resulted in a very clear sound image between the speakers. The absorption was brought along the side walls to soak up the standing wave. We wanted to keep the room symmetrical, so we spaced out the absorptive panels, winding up with a pattern where bare wall on one side was opposed by absorption on the other. A large section of absorptive wall near the left speaker created a dead corner for recording vocals. The curtain over the window Is too light to be a really broadband absorber, but it combines with the low frequency absorption of the glass to give a reasonably flat overall effect. The carpet and wooden floor interact in much the same way. We found the sound to be balanced in frequency when the walls were about one third covered with fiberglass. This left the side walls near the back of the room untreated so we added diffusion. This is provided by some homemade diffuser panels on one side and some very cluttered bookshelves on the other.

RESULT:- Familiarization with studio acoustic has been completed.