AUDIO SYSTEM [10 MARKS]

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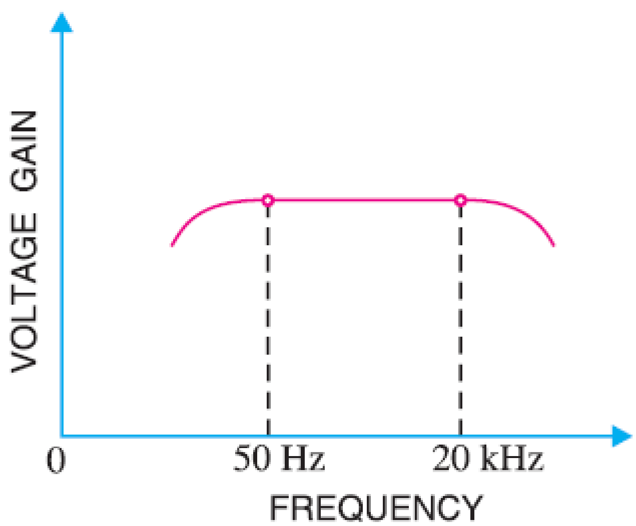
AUDIO AMPLIFIER

An audio power amplifier is an electronic amplifier that increase the strength of low-power audio signals (signals composed primarily of frequencies between 20 - 20 000 Hz, the human range of hearing) to a level suitable for driving loudspeakers. It is the final electronic stage in a typical audio playback chain.

Types of Audio systems :

 Mono Amplifier system
 Stereo Amplifier system



CHARACTERISTICS OF AUDIO AMPLIFIER

- Output power
- Distortion
- Gain
- Frequency response
- Impedance
- Sensitivity
- Signal-to-noise ratio
- Crosstalk

[1] OUTPUT POWER

The **output power of an amplifier** indicates the electrical power it is able to supply to the speakers without distorting the signal or causing damage to the equipment. This magnitude is represented by the symbol P_{out}.

[2] DISTORTION

The distortion measures the level of **unwanted signals** that appear at the output of an amplifier with respect to the input signal. Ideally the amplifier increases the **input signal** so that the output is equal but with higher magnitudes.

However, in practice an amplifier always introduces unwanted signals, so the manufacturer must indicate what **types of distortion** may occur:

Total harmonic distortion (THD), which appears when the amplifier introduces harmonics to the output signal. This value is usually below 0.1% and is indicated for a test frequency or for the entire band.

- **Intermodulation distortion** (IMD), which occurs when there are signals of different frequency at the input of the amplifier and interference occurs at the output caused by active elements such as transistors or diodes.

[3] GAIN

The gain is the **relationship between the output power and the input power** of the signal. It is measured in decibels (dB).

[4] FREQUENCY RESPONSE

Thanks to the frequency response we get to know the relationship between the input and output **signal levels** for the set of frequencies in which the amplifier works. The amplifier quality will be greater when flatter this frequency response within the audible band.

It can be indicated by a graph and also with a figure that indicates the frequency range in which the amplifier works, which is usually in the **range of the audible spectrum** (between 20 and 20,000 Hz).

There is a **linear distortion** that occurs when the frequency of the amplifier is not linear and the gain varies, leading to a distortion of amplitude.

[5]IMPEDANCE

For every amplifier there are an input impedance and an output impedance.

The input impedance must be high, so that there is a correct voltage adaptation between the sound source and the amplifier itself.

Normally, amplifiers have several inputs that offer **different impedance to adapt to different audio sources**, and the higher the better.

For common use in line and auxiliary inputs, the ranges move between 47 Kohms and 200 Kohms.

[6] SENSITIVITY

The **sensitivity of an amplifier** tells us what minimum signal level at the input can generate the **maximum output power** on the load that the manufacturer has specified.

When the level that indicates the sensitivity is exceeded, a **saturation in the output signal** is generated, and this causes the reproduced signal to become too distorted.
The sensitivity is greater the lower the specified value, and will depend in all cases on the type of input (the sensitivity is lower, for example, for mic inputs than for line inputs).

[7] SIGNAL-TO-NOISE RATIO

The signal-to-noise ratio is the difference between the level of the output signal and the noise level, measured in dB. If this parameter is high, the amount of noise introduced by the amplifier will be very low and the **sound quality** will therefore increase.
This magnitude is calculated by the difference between the signal level of the equipment working at its rated power and the **noise level** when there is no input signal. It is usually specified for a frequency of 1 kHz.

[8] CROSSTALK

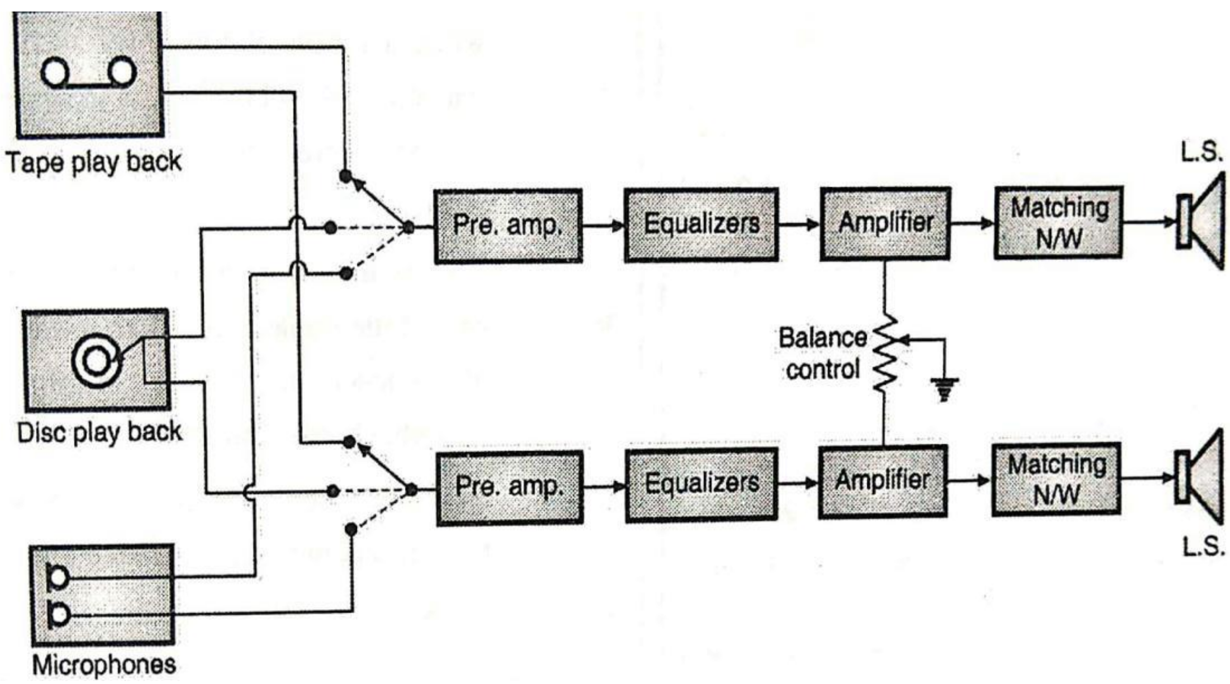
The equipment that amplifies **more than one audio channel** (for example, the stereo ones) are the only ones that can present crosstalk, since in practice the sound channels are not totally independent and the signal of one affects the output of the rest.
The manufacturer sends a test signal through one of the channels while leaving the other without signal, and measures the difference of output levels between both measurements. This figure, in dB, gives the final crosstalk measurement of the amplifier.

DIFFERENCE BETWEEN MONO AND STEREO AMPLIFIER

	MONO AMPLIFIER	**STEREO AMPLIFIER**
1	Monaural or monophonic sound reproduction is intended to be heard as if it were a single channel of sound perceived as coming from one position.	Stereophonic sound or, more commonly, stereo, is a method of sound reproduction that creates an illusion of multi-directional audible perspective.
2	**Cost** : Less expensive for recording and reproduction	**Cost** More expensive for recording and reproduction
3	**Recording** Easy to record, requires only basic equipment	**Recording** :Requires technical knowledge and skill to record, apart from equipment. It's important to know the relative position of the objects and events.
4	Audio signals are routed through a single channel	Audio signals are routed through 2 or more channels to simulate depth/direction perception, like in the real world.
5	Monaural or monophonic sound	Stereophonic sound
6	**Application** : Public address system, radio talk shows, hearing	**Application** : Movies, Television, Music players, FM radio stations

	aid, telephone and mobile communication, some AM radio stations	
7	**Channel : 1**	**Channel : 2**
8	Equalizers are not used	Contains equalizer circuit
9	Nonlinear distortion occurs.	Nonlinear distortion not more than input/output
10	**SIGNAL TO NOIZE RATIO :** Less signal to noise ratio	**SIGNAL TO NOIZE RATIO** Better than 50 dB is the S/N ratio.

BLOCK DIAGRAM OF HI-FI AMPLIFIER.



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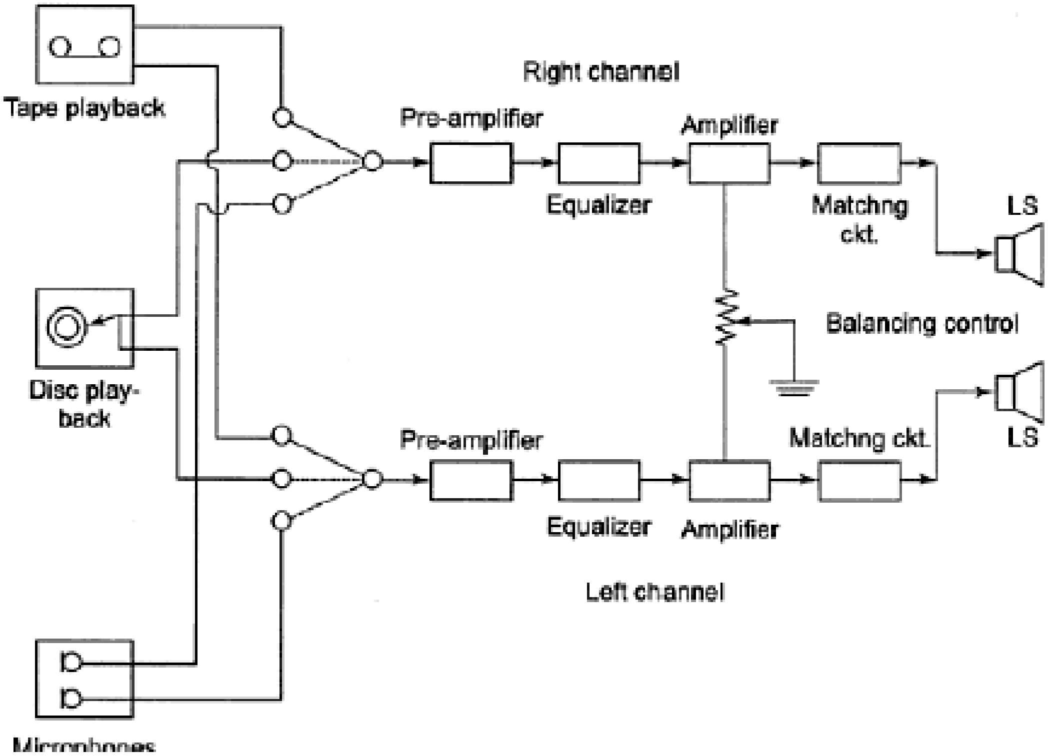


Characteristics of HI-FI amplifier:

1. Signal to noise ratio should be better than 50dB.

2. Frequency response should be flat within +-1dB.

3. Nonlinear distortion should not be more than 1%.

4. The system should possess dynamic range of at least 8dB.

5. Stereophonic effect should be provided.

6. Environmental conditions should be such as to eliminate the external noise in listening room.

Functions:

Balance Control: Two amplifiers of a stereo system, although independent of each other, are built as matched pair to give equal output for the same input. In spite of the two amplifiers being identical, there may be variations in the output of each channel due to variations in the characteristics of transistors & ICs and positioning of loudspeaker & furnishing with respect to the listener. The circuit used is called BALANCE CONTROL.

Master Gain Control: A master gain control is used for adjusting overall volume without disturbing the balance. This is achieved by using dual concentric shafts, the inner shaft adjusts the balance control & the outer shaft, the overall gain or volume of the amplifier.

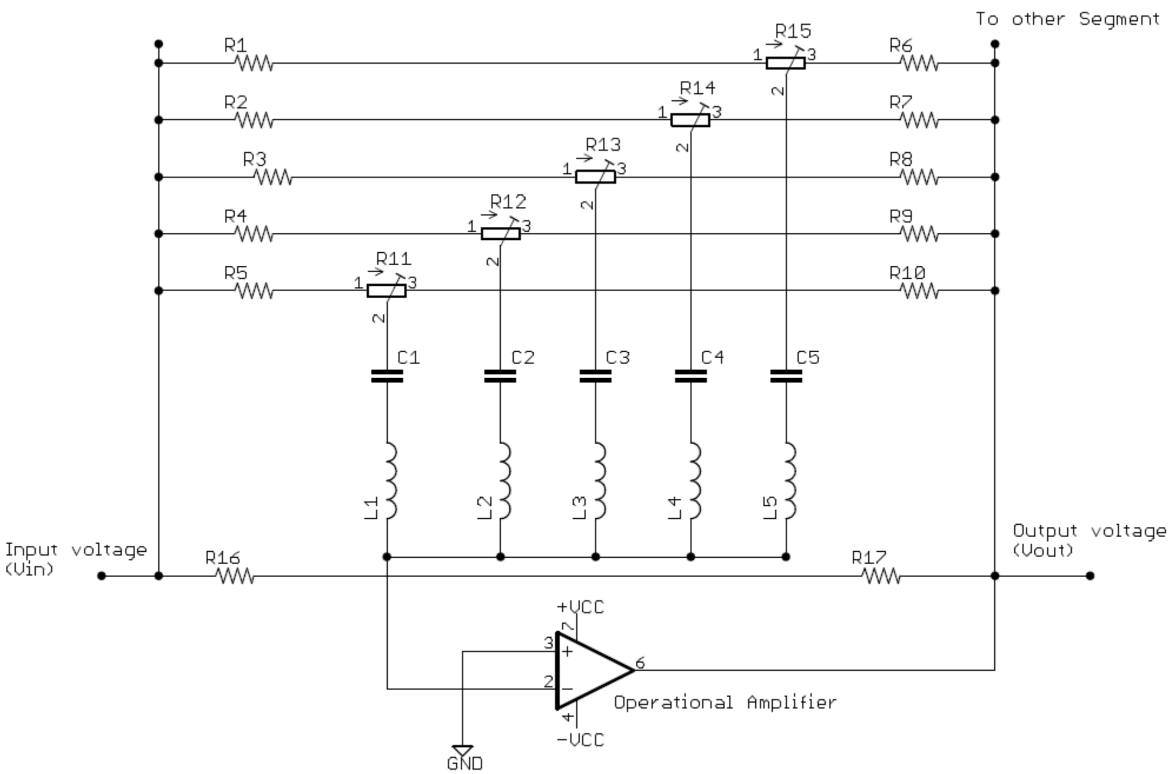
Blend Control: The stereo effect is diluted by this control when there is too much left-right effect. Diluting is done by misbalancing the two channels.

Quasi Stereo Switch: When any one channel signal is made to go into both the channels, one can use both channels & their speakers for monophonic source of signal. This is done by a switch called quasi-stereo switch.

Bass & Treble Control: It is provided to tailor bass & treble as per personal taste of listener.

Loudness Control: Sometimes music is at low level of volume. At low levels there is considerable loss in bass in reproduction. It is, therefore necessary that there should be substantial boosting of bass at low levels. Boosting at treble may be only nominal because loss at high notes is quite small. The control which provides desired boosting at bass & at treble is called Loudness Control.

GRAPHIC EQUALIZER: CIRCUIT DIAGRAM & OPERATION.



[1]Graphic equalizer is used to eliminate unwanted peaks in the frequency response of audio systems.

[2] In five point configuration, the graphic equalizer breaks up an audio input signal into five different bands covering the range of human hearing.

[3]Once this is completed, the signal in each band can be adjusted to provide best sound. The center frequencies f1, f2, f3, f4 and f5 of the frequency bands of the graphic equalizer are usually fixed at pre-set values.

[4]Once these bands are added back together, they are passed through an amplifier which increases the amplitude of the signal to the point where there is enough power that can be heard through an ordinary speaker.

[5] The graphic equalizer consists of an amplifier for every segment of octave band. Such amplifiers are connected in parallel to cover the complete frequency range.

[6] The individual gains of these amplifiers are adjusted such that the required frequency response is obtained. Using five amplifiers for five octaves of frequency may be very expensive.

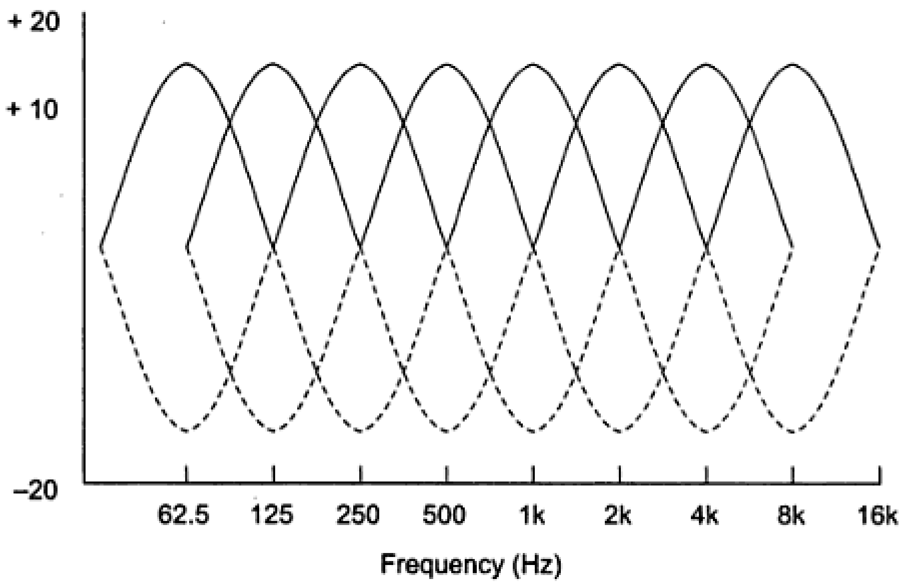


Fig. *Boost/cut for bands at frequencies centered one octave wide*

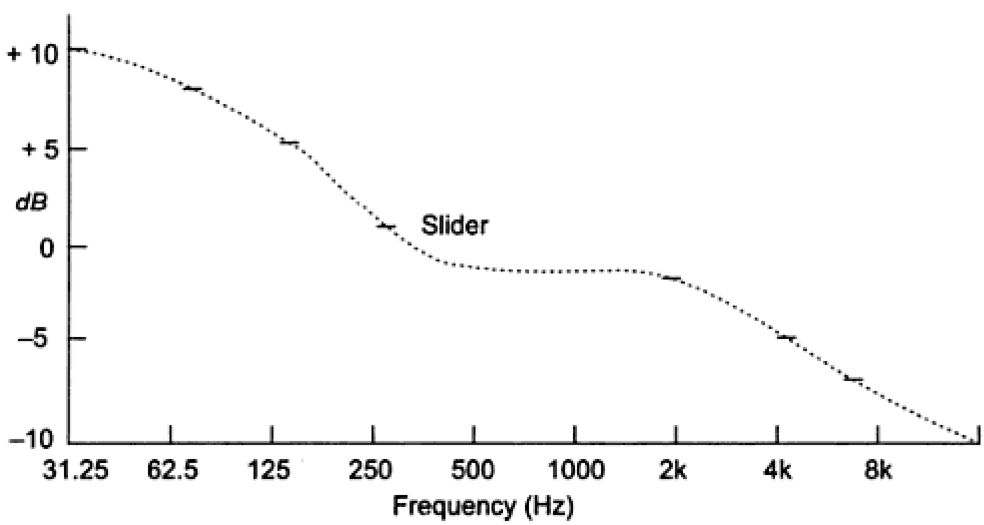


Fig. *Position of sliders of graphic equaliser to get a specified response curve (dotted line)*

DOLBY NR RECORDING SYSTEM.

Dolby NR is a noise reduction system developed by Dolby Laboratories for use in analog magnetic tape recording.

It works by companding i.e. reducing the dynamic range of the sound during recording and expanding it during playback.

It was one of the most important innovations that made fidelity practical on cassette tapes which normally have high noise because of the slow speed and narrow tape format created initially for compact voice recorders, and is common on stereo tape players and recorders to the present day.

Dolby A was the company's first noise reduction system, presented in 1966. It was intended for use in professional recording studios, where it became commonplace, gaining wide spread acceptance at the same time that multi track recording became standard. The input signal is split into frequency bands by four filters with 12 dB per octave slopes, with cutoff frequencies (3 dB down points) as follows:

- Low–pass at 80Hz; (Improvement in SNR with respect to hum & rumble.)
- Band–pass from 80 Hz to 3 kHz; (Deals with mid band noise.)
- A high–pass from 3 kHz; (Improvement in SNR with respect to hiss & modulation noise.)
- High–pass at 9 kHz. (Improvement in SNR with respect to hiss & modulation noise.)
- The output of four separate units is added. All this is done in side branch, and this branch is known as differential network. The output of differential network goes to the main branch as shown in fig. the output of adder is the Dolby processed signal.
- In playback, the differential network separates out the boosted signals in the side branch & subtracts from the input signal as shown in fig.

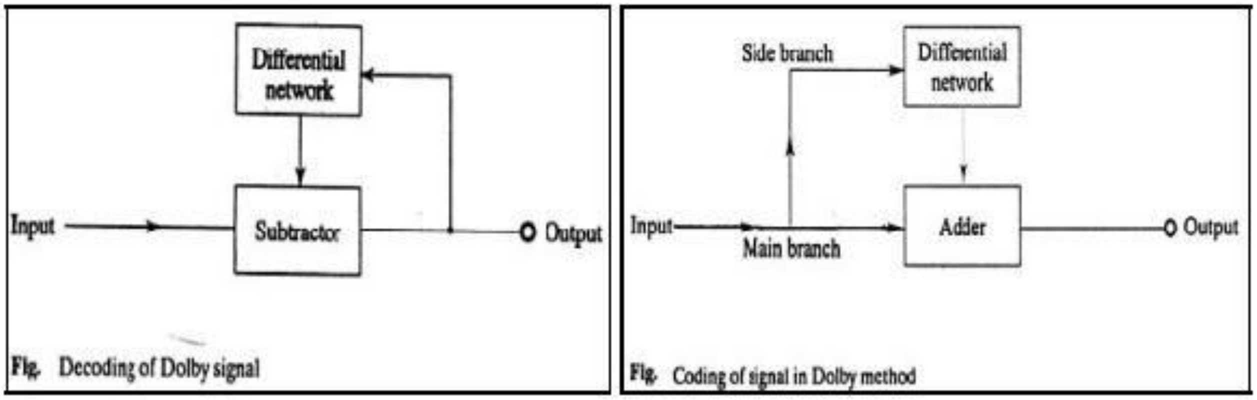


Fig. Decoding of Dolby signal

Fig. Coding of signal in Dolby method

Figure: Dolby A method

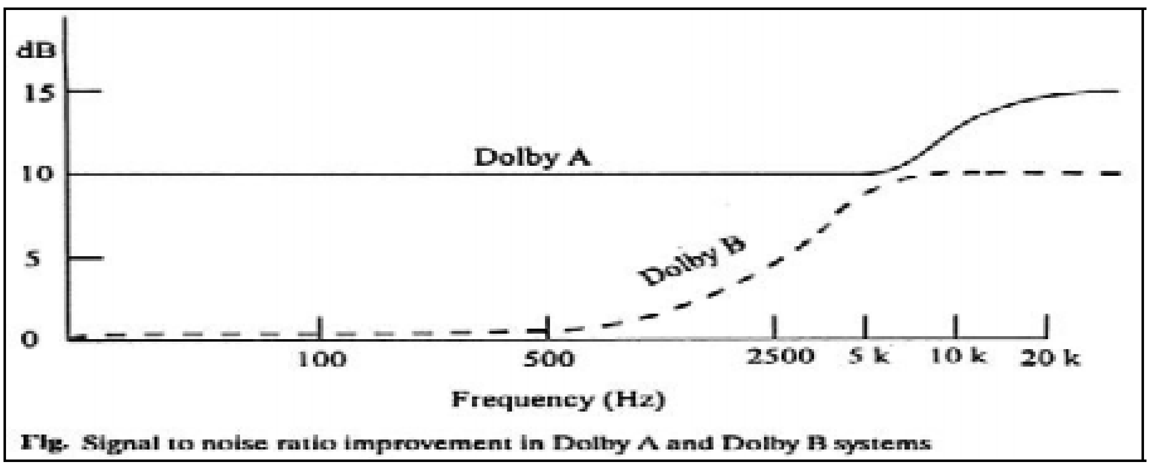


Fig. Signal to noise ratio improvement in Dolby A and Dolby B systems

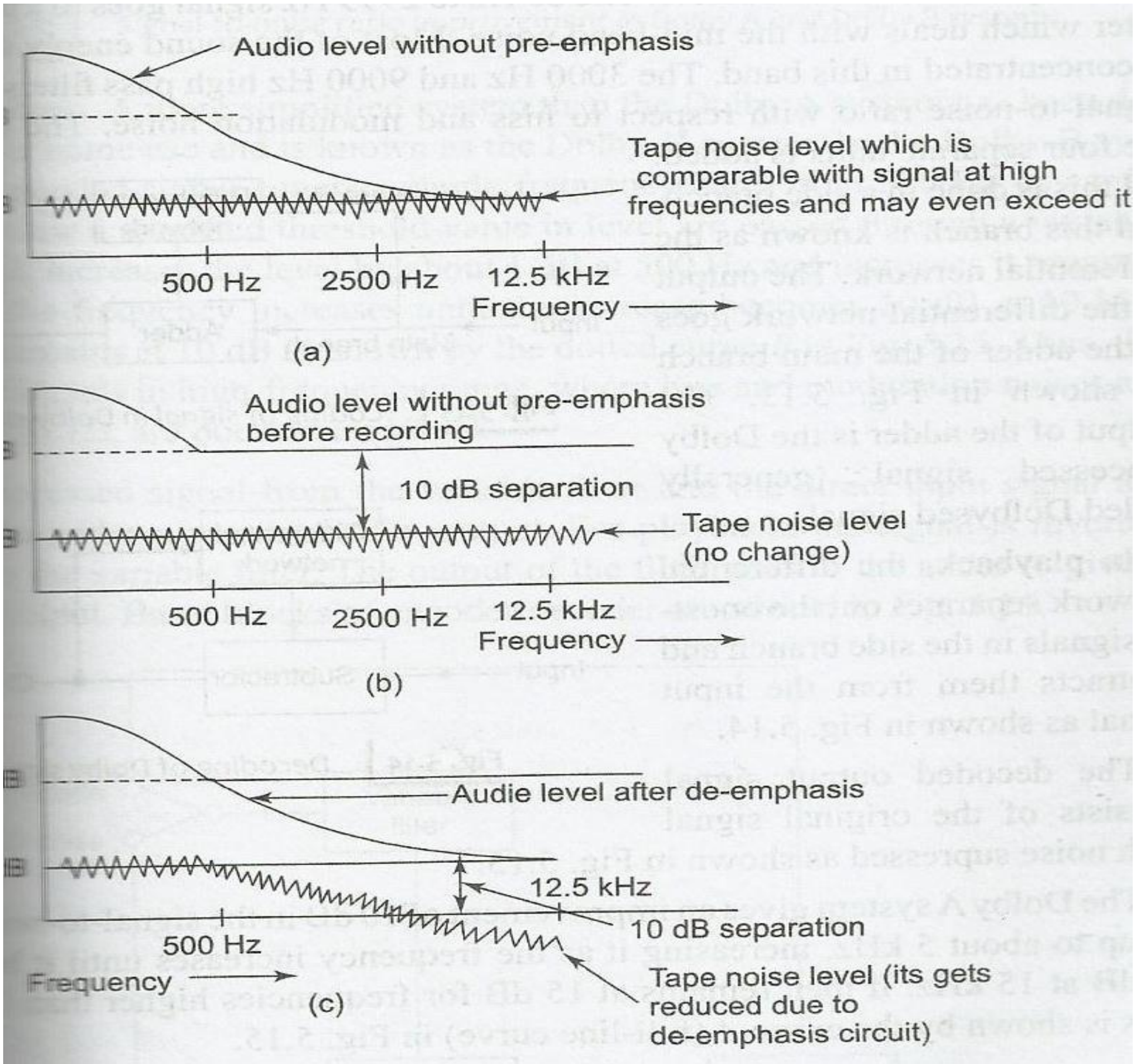


Fig. 5.12 *Reduction of noise by 10 dB in Dolby system*
(a) Position without pre-emphasis
(b) Position after pre-emphasis
(c) Position after de-emphasis

Explanation:
Five point configuration, the graphic equalizer breaks up an audio input signal into five different bands covering the range of human hearing. Once this is completed, the signal in each band can be adjusted to provide best sound.

The center frequencies f1, f2, f3, f4 and f5 of the frequency bands of the graphic equalizer are usually fixed at pre-set values. Once these bands are added back together, they are passed through an amplifier which increases the amplitude of the signal to the point where there is enough power that can be heard through an ordinary speaker.

- The circuit diagram of graphic equalizer is shown in figure. The graphic equalizer consists of an amplifier for every segment of octave band. Such amplifiers are connected in parallel to cover the complete frequency range.

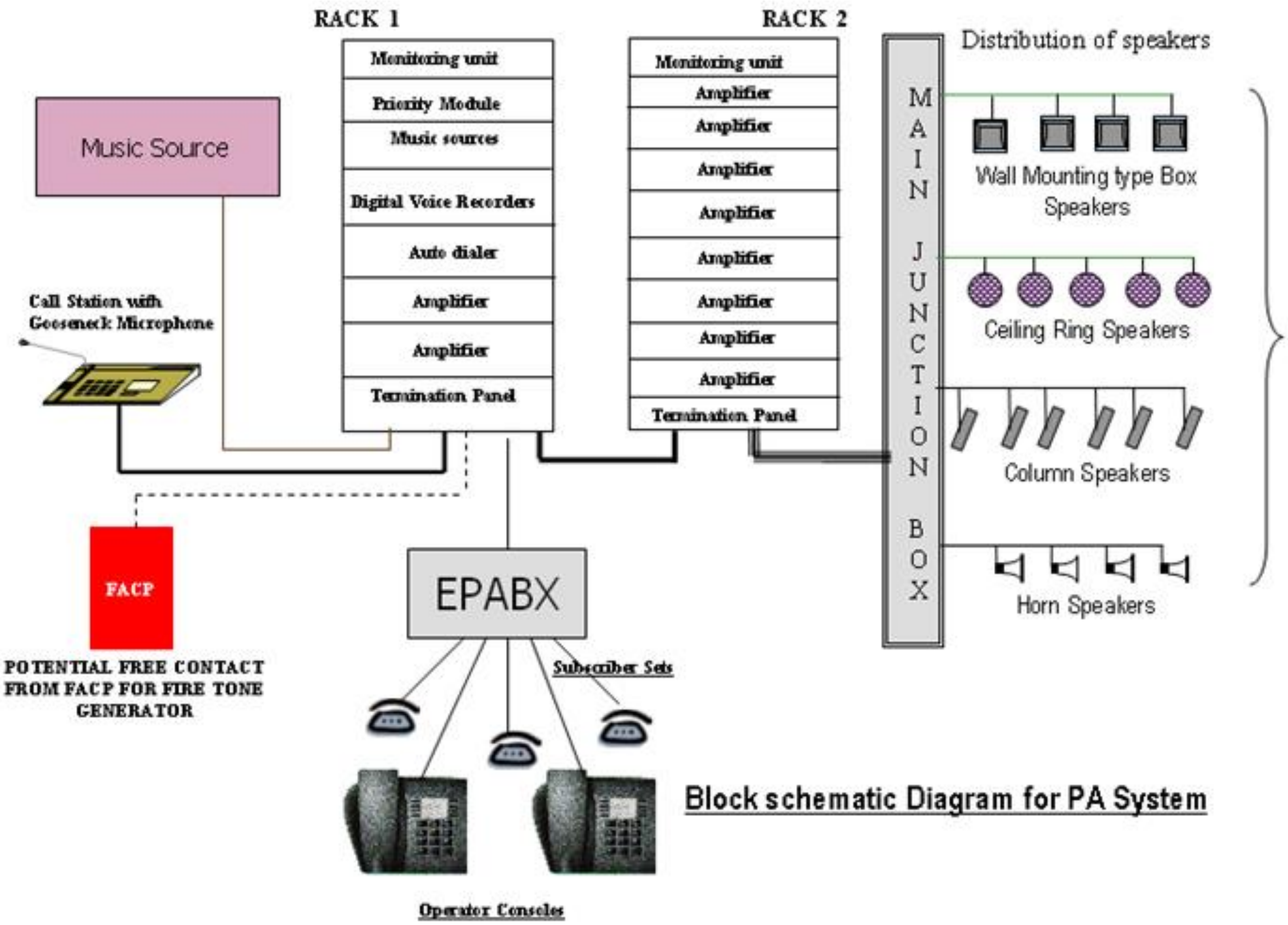
- The individual gains of these amplifiers are adjusted such that the required frequency response is obtained. Using five amplifiers for five octaves of frequency may be very expensive. Hence amplifier shown in figure is normally used.

- The figure shows, there is only one amplifier with multiple feedback paths. There are gain controls and LC tuned circuits in every feedback path. Thus the gain of every octave band is separately adjusted by the corresponding feedback path gain.

The center frequency of the octave band is selected by inductors L1, L2, L3, L4 and L5. The gains of individual octave bands are adjusted by potentiometer controls in the feedback path.

- The combination of individual control setting for various octaves will provide the required frequency response. The peaks at individual octave bands overlap each other. Hence, phasing distortion needs to be avoided. This distortion occurs due to interaction between the overlapping peaks. The slider controls of each octave band can boost or cut the signals from +15 dB to – 15 dB.

PUBLIC ADDRESS SYSTEM



Block schematic Diagram for PA System

Public Address System

When a large gathering of people is to be addressed, the sound must be amplified so that people away from the stage can listen to it comfortably. This type of system is called as Public Address system or P.A. system.

REQUIREMENTS OF PUBLIC ADDRESS SYSTEM:

1. It must avoid the acoustic feedback

2. Distribute the sound intensity uniformly.

3. Reduce reverberations.

4. It must use proper speaker orientation.

5. Select proper microphones and loud speaker.

6. It should create a sense of direction.

7. Loud speaker impedances should be matched properly.

8. Proper grounding should be provided.

9. Use closed ring connection for loud speakers.

APPLICATION
Passenger Amenity

For giving the detailed information about the train arrivals, departures, late running if any, and location of trains and any other important information related to Railway users.

Marshalling Yards

For communication between Yard Master and Shunting men through paging and talk-back system regarding formation and reception or dispatch of trains.

Breakdown train Emergency Equipment

The P.A. System in Accident Relief Train must be kept in working condition for guide the passengers and staff in rescue operations at the site of accident.

Special functions

Local Minister, G.M., etc., officials and VIPs may address some important functions such as Railway Week, felicitations, Scouts and Guides rally, some social work meetings, cultural programmes etc. a quality P.A. System needs to be installed.

Railway Workshops

Providing announcements to workshops staff when required and also for entertainment music during lunch hours.

PA system WORKING

The goal of a PA system is to provide "public address", or a way to transmit audio communication to a group. This transmission can begin with a microphone, which is a device that can assist in magnifying an audio source's volume. A microphone is classified as a device that transforms sound into an electrical signal. Microphones used in PA systems are usually dynamic or condensers. Dynamic microphones are more rugged than condensers and are able to withstand the elements a little better. Condenser microphones utilize their own power source and produce better-quality audio signals, but can sometimes be so sensitive that they also receive background noises.

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Microphones & Mixers

If a microphone or another audio source is used in a PA system, it is plugged into a mixer. The mixer allows for multiple channels of sound to be broadcast at different levels. This can help prevent feedback if a PA system will involve multiple microphones, instruments, or other audio sources. For example, someone may use a PA system to sing along with a pre-recorded CD. The CD track could be plugged into one channel, while the singer's voice would be broadcast through another.

Amplifier

The amplifier is usually side by side or incorporated with the mixer. The amplifier's job is to magnify the audio signal by manipulating its varying frequency qualities. Gain in an amplifier refers to the decibel level of the sound coming out of the speakers. Another quality, Output Dynamic Range, refers to the loud and soft levels of a specific channel's sound. Too low of an output could result in just broadcasting noise; too high of an output can lead to distortion

Loudspeakers

The loudspeaker is the output end of the PA system, transforming the electrical signal back into sound that a group can hear. A good loudspeaker system will separate the differing sound frequencies and broadcast them through different sound channels for better quality output. Parts of a loudspeaker system include a woofer, which broadcasts low frequency sounds; and tweeters, the drivers of high frequency output. Some speakers also contain crossover systems that are responsible for separating these frequencies, reducing the electrical noise that can sometimes accompany a PA system broadcast.

CONTROLS AVAILABLE ON HI-FI AMPLIFIER

- **Balance control**
- **Master Gain Control**
- **Blend Control**
- **Quasi Stereo Switch**
- **Loudness control**
- **Bass and treble control**

Balance Control:

 Two amplifiers of a stereo system, although independent of each other, are built as matched pair to give equal output for the same input. In spite of the two amplifiers being identical, there may be variations in the output of each channel due to variations in the characteristics of transistors & ICs and positioning of loudspeaker & furnishing with respect to the listener. The circuit used is called *BALANCE CONTROL*.

 A simple circuit is shown in fig. The balance control is a potentiometer. When it is set in
the center, the current through LED1 & LED2 should be identical, if the signals in the left & right channels are equal. In that case both LED will be equally bright.

 In case of any inequality, the two brightness level will also become unequal. When balance
control is moved down, the output of the left channel will increase while that of right one will decrease, and vice-versa when moved up.

Master Gain Control:

 A master gain control is used for adjusting overall volume without disturbing the balance. This is achieved by using dual concentric shafts, the inner shaft adjusts the balance control & the outer shaft, the overall gain or volume of the amplifier.

 A typical master gain control circuit is shown above. R1 is adjusted for balancing two channels & then R2 & R3 are adjusted for increasing or decreasing the volume of the channels. R2 & R3 are ganged.

Blend Control:

 The stereo effect is diluted by this control when there is too much left-right effect. Diluting is done by misbalancing the two channels.

 It is shown in fig. above; blend control potentiometer is set at zero resistance for balanced output. For disturbing the balance, this is advanced further to reduce gain of the left channel.

 Although blending can be done by balance control also, but once set, the balance control is not disturbed.

Quasi Stereo Switch:

 When any one channel signal is made to go into both the channels, one can use both channels & their speakers for monophonic source of signal. This is done by a switch called quasi-stereo switch.

Bass & Treble Control:

 It is provided to tailor bass & treble as per personal taste of listener.

Loudness Control:

 Sometimes music is at low level of volume. At low levels there is considerable loss in bass in reproduction. It is, therefore necessary that there should be substantial boosting of bass at low levels. Boosting at treble may be only nominal because loss at high notes is quite small. The control which provides desired boosting at bass & at treble is called LOUDNESS CONTROL.

 It boost audio by +12dB at 50Hz & +3dB at 10 KHz. The loudness control should be used only when sound level is low.