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Audio and Video Systems

Principles, Maintenance and Troubleshooting

Second Edition

About the Author

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Prof. Gupta is a Chartered Engineer (Electronics and Radio Engineering) of the Council of Engineering Institutions, UK and MIERE, MIEEE and FIETE. He took his Master's degree in Science with specialization in Radio Communication. Post M Sc., he underwent two special courses of one semester each in quality evaluation and maintenance of radars and remote sensing electronic equipment. He was sponsored by the Government of India for advanced training in Budapest in quality control in production and maintenance of VHF/UHF equipment and for study of new techniques used in radio control room in London.

He has published 12 papers in various internationally reputed journals. He received the Gowri Memorial Award of IETE for one of these papers and the State Award for another. He has also presented papers in the national seminars held at the University of Roorkee, Engineering University, Jodhpur and IETE. He has authored three books, all published by Tata McGraw Hill, *Audio and Video Systems* (published in 1995), *Electronic Instruments and Systems* (published in 2001) and *Television Engineering and Video Systems* (published in 2006 and the international edition from Singapore was published in 2006).

He has been associated with the IETE since 1970; was Chairman of Lucknow Centre for 1978–79 and Jaipur Centre for 1980–81. He has been chairman and member of several national and state-level committees, pertaining to technical standards, telecommunication planning and technical education. In recognition to his work in these fields, he has been awarded medal for meritorious service by the President of India, State Award for outstanding work by the Chief Minister of UP and a medal for valuable service in Asian Games 1982 (held in India) by the Games Organizing Committee.

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Tata McGraw Hill Education Private Limited

NEW DELHI

McGraw-Hill Offices

New Delhi New York St Louis San Francisco Auckland Bogotá Caracas
Kuala Lumpur Lisbon London Madrid Mexico City Milan Montreal
San Juan Santiago Singapore Sydney Tokyo Toronto

Tata McGraw Hill

Published by Tata McGraw Hill Education Private Limited,
7 West Patel Nagar, New Delhi 110 008

Audio and Video Systems: Principles, Maintenance and Troubleshooting, 2e

ISBN (13): 978-0-07-069976-2
ISBN (10): 0-07-069976-3

Managing Director: *Ajay Shukla*

Head—Higher Education Publishing: *Vibha Mahajan*
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Typeset at Text-o-Graphics, B1/56 Arawali Apartment, Sector 34, Noida 201301 and printed at Pashupati Printers Pvt. Ltd., 1/429/16, Gali No. 1, Friends Colony, Industrial Area, G.T. Road, Shahdara, Delhi 110 095

Cover Printer: Rashtriya Printers

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Contents

Preface

With the widespread use of audio and video equipment in homes, the subject of audio and video engineering was introduced in engineering diploma and degree-level courses by the boards of technical education of various Indian universities in the 1990s. This book is intended to explore the basic principles of audio and video systems and deals with all aspects of troubleshooting and maintenance of various audio and video systems. This book covers the syllabus of Audio and Video Systems, Entertainment Electronics and Consumer Electronics of the diploma and the degree-level courses, and also explains the troubleshooting procedure with the help of flowcharts while giving the corrective maintenance tables for service engineers and technicians.

Aim of this Revision

The first edition of the book, published in 1995, received good response from teachers and students. The feedback indicated that inclusion of some latest topics like VCD, DVD, DTH TV, Cable TV, Pay TV, Solid-State Video Camera, Solid-State Display Screens and Compression Techniques would make the book more useful, as these topics are now part of the revised syllabi of many boards and universities. Hence, the need was felt for a revised edition which would include these topics. The revised edition now covers all these important topics.

Target Audience

The book can be used by undergraduate students of Electronics and Communication Engineering (ECE) and Polytechnic students (ECE). The book will be helpful to the students preparing for diploma and degree examinations as well as to the service engineers and technicians engaged in troubleshooting and repair of widely used consumer equipment.

New to this Edition

Two new chapters on 'Television Fundamentals' and 'DVD' are included in this revised edition for up-to date coverage. The topic on 'Television Standards' is now divided into two separate chapters: Television Fundamentals (Chapter 13) and Television Standards (Chapter 14). Similarly, 'Video Recording and

Reproduction' is also divided into two chapters: Video Recording on Magnetic Tapes (Ch 15) and DVD (Ch 16). Some additional topics like Speech Synthesis and Helmholtz Resonator (Chapter 1), Directivity of Microphones, Precautions in Using Microphones (Chapter 2), Column Speakers, Loudspeaker Cables (Chapter 3), Digital Audio Tape (Chapter 5), and Concept of PMPO (Chapter 7) are also added for enhanced coverage. The latest systems of compact disc for audio recording, CED and Laser Vision for video recordings, Dolby system and special effect tone controls, sound synthesis and, electronic music, HDTV, solid state camera and optical memory disc have also been included to make the students conversant with new and revolutionary techniques in audio and video systems. Summary has been added at the end of each chapter.

Chapter Organization

The book contains 17 chapters. For audio, it covers characteristics of sound, microphones, loudspeakers, sound recorders, amplifiers, noise and distortion, high fidelity, stereophony, PA system planning and concepts of reverberation. For video, it covers TV standards and video recording. The chapters are organized as follows:

- Chapter 1: It covers general characteristics of sound
- Chapter 2: Microphones are presented in detail in this chapter. Discussion on Supercardioid and Hypercardioid Microphones is included in this chapter.
- Chapter 3: It presents a detailed account of working principles and different types of Loudspeakers.
- Chapter 4: The principle of Gramophone Disc Recording and Disc Reproduction is explained in the chapter.
- Chapter 5: This chapter explores the fundamentals of Magnetic Recording and Biasing.
- Chapter 6: It is devoted to the Optical Recording of Sound. The Compact Disc and Playback Process are discussed in detail for better understanding.
- Chapter 7: The chapter deals with Audio Amplifiers. It has detailed coverage on the Amplifier Circuits and Negative Feedback in Amplifiers.
- Chapter 8: The concept of Noise and Distortion is explained in this chapter.
- Chapter 9: Different requirements for Hi Fidelity (Hi-Fi) system and Flat Frequency response is presented in this chapter.
- Chapter 10: This chapter focuses on Stereophony, Stereo Systems and Stereophonic recording.

- Chapter 11: It deals with requirements of Public Address System. The planning to install Public Address System for different situation is discussed in detail.
- Chapter 12: The chapter presents detailed coverage on Acoustic Reverberation, Acoustical features and Design of Auditioriums.
- Chapter 13: The fundamentals of TV is explained in this chapter with the help of block diagrams of TV transmitters and receivers.
- Chapter 14: TV Standards followed in different countries has been explained in this new chapter.
- Chapter 15: This chapter is devoted to Video Recording on Magnetic tape. Functioning of VCR and VCP is explored in easy manner.
- Chapter 16: Latest video disk recording methods like VCD and DVD is discussed in this chapter.
- Chapter 17: It gives extensive coverage of troubleshooting techniques used in modern electronic equipment. Troubleshooting in DVD has been added. Also, maintenance policy, MTBF, MTR and servicing aids have been added. This chapter now covers complete syllabus of the subject of troubleshooting in electronic equipment.

Pedagogy

A large number of solved examples, practice questions, objective-type questions and short-answer-type questions are given in the book for better comprehension of the concepts.

Acknowledgements

I express my heartfelt thanks to the following respected scholarly reviewers who gave valuable suggestions, most of which have been included in this edition.

Akhilesh Verma	*Government Polytechnic Pauri Garhwal, Uttarakhand*
Ashutosh Mishra	*Shankara Institute of Technology Jaipur, Rajasthan*
M Bhoopathi	*Raghavendra Polytechnic Namakkal, Tamil Nadu*
S Thenappan	*MVJ Polytechnic Bangalore, Karnataka*
V B Baru	*Sinhgad College of Engineering Pune, Maharashtra*
Nikhil Gala	*Mukesh Patel School of Technology, Management & Engineering, NMIMS University Mumbai, Maharashtra*

I am also thankful to the entire editorial staff of Tata McGraw Hill Education, who took up the revision of this edition diligently, efficiently and with great devotion.

Feedback

Suggestions from students as well as teachers and practising engineers are most welcome. Please send your feedback to my email id:
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R G GUPTA

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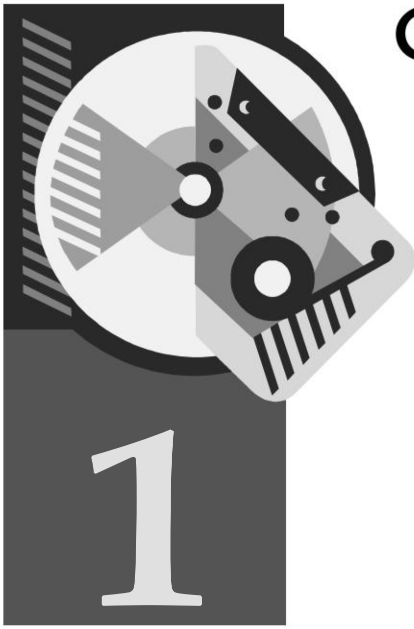


Characteristics of Sound

1.1 NATURE OF SOUND

External to the ear, sound is a longitudinal wave motion consisting of a train of compressions and rarefactions travelling in a medium. When sound waves strike the eardrum, these are converted into electrical signals. The auditory nerves carry these signals to the brain which interprets them into what we call sound.

Being a wave motion, sound has all the characteristics of a wave, i.e., amplitude, frequency, velocity, wavelength and phase. The intensity of a pure tone can be expressed by a sine wave as in Fig. 1.1. Its parameters are defined below.

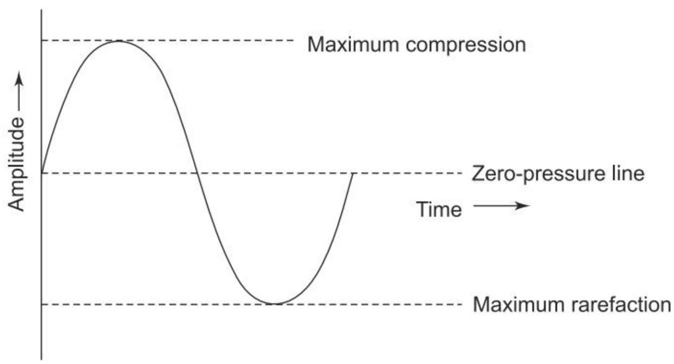


Fig. 1.1 *Sine wave representation of one cycle of pressure variation due to sound*

Amplitude It can be defined as the intensity of compressions and rarefactions produced in the medium.

Frequency It is defined as the number of successive compressions and rarefactions occurring in one second, and is expressed in hertz (or simply Hz). The frequency range of audible sound waves is 16 Hz to 20000 Hz.

Time Period (T) The time taken in completing one cycle is called time period. The frequency of f hertz means that f cycles are completed in 1 second. Therefore, 1 cycle would be completed in $1/f$ second. Hence, the time period T in seconds is given by Eq. 1.1.

$$T = \frac{1}{f} \text{ second} \tag{1.1}$$

Velocity It means the distance travelled in one second. Sound waves move in air with a velocity of 344 metres per second at 20°C (degree celsius) or 332 metres per second at 0°C. Equation 1.2 gives the relationship between velocity and temperature.

$$v_2 = v_1 \left(\frac{T_2}{T_1} \right)^{1/2} \tag{1.2}$$

where, v_1 = velocity at T_1 degree kelvin.
v_2 = velocity at T_2 degree kelvin.

Velocity of sound also depends on medium. While the velocity of sound is 344 m/s in air, it is 5500 m/s in steel, 5200 m/s in glass, 3500 m/s in wood and 1480 m/s in water at room temperature.

The change in velocity per degree change in temperature is 0.6, as calculated in Example 1.1.

Example 1.1 *Prove that the approximate increase in velocity is 0.6 m/s per degree celsius at room temperature.*

Solution

Let the velocity v_1 be 344 m/s at room temperature of 20°C or 293 K. Then, v_2 at (293 + 1) degree kelvin is given by

$$v_2 = v_1 \left(\frac{293+1}{293} \right)^{1/2}$$

$$= 344 \left(1 + \frac{1}{293} \right)^{1/2}$$

$$\cong 344 \left(1 + \frac{1}{2 \times 293} \right)$$

or, $$v_2 = 344 + \frac{344}{586} = 344 + 0.6$$

Thus, increase in velocity is equal to 0.6 m/s

Wavelength The length of space travelled by one cycle of variation is called wavelength and is represented in metres. Equation 1.3 gives the relationship between frequency (f in Hz), wavelength (λ in metres) and velocity (v in metres per second).

$$v = f\lambda \tag{1.3}$$

Derivation of Eq. 1.3 is as follows

$$v = \frac{\text{distance}}{\text{time}} = \frac{\text{distance covered in one cycle}}{\text{time taken by one cycle}} = \frac{\lambda}{T} = f\lambda$$

Calculation of wavelengths at different temperatures is illustrated in Example 1.2.

Example 1.2 *If the velocity of sound at 0 degree celsius is 332 m/s, calculate; (a) velocity at 20°C, and (b) wavelength for sound of 400 Hz at (i) 20°C, and (ii) 0°C.*

Solution

0°C = 273 K, and 20°C = 293 K

Applying Eq. 1.2,

$v_2 = 332\ (293/273)^{1/2}$

$= 332 \times 1.036 = 344$ m/s

Applying Eq. 1.3,

$\lambda = v/f$

At 20°C, $v = 344$ m/s

Therefore, $\lambda = 344/400 = 0.86$ m

$= 86$ cm.

at 0°C $\lambda = 332/400 = 0.83$ m

$= 83$ cm.

Phase This indicates the state of motion at a particular instant relative to some reference. It is expressed in terms of angle, presuming that one complete cycle is equal to a phase difference of 360 degrees. It can also be expressed in fractions of a wavelength λ, or of the time period , T. The phase difference of a quarter cycle can be mentioned as the phase difference equal to $\lambda/4$ or $T/4$.

Phase difference can be between two points of the same wave, or it can be at the same point between two different waves, as illustrated in Figs 1.2 and 1.3.

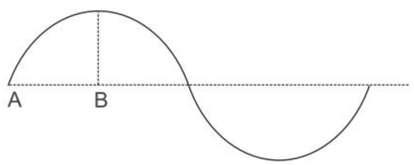


Fig. 1.2 *Phase difference between different points A and B in the same wave*

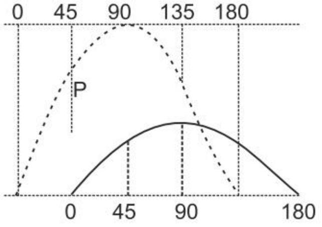


Fig. 1.3 *Phase difference between two waves at the same point, P*

Phase difference at points A and B in Fig. 1.2 in the same wave is 90°, while at the same point P in two waves in Fig. 1.3 is 45°. Phase angles for the dashed curves are at the top, while lower curves are on the bottom line. When the lower curve at P is at 0°, the upper curve at the same point is at 45°. Thus, the upper curve is leading in phase by 45°. If the two waves are opposing each other at every point, they shall have a phase difference of 180°, as shown in Fig. 1.4. The waves having a 180° phase difference cancel each other's effect. For a phase difference of 0° between two waves, as shown in Fig. 1.5, there would be reinforcement of amplitudes.

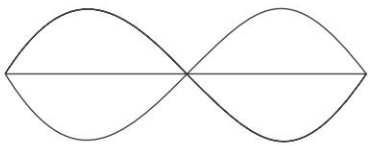


Fig. 1.4 *Phase difference of 180° between two waves*

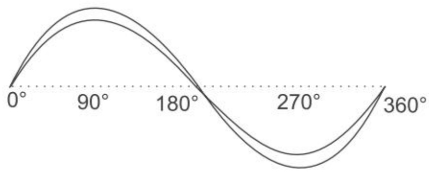


Fig. 1.5 *Phase difference of 0° between two waves*

1.2 PRESSURE AND INTENSITY OF SOUND WAVES

Sound waves produce variations of pressure in the medium. For voice and musical instruments, the medium is air. These variations are compressions and rarefactions in quick succession. Greater the intensity of sound, greater will be the compression and consequent rarefaction. Amplitude of a sound wave is, therefore, represented in units of pressure, i.e., newton per metre square (N/m^2). This unit of pressure is designated as pascal (or simply Pa). Sound pressure variations are also represented in microbar (or dynes, per sq.cm). One Pa is equal to ten microbars.

In terms of energy, the intensity of sound waves is defined as the average rate of flow of sound energy through a cross-sectional area of one square metre at right angles to the direction of motion. Intensity of sound energy is represented in watts per metre square (W/m^2), and is proportional to the square of amplitude. Intensity of sound is generally indicated in decibels (dB) with reference to the threshold of hearing. Decibel is defined as 10 times the logarithm of the ratio of two powers, or 20 times the logarithm of the ratio of two voltages, or currents or pressure values as indicated in equations 1.4 and 1.5.

$$\text{Decibel (dB) for powers, } P_1 \text{ and } P_2 = 10 \log \frac{P_1}{P_2} \qquad (1.4)$$

Decibel (dB) for amplitudes, A_1 and A_2 representing voltages,

$$\text{or currents or pressure amplitudes } = 20 \log \frac{A_1}{A_2} \qquad (1.5)$$

A decibel scale requires a reference for representing sound levels. For sound pressures, the threshold of hearing (given in Table 1.2) has been taken as reference. But for the voltage signals, 0.775 V is taken as the reference voltage because it gives 1 mW power into 600 Ω impedance, which is the resistance of telephone wires of long length. Volume Unit (VU) meters which read average value of the recording level, and the Peak Programme Meters (PPM) which read peak values, use 0.775 V as reference for the dB scale.

dBA Loudness is a subjective effect. Loudness of a sound depends upon how an individual feels it. A measuring device should incorporate a correcting circuit which gives the same frequency response as normal ear has. Such a circuit is used in sound level metres and the output is represented not in dB but in dBA, which means that the logarithmic value has been modified by an acoustic circuit, which matches the subjective assessment of loudness.

Intensity of Some Typical Sounds When sound pressure is 20×10^{-6} Pa, it gives just audible sound and is called 'threshold of hearing'. This much sound pressure pertains to 1 picowatt/m^2 of sound intensity. The sound-pressure level at which pain is felt is 63 Pa. Its intensity is 10 watt/m^2. This intensity is called '*threshold of pain*' or '*threshold of feeling*'.

The sound pressure and intensities between threshold of hearing and pain are given in Table 1.1.

Table 1.1 *Intensity of various types of sound*

Type of sound 1	Pressure in Pa or N/m^2 2	Pressure in microbar 3	Intensity in W/m^2 4	Intensity in dB over threshold of hearing 5
1. Threshold of hearing	20×10^{-6}	200×10^{-6}	10^{-12}	0
2. Rustle of leaves	63×10^{-6}	630×10^{-6}	10^{-11}	10
3. Whisper	20×10^{-5}	200×10^{-5}	10^{-10}	20
4. Average residence	20×10^{-4}	200×10^{-4}	10^{-8}	40
5. Soft note of violin at 3 metres	31×10^{-4}	310×10^{-4}	2.5×10^{-8}	44
6. Ordinary conversation	63×10^{-4}	630×10^{-4}	10^{-7}	50
7. Sound of average automobile	20×10^{-3}	200×10^{-3}	10^{-6}	60
8. Normal speech	0.1	1.0	0.25×10^{-4}	74
9. Heavy traffic	0.2	2.0	10^{-4}	80
10. Thunder	2.0	20	10^{-2}	100
11. Loud rock and roll band in a small hall	20	200	1.0	120
12. Threshold of pain	63	630	10	130

Example 1.3 *The pressure and intensity of sound of an automobile are 20×10^{-3} Pa and 60 dB respectively. Calculate (i) pressure of sound for threshold of hearing, and (ii) pressure of sound for normal conversation level (74 dB).*

Solution

(i) Let P_0 be the pressure for threshold of hearing.

$$\text{Then,} \quad 20 \log 10 \frac{20 \times 10^{-3}}{P_0} = 60$$

$$\text{or,} \quad \frac{20 \times 10^{-3}}{P_0} = 10^3$$

Therefore $P_0 = 20 \times 10^{-3}/10^3$
$= 20 \times 10^{-6}$ Pa

(ii) 60 dB corresponds to 20×10^{-3} Pa

Therefore, 80 dB will correspond to 20×10^{-2} Pa = 0.2 Pa

Reduction in pressure of 6 dB halves the value.

Therefore, 74 dB = 0.2/2 Pa
= 0.1 Pa

1.3 SENSITIVITY OF HUMAN EAR FOR SOUND

The human ear is very sensitive to sound intensity. It can detect sound intensity as low as 0.1 pW/m^2 (or 10 dB below the threshold of hearing). This much intensity of sound moves the ear-drum diaphragm to 10^{-12} metre or one-millionth part of a micron, which is of the order of atomic dimensions.

Further, the ear is sensitive, not to the absolute values of intensity, but to the ratios (or dB). The sound power generated by a large orchestra is a fraction of a microwatt at the softest tones and about a thousand milliwatts at the loudest ones. Similarly, speech during whispering is in picowatts, and while shouting, it is several milliwatts. It is not necessary for a sound-reproducing system to produce sound of the same magnitude of power as at the source, but the reproducing system should be capable of handling the maximum and minimum power in the same ratio. For example, if a source delivers power from 100 μW to 1000 μW, the reproducing system may give power output in the same ratio of 100:1000 (or 1:10), and therefore, the actual output may be 100 μW to 1000 μW, or even 1 watt to 10 watt, or any other values but in the same ratio.

Although the human ear is quite sensitive to sound, it cannot distinguish a difference of intensity of less than 1 dB between two sounds. The minimum level which can be comfortably detected over the threshold of hearing is 3 dB for speech or music. The ear can introduce the sum and difference of two tones, as it is non-linear as in the mixer stage in electronic communication circuits.

The National Physical Laboratory, UK, made a detailed study of loudness levels and their dependence on age. This study revealed important characteristics of the human ear, which are mentioned as follows:

1. The ear is most sensitive from 3 kHz to 4 kHz for all ages.
2. Sensitivity of the ear decreases with age for high frequencies. The ear has good sensitivity from 500 Hz to 10000 Hz for youngsters, but from 500 Hz to 5000 Hz only for old people. Children's ears are sensitive for up to 20000 Hz.
3. Sensitivity of the ear for all ages decreases as frequencies decrease below 500 Hz (for low and medium volumes of sound, the decrease is spectacular).

Some other important characteristics of the ear–brain combination are the following:

1. Ears can judge the direction of the source of sound because of the phase difference between sounds reaching two ears simultaneously.
2. Ears can distinguish between sounds with a frequency difference as low as 3 Hz. Variation of pitch of 0.5 Hz in the same sound can also be felt by the ears (as in wow type of distortion in recording and reproducing systems).
3. Intensity difference of as low as 1 decibel can be distinguished by the ear for steady sound, and 3 dB for music.
4. The louder sound suppresses the weaker sound. It is called ***masking effect***.
5. The ears judge the direction of sound from the sound which reaches the ear first, even if it is weaker. This is called ***Haas effect***. It is valid up to a delay of 40 ms. Beyond that, it appears as an echo. It is also called the ***precedance effect***.
6. While the human ear can detect the relative phase changes between two different sounds, it is insensitive to phase changes in the same sound.
7. The ear–brain system discriminates against sounds which it regards as unimportant. The brain hears what it thinks it ought to hear. It means that in a live programme, the background noise may not be much perceptible and hence is acceptable. One can focus on the desired sound despite background noise and conversations. This effect is called ***cocktail party effect***. Microphones in a stereo system cannot provide this effect as they will receive both wanted and unwanted sound equally well. Also, in the playback of the recorded programme, the background noise will be perceptible.
8. The brain associates sound with the visible movements. The sound of an aeroplane from an amplifier will appear as coming from a height due to this effect.

1.4 FREQUENCY OF SOUND WAVES

The audible frequencies are 16 Hz to 20000 Hz. Two factors that determine the frequency range required for satisfactory transmission of speech are

1. Intelligibility or articulation
2. Energy

It has been established that low frequencies contain most of the energy, while high frequencies give better articulation. Based on this aspect, the frequency of 300–3400 Hz is considered quite adequate for telephone speech. This range contains about 80% of the total energy and gives about 80% articulation. However, speech transmission for normal entertainment requires a bandwidth of 80 to 5000 Hz and for high fidelity entertainment programme, the frequency range should be atleast 40–15000 Hz.

1.5 OVERTONES AND TIMBRE

Sound waves produced by speech and musical instruments are not pure sine waves but, they are complex waves consisting not only of the fundamental frequencies (tones) but also of their harmonics, and other frequencies. Frequencies other than the fundamental are called ‘overtones’. The proportion of tones and overtones in a sound form the special characteristic by which a particular sound can be recognised. When we hear the sound of a relative or a friend, we automatically recognise him/her even if the person is not visible to us. This quality of sound is called ‘timbre’ and is related to the proportion in which overtones are present in the sound.

Human beings have fundamental speech frequencies between 110 to 1500 Hz. Women’s speech contains higher frequencies from 220 to 1500 Hz, while men’s contain from 110 to 1000 Hz. The overall frequency range including overtones is 110 to 8000 Hz for men, and 220 to 10000 Hz for women. Female voices are low in intensity because of lack of low frequency contents below 220 Hz, but is better in articulation (clarity). Range of frequencies for a whistle is 1000–20000 Hz, for harmonium, it is 150–16000, for flute, 170–15000, and Piano 30–16000 Hz.

1.6 INTERVALS, OCTAVES AND HARMONICS

Interval is the ratio of two frequencies. For example, the interval between 400 Hz and 100 Hz is 4, and that between 400 Hz and 500 Hz is 0.8.

An interval of 1:2 is called an ‘Octave’. One octave of 100 Hz will be 200 Hz, or we can say that the frequency range of 100 to 200 Hz is one octave wide. Two octaves of 100 Hz will be 400 Hz. Mathematically, an octave for two frequencies f_1 and f_2 is defined by Eq. 1.6.

$$\text{Number of octaves } = \log_2\left(\frac{f_2}{f_1}\right) \tag{1.6}$$

Harmonic is an integer ratio between two frequencies. There is a clear distinction between octaves and harmonics. For example, with respect to 100 Hz, a frequency of 200 Hz will be a 2nd harmonic, but in terms of octaves, it will be one octave wide. Similarly, a frequency of 400 Hz will be a 4th harmonic, but it will be two octaves wide. The harmonics are always integer multiples of a fundamental frequency, but octaves can be in fractions also. For example, 140 Hz relative to 100 Hz will be 0.5 octave (because $\log_2 140/100 = 0.5$).

The term overtone’ describes all frequencies higher than the fundamental, including harmonics. The ear judges intervals (i.e., frequency ratio) and not the actual difference in frequency. A change from 500 to 1500 Hz will be recognised to be the same as a change from 2000 to 6000 Hz, although the difference in the first case is 1000 Hz only, while in the second case it is 4000 Hz.

Example 1.6 *Calculate interval, octaves and harmonics for a frequency range of 62.5 Hz to 1 kHz.*

Solution

Interval = 62.5: 1000 = 1: 16

Octaves = $\log_2 16$

$= \log_2 2^4$

$= 4$

1 kHz is a 16^{th} harmonic of 62.5 Hz.

Example 1.7 *Calculate interval and harmonic frequency for frequencies of 50 Hz and 125 Hz.*

Solution

Interval = 50: 125

= 2: 5

Harmonic: 125 Hz is not a harmonic of 50 Hz because a harmonic has to be an integer multiple of the fundamental frequency.

1.7 PITCH

Pitch is a characteristic of sound mainly related to frequency. When the sound consists of a pure tone (no harmonics), pitch is determined by frequency alone. But in speech and music, the pitch of sound depends not only on frequency but on intensity as well. The standard pitch is a sound of 440 Hz.

Relative phases of component tones of a complex wave do not affect the ear's perception of quality or timbre. Hence it can be said that the human ear is insensitive to phase changes.

1.8 RESONANCE EFFECT IN SOUND SYSTEMS

All sound transducers (which convert sound into electricity like microphones and cartridges, or which convert electricity into mechanical vibrations like loudspeakers and recording units) have mechanical parts capable of vibrating. All these systems have a natural resonant frequency depending on the mass and compliance of the vibrating system. Similarly, the absorbing and reflecting surfaces can have a natural resonant frequency.

The resonant frequency of such systems is given by Eq. 1.7.

$$f = \left(\frac{1}{2}\right)\left(\frac{S}{M}\right)^{\frac{1}{2}} \tag{1.7}$$

where, f = resonant frequency in Hz

S = stiffness = 1/compliance

M = mass

The concept of stiffness and mass in a mechanical system is illustrated in Fig. 1.6. As compared to an electrical resonant circuit, mass (M) is equivalent

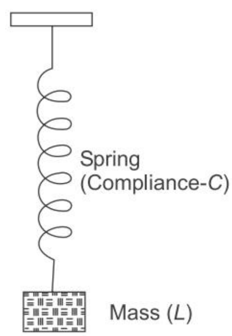


Fig. 1.6 *Components of mechanical resonant system*

to inductance, and stiffness (S) is equivalent to 1/capacitance. The reverse of stiffness is called 'compliance' which will be equivalent to capacitance. Thus, resonant frequency is directly proportional to the square root of stiffness and is inversely proportional to the square root of mass.

1.9 HELMHOLTZ RESONATOR

It is an enclosure with a narrow port called neck, shown in Fig. 1.7.

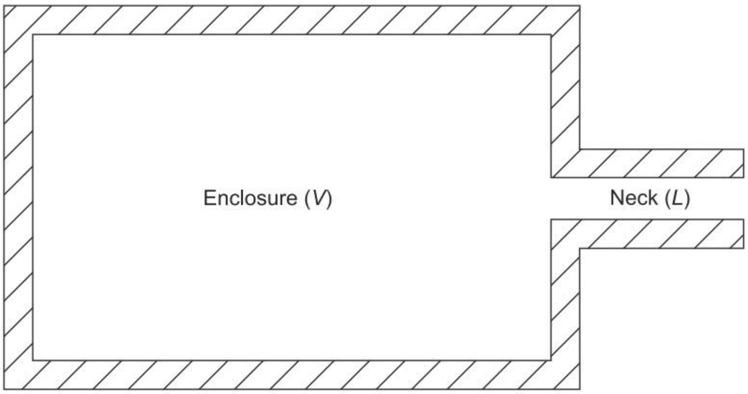


Fig. 1.7 *Helmholtz resonator*

The mass m of air in the neck works like a weight attached to a spring. The volume V of the air inside the enclosure works as a spring (compliance). Mass of the air in the neck will move up and down like a weight attached to a spring. The resonant frequency of this to-and-fro motion is given by Eq. 1.8.

$$f = 54\sqrt{A/(LV)} \text{ Hz} \tag{1.8}$$

where, f = frequency in hertz
A = area of cross-section of the neck
L = length of the neck
V = volume of the enclosure

Low resonant frequencies can be obtained with Helmholtz resonators of relatively small size. This has importance in many acoustical designs, like bass-reflex loudspeakers. If there is some absorbing material in the neck, the device becomes a sound absorber at its resonant frequency.

1.10 REFLECTION AND DIFFRACTION OF SOUND WAVES

Like light waves, sound is also reflected and diffracted. Reflection causes reverberation. Diffraction causes bending around obstacles. As the wavelength of sound is much larger than that of light, bending is quite obvious around small obstacles like casing of a microphone or a loudspeaker. Higher audio frequencies have smaller wavelengths and therefore suffer less diffraction, resulting in shadows. But lower audio frequencies, being of larger wavelength, do not cause shadows and pass easily around obstacles. This results in frequency discrimination in musical instruments, microphones and loudspeakers. Pressure-type microphones (discussed in Chapter 2), have omnidirectional pattern at audio frequencies due to the effect of diffraction. Also, this gets important consideration in the design of baffles for loudspeakers and the shape of musical instruments. The human head works as an obstacle of about 20 cm width between the two ears. So the sound coming from the side of the left ear will have to diffract around the head to reach the right ear, and vice versa. Again, this diffraction will be more at lower audio frequencies than at higher ones.

SUMMARY

- ***Sound***. is a longitudinal wave motion consisting of a sequence of compressions and rarefactions travelling with a velocity in a medium of transmission. Its velocity is 332 m/s at 0°C and frequency range is 16 Hz to 20 kHz. Intensity of sound varies from 20 dB over 1 pW for whisper to 120 dB for loud rock and roll band. Intensity of 130 dB is called threshold of pain.
- **Bandwidth** of sound waves are
  - *(1) For telephone speech* 300 to 3400 Hz
  - *(2) For normal programmes* 80 to 5000 Hz
  - *(3) For high fidelity* 40 to 15000 Hz
- It has been established that high frequencies contain low energy but high intelligibility, and low frequencies contain high energy but low intelligibility. The frequencies of women's voices are higher than those of men.
- **The human ear** converts sound pressure variations into electrical signals which are fed to the brain through auditory nerves. The brain interprets the signals into what we hear. The ears's response to sound is logarithmic.
- **Interval** is the ratio of 2 frequencies. An interval of 1: 2 is called one octave.
- **Harmonics** are integer multiples of a fundamental frequency. The sound of all musical instruments consists of fundamental and its harmonics.
- Frequencies higher than the fundamental frequencies are called **overtones**. Thus, overtones include harmonics also. Composition of fundamental and harmonics in a sound

give individuality to the sound, and this quality is known as timbre.

- **Pitch** is a characteristic of sound mainly related to frequency. In pure tones without harmonics, it represents frequency alone, but with harmonics it is related to intensity also in addition to frequency.
- **Resonance effect** Mass and compliance of the vibrating systems give rise to a resonant frequency whose value depends on the mass and compliance of the vibrating parts.
- **Helmholtz resonator** It consists of a large enclosure and a narrow neck. Air in the enclosure works as a spring and that in the neck as load. This acts as a good absorber of sound.

Review Questions

1. Define amplitude, frequency, phase and wavelength for a sound wave. Derive the relationship between velocity, frequency and wavelength?
2. Explain how intelligibility and energy contained in a sound wave are related to the frequency. Explain that a bandwidth of less than 4 kHz is sufficient for telephone. Write down bandwidths for telephone speech, normal programmes and high-fldelity programmes.
3. Explain the terms

   1. Interval 2. Octave 3. Harmonics 4. Overtones
4. Explain the terms:
   1. Timbre
   2. Pitch
   3. Threshold of hearing
   4. Threshold of pain
5. Write short notes on the following.

   (a) Haas effect

   (b) Cocktail party effect
6. Write down the characteristics of sound as perceived by brain.
7. Write a short note on Helmholtz resonator.

Short-Answer Questions

1. Why is the human ear known as very sensitive to sound intensity?
2. How does the ear decide the direction of the source of sound?

3. What do you understand by the term threshold of hearing?
4. How do ears behave with changes in phase in sound?
5. What are the special effects that are produced by brain, not by ear in sound systems?
6. Why is a bandwidth of 300–3400 Hz considered sufficient for telephones?
7. How are people able to recognize the voice of a friend or a relative, though frequency range of sound is common (16 Hz to 20000 Hz) for all.
8. Why is the female voice of low intensity but higher clarity as compared to male voice?
9. What is the difference between interval and harmonic in sound?
10. What role is played by Helmholtz resonator in auditoriums?
11. Both phenomena, reverberation and echo, are caused by reflection; what makes them different?

Multiple-Choice Questions

1. What kind of wave motion is sound?
   (a) Longitudinal
   (b) Transverse
   (c) Electromagnetic
   (d) Pulse type
2. What is the velocity of sound at 0°C?
   (a) 344 m/s (b) 332 m/s
   (c) 0 (d) 3×10^8 m/s
3. What is the intensity of sound for threshold of hearing ?
   (a) 0 (b) 1 pW/m^2
   (c) 1 mW/m^2 (d) 1 W/m^2
4. What is the intensity of sound for threshold of pain?
   (a) 1 mW/m^2 (b) 1 W/m^2
   (c) 10 W/m^2 (d) 100 W/m^2
5. As compared to the energy in low audio frequencies, energy in high audio frequencies is ___
   (a) more (b) less
   (c) equal (d) uncertain
6. Sound of 60 dB intensity at 40 Hz gives the same loudness as sound of 0 dB at 1000 Hz. Then, loudness in phons at 40 Hz is
   (a) 0 (b) 40
   (c) 60 (d) 100
7. Sone is a unit of
   (a) increase in loudness
   (b) quality of sound
   (c) pitch
   (d) timbre
8. Frequency range for telephone speech is
   (a) 16 Hz to 20 kHz
   (b) 100 Hz to 1000 Hz
   (c) 1500 Hz to 2500 Hz
   (d) 300 Hz to 3400 Hz
9. How many octaves wide is one musical scale?
   (a) 0 (b) 1
   (c) 2 (d) 3
10. With respect to 1000 Hz, a two-octave wide note is

(a) 1000 Hz (b) 2000 Hz
(c) 3000 Hz (d) 4000 Hz

11. With respect to 1000 Hz, the 2nd harmonic is ___.
(a) 500 Hz (b) 1000 Hz
(c) 2000 Hz (d) 4000 Hz

12. Electrical equivalence of mechanical phenomena of mass is
(a) resistance (b) capacitance
(c) inductance (d) emf

Numerical Problems

1. If velocity sound is 332m/s at 0°C, find its value for 273°C.
2. Find the wavelength of sound of 1000 Hz presuming its velocity to be 344 m/s.
3. Intensity of sound pressure of 20 $\times 10^{-6}$ Pa is 0 dB. What will be its intensity for pressure of 0.1 Pa?
4. In Q 3 calculate sound pressure for thunder whose intensity is 100 dB.
5. What will be the loudness level in phon at 100 Hz for a sound of 80 dB intensity which gives the same loudness as a sound of 40 dB at 1000 Hz?
6. What is the value of loudness in sone for Q 5?
7. Calculate sound interval in respect to 100 Hz for the following sound frequencies:
   150 Hz, 200 Hz, 300 Hz, 800 Hz, 10000 Hz
8. Calculate octave value in Q 7.
9. Calculate harmonics in Q 7.
10. In a musical scale two notes are of 16 and 18 Hz., respectively. Calculate interval between two notes.

Answers

Short-Answer Questions

1. It can detect sound intensity as low as 0.1 picowatt per meter square which requires the ear drum to move 10^{-12} metre or one-millionth of a micron, which is of the order of atomic dimensions.
2. The ear judges the direction of sound from the first sound it receives even if that sound is quite weak.
3. The threshold of hearing is the lowest intensity of sound which is audible to human system. Its value is 1 picowatt per meter square.
4. Ears are insensitive to phase changes in the same sound. Hence ears are immune to phase noise. However, ears can detect relative phase changes between two different sounds.
5. The special sound effects produced by brain are masking, hass, cocktail party effect and precedence.

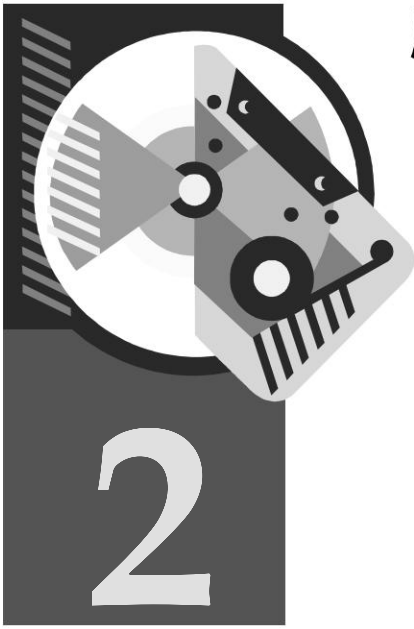
6. In a telephone conversation, the emphasis is that the voice being heard should be comprehensible. It need not be of high quality as required in an entertainment programme. The range 300–3400 Hz contains adequate energy for volume and sufficient articulation for clarity. The low bandwidth allows a large number of conversations on a single carrier channel.
7. A particular sound is recognized by the overtones and harmonics contained in a voice along with the fundamental frequency. Content of overtones and harmonics is the characteristic of an individual.
8. Female voice consists of higher frequency than male voice. The higher frequencies contain less energy but higher articulation.
9. Both are ratio of two frequencies, but, an 'interval' can take even the fractional value of ratio while harmonic will take only the integer value. For example, two frequencies of 100 Hz and 150 Hz have an interval of 1:1.5 but they are not related harmonically.
10. Helmholtz resonator is resonant at low audio frequency. Hence, it can boost low frequency, and if its neck is provided with some absorbing material, it will absorb low frequencies. Thus, it can be suitably used in an auditorium.
11. When the difference between a direct wave and the reflective wave is more than 40 ms, they will appear as distinct sounds, which is called echo. If the difference is less than 40 ms, both sounds will blend with each other and will appear as reverberation.

Multiple-Choice Questions

1. (a)	2. (b)	3. (b)
4. (c)	5. (b)	6. (a)
7. (a)	8. (d)	9. (b)
10. (d)	11. (c)	12. (c)

Numerical Questions

1. (465 m/s) 2. (34.4 cm)
3. (74 dB) 4. (2Pa)
5. (40 phon) 6. 1 sone
7. (1.5, 2, 3, 8, 100)
8. ($\log_2 1.5$, 1, $\log_2 3$, 3, $\log_2 100$)
9. (Nil, 2nd, 3rd, 8th, 100th)
10 (8:9)



Microphones

2.1 INTRODUCTION

Microphone is a transducer which converts sound pressure variations into electrical signals of the same frequency and phase and of amplitudes in the same proportion as in pressure variations. The electrical signals in the audible range are called audio signals. (Microphone, in short, is sometimes written as mic or mike).

A microphone is the first link in sound recording and transmission systems. Audio signals can be used to cross the barrier of time (by recording) and the barrier of distance (by radio transmission)

2.2 CHARACTERISTICS OF A MICROPHONE

The quality of a microphone is determined by the following characteristics:

- Sensitivity
- Signal-to-noise ratio
- Frequency response
- Distortion
- Directivity
- Output impedance

These characteristics are defined as under:

Sensitivity It is defined as output in millivolts (or in dB below 1 volt) for the sound pressure of 1 Pa (or 10 microbars) at 1000 Hz. As the normal level of speech provides a sound pressure of 1 microbar ((or 0.1 Pa), the sensitivity based on this criteria for 1 microbar pressure (or 0.1 Pa) level would be one-tenth the value for 1 Pa pressure.

(Some manufacturers quote the sensitivity in terms of dBm, i.e., power output in dB below 1 milliwatt.)

Example 2.1 *Sensitivity of a microphone is 120 dB below 1V. Calculate its output.*

Solution

120 dB below 1V means –120 dB with respect to 1V. Let E_0 be the output of the microphone in volts, then

$$20 \log \frac{E_0}{1} = -120$$

or, $\log E_0 = -6$

or $E_0 = 10^{-6} = 1\ \mu V$

Signal-to-noise Ratio Some noise (called self-noise or thermal noise) is generated inside the microphone due to resistance of the circuit, built-in transformer, etc. It is represented in terms of the *sound pressure level* (SPL) that would give the same output as the noise output. The output is measured by passing it through a weighting filter which accounts for the reduced sensitivity of the ear at high and low audio frequencies. The acoustically weighted output is represented in dBA. Instead of quoting the noise alone, manufacturers quote signal-to-noise ratio. It is defined to be the ratio in dB of the output (with SPL of 1 Pa) to the output in the absence of sound.

Example 2.2 *The output of a carbon microphone is 100 mV with a sound pressure of 1 Pa. In the absence of sound, the output reduces to 10 mV. Calculate signal-to-noise ratio.*

Solution

$$\frac{S}{N} = 20 \log \frac{\text{output in the presence of sound}}{\text{output in the absence of sound}}$$

$$20 \log \frac{100}{10} = 20\text{dB}$$

Frequency Response The frequency response of a microphone is defined by the bandwidth of audio frequencies in the output of the microphone within ±1 dB of the output at 1000 Hz. Although the complete audible frequency range of sound waves is 16 to 20000 Hz, a microphone which gives flat response within ± 1 dB for frequencies of 40 to 15000 Hz is considered good for high fidelity audio systems. Lower bandwidth of 80 to 5000 Hz is acceptable for radio broadcast. For telephones, bandwidth of 300 to 3400 Hz is used.

The mass of the vibrating system and its compliance are equivalent to electrical inductance and capacitance, respectively. Mass causes attenuation at high and compliance at low audio frequencies. Also, due to movable parts, there is a natural resonant frequency for a microphone, and the signal is boosted at the resonant frequency.

Frequency responses of various microphones are given in Fig. 2.1.

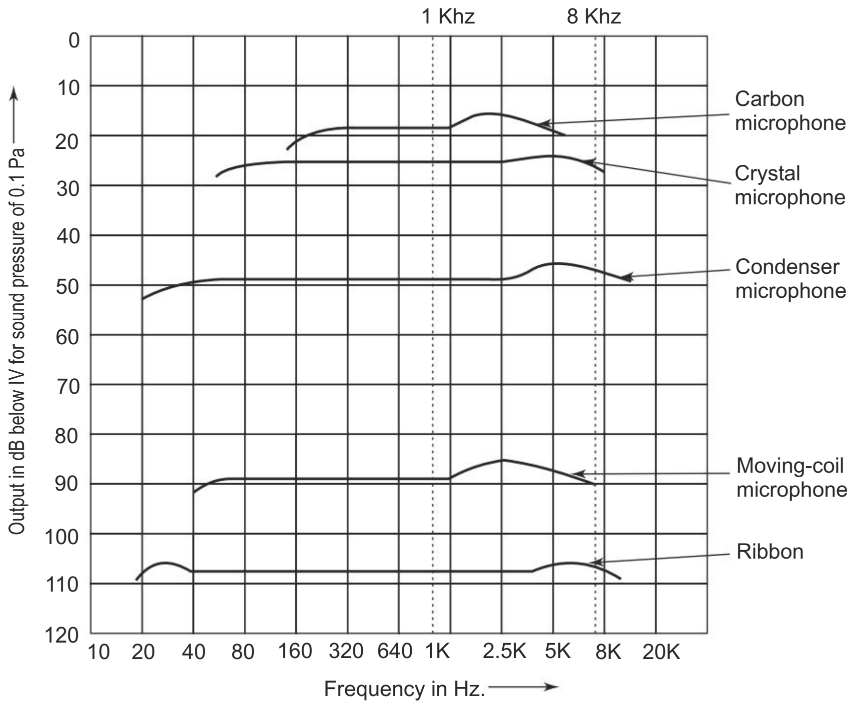


Fig. 2.1 *Typical frequency response of different microphones*

Distortion Besides frequency distortion (uneven frequency response) described above, there are two types of distortions in microphones, namely, non-linear distortion, and phase distortion.

Non-linear Distortion This distorts the amplitude of the audio signal, which results in production of such harmonics in the output that are not present in the input sound. For quality microphones, such distortion should be less than 5%. For high-fidelity sound systems, distortion should not be more than 1%.

Phase Distortion This may cause change of phase relationship between different components of a complex sound wave. Phase distortion occurs when multiple microphones are used causing relative path difference from the source of sound.

Directivity The directivity of a microphone is defined with the help of a polar diagram. The angle for half-power points in a polar diagram represents directivity of a microphone, as shown in Fig. 2.2. Maximum power is in the axial direction of the microphone towards source of sound. When the microphone's axis deflects away from the source of sound, power output is reduced.

Mathematically, directivity is defined as the ratio of actual output when placed in a direction of maximum response to the output which an omnidirectional

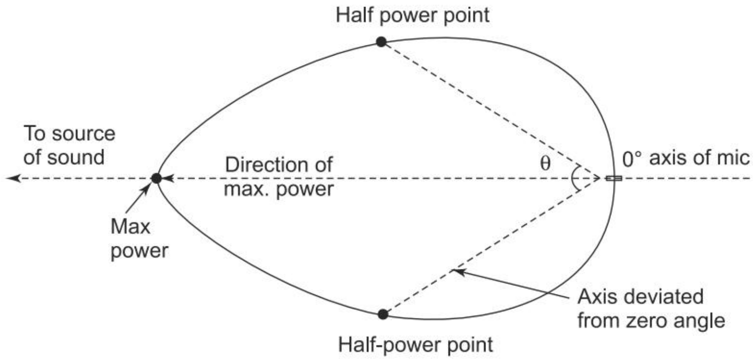


Fig. 2.2 *Directivity in terms of angle between half-power points*

microphone in the same direction would have given, keeping the intensity of sound constant. Directivity, D, is given by Eq. 2.1.

$$D = \frac{E}{E_o} \tag{2.1}$$

where, E = actual output in the direction of maximum output

E_o = Output in that direction had the microphone been omnidirectional

When represented in dB, directivity would be 20 log D.

Example 2.3 *The output of a directive microphone is 0.3 mV, while the output of an omnidirectional microphone, placed under identical condition with respect to source is only 3 μV. Calculate the directivity of two microphone in dB.*

Solution

0.3 mV = 300 μV

Directivity of the directive microphone

$$= 20 \log \frac{E}{E_o}$$

$$= 20 \log \frac{300}{3}$$

= 40 dB

Directivity of an omnidirectional microphone will always be 0 dB

(as it is equal to 20 log $\frac{E_o}{E_o}$ = 20 log 1 = 0)

Microphones with the following directivities are used in practice:

- Omnidirectional
- Bidirectional or figure-of-eight
- Cardioid or heart-shaped
- Supercazrdioid
- Hyphercardioid

In a microphone, sound pressure variations make a light diaphragm to move to and fro either by pressure inflicted on it, or by causing pressure gradient on its two opposite surfaces. All pressure types of microphones give omnidirectional pattern, as shown in Fig. 2.3(a). The pressure gradient types result in bidirectional pattern. This pattern is illustrated in Fig. 2.3(b). Zero degree represents axis of the microphone pointing sharply towards the source of sound. In Fig. 2.3(b) directivity is represented by Figure-of-eight, while circles have been drawn for showing directive gain at other angles. Moving coil, crystal, capacitor and carbon microphones (treated in detail in Sections 2.4, 2.6, 2.7 and 2.8) are all pressure-type microphones. Ribbon microphone (treated in Section 2.5) is of pressure gradient type.

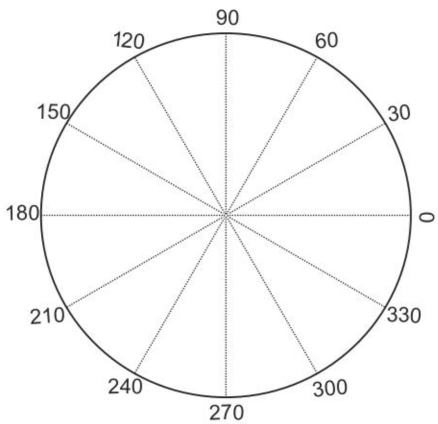


Fig. 2.3(a) *Omnidirectional pattern*

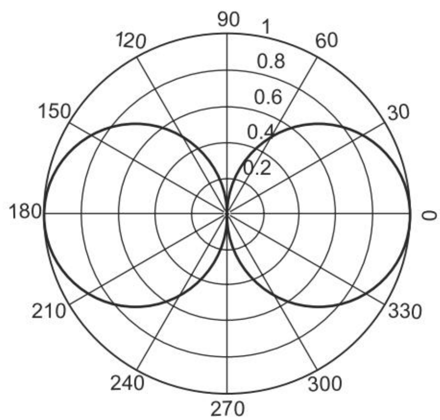


Fig. 2.3(b) *Bidirectional pattern*

Cardioid patterns are obtained by combining an omnidirectional microphone with a bidirectional microphone in series, mounted in the same housing. When the two elements contribute equally (50:50), the resultant pattern is a cardioid (heart-shaped) pattern as shown in Fig. 2.4

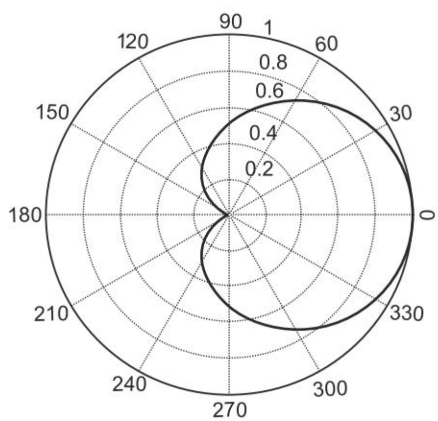


Fig. 2.4 *Cardioid pattern*

The cardioid or heart-shaped pattern so obtained is suitable for orchestras and court scenes in dramas where a large number of persons are present. Due to the practically weightless ribbon, it gives the best result with ensembles and orchestras, as the light ribbon can respond instantaneously to the minute change in the complicated waveform of such music. In this type of microphone, there is null in the rear with maximum in the front and adequate amplitude on the sides. Directive gain is maximum at 0°, but at 90° it is not zero, but is a little more than 0.5.

When the ratio of contribution of the omnidirectional and bidirectional microphones is not 50–50, other cardioid patterns result. These are

1. Supercardioid
2. Hypercardioid

A *Supercardioid pattern* is the result when contributions of omnidirectional and bidirectional elements are 37 and 63 per cent, respectively. This pattern is illustrated in Fig. 2.5.

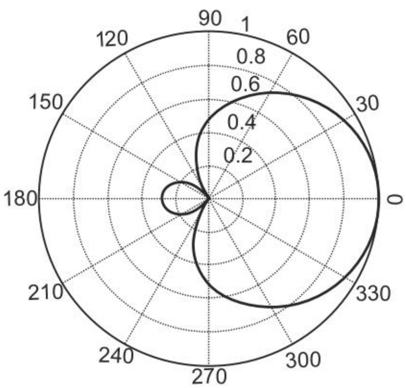


Fig. 2.5 *Supercardioid pattern*

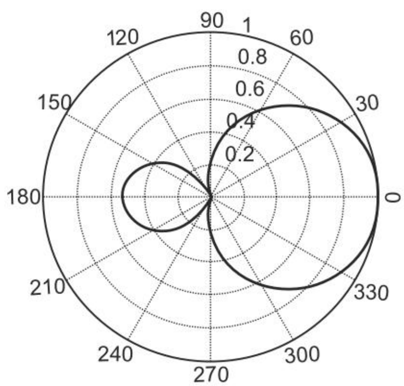


Fig. 2.6 *Hypercardioid pattern*

In a supercardioid pattern, the front lobe is narrower than the cardioid, and also there is a narrow lobe in the rear. There are two nulls on the sides. This pattern is useful in handling court scenes in dramas.

A hypercardioid pattern is obtained when the contributions of the omnidirectional and bidirectional elements are 25% and 75% respectively. The Hypercardioid pattern is illustrated in Fig. 2.6.

In a hypercardioid pattern, the front lobe is narrower and the rear lobe is broader as compared to the supercardioid type. There are two nulls on the sides.

A hypercardioid pattern is capable of providing a more natural open and less focused sound. It is easier to pick up multiple people with one stand. It picks up even if it is not pointing precisely. It is very suitable for recording group discussions and conferences.

Polar diagrams, illustrating various types of directivity patterns as in Fig. 2.3 through 2.5 are governed by the following mathematical expressions, taking R as the amplitude, varying with angle θ from the axis of the microphone.

Omnidirectional: $R = \cos 0^{\circ}$ (constant in all directions)

Bidirectional: $R = \cos \theta$ (zero output at 90°off-axis)

Cardioid: $R = 0.5 + 0.5 \cos \theta$ (Heart-shaped)

Supercardioid: $R = 0.37 + 0.63 \cos \theta$

Hypercardioid: $R = 0.25 + 0.75 \cos \theta$

Some people treat (erroneously) supercardioid and hypercardioid microphones as one and the same type. This is not so, as is vident from different mathematical expressions for the two. In the hypercardioid, the pressure component is 25% and the pressure-gradient component is 75%, while in supercardioid these are 37% and 63%, respectively. This difference results in hypercardioid pattern becoming narrower in the front and its rear lobe becoming broader, as compared to supercardioid. Table 2.1 compares characteristics of different patterns.

Table 2.1 *Comparison of different types of directivities*

PARAMETER	OMNIDIRECTIONAL	BIDIREC-TIONAL	CARDIOID	SUPERCAR-DIOID	HYPER CARDIOID
Maximum rejection angle from axis	No rejection in any direction	90º and 270º	180º	125º and 235°	110º and 250 º
Rejection at 90º	Nil	maximum	6 dB	9 dB	12 dB
Rejection at 180º	Nil	Nil	maximum (null in the rear)	12 dB	6 dB
Mathematical model	$R = \cos 0^{\circ}$	$R = \cos \theta$	$R = 0.5 + 0.5 \cos \theta$	$R = 0.37 + 0.63 \cos \theta$	$R = 0.25 + 0.75 \cos \theta$
Nulls	Nil	2	1	2	2
Applications	PA system. Being omni-directional it is suitable for group discussions also	Face to face dialogues in dramas	orchestras and ensembles	Court scenes in dramas	Capable of providing more natural open and less focused sound for group discussion and conferences.

Output Impedance A microphone has an output impedance which is represented in ohms. This is an important parameter which is used to determine which type of matching transformer would be needed to transfer the power efficiently from the microphone to the transmission line and then to the amplifier.

Some microphones like dynamic microphones, have quite low output impedance (only a few ohms), and therefore have a built-in step up transformer to match the line impedance.

(When there is a built-in transformer, manufacturers generally quote output in millivolts available across the secondary of the transformer).

2.3 REQUISITES OF A GOOD MICROPHONE

A good microphone should have high sensitivity, high signal-to-noise ratio, flat frequency response over most of the audible frequency range, natural resonant frequency outside the audible range, and very low distortion. It should have correct output impedance (with or without built-in transformer) to match with the line impedance. The directivity of the microphone should be such as to meet the requirement of application.

From the point of view of impedance, the microphones are divided into three categories:

1. Very low impedance (from a fraction of an Ω to about 50 Ω)
2. Medium impedance (100–600 Ω)
3. High impedance (750 kΩ and higher)

Ribbon and moving-coil types are dynamic microphones and need a step up transformer to increase the impedance looking into the circuit. Static microphones (crystal and capacitor types) are high-impedance microphones, needing an emitter–follower amplifier for matching. Carbon microphone are also static type and have medium impedance.

In a shielded cable, the shield and the conductor act as a shunt capacitor which attenuates high audio frequencies. This attenuation is obviously more effective in high–impedance microphones.

2.4 MOVING-COIL MICROPHONE

Basics The moving-coil microphone (also called dynamic microphone) uses the principle of electromagnetic induction. When sound pressure variations move a coil placed in a magnetic field, there is a change of magnetic flux passing through the coil. An emf is, therefore, induced in the coil and this emf forms output of the microphone. (Due to similarity in construction, a moving coil loudspeaker can also work as a moving-coil microphone. The same unit is often used both as microphone and loudspeaker in office intercom systems.)

Construction The main components of a moving-coil microphone are a magnet, diaphragm and coil. These are shown in Fig. 2.7.

The *magnet* is a permanent magnet of pot type with a central pole piece (south pole) and the peripheral pole piece (north pole). This type of magnet gives a uniform magnetic field in the gap between the pole pieces.

The diaphragm is a thin circular sheet of non-magnetic material and is of light weight. It is slightly domed for extra rigidity. It is fixed to the body of the magnet with the help of springs. The springs provide compliance (equivalent to electrical capacitance) to the motion of a diaphragm. The mass of the diaphragm and coil assembly provide inductive effect.

The coil is wound on a cardboard cylinder which is attached to the diaphragm. The coil is a single layered, thin enamelled wire.

A protective cover (a metal grill) is used to save the delicate diaphragm and coil assembly from being mishandled. A silk cloth partition is used to separate the upper chamber from the lower chamber. A small tube is used in the lower chamber to give access to the free atmosphere.

The mass of the diaphragm restricts the high audio frequency output, and the stiffness (capacitive reactance) caused by the springs' compliance, restricts the low audio frequency output. The electrical equivalent circuit for a morning coil microphone is shown in Fig. 2.8.

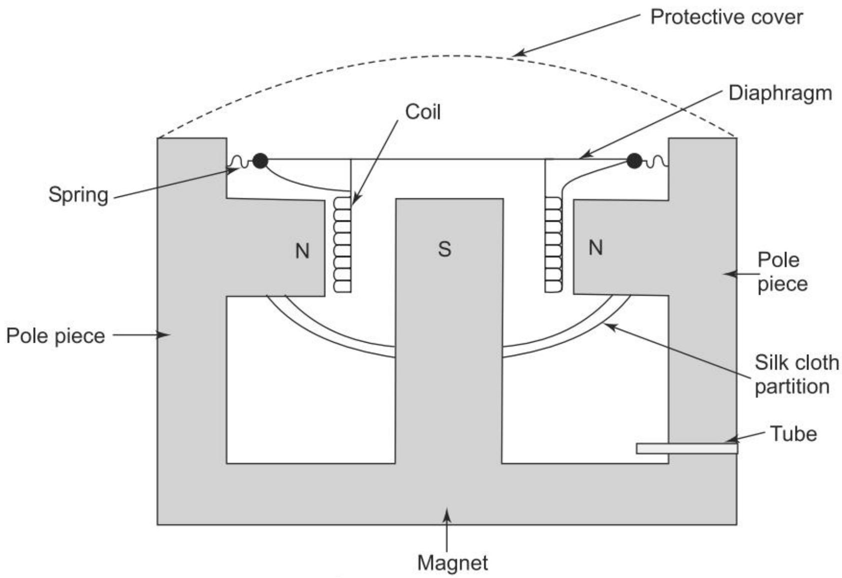


Fig. 2.7 *Moving-coil microphone*

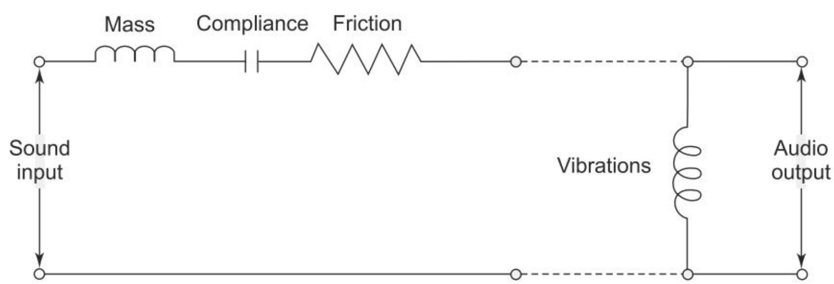


Fig. 2.8 *Electrical equivalent of moving coil microphone*

Principle of Working When sound waves strike the diaphragm, it moves and hence, the coil moves in and out in the magnetic field. This motion changes the flux through the coil, which results in an emf being induced in the coil due to electromagnetic induction. The value of this emf depends on the rate of change of flux and hence on the motion of the coil. The displacement of the coil depends on the pressure of sound waves on the diaphragm. Thus, it is a pressure microphone. The induced voltage, e, across the coil of the microphone is given by Eq. 2.2.

If B is the flux, density in tesla (or Wb/m^2), 1, the length of the coil in metres, v, the velocity of the diaphragm (and hence coil) in m/s then

$$e = \frac{d\phi}{dt} = B \times \text{change of area per second}$$
$$= B \times \text{length of conductor} \times \text{distance moved per second}$$
$$= B \times l \times v \qquad (2.2)$$

Thus, the induced emf is directly proportional to velocity which, in turn, is proportional to frequency of sound pressure variations.

Characteristics of the Moving-Coil Microphone (Typical Values)

Sensitivity 30 microvolts (or 90 dB below 1 volt) for sound pressure level of 0.1 Pa. A step-up transformer is used to match the cable impedance. The output across the secondary is about 90 microvolt.

Signal-to-noise ratio 30 dB.

Frequency Response 60 Hz to 8000 Hz for ±1 dB. (Low frequency response is further improved up to 40 Hz by using a small tube in the lower chamber of the microphone to give access to the free atmosphere.) It has a natural resonance between 3 and 4 kHz. Therefore, it is more prone to noise and feedback.

Distortion Less than 5%.

Directivity The moving-coil microphone is basically omnidirectional as it is a pressure microphone. Sound waves strike the diaphragm from the front, but sound wave, coming off the axis from sides or even from the back can reach the front by diffraction, Diffraction effect is reduced at higher audio frequencies and therefore, the microphone is truly omnidirectional at the lower audio frequencies only. It achieves directivity at higher audio frequencies, as illustrated in Fig. 2.9. The figure shows omnidirectional pattern at 100 Hz, but at higher frequencies of 1000 Hz and higher, the pattern does not remain omnidirectional.

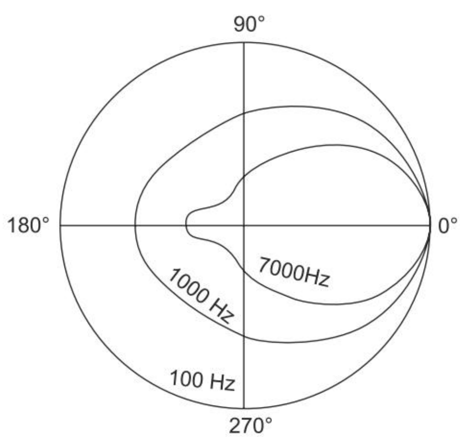


Fig. 2.9 *Directivity of moving-coil microphone vs. frequency*

Output Impedance Its output impedance is quite low and is about 25 ohms. Hence, to match a line of 200 ohm, a step-up transformer of about 3:1 turns ratio is built in the microphone case. This increases the output voltage to about 90 μV.

Other Features

(1) As the magnet is quite heavy, it can be encased in a heavy protective case, and hence it is very robust and reliable.

(2) It does not need external bias.

(3) It can be spoken into from a distance of about 25 cm.

(4) Its cost is not high. It is lower in cost than ribbon and capacitor microphones, but higher than crystal and carbon microphones.

Applications Moving coil microphone is very widely used, and is suitable for use in public address systems and broadcast studios. When it is coupled with a ribbon microphone in series, its cardioid pattern makes it useful for dramas and concerts.

2.5 RIBBON MICROPHONE

Basics In a moving-coil microphone, high frequency response is poor because of the mass of the diaphragm (including coil assembly). Mechanical mass is equivalent to electrical inductance, and this attenuates the higher frequencies. Hence, to increase the frequency response for high audio frequencies, a very light aluminum ribbon is used in place of a diaphragm and coil assembly. The ribbon acts as a conductor (placed in a magnetic field) as well as a diaphragm. Such a microphone is called a ribbon microphone.

Construction The main parts of a ribbon microphone, shown in Fig. 2.10, are permanent magnet, and ribbon conductor.

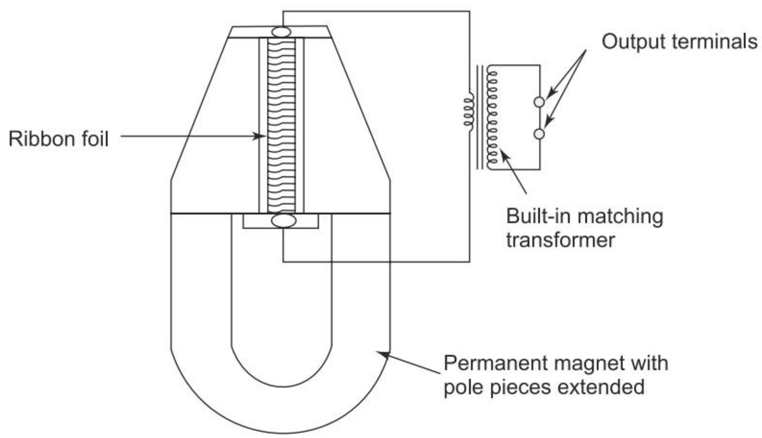


Fig. 2.10 *Ribbon microphone*

The *permanent magnet* is a specially designed horse-shoe magnet with extended pole pieces. It provides a strong magnetic field.

The *ribbon* is a light aluminum foil. It is corrugated at right angles to its length to provide greater surface area. The main feature is the lightness of the ribbon, which is only about 0.2 mg in weight, less than 1 micron thick and about 20 mm long and 3 mm wide. It is suspended in the magnetic field of the permanent magnet and the stiffness of suspension is small.

The whole unit is enclosed in a circular or rectangular baffle. The shape of the baffle is not purely circular or rectangular, but is rather irregular and depends on the structure of the magnet.

Principle of Working When the ribbon (an electric conductor), placed in a magnetic field, is made to move at right angles to the magnetic field by the force of sound pressure, there is a change of magnetic flux through the ribbon conductor. Due to this change, an emf is induced across the ribbon. This emf is proportional to the rate of change of magnetic flux which, in turn, is proportional to the force of sound waves striking the ribbon.

In the ribbon microphone, the driving mechanical force is proportional to the difference of the pressures acting on two sides of the ribbon, or the particle velocity of the sound waves. Hence it is also called ***Pressure Gradient*** microphone or ***Velocity*** microphone.

Characteristics of Ribbon Microphone

Frequency Response of Ribbon Microphone In a microphone, a high-frequency signal is attenuated due to inductive effect of mass of the moving system, and a low-frequency signal is attenuated due to low compliance (mechanical capacitance) or high stiffness (capacitive reactance). In the ribbon microphone, the mass of the moving system is very small (only about 0.2 mg) and hence mechanical inductance (M) is low resulting in good response for high audio frequencies up to 12000 Hz. The stiffness of the suspension of the ribbon is also small and results in good response for low frequencies, almost as low as the lowest audible frequency. The overall frequency response of ribbon microphone is 20 to 12000 Hz for ±1 dB. The natural resonant frequency is below 20 Hz and so does not lie in the audible range.

Directivity of Ribbon Microphone As the force of sound waves can be applied to the ribbon from the front as well as from the rear side, this type of microphone is bi-directional and its polar pattern is a figure of eight as shown in Fig. 2.11. Dotted curve represents a pattern for 1000 Hz, dashed curve for 5000 Hz and the solid curve for 10000 Hz. Patterns inside the circle are of ribbon microphone, while the outside circle represents omnidirection of pressure-type microphones and has been shown for comparison.

Some manufacturers have designed a ribbon microphone in such a way that the upper part is velocity operated and the lower part is pressure operated. As the two sections are from a single continuous ribbon, they are in series and give cardioid patterns.

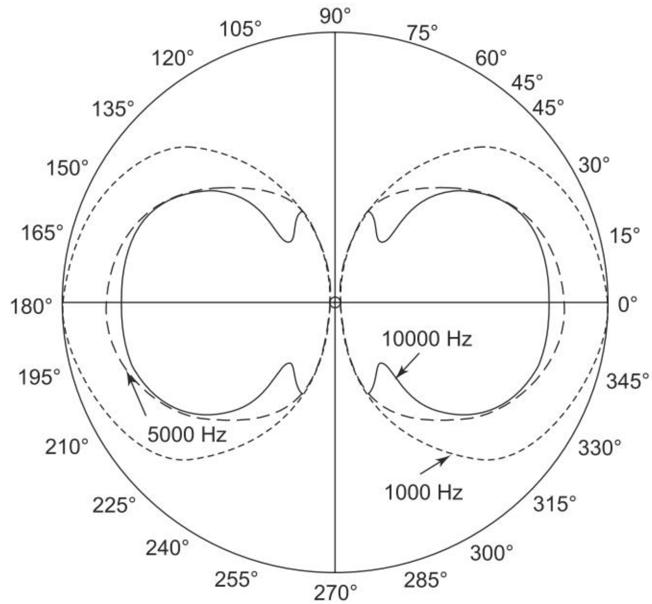


Fig. 2.11 *Directivity of ribbon microphones including frequency effect*

Impedance Impedance of a ribbon microphone is only a fraction of an ohm (about 0.25 Ω). It has a built-in step up transformer with a secondary to primary turns ratio of about 30:1 to match a cable impedance of 200 Ω.

Sensitivity The sensitivity of a ribbon microphone is less than that of a moving-coil type because the length of the ribbon is less than the length of the coil wire of a moving-coil-type microphone. It is about 3 μV or 110 dB below 1 V for a sound pressure of 0.1 Pa. Although low, this output is across a very low impedance. At the secondary of the matching transformer (30:1 turns ratio) the output would be about 90 microvolt, and hence is comparable with that of the moving-coil microphone.

Signal-to-noise Ratio Due to the directional properties, the ribbon microphone has much less background noise than the moving-coil type. Hence signal-to-noise ratio is higher than the moving coil microphone. It is about 50 dB. Therefore this type of microphone is more suitable than moving coil type for high fidelity systems.

Distortion Low (about 1%).

Other Features

1. The ribbon, being very light, is delicate and can be easily damaged by dropping. Hence it requires careful handling. A ribbon microphone,

which will rub on the magnetic pole pieces. The result is thin sound.

2. A dc current through ribbon from a battery of a meter can strain the ribbon. Hence resistance or continuity of ribbon should not be measured with a meter. It should be checked by visual inspection.
3. A ribbon microphone should not be blown into. It will cause strain in the ribbon. It is sensitive to the movement of air near it and to the shock vibrations. It is also sensitive to breathing. It should, therefore, be kept about 50 cm or more away from the source of sound. As it discriminates against background noise by a factor of 3, the speakers can stand at 1.7 times the distance prescribed for pressure microphones. From a distance of 50 cm, signal-to-noise ratio of this microphone will be the same as from lower distances for a moving-coil type. Bass output is increased if the speaker is close to the microphone.
4. The slightest wind causes rumble noise in the output.
5. It has excellent response for the starting transients produced by the musical instruments.
6. It does not need external bias.
7. Its cost is high as compared to the moving-coil type.

Applications It is very suitable for dramas. Its figure-of-eight polar diagram allows actors to talk face to face which is more natural than standing side by side. Further, the two dead sides in a ribbon microphone give an impression of large space because the voice of a person standing in the dead area, even if quite close to the microphone, will not be picked up directly, but will reach the microphone through reflections from the side walls and hence will sound distant.

Cardioid Microphones When a ribbon microphone and a moving-coil microphone both are mounted in the same housing, connecting them in series, a cardioid (or heart shaped) pattern results. This is shown in Fig. 2.12. Such microphones are suitable for orchestras and court scenes in dramas where a large number of persons are present. Due to the practically weightless ribbon, it gives the best result with ensembles and orchestras because the light ribbon can respond instantaneously to the minute changes in the complicated waveform of such music.

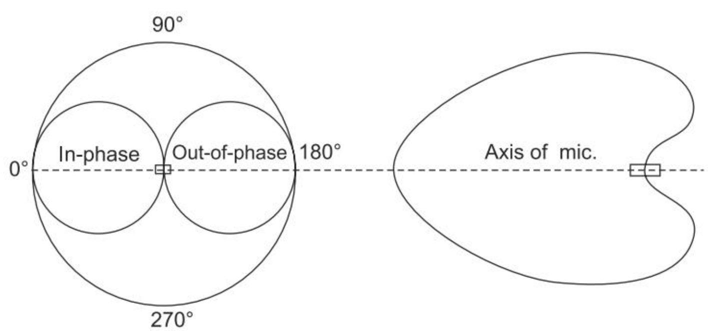


Fig. 2.12 *Resultant directivity of combining the moving coil and ribbon microphones*

2.6 CRYSTAL MICROPHONE

A crystal microphone is based on the principle of ***piezoelectric Effect***, which produces difference of potential between the opposite faces of some crystals when these are subjected to mechanical pressure. The crystals which show this effect are quartz, tourmaline, rochelle salt and ceramic. Rochelle salt has high piezoelectric effect but is susceptible to moisture. Also, it cannot withstand high temperature of more than 50°C in outdoor use. Quartz and tourmaline have low piezoelectric effect. Ceramic is most suitable for crystal microphones as it is not susceptible to moisture and can also withstand high temperatures up to 100° C.

When pressure is applied to the crystal, it deforms and momentary displacement of charge takes place within the crystal structure. This creates a difference of potential between its two surfaces.

Construction A typical crystal microphone is shown in Fig. 2.13.

The crystal is cut along certain planes to form a slice. Metallic foil electrodes are attached to the two surfaces to carry the potential difference to the output terminals.

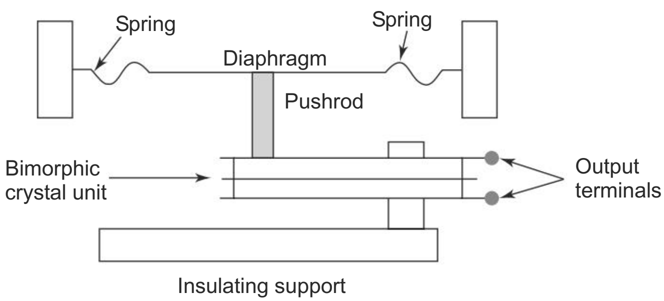


Fig. 2.13 *Crystal microphone*

Two thin crystal slices, suitably cut, are placed in an insulating holder with an air space between them. A large number of such elements are combined to increase the emf. A diaphragm, made of aluminium, is attached to the crystal surface through a push rod. The whole unit is encased in a protective case. There is a protective mesh cover (not shown in the figure) over the diaphragm.

Functioning When there is a sound wave of compression, it compresses the crystal. In case of rarefaction, the converse takes place and the crystal is extended and is under tension. Due to this compression and extension, a varying potential difference is generated which is proportional to the mechanical pressure applied to the crystal by the sound waves (it is, therefore, 'pressure microphone'). The crystal elements are connected in such a way that the potential differences developed in the elements are added up and we get a good voltage output (about 50 mV) for feeding to the amplifier.

Characteristics of a Crystal Microphone

Sensitivity The crystal microphone has good sensitivity, about 50 mV (or 26 dB below 1 volt) for 0.1 Pa pressure.

Signal-to-noise Ratio It is not prone to pick up background noise. Generation of noise inside the microphone is also low. Hence its signal-to-noise ratio is high, about 40 dB.

Frequency Response 100 – 8000 Hz for ±1 dB

Distortion Low, about 1%

Directivity Omnidirectional

Output Impedance High, about 1MΩ

Other Features

1. It is not as rugged as a moving-coil one, but is more rugged than a ribbon type.
2. It can be spoken into at close range.
3. Its high impedance requires relatively short leads to the input circuit of the amplifier to prevent loss of higher audio frequencies, or pick-up of hum. The leads must be well shielded and not more than about 6 metres long.
4. The mixer circuit will load it and cause severe loss of bass. Hence, it cannot be used in a multi-microphone system.
5. Unlike a moving-coil microphone and ribbon microphone, it has no frequency discrimination with direction.
6. It does not need a bias supply.
7. It should not be exposed to direct sunlight for a long time.
8. Its cost is low.

Applications It is used for the following purposes:

1. Home recording systems
2. Amateur communication
3. Mobile communication

However, due to variation of the acoustic characteristics, this type of microphone is not used in broadcasting and recording studios.

2.7 CAPACITOR (OR CONDENSER) MICROPHONE

Principle When the capacitance of a capacitor changes, the quantity of charge on the capacitor remains the same, but the level of voltage changes.

It is like water level in a vessel shown in Fig. 2.14. For fixed quantity of water, its level in a vessel would depend on the size of the vessel.

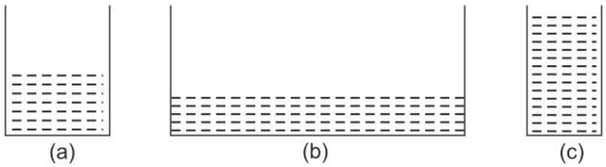


Fig. 2.14 *Levels of water in vessels of different capacities*

As the capacity of the vessel in Fig. 2.14 (b) is larger than Fig. 2.14(a), the level of water decreases in Fig. 2.14(b) as compared to Fig. 2.14(a). Similarly, the capacity of a vessel in Fig. 2.14(c) is smaller than Fig. 2.14(a), the water level is higher in it, as compared to Fig. 2.14(a). Here water level represents the voltage across the capacitor plates. Change of voltage level for the fixed quantity of charge is in accordance with Eq. (2.3).

$$V = \frac{Q}{C} \tag{2.3}$$

where, V = voltage across the capacitor in volts
Q = charge in coulombs
C = capacitance in farads

Equation 2.3 shows that if C increases, V will decrease and if C decreases, V will increase.

The diaphragm of the microphone acts as one plate of the capacitor. The other plate, called the *back plate*, is fixed. When sound pressure moves the diaphragm in, the capacitance increases, and when it moves out, the capacitance decreases. The change in capacitance results in change in voltage. The capacitor microphone is a pressure microphone, as sound waves coming from all sides strike the diaphragm on the front side only.

Construction A capacitor microphone is shown in Fig. 2.15. It consists of a light-weight metal diaphragm (generally aluminium) which is suspended above a fixed metal back plate.

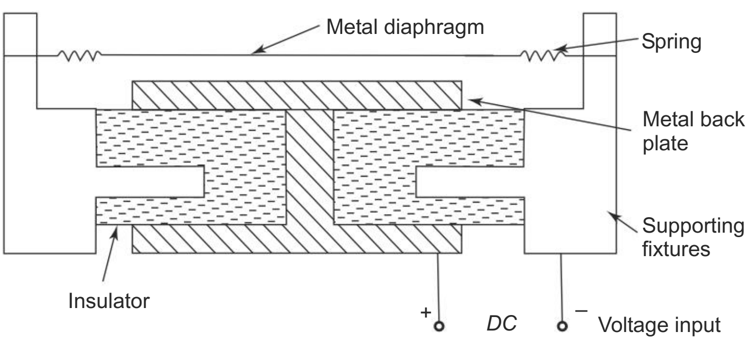


Fig. 2.15 *Capacitor microphone*

The metal diaphragm and the fixed metal plate are near each other and form a capacitance of a few picofarads. A fixed d.c.voltage of about 50 to 100 volts is applied between the backplate and the movable diaphragm plate. The diaphragm is in stretched condition as it remains attached to the supporting fixtures with the help of spider springs. The two plates are insulated from each other. The capacitance of this microphone is about 30 pF.

Functioning When sound waves strike the diaphragm, it moves. During compression, it moves towards the fixed back plate and increases capacitance. During rarefaction, it moves away from the back plate and therefore decreases capacitance. The change in capacitance changes the dc voltage across the capacitor plates. As distance between the plates changes, its capacitance changes as per Eq. 2.4.

$$C = \frac{kA}{d} \tag{2.4}$$

where, k is the dielectric constant of the medium between the plates, A, the area of cross section of the plates and d, the distance between the plates.

The voltage V between the capacitor plates is equal to Q/C.

In terms of the capacitor parameters of Eq. 2.4,

$$V = \frac{Q}{kA/d} = \frac{qd}{kA}$$

As Q, k and A are constants for a capacitor microphone,

$Q/(kA)$ is constant = say, K

Hence, $$V = Kd \tag{2.5}$$

Differentiating Eq. 2.5,

$$\delta\upsilon = K.\ \delta d$$

Characteristics of Capacitor Microphone

Sensitivity The output is very low and an amplifier is built-in inside the microphone case. The amplifier output is about 3 mV (about 50 dB below 1 V) at a sound pressure of 0.1 Pa or 1 μ bar.

Signal-to-noise Ratio High, about 40 dB.

Frequency Response Excellent, 40 Hz to 15 kHz for ±1 dB. Its frequency response is so good that it is used as *standard* microphone against which other microphones are calibrated and loudspeakers are tested. It is therefore used in sound level meters. Its natural resonant frequency is about 6000 Hz.

Distortion Low, about 1%

Directivity Omnidirectional

Output Impedance High, about 100 MΩ

Other Features

1. It needs an external dc bias supply.
2. It is delicate because of the narrow separation between the moving plate (diaphragm) and the fixed back plate.
3. It cannot withstand excessive heat. Moisture is also harmful as the condensation causes a crackling sound.
4. It is costly because of the necessity of a dc bias.

Applications

1. It is used as a standard microphone for calibrating other microphones.
2. It is used in sound level meters.
3. It is used in professional high-fidelity recording.

2.8 ELECTRET MICROPHONE

External dc bias in a capacitor microphone makes it costly and unsuitable for field work. In electret microphone, the external dc bias is dispensed with. Its constructional features are shown Fig. 2.16.

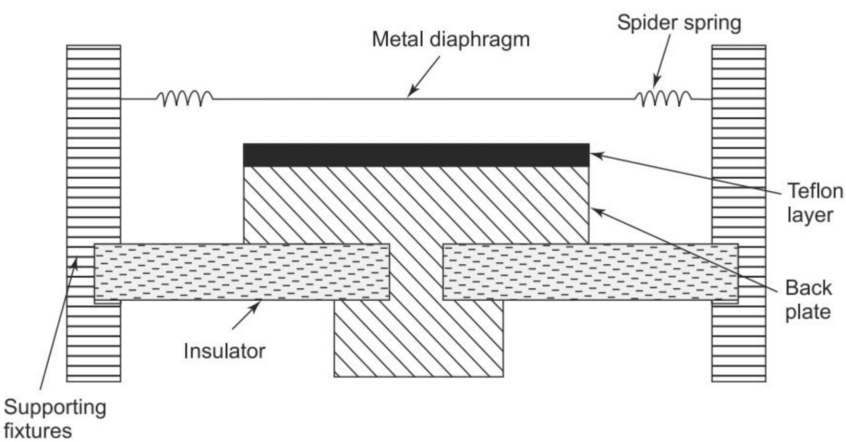


Fig. 2.16 *Electret microphone*

The electret microphone is also a capacitor microphone, but it has built-in charge. Insulating materials can trap a large quantity of fixed charge and can retain it indefinitely. The insulating material used is Teflon. The back plate of the microphone is coated with a thin layer of Teflon. The thin layer is charged negatively at the time of manufacturing. This negative charge remains trapped for a long period. The -ve charge induces +ve charge on the diaphragm. The positive charge on the diaphragm and negative charge on the Teflon establish an electric field across the gap of the capacitor plates. The charge results in a terminal voltage. When the capacitance changes due to sound pressure, the charge tends to remain constant and hence the terminal voltage changes.

It has the same characteristics as capacitor microphone except that it does not need external bias supply and is less costly. It is also sensitive to temperature and humidity which cause leakage of charge. It is used in sound level meters.

As the electret microphone is cheap, has good frequency response, is rugged, and does not need bias supply, it is also used in small PA systems for clubs and small halls, to keep the cost low. It being very light, is also used as tie clip microphones for lecturers and as radio (wireless) microphones in sports meets.

2.9 CARBON MICROPHONE

Principle When fine carbon granules enclosed in a case are subjected to variations of pressure, the resistance of the granules changes. When such a device of carbon granules is connected in series with a load through a dc supply, the current through the load will vary in accordance with pressure variations on the carbon granules.

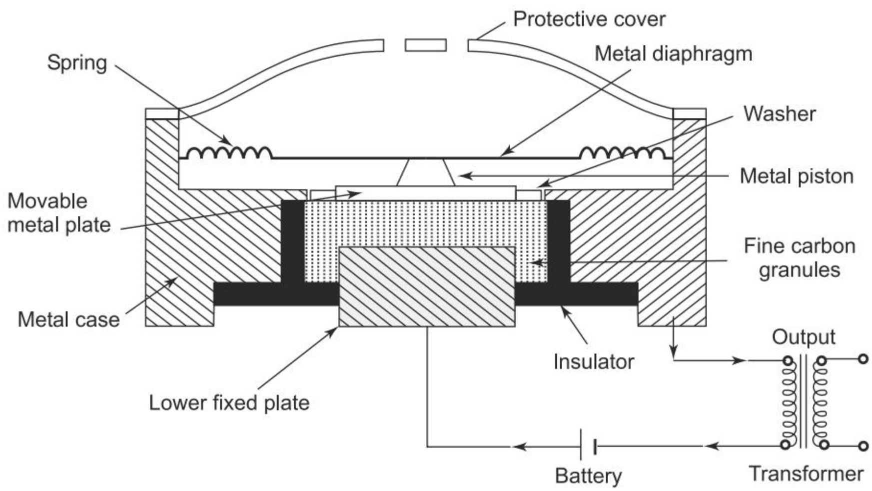


Fig. 2.17 *Carbon microphone*

Construction The construction of a carbon microphone is shown in Fig. 2.17.

Fine carbon granules are enclosed between two metal plates. The upper plate called diaphragm, is attached to a movable metal plate through a metal piston or plunger. The lower metal plate is fixed and is insulated from the diaphragm. A protective cover with holes is used to protect the unit.

A battery is connected between two metal plates. When the load is connected, current flows through the carbon granules and the load. Path of the current passes from the +ve battery terminal through the fixed lower plate, the resistance of carbon granules, movable metal plate, metal casing, and output transformer, as shown in Fig. 2.17. The purpose of the output transformer is to eliminate dc content of the microphone.

Functioning When sound waves strike the diaphragm, it moves to and fro. During compression condition, it presses the carbon granules and during rarefaction, it loosens them. When carbon granules are pressed, the resistance decreases and hence the current through the circuit increases. When carbon granules loosen, the resistance increases, decreasing the current through the circuit. In the absence of sound, a steady current flows. Thus, sound waves superimpose a varying current, or audio current on the steady dc current.

The net resistance of the carbon granules is given by Eq. 2.5.

$$r = R_0 + \delta r \tag{2.5}$$

where, r = Net resistance in ohms.

R_0 = Steady resistance in ohms for no sound

δr = Variation of resistance due to sound pressure (it will have positive as well as negative value)

The development of a voltage V across a load resistance R_1 is illustrated in Fig. 2.18.

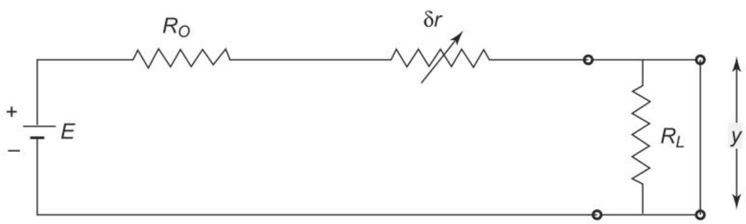


Fig. 2.18 *Equivalent circuit of a carbon microphone*

If E is the battery voltage and R_L, the load resistance then, $R_o + R_L$ = constant for constant load = say, R steady current I_o in the absence of sound-pressure variations $= \dfrac{E}{R}$.

When sound-pressure variations cause a change in the resistance R by δr then, the instantaneous current, I is given by Eq. 2.6.

$$
\begin{aligned}
I &= \frac{E}{R + \delta r} \qquad (2.6)\\
&= \frac{E}{R\left(1 + \dfrac{\delta r}{R}\right)}\\
&= \frac{E}{R}\left(1 + \frac{\delta r}{R}\right)^{-1}
\end{aligned}
$$

As δr is very small compared to R, $\dfrac{\delta r}{R}$ is a small quantity. Applying the binomial theorem, neglecting higher powers of $\delta r/R$, we get

$$I = \frac{E}{R}\left(1 - \frac{\delta r}{R}\right)$$

$$= I_o\left(1 - \frac{\delta r}{R}\right) \tag{2.7}$$

Equation (2.7) shown that the change in current, and hence the change in voltage across the load is proportional to the change in resistance (δr) of the carbon granules, and hence proportional to the pressure variations due to sound. When pressure increases, δr decreases and so I increases and the output voltage across the load increases.

Characteristics of a Carbon Microphone

Sensitivity Very high. The output of a carbon microphone is about 20 dB below 1V (i.e., about 100 mV).

Signal-to-noise Ratio Poor. Random variation of resistance of carbon granules generates a continuous hiss.

Frequency Response Carbon microphones have a frequency response of 200 to 5000 Hz, and therefore are unsuitable for high fidelity work. The resonance peak is at 2000 Hz and overall frequency bandwidth is usually up to 5 kHz.

Distortion High. The content is rich in harmonics unless variation in resistance (δr) is a very small percentage of steady resistance R. Distortion is of the order of 10%. Also, carbon granules have a tendency to stick to each other which further increases the distortion.

Directivity A carbon microphone is substantially omnidirectional. However, high frequency response over 300 Hz falls beyond an angle of 40° from the front of the microphone.

Output Impedance It is about 100 Ω.

Other Features

- It is mechanically very rigid.
- It is prone to moisture and heat.
- It is small in dimensions.
- Cost of the microphone is the lowest of all other microphones.

Applications Due to limited frequency range, it is useful only in telephones. It is also sometimes used in portable radio communication sets.

2.10 COMPARISON OF VARIOUS TYPES OF MICROPHONES

Parameters of different microphones have been compared in Table 2.2.

Table 2.2 *Comparison of microphones*

Parameter	Moving Coil	Ribbon	Crystal	Capacitor	Carbon
Sensitivity In μV for sound pressure of 1 μ bar.	30 without transformer, 90 with transformer	3 without transformer, 90 with transformer	50×10^3	3×10^3	100×10^3
Self-noise	Lower than carbon	Lower than moving coil	Lower than ribbon	Lowest	Highest
Noise pick-up	High	Low	Higher than ribbon	Low	Lowest
Frequency response for ±1 dB	60–8000 Hz	20–12000 Hz	100–8000 Hz	40–15000 Hz	200–5000 Hz
Natural resonance frequency	between 3 to 4 kHz	20 Hz	6000 Hz	6000 Hz	2000 Hz
Distortion	5%	1%	1%	1%	10%
Basic Directivity	Omni-directional	Figure-of-eight (Bidirectional)	Omni-directional	Omni-directional	Omni-directional
Output impedance	Low, 25 Ω	Lowest, 0.25 Ω	high, 1M Ω	Highest, 100 MΩ	Medium, 100 Ω
Ruggedness	Rugged	Most delicate	Less rugged than moving coil	Delicate	Most rugged
Effect of temperature	Negligible	Negligible	Yes	Negligible	Yes
Effect of moisture	Nil	Nil	Yes	Nil	Yes
Bias supply	Not required	Not required	Not required	Required for capacitor type but not required for electset type	Required
Distance of Speaker	25 cm	50 cm	Close	Close	Close
Size	Big	Big	Small	Big	Small
Cost	Medium	High	Low	High	Lowest
Applications	PA system, broadcast, Music. In cardioid form, suitable for group discussions, orchestra and conferences.	Drama, music, broadcast. In cardioid form it is suitable for group discussions and orchestra.	Home recording system, amateur communication	Professional recording, calibration., sound-level meters. Electret type is used in small halls, and clubs.	Telephones, Cheap portable communication sets.

2.11 SPECIAL MICROPHONES

Lavalier Microphone A small transducer suspended on the chest by means of a chord around the neck is called Lavalier microphone. It is a small moving-coil type, specially designed to work as Lavalier microphone. It has its applicability where mobility is necessary, for example, for a lecturer.

Tie-clip Microphone It is an electret type tiny microphone which can be clipped on to a tie, lapel or any other convenient part of the clothing. An external amplifier made on a tiny chip of silicon is used inside the microphone. Even with a tiny amplifier and its cell, it is very light.

Radio (Wireless) Microphone It uses a small frequency-modulated VHF transmitter of low power (a few milliwatts). Cable from amplifier to microphone is not needed. The signal is received by a VHF receiver placed at a suitable distance, then amplified and fed to the loudspeakers. It is useful in sports for oath-taking ceremony.

Noise Cancelling Microphone In this type of microphone, two matched transducing elements are used. These are mounted a few centimetres apart in the microphone housing and are connected together in the opposite phase. Sound which originates 50 cm (or more) away from the microphone will actuate both transducers equally. The output being in opposite phase will cancel. Sounds which originate within a few centimetres will affect the nearer element more and hence the two outputs will be unequal, and so will not completely cancel each other, resulting in signal output. The speaker has to use the microphone very close to his lips. Such microphones are suitable for use in noisy environments like sports meets etc.

Shot-gun Type Microphone It is the most directional microphone with the main lobe in the front and only very small lobes to the left, right and rear. This type of microphone is therefore not sensitive to the sides and the rear. The special sharp pattern is produced when the diaphragm is placed at the end of a long tube with slots cut along the sides. This is used in field recording of wildlife and also in outdoor TV interviews in noisy environment.

Fibre Optical Microphone It works on the principle of sensing changes in the intensity of light and converting these changes into electrical signals. Laser light from an LED travels through an optical fiber to illuminate the surface of a tiny reflective diaphragm. The diaphragm position variations, when sound pressure variations are incident on it, cause change in the intensity of reflected laser beam, which, then falls onto a photodiode detector The photodiode converts these variations into electrical signals.

No metal components are used in this type of microphone. It is immune to electromagnetic interference and radio frequency inference. Such a microphone is used in Magnetic Resonance Imaging (MRI) equipment environment for communication with the patient. It is resistant to environment parameters like temperature and moisture and is excellent for noise cancellation. However, it is costly.

Digital Interface Microphone A microphone when provided with a digital interface can directly give a digitized audio signal at the output (instead of first producing analog output and then digitizing the same by using sampling and quantization external to the microphone). Built in digital interface devices have become possible due to availability of switchable ICs. Miniature microphones, using crystals, can be used as digital hearing aids, in which microphone is so designed as to follow the ear's contours. Cells for IC amplifier may be button-size dry cells.

2.12 PRECAUTIONS WHILE USING MICROPHONES

A microphone being the first stage in sound amplifier and recording systems, has an important role in getting a replica of the original sound or the artistic sound (a result of mixing and editing techniques). Hence its correct use needs no emphasis. The following precautions should be taken while using microphones:

Selection of Microphones All microphones are not suitable for all applications. Hence selection of a microphone should logically depend on the application. For example, a ribbon microphone is suitable for dramas, but is not suitable for group discussions or for interviewing people outdoor. An omnidirectional moving-coil microphone will be suitable for outdoor interviews. Cardioid type is suitable to eliminate the background noise. Capacitor microphone is suitable for hi-fi work in studio where moisture can be controlled. For a solo presenter, a small tie-clip microphone or a hand-held microphone is suitable. For oath-taking ceremonies in sports, a wireless microphone is more suitable.

Elimination of Overload Distortion Excessive sound levels at the microphone cause overload distortion. To avoid it, the microphone should not be closer than 20 cm from the speaker. Greater distances are needed for musical instruments. A microphone should see the whole adstrument and hence ought not to be too close. For piano, a distance of 2 to 3 m is adversable. Sometimes, it may be desirable to keep two or three microphone near the instrument to cover each section of the string.

The rule of thumb is that the microphones should be so placed that they are looking in the direction that an audience would look.

Elimination of the Wind Effect Wind and even breath affect the signal-to-noise ratio of the microphone. These appear as disturbing blasts in playback. A suitable wind shield reduces the effect. Such a shield is supplied as an accessory to the sensitive microphones, susceptible to wind effect. In the absence of a shield, a clean handkerchief may be wrapped round the diaphragm side and may be held in position with a rubber band.

Rumble Noise produced subconsciously by a nervous foot or hand may be transmitted to the microphone through its stand. Such noise is very annoying when reproduced in playback. A suitable support of thick padding into which the stand may sink slightly (but does not compress it) reduces rumble.

Cock-tail Party Effect The brain discriminates against sounds which are regarded as unwanted or unimportant. This is known as cock-tail party effect. Hence while recording, the background noise is subconcionsly ignored, but during playback it appears and is irritable. The remedy for this is to use a directional microphone, a cardioid type or supercardioid type. The dead side of the microphone is pointed towards the source of noise.

Cable Laying The cable should preferably be laid out on the floor in such a way that tripping over by anybody may not occur. At doorways, cables can be carried on the top or may be laid under mats or carpets. There should be no loop, as a loop can be caught by a person moving around. Cable going to the microphone stand should be taped at the bottom of the stand and at the bottom of the table leg.

Microphones and cables should be as inconspicuous and tidy as possible. The audience wants to see performance and not a show of equipment.

Feedback A microphone should be so placed that sound from the loudspeakers does not reach the microphones to eliminate occurrence of howling.

SUMMARY

☞ **A microphone** is a transducer which converts sound pressure variations into electrical signals of the same frequency and phase and of amplitudes in the same proportion as in the original pressure variations. A good microphone should have high sensitivity, high signal-to-noise ratio, flat frequency response for audible frequencies, low distortion, matching impedance and directivity as per requirement. There are 5 directivity patterns, namely, omnidirectional, bidirectional, cardioid, supercardioid and hypercardioid.

☞ **Moving-coil Microphone** When sound-pressure variations move a coil placed in a magnetic field, magnetic flux through the coil changes. This induces emf across the coil. Its sensitivity is 90 dB below 1 V, signal-to-noise ratio of 30 dB, frequency response of 60 Hz to 8000 Hz, distortion less than 5%, directivity omni-directional (but dependent on frequency) and impedance 30 Ω. It is most widely used in PA systems and broadcasts.

☞ **Ribbon Microphone** It uses a light ribbon of aluminium in place of heavy coil. The driving force is equal to the difference of pressure on two sides. Hence, it is called a *pressure gradient microphone*. The emf is proportional to the rate of change of velocity and hence it is also called *velocity microphone*.

- ☞ Due to light weight of the ribbon, the high frequency response is excellent and is up to 12000 Hz. Its directivity is bidirectional or a figure of eight (8). Its impedance is less than 1 Ω without a transformer, and hence a matching transformer is essentially required. It has a sensitivity of 110 dB below 1 V, signal-to-noise ratio of 50 dB. Its main application is in dramas and studios. In combination with an omnidirectional microphone, it gives a cardioid pattern of directivity.
- ☞ **Crystal Microphone** It works on the principle of piezoelectric effect. When a ceramic crystal is subjected to mechanical pressure, emf is generated across opposite faces.
- ☞ Its sensitivity is 26 dB below 1 V, signal-to-noise ratio of 40 dB, frequency response of 100 to 8000 Hz, low distortion, omnidirectional directivity and MΩ impedance. It is used in home recording systems and in communication sets.
- ☞ **Condenser Microphone** Its working is based on the principle that $V = Q/C$. When C changes due to sound-pressure variations on the movable plate of a capacitor, V also changes. Hence, it gives output voltage corresponding to change of capacitance and consequently the change in pressure variations.
- ☞ It has the best frequency response (40 Hz to 15 kHz), 40 dB signal-to-noise ratio, low distortion, and 100 MΩ impedance. Its sensitivity is very low and hence, an amplifier is in-built which gives an output of 50 dB below 1 volt. The directivity pattern is omnidirectional. It is used as standard for calibrations and also in professional high-quality recording. Its main disadvantage is that it needs dc voltage of about 200 volts to charge the capacitor.
- ☞ Another version of a condenser microphone is the ***electret microphone***, which has in-built charge and hence does not need external battery. Its characteristics are similar to that of the condenser microphone. It is used in small PA systems as in clubs. It is also used as ***tie-clip microphone.***

☞ **Carbon Microphone** When fine carbon granules are enclosed in a case and subjected to variations of pressure, the resistance of the granules changes. When a small battery is connected in series with the microphone and load, the varying current through the load gives signal output.

☞ It has the highest sensitivity (20 dB below 1 volt). Signal-to-noise ratio is poor. The frequency response is 200 to 5000 Hz only. Distortion is high and impedance is about 100 Ω. The directivity pattern is omnidirectional. It is used in telephones.

☞ There are several other microphones for specific applications. These are Lavalier, tie clip, radio, noise cancelling, shot gun, optical fibre and digital interface.

Review Questions

1. Explain the parameters on which quality of a microphone depends. Give the values of these parameters for moving-coil, ribbon, crystal, condenser and carbon microphones.
2. Explain the working principle of a moving-coil microphone with the help of a neat sketch. Give typical values of its parameters and mention its applications.
3. With the help of a neat sketch, explain the principle of working of a ribbon microphone. Why is it known as a velocity microphone? Explain the directivity pattern of a ribbon microphone and discuss how cardioid pattern can be obtained. Mention its applications.
4. Distinguish between cardioid, supercardioid and hypercardioid patterns and specify their applications.
5. With the help of a neat sketch, explain the functioning of a crystal microphone. Give values of its parameters and applications. Explain why ceramic crystal is more suitable than rochelle salt crystal.
6. Explain the working of a condenser microphone with the help of a neat diagram. What is its main disadvantage and how is it eliminated by an electret microphone. Draw its frequency response curve and mention its applications.
7. Describe the construction of a carbon microphone with the help of a neat figure, and explain its functioning. Give its parameters and applications.
8. Write short notes on the following:
   (1) Lavalier microphone
   (2) Tie clip microphone
   (3) Wireless microphone
   (4) Noise-cancelling microphone
   (5) Digital interface microphone
   (6) Shot gun microphone

9. Compare the parameters of moving-coil, ribbon, crystal, condenser and carbon microphones. Discuss applications of each type.
10. Discuss the precautious which should be taken while using microphones.

Short-Answer Questions

1. What is the importance of a microphone in amplifiers, recorders and broadcast systems?
2. Define sensitivity of a microphone. What is its SI unit?
3. What is the reason that a microphone is unable to produce a complete audio range of frequencies?
4. How are different cardioid patterns obtained?
5. Why is a moving coil microphone not omnidirectional at higher audio frequencies?
6. Why is a capacitor microphone most suitable for sound-level meters?
7. Why is a carbon microphone not suitable for public address amplifier systems?
8. Which microphone is most suitable to cover court scenes in dramas?
9. Why is a built-in amplifier needed for a capacitor microphone?
10. What phenomenon makes a loudspeaker howl?

Multiple-Choice Questions

1. What makes a microphone omnidirectional?
   (a) Reflection
   (b) Refraction
   (c) Diffraction
   (d) Pressure gradient
2. On what principle does a moving-coil microphone work?
   (a) emf induction
   (b) Motor
   (c) Amplifier
   (d) Oscillator
3. Which of the following microphones is not a pressure microphone?
   (a) Moving coil
   (b) Ribbon
   (c) Capacitor
   (d) Crystal
4. In a ribbon microphone, what of the following is responsible for frequency response being higher than a moving-coil type?
   (a) Mass (b) Compliance
   (c) Friction (d) Magnet
5. Which of the following microphones is called a *velocity microphone*?
   (a) Carbon (b) Moving coil
   (c) Capacitor (d) Ribbon

6. Which microphone's directivity is like a figure-of-eight?
   (a) Moving coil
   (b) Ribbon
   (c) Crystal
   (d) Capacitor
7. What is the impedance of a ribbon type microphone?
   (a) Less than 1Ω
   (b) 100 Ω –1000 Ω
   (c) 1 m Ω
   (d) 100 m Ω
8. Which microphone uses piezoelectric effect?
   (a) Moving (b) Ribbon
   (c) Crystal (d) Capacitor
9. Which crystal is the best for a crystal microphone?
   (a) Quartz
   (b) Tourmaline
   (c) Rochelle salt
   (d) Ceramic
10. Which type of microphone has the best frequency response?
    (a) Capacitor (b) Crystal
    (c) Carbon (d) Moving coil
11. Which microphones need an external dc voltage?
    (a) Moving coil and ribbon
    (b) Crystal and electret
    (c) Capacitor and carbon
    (d) Ribbon and hypercardioid
12. Cardioid microphone is obtained by a combination of the following two microphones.
    (a) Carbon and condenser
    (b) Ribbon and moving coil
    (c) Crystal and condenser
    (d) Carbon and crystal
13. Which microphone has the highest sensitivity?
    (a) Moving coil
    (b) Capacitor
    (c) Crystal
    (d) Carbon
14. Which microphone has the lowest frequency range?
    (a) Moving coil
    (b) Capacitor
    (c) Crystal
    (d) Carbon
15. What is the cause of rumble noise in PA system?
    (a) Nervousness of the speaker
    (b) Wind
    (c) Breathing sound
    (d) Bad microphone
16. Which microphone is the best for face-to-face dialogues in dramas?
    (a) Moving coil
    (b) Ribbon
    (c) Crystal
    (d) Capacitor

Numerical Problems

1. Sensitivity of a microphone is 80 dB below 1 volt for sound pressure of 0.1 Pa. Find out its output in microvolts.
2. Presuming linear relationship, calculate sensitivity of the microphone of Q.1 for 1 Pa pressure.

3. The output of directive microphone in the best direction is 1 mV, when the output of an omnidirectional microphone placed under identical conditions with respect to the source is only 100 μV. Calculate the directivity of the directive microphone in decibels.
4. A microphone placed at 10 metres from a source which is radiating 1 watt, gives maximum output. When the direction of a microphone is changed by 90°, the power of the source has to be increased to 10 watts to get the same output from the microphone. Calculate microphone's directivity in dB in optimum direction with respect to 90° from the optimum.
5. The resistance of carbon microphone changes by ±50 Ω by the incident sound wave. The dc resistance of the microphone and load are 150 Ω and 10,000 Ω, respectively. Battery voltage is 1 volt. Calculate peak-to-peak audio output in millivolts.

Answers

Short-Answer Questions

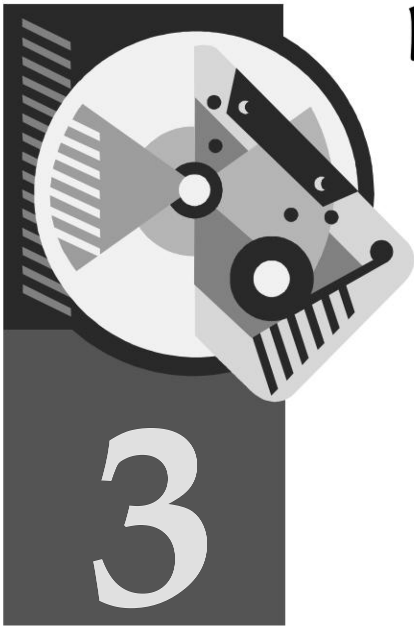
1. Microphone is the first link in any PA system or audio recorders or communication systems.
2. Sensitivity is defined as output in millivolts (or in dB below 1V) for the sound pressure of 1 Pa. Its SI unit is mV per pascal.
3. Mass, compliance and friction behave like inductance, capacitance and resistance, respectively of an acoustic circuit. These attenuate higher and lower frequencies and Q of the microphone circuit.
4. Different cardioid patterns are obtained by combining a moving-coil microphone with a ribbon microphone.
5. Omnidirectional property is achieved by diffraction of sound waves around the case, Low frequencies diffract easily, Diffraction is reduced at high audio frequencies and hence the moving coil is not omnidirectioal at higher audio frequencies.
6. Capacitor microphones frequency response is excellent, 40 Hz to 15 kHz. This makes it most suitable for sound level meters.
7. Frequency response and distortion of a carbon microphone is the worst, and hence it is not suitable for PA system.
8. Cardioid microphone, a combination of moving coil and ribbon is most suitable to cover court scenes.
9. Capacitor microphone's output is very low, and hence a built in amplifier is needed.
10. Feedback from loudspeaker to the microphone in PA system causes howling.

Multiple-Choice Questions

1. (c)	2. (a)	3. (b)
4. (a)	5. (d)	6. (b)
7. (a)	8. (c)	9. (d)
10. (a)	11. (c)	12. (b)
13. (d)	14. (d)	15. (a)
16. (b)		

Numerical Questions

1. (100 μV)	2. (1 mV)
3. (20 dB)	4. (10 dB)
5. (9.7 mV)	



Loudspeakers

3.1 CHARACTERISTICS OF LOUDSPEAKERS

A loudspeaker is a transducer which converts electrical signals of audio frequency into sound waves of the same frequency. It is also called output transducer or reverse transducer. A loudspeaker's performance is determined by the following characteristics:

1. Efficiency It is defined as the ratio of output sound power to the input audio (electrical power). Its value depends on proper matching of the mechanical impedance with acoustical impedance of the air volume being disturbed. (Some manufacturers quote the efficiency in terms of sensitivity which is defined to be the input signal required to give a sound pressure level of 0.1 Pa or 1 microbar at a distance of 1 metre from the loudspeaker.)

2. Noise The unwanted sound, not contained in the input signal but present in the output of a loudspeaker is called noise produced by the loudspeaker (the mechanical parts may vibrate at some resonant frequency, causing noise). What is more important is signal-to-noise ratio (S/N or SNR) of the system which is defined as ratio of 'signal output' to the 'output of noise in the absence of signal'.

3. Frequency Response It indicates the loudspeaker's response for the audible frequency range of sound. Ideally, the response of a loudspeaker should be flat within ±1 dB for the frequency range of 16 Hz to 20 kHz. However, due to mass of the diaphragm assembly, high frequencies are attenuated; and due to series compliance, low frequencies are attenuated. Moreover, the movable system may have some natural resonant frequency within the audible range and the output at that frequency will be emphasised.

4. Distortion Any change in frequency, phase and amplitude complexion of the output sound as compared to the input audio signal is called distortion. Frequency and phase distortions may result due to mass and compliance effect. Amplitude or non-linear distortion will result due to non-uniformity in the magnetic field in which the coil moves.

5. Directivity It is the ratio of actual sound intensity at a point (in the direction of maximum intensity) to the sound intensity that would have been available there, had the loudspeaker been omnidirectional.

6. Power It is the maximum audio power (indicated in watts) for which it is designed. Power more than the maximum will damage the speaker.

7. Impedance The input impedance of the loudspeaker is represented in ohms and is an important parameter, as its matching with the impedance of source amplifier is necessary for the optimum efficiency.

For installations, requiring long leads, a fixed line voltage is specified for the system. It is 100 volt in India and 70 volt in USA. It means that the amplifier at the peak output power will give 100 volt to the loudspeaker line connected to it. Impedance of the loudspeaker is low, and therefore the line voltage is reduced by using a step-down transformer at the loudspeaker end. This method reduces loss of power in long leads. The impedance R ohms of the amplifier, giving a peak output power of P watts can be calculated by Eq. 3.1 for the 100 V-System.

$$R = \frac{100 \times 100}{P} \tag{3.1}$$

This aspect has been further clarified in Example 3.1.

Example 3.1 *A long cable used for a loudspeaker has a finite resistance of 1 Ω.The amplifier used delivers, 100 W peak power at 10 A and 10 V at the primary of an output transformer. Calculate power loss in the cable for (a) 100 V line system and (b) non-100 V line system.*

Solution

(a) Line voltage = 100 V. Hence, a step-up transformer is used to transform 10 V of the emplifier to 100 V at the line.

Therefore current I in the line

$$= \frac{100\text{ W}}{100\text{ V}} = 1\text{ A}$$

Hence, loss of power in the cable $= I^2 R = 1 \times 1 = 1$ W. Thus, power delivered to the loudspeaker is 99 W.

(b) Power in the secondary = 100 W

Voltage at the secondary = 10 V (step-up transformer not needed)

$$\text{Current in the cable} = \frac{10}{1\Omega} = 10\text{ A}$$

Therefore, loss in the cable $= I^2 R = 10^2 \times 1 = 100$ W

Thus, no power would be delivered to the loudspeaker.

Example 3.2 *Calculate the efficiency of loudspeaker, if power intensity of sound is 0.08 W/m² at 1 metre distance from the loudspeaker, when the amplifier delivers 25 watts.*

Solution

$$\frac{P}{4\pi r^2} = 0.08$$

(where P is the acoustic power given by the loudspeaker)

As $r = 1$ m,

$$P = 0.08 \times 4\pi = 1 \text{ W}$$

Hence,

per cent efficiency

$$= \frac{1}{25} \times 100 = 4\%$$

3.2 MOVING-COIL CONE-TYPE LOUDSPEAKER

Principle The moving-coil loudspeaker works on the principle of interaction between a magnetic field and current in the same way as an ac motor works.

A coil, called voice coil, is placed in a uniform magnetic field. When the audio current passes through the voice coil, there is an interaction between the magnetic field and the current, resulting in a force working on the movable coil. This force is proportional to the audio current, and hence causes vibratory motion (motor like action) in the coil, which makes a conical paper diaphragm to vibrate and produce pressure variations in air, resulting in sound waves.

The force on the coil due to interaction between the current through coil and the magnetic field is given by Eqs. 3.2 and 3.3.

$$F = Bli \sin \alpha \tag{3.2}$$

where, F = force in newtons, B = flux density in tesla
l = length of the coil wire in metre
i = current in amperes
α = angle between the coil and the field

Normally, $\alpha = 90°$, and hence

$$F = Bli \tag{3.3}$$

When the motion of the coil is small, it will remain within the region of uniform flux density. Thus, for small motion of the coil, the output is linear.

(When the coil moves in the magnetic field, a back emf is produced due to electromagnetic induction. This makes motion of the coil steady.)

Construction Construction of a cone-type moving-coil loudspeaker is shown in Fig. 3.1

The moving-coil loudspeaker consists of a voice coil (single layer winding of fine enamelled wire), wound on a cardboard or fibre cylinder. Audio current is fed to the loudspeaker through two terminals. The coil is placed in a magnetic field.

The magnet is a pot-type permanent magnet which has a central pole (south pole) and a peripheral pole (north pole). The magnet is made of high-grade magnetic material, like alnico (composition of alnico is 18% aluminium, 10% nickel, 12% cobalt, 6% copper and 54% iron) which retains the magnetism extremely well. The magnet is so shaped as to give strong radial magnetic field in the annular space between the central and peripheral poles. The voice coil is free to move in the annular space having strong and uniform magnetic field. Because of the use of a permanent magnet, it is also called 'permanent magnet moving-coil (PMMC) type speaker.

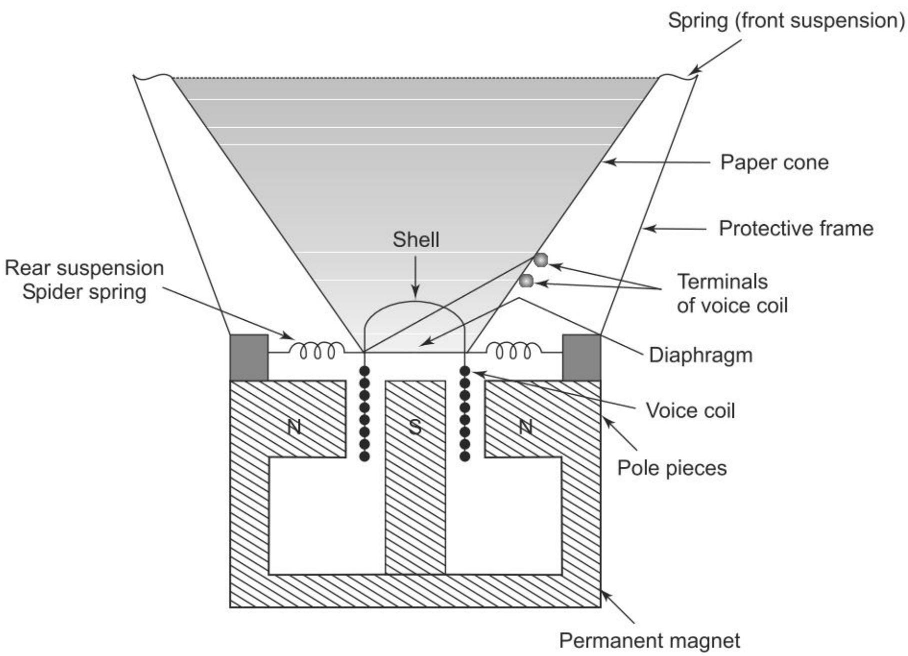


Fig. 3.1 *Moving-coil (cone-type) loudspeaker*

The coil is attached to a conical diaphragm, made of paper or parchment. It is called 'paper cone'. The cone is corrugated having circular corrugation to increase the surface area for better efficiency. A flexible strip of rubber round its periphery is used to support it. The spider springs are used to support the complete diaphragm and also provide the required stiffness to restrain the motion. The spiders also keep the coil centered, so that the cone moves forward and backward only.

Leads from the voice coil are cemented to the cone surface. From there, these are brought to the terminals mounted on the metal frame or basket.

Functioning When an audio current flows through the voice coil placed in a magnetic field, a force equal to *Bli* newtons acts on the coil and moves it to and fro. The paper cone attached to the coil also moves and causes compression and rarefaction cycles in the air. Thus, audio current is converted into sound waves. The equivalent circuit of the cone speaker is shown in Fig. 3.2.

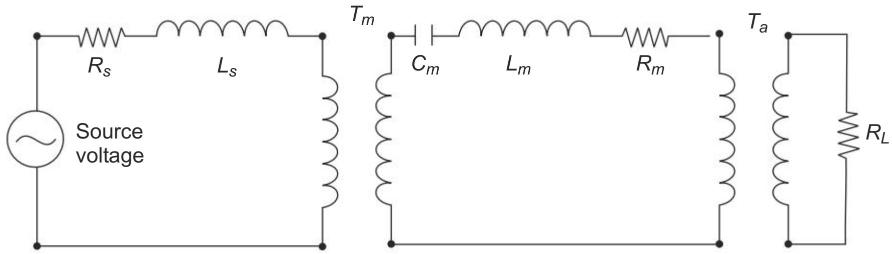


Fig. 3.2 *Equivalent circuit of the cone speaker*

There are two transformations. One is electromechanical and the other is mechano-acoustical. The electromechanical transfer, represented by transformer T_m, transfers force (produced by the source current in inductance L_s of the voice coil and the associated resistance R_s) to the movable mechanical parts (voice coil, diaphragm, springs and cone). Mass, compliance and friction of the moving parts are represented by L_m, C_m and R_m which are analogous to inductance, capacitance and resistance, respectively. The mechano-acoustical transfer, represented by transformer T_a transfers motion (forward and backward vibration of the cone) to the air-mass, which with its mass, compliance and viscosity (analogous to inductance, capacitance and resistance), acts as load and is represented by R_L. At low audio frequencies, L_m is negligible and the output depends on the compliance, C_m. At high audio frequencies, C_m is negligible and the output depends on L_m. So the high-frequency speakers (tweeters) are of low mass, and the low-frequency speakers (woofers) are of high compliance (large size). Typical frequency response of a 20-cm sized cone-type loudspeaker is shown in Fig. 3.3.

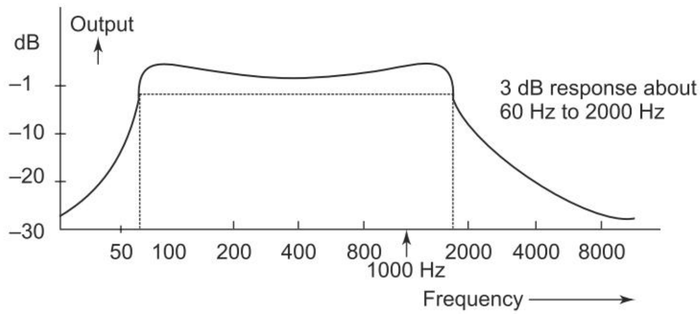


Fig. 3.3 *Frequency response of a 20 cm cone-type loudspeaker*

Direct Radiating Type The whole paper in a cone-type loudspeaker acts as a diaphragm and causes pressure variations direct in the listeners' area. Hence it is called 'direct radiating type loudspeaker'.

Characteristics of the Cone-type Speaker

Efficiency The efficiency of a cone speaker is quite low, about 5 per cent only. The poor efficiency is due to the fact that it acts as a direct radiator, and so there is complete mismatch between the low acoustic load presented by the large volume of air and the high mechanical load presented by the voice coil and cone assembly.

Signal-to-noise Ratio It is 30 dB or better.

Frequency Response It is restricted to mid-frequencies only. Frequency response drops at low and high audio frequencies for a typical loudspeaker. However a massive loudspeaker (called woofer) for low frequencies and small size speaker (called tweeter) for high frequencies can be designed. 3 dB frequency response of a typical speaker is from 60 Hz to 2000 Hz. Low-frequency woofer speakers with baffles will give frequency response up to 30 Hz. High-frequency tweeters extend the higher frequency response to 10 kHz or even higher.

Distortion Non-linearity due to non-uniformity in the magnetic flux density causes severe non linear or amplitude distortion (up to about 10%).

Directivity Basically, the loudspeaker is omni-directional. But baffles and enclosures modify the directivity so that most of the power is in the front. However, high audio frequencies are concentrated in a narrow cone about the axis of the radiator.

Impedance The effective impedance taking into account the mechanical and acoustical loads varies from 2 Ω to 32 Ω. The common impedances in commercial speakers are 4, 8 or 16 Ω (200 to 300 Ω impedances are obtained in an electrodynamic type cone speaker).

Power handling Capacity Power range of speakers lies between a few milliwatts (for 2 cm speaker) to about 25 watt for large size speakers. (Electrodynamic speakers can withstand a few hundred watts of input power.)

3.3 ELECTRODYNAMIC LOUDSPEAKER

To provide very strong magnetic field for high wattage speakers, an electromagnet is used instead of a permanent magnet. The working principle of an electrodynamic speaker is the same as that of a permanent magnet type. Its construction is shown in Fig. 3.4.

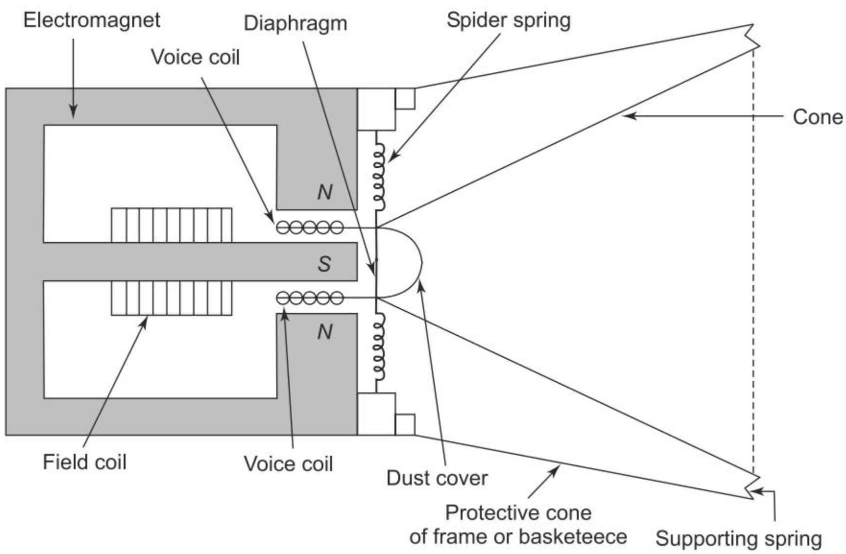


Fig. 3.4 *Electrodynamic loudspeaker*

Loudspeakers of more than 25 watt and up to a few hundred watts are of the electrodynamic type.

The strong and steady magnetic field is produced by a large field coil wrapped around a core. The shape of the magnet is pot type with the south pole in the centre and the north pole in the periphery. The special shape of the core allows magnetic flux to remain concentrated in the annular gap between pole pieces.

The ***voice coil*** is wound on fibre or aluminium (to keep it light in weight). It is placed in the annular gap. The audio signal from the amplifier's output transformer is applied to the voice coil. This signal causes a varying magnetic field. The resultant interaction between the two magnetic fields (one due to electromagnet and the other due to audio current in the voice coil) produces mechanical vibrations (motor action) in the coil assembly, which correspond to the audio signals.

The vibrations of the coil are transmitted to the attached cone which create sound waves in the air in the listeners' area, and hence radiate sound energy directly.

Advantages

1. Higher power can be obtained
2. Frequency response is better (40 Hz to 5000 Hz)

Disadvantages

1. Power supply needed for field coil
2. Heavier weight for the same amount of magnetic field
3. Costlier

3.4 HORN-TYPE OR INDIRECT RADIATING TYPE LOUDSPEAKER

Principle A horn-type loudspeaker uses a moving-coil placed in a magnetic field (similar to paper cone-type), but instead of radiating acoustic power direct in open space of the listeners' area, the power is first delivered to the air trapped in a fixed non-vibrating tappered or flared horn, and from there to the air in the listeners' area. Thus, it radiates sound power to the air in space not direct from the diaphragm but indirectly through the horn. This is the reason why the horn-type loudspeaker is called *indirect radiating loudspeaker*. The horn does acoustically what the cone does mechanically. The horn acts as an acoustic transformer. This allows better impedance-match between low impedance of the free air and the high impedance of the vibrating voice coil assembly, which results in increased efficiency. The efficiency of a horn-type speaker is 30–50% as against only 5% efficiency of cone-type speaker.

Construction The horn is a tapered enclosure whose diameter increases from a small value at one end (called *throat*) to a large value at the other end, (called *mouth*). There is an air chamber trapped between throat and diaphragm. The chamber is lined with sound-absorbing material like loose felt. The drive unit is similar to a direct radiating type except that the paper cone is not present. A basic horn (called exponential horn) type speaker is shown in Fig. 3.5.

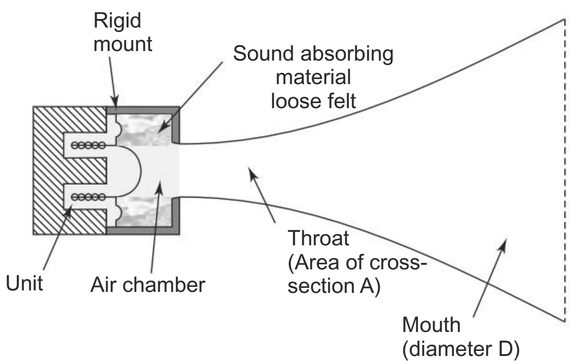


Fig. 3.5 *Horn-type loudspeaker*

The exponential horn is a straight circular tube whose cross-sectional area increases logarithmically along its length from the throat onwards.

The horn acts as a high-pass filter. The cut off frequency f_c is given by Eq. (3.4)

$$f_c = \frac{C\,A}{2\pi\,V} \tag{3.4}$$

where, C = velocity of sound,

A = area of cross section of throat, and

V = volume of air chamber trapped between throat and diaphragm

In terms of diameter of the mouth, the lowest frequency which can be reproduced by a horn-type speaker is given by Eq. 3.5.

$$f_c = \frac{170}{d} \tag{3.5}$$

where, f = frequency in Hz

d = diameter of the mouth in metre

At high frequencies, distances of different points on the diaphragm from the horn will not be equal, causing phase difference and hence resultant cancellation. To overcome this difficulty, special chambers have been developed as illustrated in Fig. 3.6.

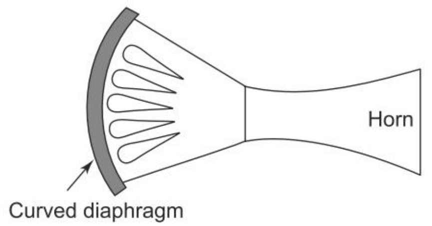


Fig. 3.6 *Special chamber to compensate for inequal distances of the points on diaphragm for horn*

To improve low-frequency response, we need-horn of large size. The length of the horn may be as big as 2 m and the diameter of the mouth is 1 metre. Such large dimensions for a horn are unwieldy, and hence the horn structure is folded back in itself to conserve physical space (low frequency response is improved by a wide mouth and high frequency response by a small throat).

A low frequency horn for hi-fi system is shown in Fig. 3.7. It contains a cone loudspeaker with a horn. The front of the cone faces away from the audience, but the enclosure of the cone and the exponential horn are so placed that the output is directed towards the listeners. Such a horn can be placed in the corner of a room to use the two walls as the extended sides of the horn.

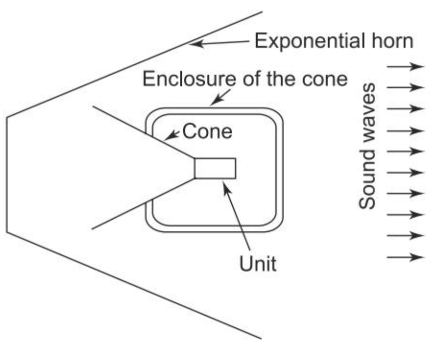


Fig. 3.7 *Horn for high fidelity*

Characteristics of Horn-type Speaker

Efficiency High, 30–50%.

Signal-to-noise Ratio 40 dB.

Directivity The directivity, frequencywise, is shown in Fig. 3.8 (pattern *A* is for 4000 Hz, *B* for 800 Hz and *C* for 400 Hz). Angle between half-power points around the horn axis is about 90°. However, the directivity of horn-

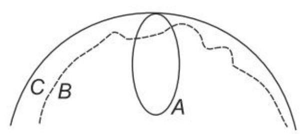


Fig. 3.8 *Directivity of horn-type speaker*

type (as also of cone-type) differs considerably from low audio frequencies to high audio frequencies. Radiation of high audio frequencies is concentrated in a narrow cone about the axis of the horn.

Frequency Response 30–10000 Hz (a tweeter horn is used to improve the frequency response further to 20 kHz).

Distortion Low, less than 5%

Impedance 16 ohms

Power-handling Capacity It is much more than a cone-type. About 100 watts can be easily accomodated in a horn-type speaker of feasible size.

3.5 COMPARISON BETWEEN CONE-TYPE AND HORN-TYPE SPEAKERS

Table 1.1 compares the characteristics of horn-type and cone-type speakers.

Table 3.1 *Comparison of cone-type and horn-type speakers*

ITEM	CONE-TYPE	HORN-TYPE
Principle of producing driving force	Interaction between two magnetic fields (motor-action).	Interaction between two magnetic fields (motor-action)
Diaphragm	Light diaphragm of aluminium which is extended in the form of a paper cone.	Light diaphragm of aluminium. No paper cone.
Transfer of mechanical energy to the air in the liseners' area in the form of acoustic energy (or sound)	The paper cone directly transfers. and is, therefore, called direct radiating loudspeaker.	Mechanical energy of diaphragm is transferred to the throat through air chamber and then to horn, and finally to the air. Thus, it is indirect transfer.
Matching of mechanical impedance of diaphragm to the acoustic impedance	No matching, it is a complete mismatch.	Matching is achieved. Low impedance of free air is matched with high impedance of diaphragm.
Efficiency	5%	40%
Frequency response	Poor, 60–5000 Hz needs woofer for lower frequencies and tweeter for higher frequencies.	Good, acts as a high-pass filter. The cut off frequency may be as low as 30 Hz. Frequency response is 30 to 10000 Hz.
Signal-to-noise ratio	30 dB	40 dB
Distortion	More than 5%	Less than 5%
Directivity	Broad cone about the axis of the radiator. It is more directive for high audio frequencies.	Same as for cone-type

Contd.

Contd.

Power-handling capacity	milliwatts to about 25 watts. Electrodynamic type can work up to a few hundred watts.	25 watts and above up to about a hundred watt.
Impedance	2 to 32 Ω for permanent magnet type and up to 200 Ω for electro-dynamic type.	16 Ω
Size	Compact	Large
Cost	Low	High.
Applications	Radio receivers, TV receivers, record players, cassette players, small audience.	PA system for large audience, Music concerts, big auditorium.

3.6 LOUDSPEAKER COLUMN OR LINE SOURCE SPEAKERS

When several drive units are mounted one above the other in a suitable enclosure, the loudspeaker form a line of source and are known as line source speakers. As the loudspeakers are in a column, one above the other, such an arrangement is also called column of loudspeaker, or more commonly a column speaker. This is shown in Fig. 3.9 for 5 speakers in the column.

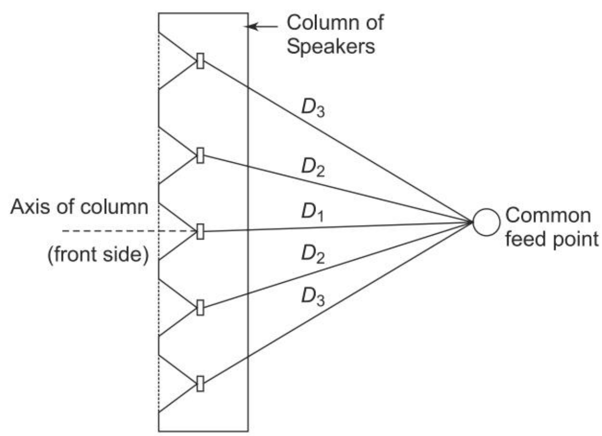


Fig. 3.9 *Column of loudspeakers fed from a common point*

The same phase difference from the common feedpoint can be ensured by making the length of the cable from common feed point to the respective units of loudspeakers to be the same irrespective of actual distances, D_3, D_2 D_1 ($D_3 > D_2 > D_1$). Length L is taken equal to the distance D_3 for all the speakers so that $L_3 = L_2 = L_1 = D_3$.

Sound waves from all the drive units are in phase on the axis of the system and, therefore, reinforce each other in the front. Off the axis, path lengths from different loudspeakers will be different, and will cause phase differences.

Differences will be such that there is reduced sound on the sides. Thus, the column would give maximum sound in the front.

3.7 BAFFLES AND ENCLOSURES

While column speakers provide excellent broad output in the front area and negligible output (outside the box and in the rear side) as the combined result of all the loudspeakers in the column. The same speakers, when used individually without an enclosure will behave differently.

There is air on both sides of the cone, the front as well as the back. Hence, the cone radiates sound from both sides. However, when the cone moves forward, there is compression of the air in the front, but rarefaction at the back. When the cone moves backwards, compression in the back and rarefaction in the front occurs. Thus, the sound waves produced in the rear air are of opposite phase (i.e., phase difference is 180° relative to the sound waves produced in the front). The sound waves from the rear leak or diffract round the sides and meet the sound waves in the front. If the path difference is small as compared to $\lambda/2$, the two waves will almost cancel each other, causing response of the speaker to drop off sharply. When the frequency of radiated waves is low, the wavelength is large and hence the path difference between rear and front sides is quite small in terms of λ. This will allow low frequency waves to arrive from rear to the front almost 180° out of phase, reducing the radiated energy substantially. To save the low frequencies from attenuation, it is necessary to increase the path difference by using a physical barrier, the baffle (at high audio frequencies, a baffle is not needed because compressions and rarefactions are very closely spaced and occur in quick succession, and hence there is no general cancellation).

A rigid flat material used to extend the edges of a loudspeaker cone is called a baffle. It is shown in Fig. 3.10. The term baffle is commonly used for a plane surface. A baffle increases the effective length of the acoustical transmission path between the front and the back of the radiator.

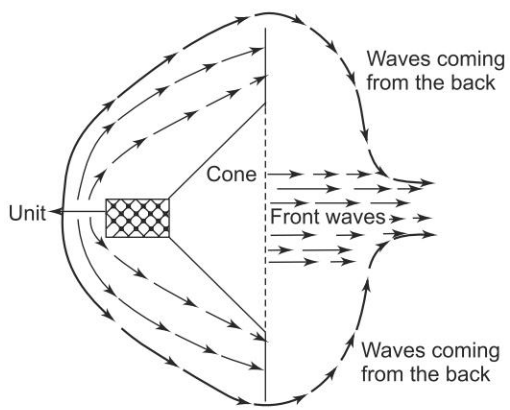


Fig. 3.10 *A basic baffle*

The baffles are made of a good sound insulating material. The material is soft to check rattle. Soft wood or cellotex is good. The loudspeaker must be securely fixed to the baffle. The baffle is also rigidly mounted to prevent rattle sound.

Types of Baffles

There are four types of baffles:

1. Finite baffle
2. Infinite baffle
3. Enclosure
4. Bass-reflex baffle

Finite Baffle The wooden cabinets as in radio receivers or TV receivers act as 'finite baffles'. Such baffles, being of finite size, are not very effective and do cause some loss of low audio frequency signals due to diffraction.

Infinite Baffle An ideal infinite baffle is the one which has infinite lateral dimensions. Such a baffle will not be feasible. However, if baffle dimensions are so large that the frequency at which the path length is $\lambda/2$, is far below the lowest frequency to be used, such a baffle can be called 'infinite baffle'. For example, let a speaker be fixed in a hole in the wall of a hall. As the dimensions of the wall are quite large, the wall will act as a natural barrier for the rear sound to come in front of the speaker. The wall will act as an infinite baffle.

A practical infinite baffle will have a huge dimension. Several designs have been developed to reduce the size. One such design is to bend the baffle back at the edges as shown in Fig. 3.11.

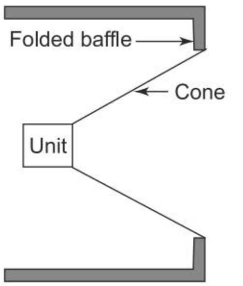


Fig. 3.11 *Folded baffle*

Enclosures When a loudspeaker is mounted in a closed box with an opening in the front (to enable the cone to transmit its vibrations to the air in front), the box serves the purpose of an infinite baffle because the waves from the back of the cone will not be able to come to the front side. Such a closed box is called an enclosure, and is shown in Fig. 3.12. Column speakers are mounted in such a closed box.

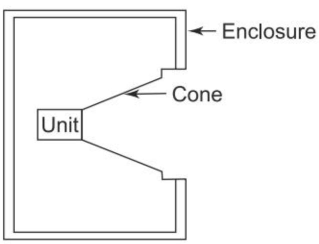


Fig. 3.12 *Basic enclosure*

The best material for enclosures is 18-mm thick plywood or particle board. Solid timber is not good because the timber grain can produce resonance at certain frequencies within the useful audio frequency range. The box must be air tight. All joints should be glued and all terminals or wires on the back panel should be properly sealed, The hole for the loudspeaker cone should be cut accurately and a good adhesive should be used around the edge of the speaker to ensure excellent sealing so that no air leaks out around it.

As there is no motion of the air from inside the box, there is no inertness (or inductive effect). But compliance (or capacitive effect) exists as pressure of the vibrating cone on the air inside the enclosure and causes pressure change in the cabinet. This capacitive effect increases the resonance frequency of the system as shown in Fig. 3.13.

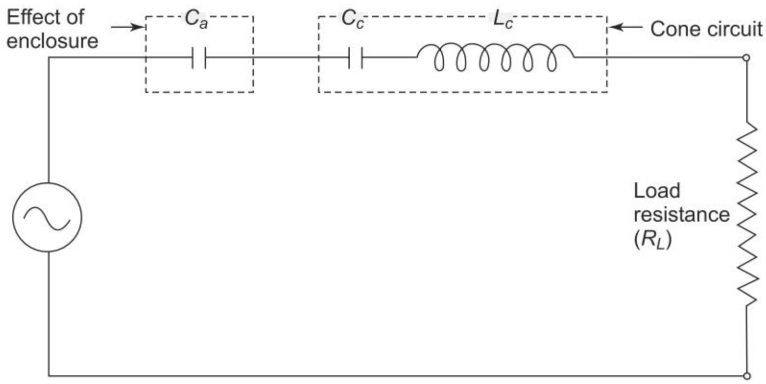


Fig. 3.13 *Effect of enclosure compliance on the resonant frequency*

C_a is the compliance of the enclosed air and is in series with the cone compliance, C_c and cone's mass (inductance) L_c. As C_a and C_a are in series, the total capacitance decreases, and hence the resonance frequency of the system increases. The box cannot be made too small, otherwise C_a would also be too small. However, a loudspeaker with very low self-resonant frequency can be mounted into a small box. About 4-litre volume of the enclosure per cm of the diameter of the cone is required to limit the increase of resonance. For a 15-cm size speaker, about 60 litres of enclosure volume would be needed.

Internal reflections in enclosures will cause the sound energy to reach the cone. These reflected waves will change the vibrations of the cone itself, causing irregularities from phase difference between the original vibrations of the cone and the vibrations due to the reflected waves. These become serious at high audio frequencies, giving a dip. At low audio frequencies they may give a boony peak in the bass response. These effects are reduced by padding the inner surfaces of the enclosure walls with sound absorbents like felt, glass, wool, sheet cork, cellulose, etc., as shown in Fig. 3.14. The absorbents eliminate the reflections. In some enclosures, the rear wall is broken up into small flat surfaces at random angles.

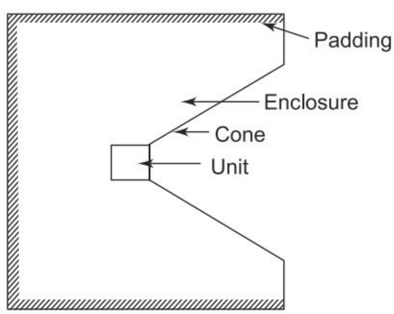


Fig. 3.14 *Enclosure with padding on the inner walls*

Bass Reflex Enclosures In this system, radiation from the back of the cone is used to strengthen the front radiation rather than weaken it. This design is shown in Fig. 3.15. Here, the port or cut is so designed that it permits flow of air from the rear to the front with additional path difference of $\lambda/2$ (or phase difference of 180°). Thus the phase difference between the back wave coming out from the port and the front wave caused directly by the cone is 180° + 180° = 360° (equivalent to zero), and therefore the two waves reinforce each other.

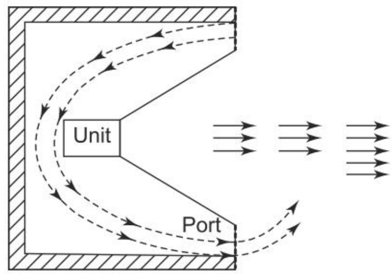


Fig. 3.15 *Bass reflex enclosure*

The port gives rise to inertness to the air and also has some friction (resistance); the acoustic circuit becomes as shown in Fig. 3.16. The circuit shows that the enclosure with the port works as a parallel resonant circuit, having compliance C_a, inertia, L_p and friction, R_p. The cone works as a series resonant circuit with compliance, C_c and inertia, L_c. The system is so designed that the two resonant circuits offset each other. The Q of the two circuits are also made equal by using sound absorbent material on the inside of the walls, or placing a grill cloth over the port. The grill increases friction and hence, resistance.

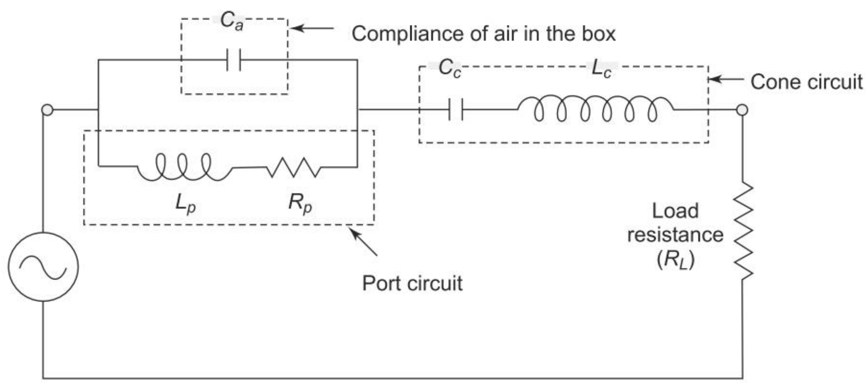


Fig. 3.16 *Equivalent acoustic circuit for bass reflex enclosure*

The enclosure's resonance frequency (with port) is expressed by Eq. 3.6.

$$f = 0.025(A/V^2)^{1/4} \tag{3.6}$$

where, f = resonance frequency in Hz
A = area of the port in sq. metres
V = volume of the enclosure in cubic metres

(Design of the port to give correct effect is quite complex and is done with the help of computers.)

Acoustic Labryinth For very low audio frequencies (16 Hz to 50 Hz), the ordinary bass-reflex enclosure will have large dimensions. To solve this difficulty, the enclosure is divided into parts by several baffles forming ducts. This lengthens the path of the wave, coming out of the port (in the enclosure) and causes additional phase difference of 180° at very low audio frequency so that the two waves in the front meet in the same phase. The ducted enclosure, shown in Fig. 3.17, is called '*Acoustic Labryinth*'.

For hi-fi, at least two speakers, called woofer and tweeter (described in Section 3.7), are mounted in the enclosure, which is generally a bass-reflex enclosure. The woofer portion is an acoustic labryinth. The enclosure contains folded absorbent material as shown in Fig. 3.18.

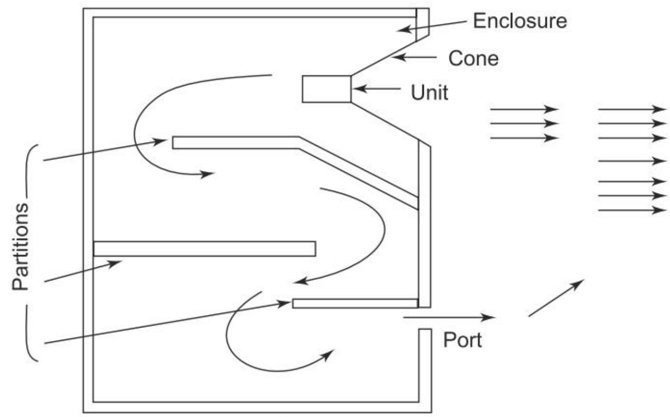


Fig. 3.17 *Acoustic labryinth*

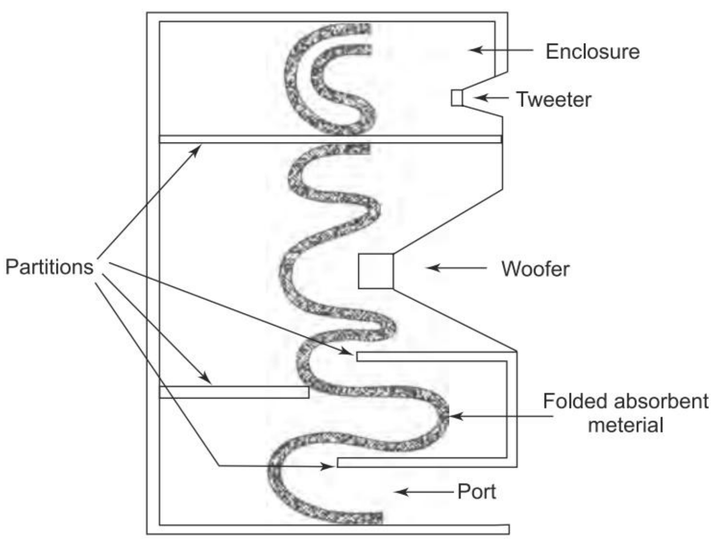


Fig. 3.18 *Hi-Fi Enclosure with port*

3.8 MULTI-WAY SPEAKER SYSTEM (WOOFERS AND TWEETERS)

A single loudspeaker cannot have flat response for the whole audio frequency range from 16 Hz to 20000 Hz, and not even for the practical hi-fi range of 40 Hz to 15000 Hz. Low frequencies are weakened by the back sound waves of reverse phase in an open-speaker. In a closed box (enclosure), the compliance (capacitive effect) of the entrapped air comes in series with the compliance of the cone system and hence increases the resonant frequency of the loudspeaker. The loss at high frequencies is due to mass (or inductive effect) of the diaphragm (including cone). Thus, a single speaker cannot produce both, the good solid bass and the smooth crisp treble. The best of them can only produce just acceptable bass and treble which will not satisfy the hi-fi requirements.

To solve the problem, the audio frequency spectrum is divided into at least 2 and preferably 3 parts. Separate speakers are designed for each part, so that each speaker has to cover only a small range of frequency. The speakers which cover low frequencies from 16 Hz to 1000 Hz are called ***woofers.*** The speakers which cover higher audio frequencies are called ***Tweeters***. Many a time, a third speaker, called ***Squawker*** is used for mid-frequency range from 500–5000 Hz, and in that case woofer works up to 500 Hz and a tweeter from 5000 Hz onwards.

Woofers and tweeters can either be separate speakers mounted in a common enclosure, or there can be a dual cone loudspeaker. The two systems are shown in Fig. 3.19 (a) and 3.19 (b) respectively.

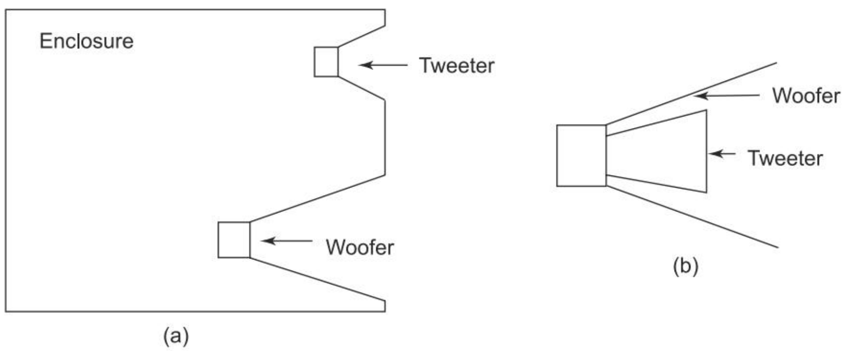


Fig. 3.19 (a) *Woofer and Tweeter in a common enclosure* (b) *Dual-cone loudspeaker*

3.9 CROSSOVER NETWORKS

When a multiway loudspeaker system is used to get flat frequency response for the entire range of audio frequencies, it is essential to have a crossover network to divide the incoming signal into separate frequency ranges for each speaker. In the absence of crossover networks, the speakers will suffer overheating and the output will be distorted when full power at frequencies outside their range is fed to them. The overall efficiency will be much reduced in the absence of crossover networks.

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

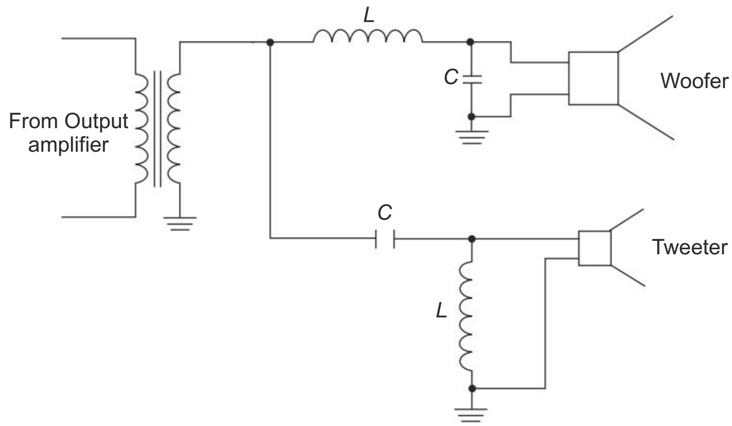


Fig. 3.20 *Basic crossover network*

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

The response curve of a typical crossover network (of Fig. 3.20) is shown in Fig. 3.21**.** It gives an attenuation of 12 dB per octave.

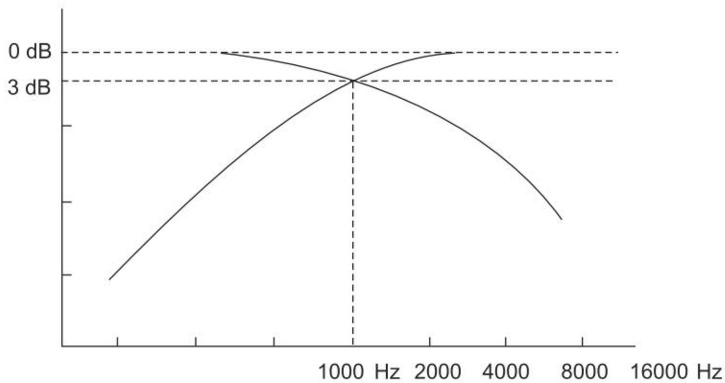


Fig. 3.21 *Response curve of basic crossover network of Fig. 3.20*

Equations 3.7 and 3.8 give the values of L and C

$$L = \frac{R_L \sqrt{2}}{2\pi f_c} \tag{3.7}$$

$$C = \frac{1}{2\pi f_c R_L \sqrt{2}} \tag{3.8}$$

where, R_L is the impedance of a loudspeaker in ohms and f_c is the crossover frequency in Hz, L is the inductance and C, the capacitance of LC circuits.

A commercial three-way divider network is shown in Fig. 3.22. In this circuit, the capacitor C_t of 1 μF in series with the tweeter prevents low and mid-frequencies from reaching the tweeter. Similarly, the inductance L_w of 5 mH in series with the woofer prevents high frequencies from reaching the woofer. Inductances L_{S-1} and L_{S-2} of 0.5 mH and 5 mH, respectively in the squawker circuit allow only mid-frequencies and prevent too low and too high frequencies from reaching the squawker. A typical divider curve for a three-way network of Fig. 3.22 is shown in Fig.3.23. A single element in filtering gives attenuation of 6 dB per octave and double elements give 12 dB per octave.

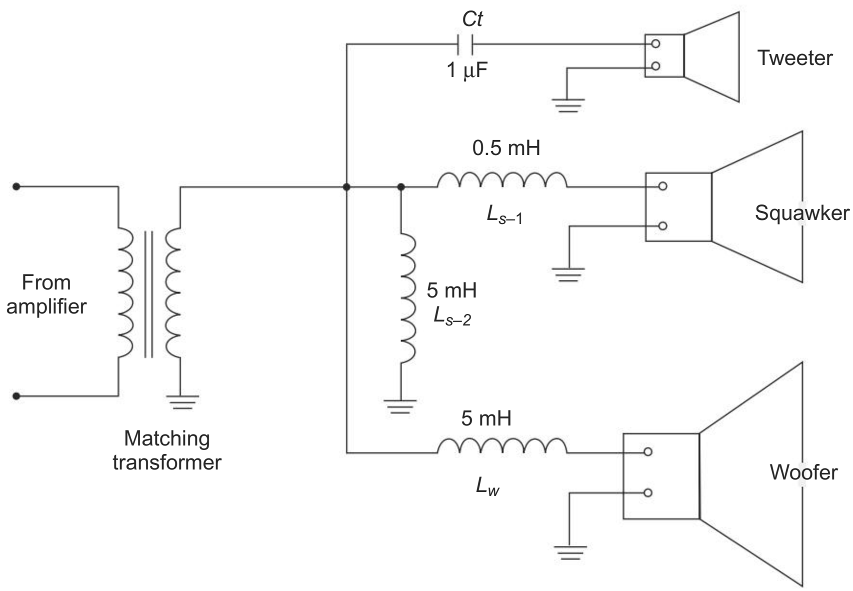


Fig. 3.22 *Three-way speaker system*

Considerations for Designing Cross-over Networks

1. The crossover frequency for a woofer–tweeter circuit is where the woofer output curve crosses the tweeter output curve. This is normally 1000 Hz. Hence, a woofer gives an output between 16–1000 Hz and a tweeter from 1000–20000 Hz for a 2 speaker system.

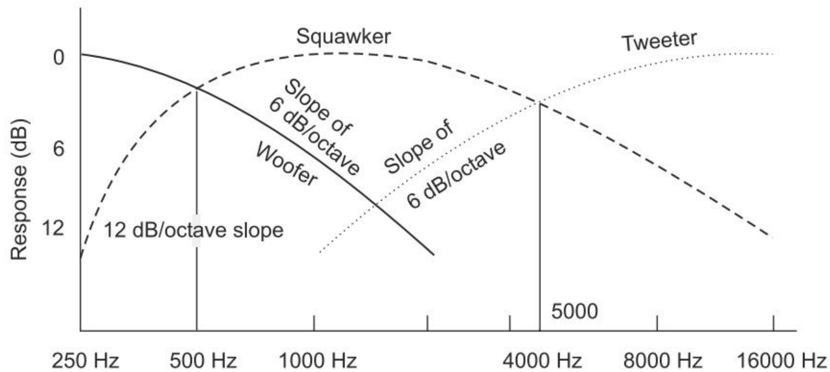


Fig. 3.23 *Response curves for a three-way speaker system*

2. Attenuation beyond the cutoff frequency for woofer and before the cutoff frequency for a tweeter should preferably be 12 dB per octave, although 6 dB per octave is acceptable for economy models.
3. For a three-way speaker system, frequency coverage to the crossover point is as given below:
   - *Woofer* 16 Hz to 500 Hz
   - *Squawker* 500 Hz to 5000 Hz
   - *Tweeter* 5000 Hz to 20000 Hz
4. Inductors and capacitances should be calculated correctly.
5. Electrolytic capacitors cannot be used as there is no polarisation dc current.

Example 3.3 *Design a crossover network to give 12 dB per octave woofer for a critical frequency of 1 kHz. Loudspeaker resistance is 8 Ω.*

Solution

$$C = \frac{1}{2\pi\ f_c\ R_L \sqrt{2}}$$

$$L = \frac{R_L \sqrt{2}}{2\pi\ f_c}$$

Substituting the given values,

$$C = \frac{1}{2\pi \times 1000 \times 8 \times 1.4} \cdot \mu\text{F}$$

$$= \frac{625}{44}\ \mu\text{F} = 14.2\ \mu\text{F}$$

$$L = \frac{8 \times 1.4}{2\pi \times 1000}$$

$$= \frac{19.6}{11000}\ \text{H} = \frac{19.6}{11}\ \text{mH}$$

$$= 1.8\ \text{mH}$$

A column speaker system, described in Section 3.6 may use multi-way speakers in place of individual speakers for hi-fi applications.

3.10 CONSEQUENCE OF MISMATCH BETWEEN AMPLIFIER OUTPUT AND LOUDSPEAKER IMPEDANCE

The maximum power transfer theorem shows that power transferred from the source to the load will be maximum when the internal impedance of the source is equal to the impedance of the load. When there is mismatch, the first consequence is that power developed in the load is reduced.

Frequency distortion also occurs under the condition of mismatch. As the load is generally inductive, it might have higher impedance at high frequencies than at low frequencies. So, if the source impedance matches at one frequency, there will be mismatch at the higher frequencies as well as at the lower frequencies. For solving this difficulty, it is necessary that the reactive component of the load is neutralised by the opposite reactance in the source, i.e., if the reactance of the load is inductive, the reactance of the source should be capacitive and vice versa. For distortionless output, the matching should satisfy the condition of Eq. 3.9.

$$R_s + jX = R_L - jX \tag{3.9}$$

where, R_s = resistance of the source

jX = reactance of the source as well as of the load

R_L = resistance of the load

The matching of source impedance with the load impedance is done with the help of a matching transformer. Relationship between the turns ratio of the secondary turns (n_2) to the primary turns (n_1) and the ratio of impedance in the secondary circuit (Z_2) to impedance in the primary circuit (Z_1) is given by Eq. 3.10.

$$\frac{n_2}{n_1} = \left(\frac{Z_2}{Z_1}\right)^{1/2} \tag{3.10}$$

Example 3.4 *If the initial resistance of a power amplifier is 2500 Ω and load resistance of the loudspeaker is 16 Ω, calculate the turns ratio.*

Solution

n_2/n_1, = $(16/2500)^{1/2}$ = 4/50 = 4:50

Thus, it will be a step-down transformer.

SUMMARY

☞ A **loudspeaker** converts electrical signals of audio frequency into sound waves.

A good loudspeaker should be efficient, should not produce noise signals, should have fairly flat frequency response over the audio range, should have low distortion, desired directivity, adequate power handling capacity and correct matching impedance. Installations using long leads use a 100-volt line system.

☞ Loudspeakers are of 2 types: (1) Moving-coil cone-type and (2) Horn-type. In a **moving-coil cone-type speaker,** a voice coil is placed in a magnetic field. When an audio current passes through the coil, there is an interaction between the magnetic field and the coil, resulting in motion of the coil. The coil is attached to a paper cone which vibrates producing sound waves in air. This type of speaker is known as **direct radiating speaker** because the whole paper cone acts as a diaphragm and causes sound-pressure variations direct in the listeners' area. Its efficiency is low (about 5%), signal-to-noise ratio is 30 dB, frequency response is 200 Hz to 5000 Hz (woofers and tweeters extend the range from 40 Hz to 10000 Hz) and the distortion is about 10%. Basically, it is omni-directional, but high frequencies are concentrated in a narrow cone about the axis of the radiator. Column of loudspeakers is designed to give higher coverage to the audience in the front. Baffles and enclosures modify the directivity of individual loudspeakers and most of the power is in the front. Its impedance is from 2 Ω to 32 Ω (4, 8 and 16 Ω speakers are more common). The power handling capacity is not more than 25 watt for large-size speakers. To increase the power-handling capacity, an electromagnet is used instead of a permanent magnet. Such speakers, known as **electrodynamic speakers,** can handle power up to a few hundred watts.

☞ In a **horn-type speaker,** acoustic power is first delivered to the air trapped in a fixed non-vibrating tapered horn and from there it goes to the listeners' area. Thus, it radiates power not directly from the diaphragm, but indirectly through the horn. Hence, the horn-type speaker is known as **indirect radiating speaker.** Conversion of audio signals into sound waves is done in the same way as in the cone-type speaker by interaction between current of the voice coil and the magnetic flux of the magnet. The horn is a tapered enclosure whose diameter increases from a small value at the diaphragm side (called **throat)** to a large value at the radiating side (called **mouth).** An air chamber is trapped between the throat and diaphragm.

The horn acts as an acoustic transformer allowing better impedance match between the free air and the voice coil/ diaphragm assembly. Its efficiency is, therefore, higher than a cone-type speaker and is 30–50%. Signal-to-noise ratio is 40 dB, frequency response is 30 to 10000 Hz (A tweeter horn extends it up to 20000 Hz), distortion is less than 5%, directivity for half-power points is about 90° for low frequencies. High frequencies concentrate in a narrow cone in the front. Its impedance is typically 16 Ω. It can handle power up to 100 watts.

☞ **Baffles and Enclosures** Sound waves in the back are out-of-phase by 180° with respect to sound waves in the front for individual speakers. Hence, if the sound waves from the back are not prevented and meet the waves in the front, the two will cancel each other, causing intensity of sound to drop off sharply. This phenomenon will be more pronounced at low frequencies. To save the low frequencies from attenuation, it is necessary to increase the path difference by using a physical barrier, called baffle. Baffles are of 4 types: (1) finite baffle as in a TV or radio receiver, (2) infinite baffle as in a loudspeaker fixed in a hole in a wall, (3) enclosure (speaker mounted in a closed box which acts like an infinite baffle) and (4) bass reflex enclosure (it contains a port through which back sound passes on to the front and cause a path delay of $\lambda/2$).

☞ **Woofers, Squawkers and Tweeters** A single loudspeaker cannot have a flat response for the whole audio frequency range, due to mass and compliance effect present in all the vibrating systems. Separate speakers are designed to cover a small band of audio frequencies. The speakers which cover low frequencies from 16 Hz to 1000 Hz are called 'woofers'. The speakers which cover high frequencies are called 'tweeters'. Sometimes, a third speaker, called a 'squawker' is used to cover frequencies from 500 to 5000 Hz. A column speaker may contain multi-way speakers in place of individual speakers for hi-fi applications.

When a multi-way speaker system is used, it is essential to have a **crossover network** to divide the incoming signal into separate frequency ranges for each speaker. These networks use the properties of inductors and capacitors.

For transfer of maximum power from amplifier to the loudspeaker, input impedance of the loudspeaker must be matched with the output impedance of the amplifier.

Review Questions

1. What are the characteristics of a good loudspeaker? Why is efficiency of an indirect radiating loudspeaker is higher than the direct radiating type?
2. With the help of a neat sketch, explain the principle of working of a moving-coil loudspeaker. Why is it called 'direct radiating type speaker'?
3. Draw a neat sketch of electrodynamic-type loudspeaker and describe its construction. Explain its functioning. What are its advantages and disadvantages relative to permanent-magnet type speaker?
4. With the help of a neat sketch, explain the principle of working of a horn-type loudspeaker. What are the advantages and disadvantages of a horn-type speaker relative to the cone-type? Why is a horn-type speaker called 'indirect radiating type'?
5. Compare the characteristics of cone-type and horn-type speakers, and indicate their applications.
6. What is a column-speaker? How is it more efficient than a system of equal number of individual speakers mounted separately without an enclosure?
7. What do you understand by baffle? Why is it needed? Name different types of baffles.
8. Draw a neat sketch of a bass-reflex baffle. Explain how a bass-reflex baffle can give higher intensity of sound than an ordinary finite or infinite baffle.
9. Explain the concept of infinite baffle. Explain how an enclosure acts as an infinite baffle. Why are inner surfaces of an enclosure padded?
10. What do you understand by woofer, squawker and tweeter speakers? Explain the necessity of crossover networks.
11. Draw a basic crossover network circuit and explain its objective with the help of characteristic curves. Why cannot electrolytic capacitors be used in crossover networks?
12. Explain the consequence of mismatch between amplifier output impedance and loudspeaker impedance. How is matching achieved?

Short-Answer Questions

1. Why is a loudspeaker called a reverse transducer?
2. Why is the output of a loudspeaker not flat for the whole range of audio frequencies?
3. What is the purpose of a 100 V line for a loudspeaker?
4. Why is PMMC type loudspeaker called direct radiating type, and horn-type, indirect radiating type?
5. Why is the efficiency of a horn-type speaker higher as compared to PMMC type?
6. Why is a loudspeaker column more suitable for a large audience?
7. What is the necessity of a baffle for a loudspeaker?
8. What is the advantage of a bass-reflex enclosure?
9. What is the difference between a woofer and a tweeter?
10. Why are electrolytic capacitors not used in a multiway speaker system?

Multiple-Choice Questions

1. On what principle does a loudspeaker work?
   (a) Motor (b) Generator
   (c) Amplifier (d) Oscillator
2. What is the efficiency of a direct-radiating type speaker?
   (a) 50% (b) 35%
   (c) 20% (d) 5%
3. Impedance of a loudspeaker is of the order of
   (a) milliohms (b) ohms
   (c) kilo ohms (d) megaohms
4. Which type of loudspeaker can handle the lowest maximum audio power?
   (a) PMMC type
   (b) Electrodynamic type
   (c) Horn-type
   (d) Multiway speaker system
5. For what purpose is a column of loudspeakers used for?
   (a) High fidelity system
   (b) Public address system for large audience
   (c) Domestic television receiver of big size
   (d) To address field units in the army
6. Multiway system of loudspeaker is used for
   (a) PA system
   (b) Conference
   (c) High-fidelity programme
   (d) Group discussion
7. A rigid flat material used to extend the edges of a loudspeaker cone is called
   (a) woofer (b) tweeter
   (c) baffle (d) squawker

8. Bass-reflex type enclosures increase the strength of
   (a) low audio frequencies
   (b) high audio frequencies
   (c) complete audio frequency range
   (d) reflected waves
9. Which type of baffle uses an extra hole in the enclosure?
   (a) Infinite baffle
   (b) Finite baffle
   (c) Bass-reflex baffle
   (d) Useless baffle
10. Which frequency response is extended by a woofer?
   (a) High frequency
   (b) Mid frequency
   (c) All frequencies
   (d) Low frequency
11. Which frequency response is extended by a tweeter?
   (a) High frequency
   (b) Mid frequency
   (c) All frequencies
   (d) Low frequency
12. Crossover frequency is that frequency
   (a) beyond which the alternative loudspeaker is automatically selected
   (b) at which cross-talk interference becomes effective
   (c) which is most common in use
   (d) which is never in use
13. A capacitor most easily passes
   (a) low frequency
   (b) mid frequency
   (c) high frequency
   (d) all frequencies
14. An inductor most easily passes
   (a) all frequencies
   (b) high frequencies
   (c) mid frequencies
   (d) low frequencies
15. How many speakers are contained in a column of loudspeakers?
   (a) One only
   (b) Two only
   (c) Three or more
   (d) one set of multiway speakers

Numerical Problems

1. Sound intensity at 1 metre from a loudspeaker is 400 mW/m^2. The audio power fed to the loudspeaker is 10 W. Calculate the efficiency of the loudspeaker. What type of speaker it could be?

   (Hint: Intensity = $P_o/4\pi r^2$)
2. An amplifier of 100 watts gives 100 volt at open loudspeaker terminals. Calculate the impedance of the amplifier.
3. Find the resistance of the amplifier if 250 W output power gives 100 V on the loudspeaker line. What would be the loss in line if the line resistance is 1 Ω.
   (Hint: Current in 100 V line system would be 2.5 A. Hence loss of power in cable $I^2R = 6.25 \times 1$ (= 6.25 W).
4. Calculate the value of L and C for tweeter to have a crossover

frequency of 1000 Hz. The impedance of tweeter is 16 ohms.

5. A loudspeaker of 4 Ω is to be connected to an amplifier of 100 Ω impedance. Calculate turns ratio (secondary to primary) of the transformer.

6. Calculate size of the horn's mouth for audio frequency of 400 Hz.

Answers

Short-Answer Questions

1. In any audio system, a microphone is the input transducer to convert sound-pressure variations into electric signal. A loudspeaker in that system converts electric variations' into sound-pressure variation at the output of the system. Hence, it is called reverse transducer as compared to a microphone.

2. Mass of the diaphragm-cone assembly works as inductance and so it attenuates high frequencies. Low frequencies are attenuated due to compliance in series.

3. When the connecting cable of a loudspeaker is lengthy, the transmission of power to the loudspeaker is more efficient if voltage is fed to the line (cable). Cable impedance is quite low (about 1 Ω) and so the voltage drop in the line would be small. It will make the transmission of power to the loudspeaker efficient. It is similar to the transmission of electric power to long distance at high voltage (say 11000 V instead of 220 V).

4. In a PMMC type of loudspeaker, the whole paper cone acts a diaphragm and causes pressure variations direct in the listener's area. Hence, it is called direct-radiating type. In contrast, a horn-type speaker delivers audio power first to the air trapped inside the inner chamber from where it goes to a non-vibrating tapped horn and then to the audience. Thus, vibrations are indirectly radiated to the outside air. This is the reason why a horn-type speaker is called indirect radiating type.

5. Due to good matching between the mechanical impedance of the horn and the acoustic impedance of the air outside.

6. Directivity of a loud speakers column is such that almost the whole sound energy is directed to the front to cover the wide listener's area. There is only a little feedback to the microphones in the rear. Also, the column is a closed box and hence sound of opposite phase existing in the rear of the cone cannot come out.

7. There is air on both sides of the cone. When compression occurs in the front, there is rarefaction behind. Thus, two variations are in opposite phase. When the air inside transmits vibrations to the air outside, the two vibrations will cancel each other. If we can intro-

duce an extra phase difference of 180º to the air coming from inside, the phase change would become 360º (or 0º) and so the air coming from inside will be in the same phase as the air outside and the two vibrations will reinforce each other to produce vibrations of high intensity to the audience.

8. The air in the rear is made to come from outside through a hole in the enclosure. The hole is so designed that the path length $\lambda/2$, gives an extra 180º phase difference, and thus total phase difference become zero. Hence, the vibrations outside and the vibrations coming from inside are in the same phase and reinforce each other.

9. A heavy speaker is good for low audio frequencies and is called woofer. A small-sized speaker is good for high audio frequencies, and is called tweeter. Both encased in a single case result in uniform frequency response from 40 Hz to 16000 Hz.

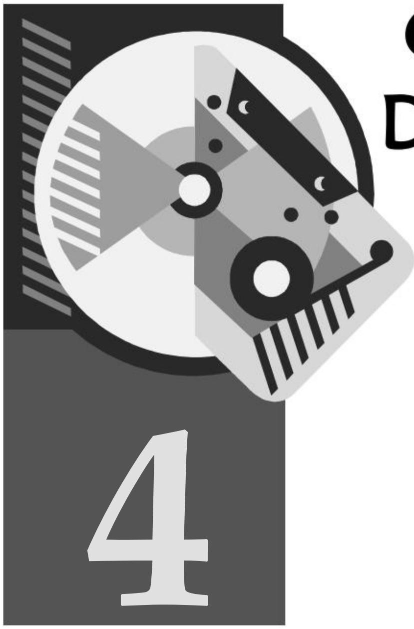
10. An electrolytic capacitor can be used in such systems only that have dc voltage in addition to ac voltage. Filters in the multiway speaker systems handle only ac, and therefore they cannot use electrolytic capacitors. When a design requires a capacitor of high value, a ceramic capacitor can be used instead of an electrolytic capacitor.

Multiple-Choice Questions

(1) (a)	(2) (d)	(3) (b)
(4) (a)	(5) (b)	(6) (c)
(7) (c)	(8) (a)	(9) (c)
(10) (d)	(11) (a)	(12) (a)
(13) (c)	(14) (d)	(15) (c)

Numerical Questions

1. (50%, horn-type)
2. (100 Ω)
3. (40 Ω, 6.25 W)
4. (3.6 mH, 7.1 μF)
5. (1:5)
6. (0.4 m)



4 Gramophone Disc Recording

4.1 PRINCIPLE OF DISC RECORDING

When an audio signal passes through a coil wound on an armature and placed in a magnetic field, the armature moves to and fro (motor action) in accordance with the audio signal. A cutting needle (called cutter stylus) is fixed with the armature, and it also vibrates. The cutter moves from the edge to the centre on a rotating disc made of wax or lacquer, and hence cuts spiral grooves on the surface of the disc, which change their positions laterally in accordance with the audio signal. In the absence of sound, the grooves shall be uniformly spaced and move spirally ending at the centre, as shown in Fig. 4.1 (a). The audio signal makes the grooves shift laterally, and thus, the sound is recorded on the disc as shown in Fig. 4.1 (b).

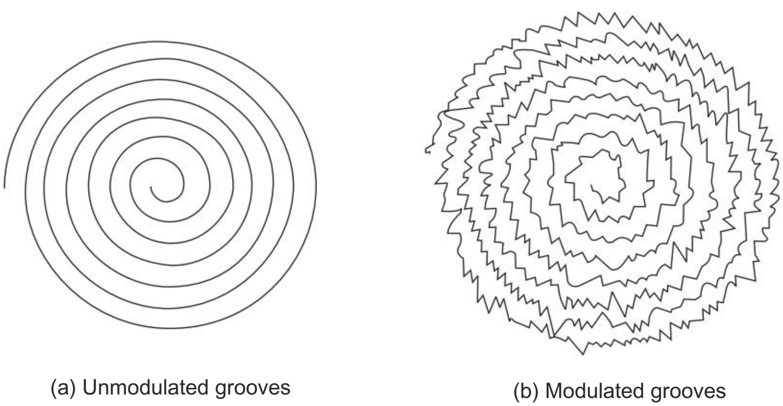


Fig. 4.1 (a) *Unmodulated grooves* (b) *Modulated grooves*

A disc-recording unit is shown in Fig. 4.2. It consists of a powerful electromagnet giving a strong magnetic field between north and south poles when a dc current is passed through the field coil. An armature of soft iron is placed between the two pole pieces. A coil is wound on the armature, and a cutter stylus is attached to the lower end of the armature. When an audio current (produced by microphone and amplified by audio amplifiers) flows in the armature coil, it produces a varying magnetic field which is superimposed on the steady field of the electromagnet. This causes the armature to vibrate in the horizontal plane (i.e., lateral vibrations). These vibrations are in accordance with the variation of the audio current, and are transferred to the cutting needle which is made of diamond.

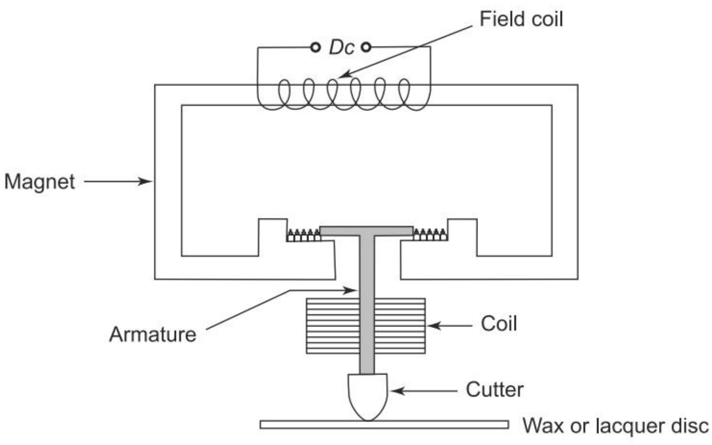


Fig. 4.2 *Disc-recording unit*

4.2 PRINCIPLE OF DISC REPRODUCTION

For reproduction, a device called a cartridge is made to change the vibrations of the playback needle (or stylus) into electrical signals which can be amplified and then converted into sound by a loudspeaker. The working principle of a cartridge is similar to that of a microphone. Like microphones, cartridges can be of many types. A magnetic cartridge is shown in Fig. 4.3.

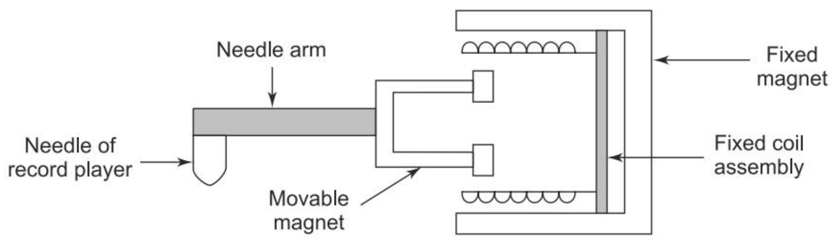


Fig. 4.3 *Magnetic cartridge*

When the needle of the record player tracks the recorded grooves on the disc, it vibrates in accordance with lateral variations of grooves. These vibrations move a magnet placed in the magnetic field of a fixed permanent magnet. Due to vibrations of the magnet, the flux density through a coil, placed in that magnetic field, changes and hence an emf is induced in the coil depending on the rate of change of flux. The electrical signal is amplified by transistor amplifiers and is fed to a loudspeaker to get the original sound.

4.3 BLOCK DIAGRAM OF A DISC-RECORDING SYSTEM

Audio signals in the output of a microphone are processed and amplified to drive a disc-recording unit to cut grooves in the lacquer compound of the disc. Figure 4.4 depicts the block diagram of a disc-recording system.

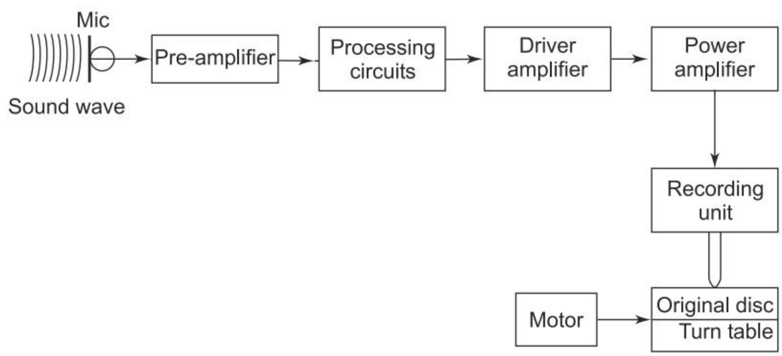


Fig. 4.4 *Block diagram of disc recording*

Function of Each Block

Microphone A high-grade microphone like condenser microphone or a specially designed cardioid microphone is used to convert the sound waves into electrical variations called audio signals.

Pre-amplifier It amplifies the weak output of the microphone. It is a low noise, high-gain amplifier to get high signal-to-noise ratio.

Processing Circuits The amplified signals are processed to de-emphasise low frequency signals and emphasise high-frequency signals to eliminate chance of over-modulation for low frequencies and improve signal-to-noise ratio for high frequencies.

Driver Amplifier The processed signals are further amplified for voltage amplification, so that high output voltage is available to drive the next stage to give adequate power.

Power Amplifier It provides power amplification to the signal. Its internal resistance is low and the output circuit has a matching transformer to match the source and load impedance with each other.

Recording Unit The output audio power from the power amplifier is used to drive the recording unit to convert audio signals into mechanical vibrations.

Cutter Stylus It cuts spiral grooves on the disc, producing a wavy pattern in the horizontal plane when audio signals cause the stylus to vibrate laterally. The stylus moves radially on a rotating disc, and hence, cuts spiral grooves from edge of the disc to its centre.

Disc It consists of a wax or lacquer coating on a metal disc. Grooves are engraved on the coating by the vibrating stylus.

Turn-table The turn-table carries the disc over itself. When the turn-table rotates, the disc placed on it also rotates.

Motor It is used to drive or rotate the disc at a constant speed. A synchronous motor is used as the speed of a synchronous motor remains locked to the frequency of ac mains which is very stable. The steady speed is necessary to reduce wow and flutter noise during playback. The connection between the motor and the turn-table may be by means of gears, friction drives and belts.

Different types of discs recording units are described in Sec. 4.4.

4.4 BLOCK DIAGRAM OF DISC-REPRODUCTION SYSTEM

A disc-reproducing system makes use of a pick-up cartridge which like a microphone, converts vibrations of a pick-up needle into electrical signals which are processed and amplified and finally fed to the loudspeaker to get converted into sound. The block diagram of a disc-reproducing system is shown in Fig. 4.5.

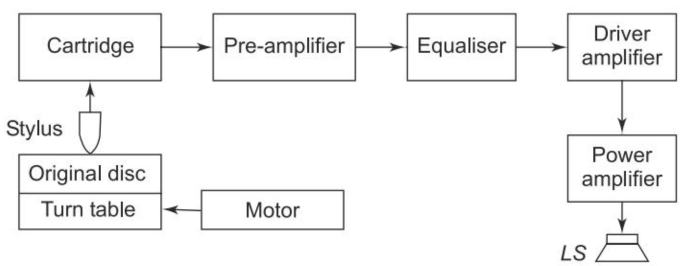


Fig. 4.5 *Block diagram of disc-reproduction function of each block*

Function of Each Block

Motor It is a synchronous motor and rotates the turn-table with steady speed. Its body vibrations are mechanically filtered to ensure elimination of rumble noise. Speed steadiness reduces wow and flutter noise.

Turn-table It carries the playback disc over itself, and hence rotates it.

Disc Playback disc, also called *record*, consists of a mixture of shellac, rags, carbon black, etc. It has the recorded grooves on its surface. When playback needle moves radially over the rotating disc, it tracks the spiral grooves.

Stylus It is the playback needle whose function is to track the grooves of the record-disc and transmit the resultant vibrations to the cartridge transducer.

Cartridge It converts vibrations received from the stylus into electrical signals of the same frequency (called audio signals). The magnetic cartridge gives signal of the order of 10 mV for grooves of 25 μm radius.

Pre-amplifier It amplifies the weak output of the cartridge.

Equaliser It equalises the signal by emphasising low notes and de-emphasising high notes. Volume control, bass and treble controls are also generally located after the pre-amplifier.

Driver Amplifier It further amplifies the signal to give sufficient input to the power amplifier to drive it for optimum power.

Power Amplifier It gives power amplification to the signal. It also has a matching transformer to match source-impedance with the impedance of the load (i.e., loudspeaker).

Loudspeaker It converts audio power into sound.

4.5 PRODUCTION OF DISC RECORDS ON MASS SCALE

The disc on which sound is originally recorded is called 'original disc'. It is necessary to make copies of it for public consumption. The records (or playback discs) are prepared in three steps as described below:

1. The original disc is made of metal or glass coated with cellulose nitrate lacquer compound. The chisel-shaped cutting stylus tip, made of diamond, removes a continuous thread of cellulose as it cuts the groove on the rotating disc. The cutter moves slowly from edge to the centre of the rotating disc and this results in spiral grooves being recorded on the lacquer. To give very smooth grooves, the stylus is heated by passing low voltage dc through a small coil of high-resistance wire around the shaft of the stylus just above the tip.
2. The wax or the lacquer compound surface of the original disc is made conducting by coating it with graphite or some conducting material. Then, by electrolysis, the disc is heavily electroplated with copper. The

copper shell (called electrotype) is removed from the wax or lacquer by heating. The copper shell has ridges in place of corresponding grooves of the original disc, and hence, it is called *negative* or *Master Negative*. The negative disc of copper is nickel plated and is backed by steel plate to give it support and hence, strength.

3. When the negative copper disc is pressed on to a suitable recording material, a positive disc, identical with the original disc, will be obtained. This process is called stamping. The positive disc material consists of a mixture of shellac, shredded rags, rotten stone, sulphate of barium, carbon black, etc. The record disc is made quite hard and smooth by pressing the disc material between the heated rollers and then cooled. Record discs can be made by the above process on a mass scale from a single copper *Master Negative*.

4.6 TYPES OF GROOVES

Grooves are of three types:

1. Hill and dale type grooves, shown in Fig. 4.6(a)
2. Lateral grooves, shown in Fig. 4.6(b)
3. Stereophonic grooves or 45° angle grooves, shown in Fig. 4.6(c)

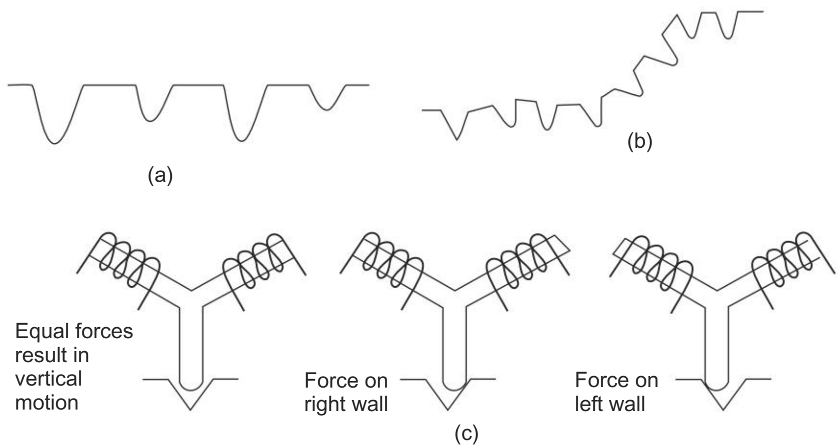


Fig. 4.6 *Different types of grooves*
(a) Hill-and-dale type, depth varies
(b) Lateral grooves, depth remains constant, lateral variation
(c) Stereophonic grooves (45° angle recording)

Hill-and-dale Type Grooves In the early days of the development of disc recording, Edison used vertical motion of the cutter stylus to make grooves in the wax. The stylus moved up and down in accordance with the audio current in the recording coil. This type of groove was called 'Hill and Dale'. The top of this groove was of 150 µm (or 6 mil) width. In this type, only depth of the groove changed the gap between grooves did not change.

Lateral Grooves The hill and dale method produced high noise and was not satisfactory. Also, hills in the groove had to lift the playback arm and cartridge. The arm and the cartridge were relatively heavy in earlier days, and therefore the hill-and-dale method resulted in excessive wear. Berliner devised a new method of recording in which the stylus moved laterally on a flat disc. These lateral grooves were of 5 μm diameter, or even less at the top. In this type, the depth of the groove remained constant, the width between grooves changed.

Stereophonic Grooves In stereophonic sound, two independent channels are to be recorded simultaneously. One of the main requirements is that there should be no interaction or coupling between the two channels. Although complete isolation is not possible in practice, two motions acting at 90° to each other can be non-interactive. Bluemlein devised a system by which two channels are recorded on two walls of the groove inclined at 45° to the horizontal (shown in Fig. 4.6c). When current in the two channels is equal, the resultant motion of the stylus is vertical. When unequal, the resultant motion is on the wall. As the motion of stylus arms is at 45° with the horizontal, the grooves so produced are called 45° grooves or stereophonic grooves. The size of the groove at the top is 6 µm (or 0.25 mil) in this type.

Coarse-grooves and Micro-grooves Old-fashioned discs could give a playing lime of about 6 minutes only, using 30 cm (12") size disc with 78 rpm speed. This much play-time was too short and could not be increased as the groove width was as large as 150 µm (6 mils). The groove of this size is called a coarse-groove. To meet the demand of longer play-time, two factors were combined to increase the duration of playback. These are (i). Decreasing the speed of rotation, and (ii). Reduction in the size of the groove.

The speed was decreased from 78 rpm to 45 rpm for extended play (EP) records and to $33^1/3$ rpm (often mentioned as 33 rpm) for long play (LP) records. Further reduction in speed was not desirable due to difficulty of resolution between cycles which shall become rather crowded on the disc.

The other factor was to reduce the size of the groove. Ultimately, the size of the groove width on top was reduced from 150 µm (6 mil) to 50 µm (2 mil) or less, and for stereophony, as low as 6 µm (or about 0.25 mil). The grooves made of such small size are called *fine-grooves* or *micro-grooves* Dimensions of a standard micro-groove are shown in Fig. 4.7.

Number of Grooves A disc of 2R diameter will have a maximum number of grooves equal to "*R*/width of a groove". (It presumes that there is no vacant space between adjacent grooves, i.e., the grooves on adjacent spirals just touch each other.)

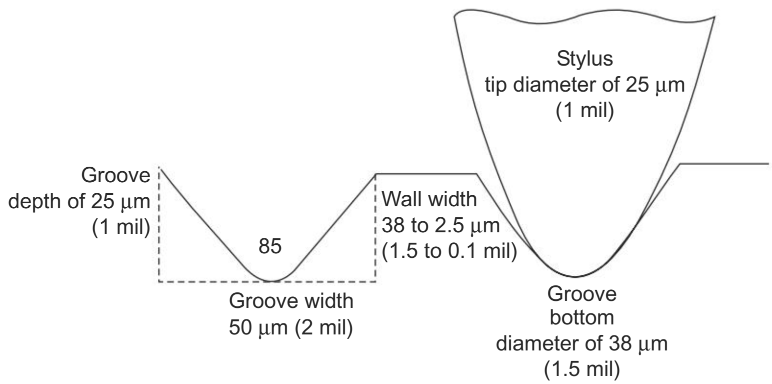


Fig. 4.7 *Micro-groove dimension*

Example 4.1 *If a groove width is 25 μm, calculate the number of maximum grooves on a 25 cm diameter disc. Also, calculate the number of grooves per cm of width (diameter of the disc).*

Solution

Max.no.of grooves

$$= \frac{\text{Radius of disc}}{\text{width of grooves}}$$

$$= \frac{25 \times 10^{-2}}{2 \times 25 \times 10^{-6}} = 5000$$

No.of grooves per cm of width

$$= \frac{\text{max No. of grooves}}{\text{disc diameter}}$$

$$= \frac{5000}{25} = 200$$

[Actual grooves would be less (about 100 grooves/cm) depending upon the space between adjacent grooves.]

4.7 CONSTRUCTION OF CUTTER STYLUS

The cutter stylus (shown in Fig. 4.8) vibrates and hence, makes grooves on the disc. These vibrations are recorded on the disc as lateral variations of the grooves.

The cutter tip is chisel-shaped or V-shaped and made of diamond. Sapphire (*neelam*) or a synthetic sapphire (fused aluminium oxide) is also used to make cutters, but diamond, although costly, is better.

The cutter is mounted on a long screw of suitable pitch. This makes the cutter move slowly from the edge to the centre along

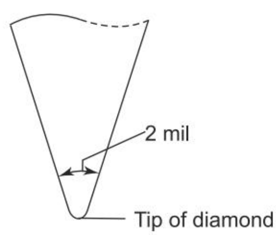


Fig. 4.8 *Cutting stylus for monophonic recording*

the radius of the rotating blank disc. Radial motion of the cutter combined with rotational motion of the disc gives spiral grooves on the disc. When an audio current is fed into the armature coil, the stylus vibrates laterally as it cuts the groove, producing a wavy pattern instead of uniformly spaced circular pattern.

The cutter stylus removes a continuous thread of cellulose as it cuts the groove, and a suction tube is fitted to remove the wax.

If the surface of the grooves cut on the disc are not smooth, a hissing sound will be produced on the playback. Hence, smoothness is an important factor in reducing the background noise of a recording. A well polished, hard and properly shaped stylus can produce reasonably smooth grooves.

The smoothness is improved considerably by heating the stylus. Heating is done by a passing low-voltage direct current through a small coil of high resistance wire around the shaft of the stylus just above the tip.

The diameter of the tip is important because the smaller the diameter, the greater will be the length of the recorded programme on the disc. But, the smaller the diameter, the greater will be the pressure on the record resulting in more wear. A compromise is made, and for monophonic records, the tip diameter is about 50 µm (2 mil) and for stereo records it is 25 µm (or 1 mil).

4.8 PLAYBACK NEEDLES

The playback needles are made of osmium or synthetic sapphire or diamond. These have the following characteristics:

Osmium Needle It is plain hardened steel and is good for only a small number of plays. It wears out after about 10 hours of play. It is the cheapest one and is used in low-quality gramophones. It is not used in hi-fi equipment.

Sapphire Needle This needle is most common. It has a much longer life than the steel or osmium needle. It can give hundreds of plays and wears out after about 50 hours of play. It is more costly than the osmium needle, but less costly than the diamond one.

Diamond Needle It is the best of all. It gives excellent performance for thousands of plays and wears out only after about 400 hours of play. It is the costliest needle. Although, it is superior to all other needles in performance and is most suitable for high-fidelity system, it is delicate and its tip can be broken by slight impact. A broken tip will ruin any type of recording. Moreover, it is the costliest of all, hence it is not used for playback.

The playback needles are also classified in accordance to their shape. These are (i) conical needle, (ii) round needle, and (iii) elliptical needle. Their shapes are shown in Fig. 4.9.

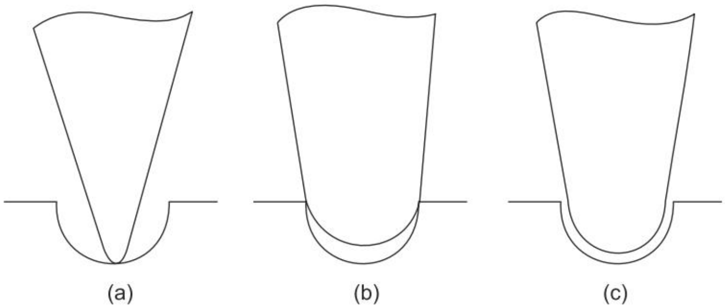


Fig. 4.9 *Reproducing needles (a) Conical tip (b) Round tip (c) Elliptical tip*

The conical or chisel-shaped needle is sharp and uses the least material and hence, is the cheapest. However, it may gouge the bottom of the groove, and is, therefore, not used in playback systems. (For recording chisel shape is suitable).

The round or spherical-shaped needle is rather blunt and does not fit very well in the groove. It may cause excessive wear on the walls of the groove, which may eventually breakdown. The round needles also cause tracking distortion. As the groove width narrows, it rises above and hence, does not maintain constant vertical position **(Pinch effect).**

The Elliptical needle It fits in the groove very well and hence is most satisfactory as a playback needle. The needle is fixed in such a manner that the long axis of the ellipse is nearly parallel to the radial which passes through the centre of the record. Due to most satisfactory tracking of the grooves (in consonance with the recording needle), there is little distortion.

The needles can also be classified as (i) monophonic and (ii) stereophonic. The *monophonic* needle mainly deals with the lateral recordings with little vertical motion, and hence, need not have high compliance for vertical motion. But a *stereophonic* needle requires high vertical compliance as the recording has vertical components. (This is the reason why a monophonic cartridge ruins the stereophonic record after only a few playings). The stereo needles are of 25 μm diameter, while the mono-needles are of 50 μm diameter. A stereo pick-up is more sensitive to vertical vibrations, and hence picks up the rumble noise from the motor more than a mono pick-up, because motor vibrations are also vertically directed.

4.9 CARTRIDGES OR PICK-UP UNITS

These are the units which pick up vibrations of the playback needles and convert the same into electrical signals (audio currents) and feed the same to the audio amplifiers of the record player for further amplification. It is like a microphone. The difference between a microphone and a pick-up cartridge is that while the microphone receives sound waves on its diaphragm, a pick-up cartridge receives the mechanical vibrations of the reproducing needle.

Pick-up cartridges can be of four types, namely, (i) moving coil or dynamic cartridge, (ii) magnetic cartridge, (iii) crystal cartridge, and (iv) capacitor cartridge. These are described below.

Moving-coil or Dynamic Cartridge This cartridge consists of a coil wound on a cardboard cylinder and is placed in a uniform magnetic field as shown in Fig. 4.10. The needle is attached to the coil assembly. As the needle moves to and fro, the coil moves in the magnetic field and hence, there is a change of magnetic flux through the coil. Thus emf is induced in the coil, depending on the rate of change of flux. The dynamic cartridge has excellent frequency response, but low output (in microvolts). Another version of moving-coil type is electrodynamic cartridge in which an electromagnet is used in place of a permanent magnet to get more output.

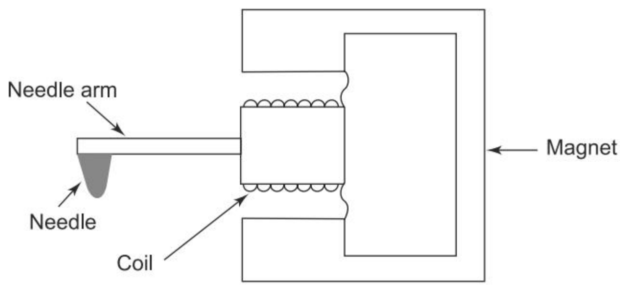


Fig. 4.10 *Moving-coil cartridge*

Magnetic Cartridge In the magnetic cartridge, the coil is fixed and a magnet is attached to the needle. The magnet moves as the needle vibrates. The movable magnet is placed in the fixed field of a permanent fixed magnet. The relative motion between the two magnets causes change in the lines of force in the space where the coil is placed. Hence emf is induced in the fixed coil. A magnetic cartridge is illustrated in Fig. 4.3 (Sec. 4.2).

The frequency range of a magnetic cartridge is good and is greater than that of a crystal pick up. Distortion is not more than 4%. The output is good, about 10 mV for a groove of 25 μm (or 1 mil) size. This is the most popular cartridge for the record player.

Crystal Pick-up Cartridge When a piezoelectric crystal is subjected to pressure variation, e.m.f. is generated across the crystal. Ceramic crystal is used for this purpose. It is shown in Fig. 4.11. Crystal pick up is made of a sandwich of two thin flat crystals with a metal foil between them. One end of the sandwich is attached to the arm to which a stylus" (or needle) is fixed. The other end of the crystal assembly has a metal foil connection on the outside of the sandwich.

The ceramic pick-up is stiff and therefore variations in the mechanical load do not greatly affect the performance. It is simple in construction and has fair frequency response. (The recording cutter units use crystal units like a motor.) It is the cheapest of all. Its output is very good, about 0.5 V. Impedance is very high, about 100,000 Ω. Crystal pick-ups do not need equalising circuits.

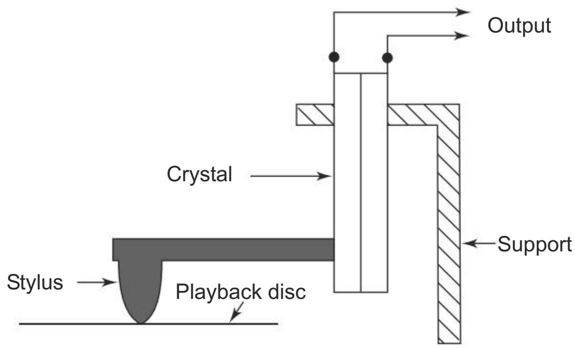


Fig. 4.11 *Crystal pick-up*

Capacitor (or condenser) Cartridge The capacitor cartridge consists of two parallel plates, one of which is movable and the other one is fixed. The playback needle is attached to the movable plate, as shown in Fig. 4.12.

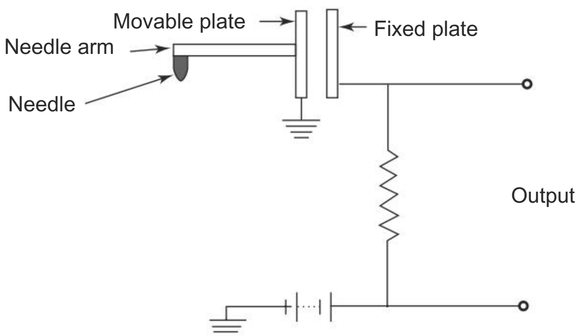


Fig. 4.12 *Capacitor cartridge*

When the needle vibrates while tracking the grooves, the movable plate of the capacitor moves to and fro, changing the capacitance, and hence, the dc voltage across the capacitance changes because the charge stored by the capacitor does not change. Voltage across the capacitor is given by Q/C, where C is charge in coulombs and C, the capacitance in farads. With change in the value of Q due to vibrations of the moving plate, and no change in *Q*/*C*, i.e. voltage across the capacitor, will change. The output voltage can be recorded across a resistance connected in series with the cartridge and a battery. The output power of the capacitor cartridge is quite low and a built-in pre-amplifier is used with it.

The capacitor cartridge has excellent frequency response, almost up to the high end of the frequency range, and its distortion is the lowest, about 1%.

4.10 EQUALISATION IN DISC RECORDING/PLAYBACK SYSTEMS

Equalisation is the process of improving signal-to-noise ratio by modifying the frequency characteristics before recording and neutralising that modification in the playback amplifiers. High dynamic range makes it necessary. The dynamic range of an orchestra is from 20 dB to 100 dB (or even more up to 120 dB in some cases). It is difficult to accommodate such large range of sound levels on a disc or a magnetic tape while recording. Further, the energy of sound contained in low-frequency notes is more than in high-frequency notes. If it is presumed that amplitude at mid frequencies is right for recording, the amplitude of low-frequencies would be too big and hence will occupy too much space on the disc and needs to be reduced before recording. The amplitude of the high frequencies on the other hand, will be comparable to the groove noise. Hence, to improve the signal-to-noise ratio, the high-frequency signals need to be emphasised before recording. (In a magnetic tape, the high intensity at low frequencies will saturate the tape and the low intensity at high frequencies will be comparable to the random variations in magnetisation.)

The Record Industry Association of America (RIAA) has set up the standard that recording be done at constant amplitude up to 1000 Hz and also from 2000 Hz onwards. In between, i.e., between 1000 Hz and 2000 Hz it should be done at constant velocity. The amplitude of sound and hence of audio increases as the frequency decreases. Hence, the amplitude of audio has to be lowered down more and more as the frequency decreases to keep it constant from 16 Hz to 1000 Hz. The amplitude of sound decreases as the frequency increases and, therefore, the amplitude of audio has to be increased more and more from 2000 Hz onwards as the frequency increases. This processing of signal is shown in Fig. 4.13.

At the playback end, the signal is restored to the original proportionalities for the whole audio-band by reversing or equalising the process, i.e., by emphasising low frequencies up to 1000 Hz and de-emphasizing the high frequencies above 2000 Hz. Constant velocity section of the recorded frequency range is also converted back to the values of the original sound. This technique of improving signal-to-noise ratio and accommodating high dynamic range is called 'Equalisation'. Figure 4.13 shows both curves, i.e., processing before recording and equalisation in the playback section.

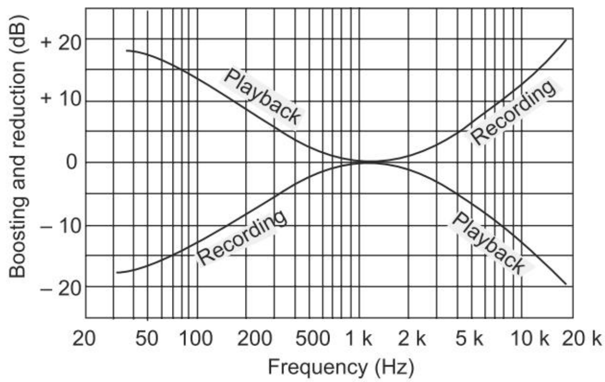


Fig. 4.13 *Equalization curves*

Example 4.2 *A signal was emphasised at 6 dB per octave from 2000 to 16000 Hz. Calculate the signal after each octave if the original signal was 1 volt.*

Solution

Signal at 2000 Hz 1 V

At 4000 (after 1 octave) = 6 dB above 1 V = 2 V (as 6dB doubles the voltage)

At 8000 Hz = 12 dB above 1 V = 4V

At 16000 Hz = 18 dB above 1 V = 8V

SUMMARY

- When an audio signal passes through a coil wound on an armature and placed in a magnetic field, the armature, experiencing motor action, will vibrate to and fro. A cutting needle, called the stylus is fixed to the armature. It cuts spiral grooves on a rotating disc made of lacquer. The grooves will change their position laterally due to audio current. For reproduction, pick-up cartridge, working like a microphone, will develop emf in its output. The emf will be in accordance with the lateral variations of grooves.
- The cutter stylus is made of sapphire or diamond and is chisel-shaped. The disc is made of metal or glass coated with lacquer. Grooves arc of 2 types:

  (1) coarse-grooves with 6 mil (150 microns) width, and

  (2) micro-grooves with 2 mil (50 microns) or less width.
- Coarse-grooves are used with 78 rpm discs and micro-grooves with 45 and 33 rpm discs.
- Recording is done on a wax disc, from which a master negative disc of copper is made by electro-plating. The

playback discs are stamped out by pressing on the master negative disc.

☞ Pick-up cartridges in playback systems are of 4 types: moving coil, magnetic, crystal and capacitance. Reproducing needles are of osmium, sapphire or diamond. The diamond needle is most durable, giving thousands of playbacks before wearing out.

☞ Considering the limitations of space and noise on the disc, high energy signals (in low frequency region) are to be attenuated to avoid overlapping and low energy signals (in high frequency region) are to be emphasised to give good signal-to-noise ratio before recording. In a playback system, the process is to be reversed to bring in the original proportionalities of sound in a reproduced programme. This processing is called equalisation.

Review Questions

1. Describe the principle of disc recording with the help of a block diagram.
2. Write down step-by-step procedure for producing records.
3. Draw a block diagram of disc recording and reproducing system and explain the function of each block.
4. Name different types of reproducing cartridges. Describe any one of them in detail with the help of a neat sketch.
5. What do you understand by coarse and micro-grooves? Give specifications of each. Explain how does speed and size of the groove affect the duration and quality of the programme that can be recorded.
6. Describe various types of cutting and reproducing needles.
7. What do you understand by 'equalisation'? Explain with neat figures, how does it improve quality of reproduced sound.

Short-Answer Questions

1. Why is lateral recording preferred to vertical recording?
2. How is audio programme recorded on a gramophone disc? (Answer briefly describing basics only.)
3. How are audio programmes reproduced from a record (give only the basics of the methods).
4. Why is diamond cutter preferred to a steel cutter in preparing gramophone records?

5. Why is diamond needle not suitable for reproducing the recorded programme?
6. What is the function of cartridge in playback system?
7. Why are micro-grooves preferred over coarse grooves?
8. What do you understand by 45° grooves?
9. How is smoothness of a groove improved?
10. What is equalisation in recording and playback system?

Multiple-Choice Questions

1. Which of the following working is like the working of a disc recording unit?
   (a) Motor (b) Dynamo
   (c) Microphone (d) Generator
2. Which of the following's working is like the working of the disc reproducing pick-up cartridge?
   (a) Motor
   (b) Dynamo
   (c) Loudspeaker
   (d) Amplifier
3. What materials the cutter stylus tip is made of?
   (a) Plastic (b) Copper
   (c) Diamond (d) Gold
4. Which disc is stamped over to make playback discs?
   (a) Negative (b) Positive
   (c) Flat (d) Slave
5. Magnetic cartridge uses the principle of
   (a) piezoelectric
   (b) emf induction
   (c) motor
   (d) amplifier
6. Which speed is not used in discs?
   (a) 33 (b) 45
   (c) 50 (d) 78
7 What is the width at the top of a coarse groove?
   (a) 50 μm (b) 100 μm
   (c) 150 μm (d) 200 μm
8. What type of grooves do modern monophonic discs have?
   (a) Lateral
   (b) Hill and dale
   (c) Both lateral and vertical
   (d) 45–45 degree type
9 What type of grooves are used in stereophonic discs?
   (a) Lateral only
   (b) Vertical only
   (c) 45°
   (d) None of these
10 What type of recording system uses 45° recording?
   (a) Monophonic
   (b) Stereophonic
   (c) Quadraphonic
   (d) Optical

Numerical Problems

1. Calculate the number of grooves per cm of width and maximum number of grooves on a disc of 25-cm radius if the width of groove is 15, μm and extra vacant spacing between the grooves is 10/μm.
2. A signal was dc-emphasised at 6 dB per octave from 1000 Hz to 31.25 Hz and was emphasised at the same rate from 2000 to 16000 Hz before recording with reference to its level at 1000 Hz. Calculate the reduction or increase in dB in the playback section at 31.25, 1000 Hz, 2000 Hz and 16000 Hz.

Answers

Short-Answer Questions

1. Lateral grooves eliminated high noise produced by hill-and-dale type vertical grooves. They also eliminated erosive wear caused by vertical grooves.
2. Current flows through the coil of disc placed in a magnetic field, the force on the coil makes a cutter vibrate on the lacquer (wax) and produces grooves of the same frequency as audio current. It is like a motor action.
3. When the needle (called stylus) of a playback cartridge moves radially over a gramophone record, it picks up vibrations from the grooves on the record. These vibrations are converted into audio signals by e.m. induction in the same way as a microphone's diaphragm produces audio signals.
4. Diamond cutter is more durable than steel cutter.
5. While diamond is used for cutting groves, it cannot be used for playback needle. Cutting grooves is one-time affair and hence costly needle can be used. But records are used on mass-scale by the people, and hence it should not be costly. Moreover, diamond tip is delicate and can be broken by slight impact. A broken tip will ruin any type recording. Hence diamond is not used in a playback cartridge.
6. The cartridge converts vibrations received from the stylus into electrical signals of the same frequency (called audio signals).
7. By reducing the size of the grooves, play time is increased.
8. For stereophonic recording, two channels are to be recorded simultaneously without any coupling (cross-talk) between the two channels Bluemlein devised a system

by which two channels are recorded on two walls of the groove inclined at 45º to the horizontal. As the motion of the stylus arms is 45º with the horizontal, the grooves are called 45º grooves.

9. A well-polished, hard and properly shaped stylus can produce reasonably smooth grooves. The smoothness is further improved considerably by heating the stylus.

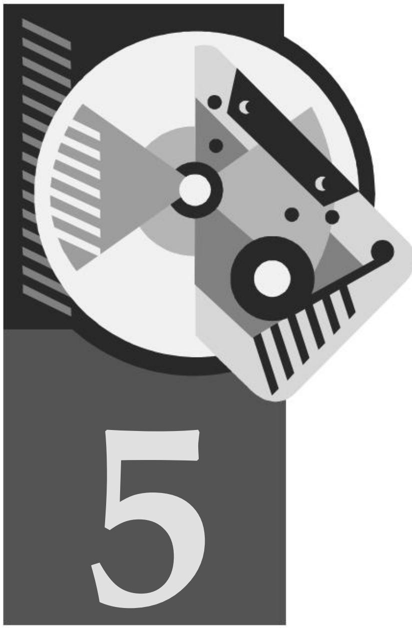
10. Equalization is the process of improving signal-to-noise ratio by modifying the frequency characteristics before recording and neutralizing that modification in the playback amplifier.

Multiple-Choice Questions

1 (a)	2 (b)	3 (c)
4 (a)	5 (b)	6.(c)
7 (c)	8 (a)	9 (c)
10 (b)		

Numerical Questions

1. (200,10000)
2. (–30,0,0, + 18)



Magnetic Recording

5.1 PRINCIPLE OF MAGNETIC RECORDING AND REPRODUCTION

Magnetic recording is storage of the sound-pressure variations in the form of elementary magnets (formed in a magnetic material), whose length and strength depend on audio signals. It was discovered by Poulsen of Denmark in 1898.

Magnetic recording is based on the principle (shown in Fig. 5.1) that certain materials (like iron oxide) when brought in a magnetic field, get magnetised and retain that magnetism permanently until altered. The various steps involved in magnetic recording are described below.

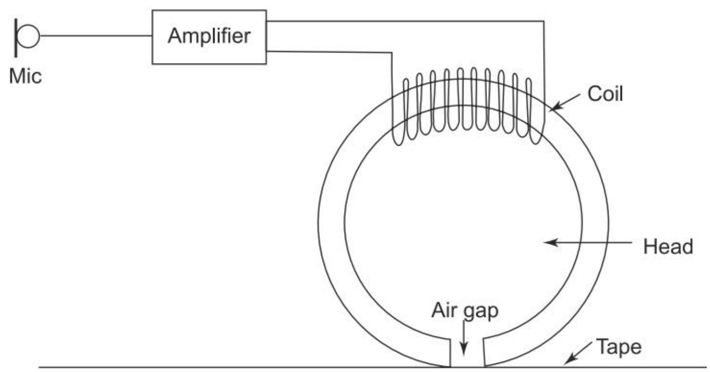


Fig. 5.1 *Magnetic recording on tape*

Sound-pressure variations are converted into electrical variations (audio signal) by a microphone. The audio output of the microphone is amplified and fed to the coil of an electromagnet. The electromagnet (called *head*) has a minute gap through which magnetic lines of force cannot pass easily due to high reluctance of air. When a tape with a coating of a magnetic material (like iron oxide) is made to pass across the gap, the lines of force get an easy path through the iron oxide which is formed into elementary magnets. The magnetic strength of electromagnet, and hence through the gap covered by the iron oxide of the tape depends on the audio current. Thus, the coating of iron oxide on the tape is magnetised in accordance with the audio current and hence, in accordance with the sound-pressure variations. The magnetism in the iron oxide can be retained for long time. This means that sound has been recorded in the form of a varying magnetic field.

In reproducing the recorded sound, the tape is again made to pass through a similar gap (or even the same gap), and it causes change of lines of force through the coil. This induces emf (audio signal) in the coil, which is in accordance with the rate of change of magnetic flux in the tape. The induced emf is amplified and is fed to a loudspeaker which converts the audio signal into sound.

Figure 5.2 shows a magnetised tape for 3 cycles of audio signal. The Figure shows that as the wavelength decreases, the length of the bar magnets formed on the tape decreases. Each cycle gives rise to two bar magnets.

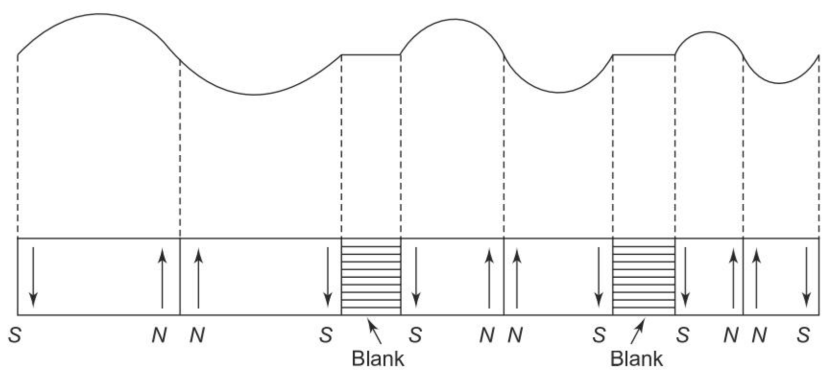


Fig. 5.2 *Magnetisation of tape by three audio cycles of different frequencies*

5.2 RECORDED WAVELENGTH, GAP-WIDTH AND TAPE SPEED

The length of tape magnetised by one cycle of audio signal is called the recorded wavelength. It is expressed by Eq. 5.1

$$\lambda = \frac{S}{f} \tag{5.1}$$

where, λ = recorded wavelength in cm

S = speed of tape in cm/s

f = frequency of audio signal in Hz

As the frequency increases, λ decreases. At high audio frequencies, the opposite poles become very near to each other as shown in the last cycle of

Fig. 5.2. This nearness of the opposite poles results in self-demagnetisation, and hence in reduction of magnetic strength. Thus high audio frequencies produce weaker bar magnets on the tape.

Effect of gap-width In recording, change in the magnetic flux on the tape is directly proportional to the audio current variations through the head's coil. But in reproducing, the audio signal induced in the coil is proportional to the rate of change of the magnetic flux on the tape. This rate of change increases as the frequency of the recorded signal increases. The induced signal increases at 6 dB/octave. However, the output does not rise indefinitely with frequency. It peaks at a maximum output voltage for a particular frequency, and then drops. The output becomes practically zero at a frequency at which the wavelength of the recorded signal becomes equal to the width of the head-gap. As the frequency increases, wavelength decreases and hence the gap-size measured in terms of wavelength increases, that is, the same gap-size becomes $\lambda/4$, $\lambda/2$, λ, and so on. The curve, relating output with gap-width (and hence with frequency), is shown in Fig. 5.3.

Figure 5.3 shows that if the gap-width is equal to $\lambda/2$, maximum output is obtained. But when it is more than $\lambda/2$, the output decreases and becomes zero when the head size is wide enough to accommodate full wavelength.

When the gap-width is λ, it allows full two magnets of equal and opposite polarity on the tape to be impressed through the coil. Two areas of equal and opposite polarity cancel the magnetic effect through the coil, resulting in 'null' output.

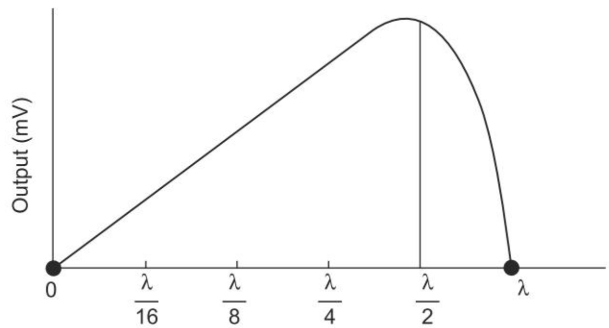


Fig. 5.3 *Effect of frequency and gap-width on the output during play-back*

Thus, the gap-size puts a ceiling on the high-frequency response. The gap-width has to be very small to give good output even at high audio frequencies. The gap size is made quite small, of the order of 10^{-5}m (a few microns).

Relationship between Gap-width, Tape Speed and Frequency The gap-width cannot be reduced beyond a certain limit. But if the speed of the tape, moving past the gap, is increased so that only one bar magnet of the tape passes across the gap in $T/2$ seconds (T = time period or $1/f$ of the recorded signal), the cancellation of magnetic effect will not take place.

The relationship between speed (S), gap-width (G) and the highest usable frequency, f_m, is expressed by Eq. 5.2.

$$f_m = \frac{S}{2G} \tag{5.2}$$

Derivation of Eq. 5.2 One cycle of variation covers distance = λ on the tape. If T is the time period in seconds (i.e., time taken in completing one cycle) then,

in T seconds, distance covered = λ

Hence, in 1 second distance covered $= \frac{\lambda}{T}$

As distance covered in 1 second is called speed,

so, speed, $S = \frac{\lambda}{T} = f\lambda \qquad \left(as\ \frac{1}{T} = f\right)$

or, $\lambda = \frac{s}{f}$

For optimum output, the gap-width should be equal to $\lambda/2$ for the highest frequency of use (f_m).

Therefore,

$$\text{Gap-width, } G = \frac{\lambda}{2} = \frac{S}{2f_m}$$

or, $$f_m = \frac{S}{2G}$$

Equation 5.2 gives the highest usable frequency of the recorded signal for peak output. Increasing the speed of the tape increases the highest frequency that can give good output. The frequency, f_n, at which output would be zero (or null) will be equal to S/G.

Example 5.1 *If the gap-width is 6 microns and the speed of tape is 4.75 cm/s, calculate the maximum frequency for recording. What will happen if tape speed becomes 19 cm/s? Comment for video frequencies.*

Solution

G = 6 microns

$= 6 \times 10^{-4}$ cm

S = 4.75 cm/s

Then, $f = \frac{4 \cdot 75}{2 \times 6 \times 10^{-4}}$

$= 0.3958 \times 10^4$

$= 3958 \text{ Hz} \simeq 4 \text{ kHz}$

If the tape speed is increased to 19 cm/s, the highest usable frequency becomes 15832 Hz. This is good enough to cover the practical audio range.

(The video frequencies are in the range of MHz which will need such high speeds as are not feasible. The difficulty is solved in two steps. Firstly, the gap-width is reduced by a factor of 10. Secondly, the head itself is rotated in opposite direction to the motion of the tape to give high relative velocity between the head-gap and tape. This aspect is explained in detail in Chapter 15 on video recording on magnetic tape).

Choice of Tape Speed If the speed is high, programmes of only small duration will be recorded, because a single cycle of audio-wave will occupy long space on the tape, as

$$\text{distance occupied} = \frac{\text{speed}}{\text{frequency}}$$

Hence, for long play tapes, low speed is preferable.

However, if the speed is very low, cycles will occupy small space, reducing resolution between adjacent cycles. The high audio frequency cycles, in particular, will merge with each other. This will give rise to distortion. Further, closeness of bar magnets of opposite polarities will weaken the magnetic strength. Also, the wavelength recorded may be of the dimension of gap size, which will nullify the output in reproduction mode. The relationship between gap-size and speed for optimum highest frequency is given by Eq. 5.2 [$f_m = S/(2G)$]. Thus, for a given gap-size, highest recordable frequency is directly proportional to speed. Higher the speed, higher is the optimum highest frequency. Hence speed cannot be too low.

In view of these considerations, a compromise is made while choosing the tape speed. The speeds which have been standardised for tapes are

- 38 cm/s
- 19 cm/s
- 9.5 cm/s
- 4.75 cm/s

Lower speeds of 9.5 and 4.75 cm/s are used for long play as in domestic tape recorders and cassette recorders. (Higher speed is needed in good quality studio tape recorders.) Further reduction in speed is not possible due to problems of poor resolution and weak output.

Cassette Nomenclature Cassettes are often marked in terms of the duration of the recorded programme. For example, a two-track cassette marked C-90 will have a programme of 45 minutes on each track. Cassette tape speed is normally 4.75 cm/s, which means the length of the C-90 tape is $4.75 \times 60 \times 45 = 12825$ cm = 128.25 m.

5.3 NEED FOR BIASING

Magnetisation of the head's core is not linearly proportional to the magnetising force. The characteristic curve showing magnetic flux density vs magnetising force (or audio current) is non-linear as shown in Fig. 5.4. The curve shows that the small signals will not be recorded on a tape and the large signals will be severely distorted due to non-linearity at small current levels between *A* and *B*.

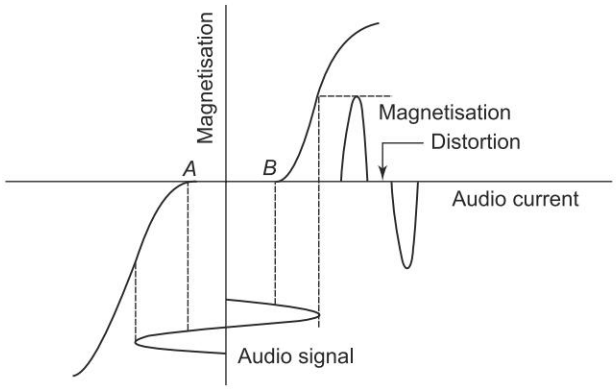


Fig. 5.4 *Distortion of signal in absence of any bias*

5.4 DC BIASING

One method to solve the problem of non-linear distortion is to provide dc biasing to the core and superimpose the audio signal on dc. This biasing will initially magnetise the core and hence the effect of non-linearity at a small current level will be eliminated as shown in Fig. 5.5. In dc biasing, the output is limited due to saturation of the core after the point *D*. Only the + ve side of the magnetising curve is being used in dc bias. Also, the noise level will increase as the suppressing effect of low signal portion of the curve (*A* – *B*) will not be present. The curve

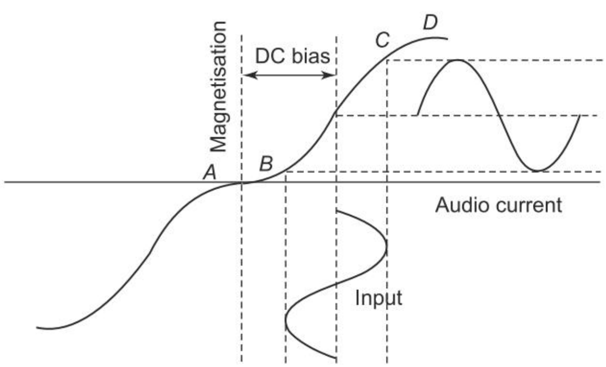


Fig. 5.5 *DC bias for recording*

is linear between *B* and *C*. Signal-to-noise ratio is only 25 dB in this type of biasing, while it should be more than 50 dB for good quality.

5.5 AC BIASING

To eliminate the defects of dc biasing, an ac signal (50 – 100 kHz) of constant amplitude and constant frequency is passed through the head coil. The audio signal is superimposed on the ac signal. The output is obtained on both sides of the magnetising curve. In fact, the tape will be magnetised in a series of pulses where peaks will correspond to the peaks of the combined waveform. An ac biasing curve is shown in Fig. 5.6. *P* is the input signal and *Q* is the output signal.

Now, whatever distortion occurs due to low signal level curve (*A*, *B*), it will occur only in ac bias wave (as shown at the point *E*) and not on the audio signal. The ac is supersonic (being much higher than 20 kHz) and hence, will not be audible during playback. Also, it can be easily filtered in the amplifier stage.

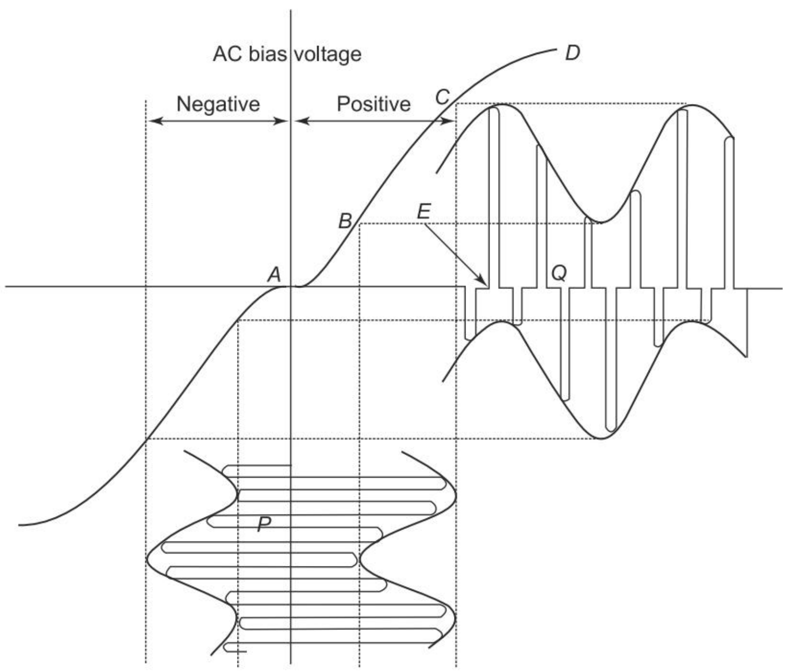


Fig. 5.6 *AC bias for recording*

AC biasing has the following advantages:

1. High audio levels can be recorded without distortion.
2. Both sides of the magnetisation curve are used, resulting in higher output.

3. Noise in the audio spectrum is much lower and signal-to-noise ratio can be as high as 60 dB.
4. Effect of head gap-width for high-frequency audio is reduced as magnetisation will be in pulses.

5.6 PARTS OF A TAPE RECORDER

Besides tape transport mechanism, a tape recorder consists of the following parts:

1. Tape
2. Recording head
3. Playback head
4. Erase head
5. Microphone and recording amplifiers
6. Playback amplifiers and loudspeaker

These parts are described individually in the following paragraphs.

Tape and Tape Materials A tape consists of a tough unstretchable non-magnetic base or backing material. The most popular material for base is mylar (a plastic). The base is coated with fine particles of a magnetic material such as ferric oxide or cobalt doped ferric oxide (Fe_2O_3) or chromium di-oxide (CrO_2). Fe_2O_3 is prone to self-demagnetisation, but the chrome type has high coercivity and hence self-destruction is very much reduced resulting in good, high audio frequency response. However, the chrome type needs stronger magnetic fields and is costly. Also, chrome tapes require 40% more bias than iron oxide tapes. Some good quality tapes use iron-alloy rather than an oxide of iron or chrome. Iron alloy has a very high coercivity and hence its high-frequency response is the best of all. The coated surface is polished to improve tape-head contact and reduce wear in the head.

The tapes come in various standard widths, like 6.3, 12.7, 25.4 and 50.8 mm (or quarter inch, half inch, 1 inch and 2 inch). The standard quarter-inch wide tape can accommodate two tracks, each 2.5 mm wide with a guard band of 1.3 mm in between. The thickness of the standard tape is about 38 micron (1.5 mil). The thinner the tape, greater will be the length of tape in a spool.

A good quality tape should have the following characteristics:

- It should not be stretchable.
- The coercivity of magnetic material should be high.
- The tape base material should be tough.
- The coating of magnetic material on the base material should be uniform.
- The base thickness should be uniform.
- Foreign particles should not be present.

- It should be well polished and impregnated with compounds that act as lubricants to reduce the friction between the tape and head.
- The tape must press the head properly.

Recording Head The recording head consists of a ring-shaped, high permeability core (mu-metal or ferrite) with a very narrow gap, about 6 microns wide, (video heads are only 0.5 micron wide). Such narrow gaps of molecular dimensions are obtained by using fabrication methods of semiconductors to grow thin layers of silicon-dioxide. The magnetic reluctance of silicon-dioxide is high which means that the gap's permeability is low. When a magnetic tape moves past the head, the magnetic lines of force can go through the tape much more easily than through the gap.

High permeability of the core gives high flux density at the gap and gives good recording. Permalloy (Nickel–iron–molybdenum alloy) and ferrite (ceramic material sintered at high temperature, composed of the oxides of iron, manganese, nickel and zinc) are suitable as core material. Heat treatment improves performance by bettering magnetic property and by giving more hardness and hence high wear resistance. The gap is cut in the flattened surface of the core and parallelism of the faces is strictly maintained.

A coil is wound on the soft iron core of the head. When audio current (flows through the coil, the core is magnetised and the variation in magnetisation is in accordance with the audio variations of current. Record head coils have an inductance between 0.5 to 7 mH and are designed to take up maximum operating level current.

Playback Head In domestic low-priced recorders, the record head itself acts as a playback head. The varying magnetic field of the tape cuts the coil winding and induces emf due to electromagnetic induction. This emf is processed and amplified and then fed to the loudspeaker.

However, the professional recorders use a separate head for playback.

High permeability core like permalloy or ferrite gives good output in the playback. Also, high permeability confines the flux within the core. The playback head has narrower gap (4 microns or less) for better high-frequency response. Inductance of the playback head is about 10 times that of the record head. The wire is thin as only small current is to be handled by it. Except these three variations (i.e. gap size, inductance and conductor size), the playback head is similar to the record head. The output is about 1 mV.

Erase Head It is also a core of high permeability with a gap. A coil is wound round the core, giving an inductance of about 40 to 80 mH. The gap-width in the erase head is much wider than the record head and covers bar magnets of the tape for demagnetisation. A radio frequency current is passed through the coil, and it demagnetises the tape. The erase field should be such that the magnetic recording is reduced in amplitude by at least 75 dB.

The erasing frequency is 5 times or more the highest audio frequency. Thus, it is about 100 kHz for hi-fi tapes and about 40 kHz for other tapes. Power of 2 to 4 watts is required for erasing. The head is adequately screened from the other heads.

Microphone and Recording Amplifiers A crystal microphone is used in low-priced tape recorders and cassette recorders. Its cost is low, frequency response is up to 8 kHz, and sensitivity is higher than the moving-coil ones. In professional-grade tape recorders, a capacitor microphone is used because of its flat response of up to 15 kHz, high sensitivity (due to an in-built amplifier), low noise and low distortion.

The record amplifier accepts the audio output of the microphone and amplifies it to drive the head. The signal-to-noise ratio of the amplifier is high, about 85 dB in a professional-grade equipment. The amplifier stage consists of a pre-amplifier and a power amplifier. Processing of the signal for de-emphasising of high intensity at low frequencies and emphasising low intensity at high frequencies is done at the pre-amplifier stage. Level control or gain control is also located here.

Playback Amplifiers and Loudspeakers Playback amplifiers have equalisation circuits, bass/treble tone control circuits and volume control in the pre-amplifier stage. The output of the pre-amplifier goes to the driver amplifier and then to the power amplifier which gives the desired power and drives the loudspeaker.

Loudspeakers are generally of the cone type. In high-quality systems, a column having woofers, squawkers and tweeters with proper crossover network, is used.

In cassette recorders, recording and playback amplifiers are common.

5.7 TAPE TRANSPORT MECHANISM

The tape needs to be moved steadily across the head-gap for recording and playback. The standard direction of travel is from left to right as seen from the opposite of the front of tape heads.

In the studio type, separate motors are used to drive the capstan, supply spool and take-up spool. But in the domestic type, only a single motor serves the purpose as shown in Fig. 5.7. It drives the capstan and then with the help of pulleys and belts, spools are driven.

The tape transport mechanism consists of the following components:

1. Motor,
2. Capstan and press roller,
3. Flywheel,

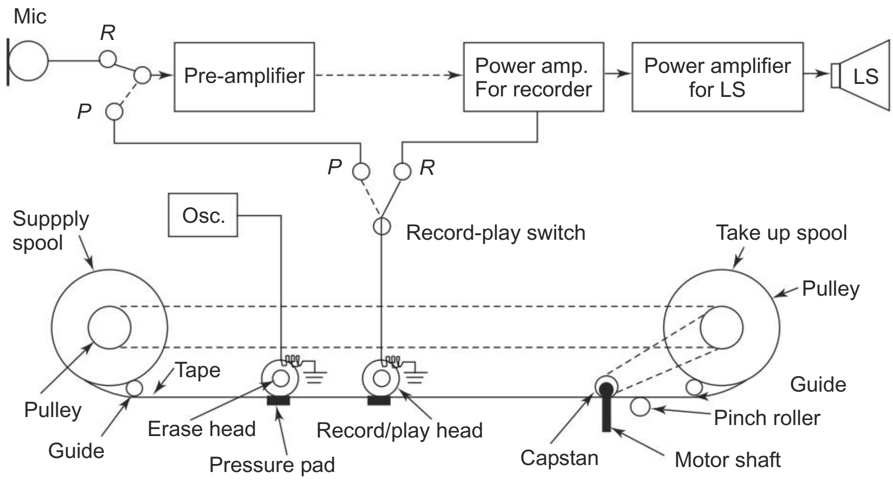


Fig. 5.7 *Transport mechanism for cassette recorder*

4. Tape guides, and
5. Spools.

These are described below:

Motor The motor used in a tape-transport mechanism is of synchronous type. A synchronous motor has its speed locked to the frequency of the supply voltage and therefore, maintains constant speed irrespective of variation of supply voltage or load. This steadiness results in reduction of wow and flutter distortion. In good tape machines, wow and flutter are not more than 0.2%. Speed of motor for fast rewind and fast forward can be changed with the help of speed gears.

Motor rotations are transferred to the capstan flywheel assembly by means of a rubber belt. It prevents motor vibrations from reaching the capstan and thus reduces the rumble noise.

Capstan and Press (or Pinch) Roller Capstan is a spindle, machined accurately, and pulls the tape past the heads. The tape is pressed against the capstan by means of a rubber-covered pinch roller.

Flywheel It is a heavy wheel made of metal and is fitted to the capstan shaft. This damps minor variations in the speed. It should be free from any tendency to vibrate, because any vibration here will cause a rumbling problem.

Tape-guides These provide the desired tension in the tape and keep it in the correct position. The angle round which the tape should turn at any point in the transport should not be excessive. All the bearings over which the tape passes must be of high quality.

Spools There are two spools. While in use, one spool feeds the tape to the other spool, and hence, the spools are known as (i) Feed spool or supply spool, and (ii) Take-up spool.

Indication of the position of any recorded signal is achieved by fitting a rotation counter to the take-up spool's spindle.

5.8 ADVANTAGES AND DISADVANTAGES OF TAPE RECORDING

Advantages

1. There are no vibrating mechanical parts (like stylus of discs) and hence, it is immune to wear, and the quality does not deteriorate on playing. A tape can be played thousands of times without degradation.
2. Editing and dubbing is easy.
3. Recording can be monitored simultaneously and can also be used for immediate playback. (Immediate playback is not possible in disc recording. A negative is to be made first from the original recording and then positive plate for playback.) There is no master negative system and recording is direct on the tape itself.
4. The same tape can be used again and again for recording different programmes by simply erasing the previous programme.

Disadvantages

1. The life of a recorded tape will not be as long as that of a negative copper plate in the vinyl disc system. The negative copper plate can be preserved for thousands of years.
2. Modulation noise is present in tapes, while there is no such noise in discs.
3. Transient response in tapes is not as good as in discs.
4. For playing a reel-to-reel tape, the reel must be carefully keyed into position and the tape threaded through the guides. This is a time-consuming operation. Contrary to it, a disc can be easily put on a turn-table and the pick-up placed on it. Thus, music is obtained with little delay on the disc. However, this defect is not present in cassettes.
5. Copying from one tape to the others is not as easy as production of playback discs from the master plate. Moreover, multiple copying causes severe degradation.
6. To search a portion of a programme a tape is to be run from the very start. This is cumbersome.

Advantages of tape recorder outweigh its disadvantages with respect to gramophone records, and hence tape recorders became more popular, particularly in the form of cassette recorders. However, for preserving a recording, tape recordings are transferred to disc recording and preserved in the form of negative copper plates.

Nowadays compact discs using laser beams have replaced wax discs and even tapes. Digital recording and reproduction on compact discs have changed the scenario completely in the field of recording and reproduction. These are described in Chapter 6 (Sections 6.4 through 6.7). Digital video discs (DVDs) have been described in Chapter 16.

5.9 BLOCK DIAGRAM OF TAPE RECORDER

Figure 5.8 gives a block diagram of a magnetic recoding and reproduction system. It shows a high quality system having separate amplifier stages for recording and playback. (In domestic cassette recorders, common switchable amplifier stages are used for recording and reproduction.)

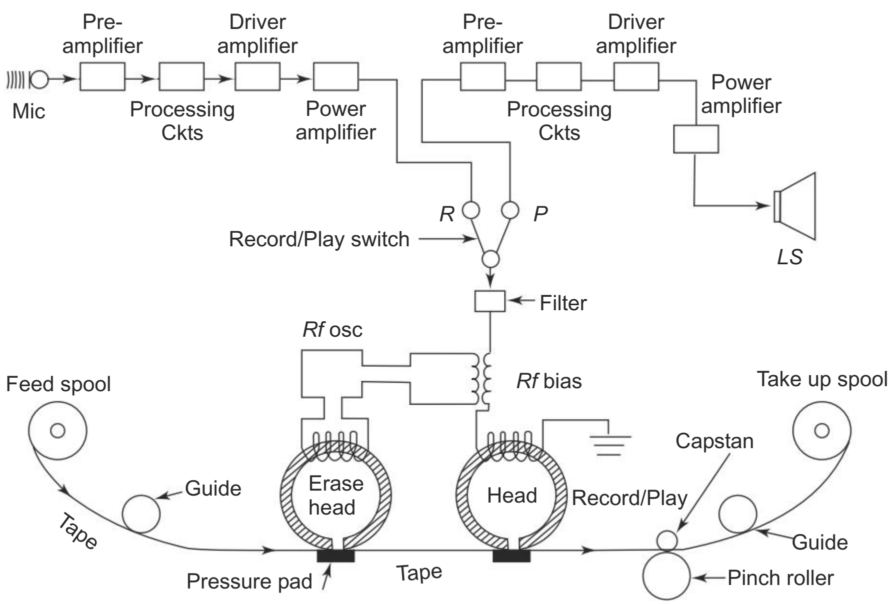


Fig. 5.8 *Block diagram of a tape recorder*

Function of Each Block

Microphone Sound waves strike the diaphragm of the microphone which converts the sound-pressure variations into electrical signals, called audio signals.

Pre-amplifier It amplifies the weak output of the microphone. Its noise figure is low.

Processing Circuits These circuits in the record section control the gain and level of recording and also provide de-emphasis and pre-emphasis for low-frequency and high-frequency audio signals, respectively.

Driver Amplifier It gives further voltage amplification to the signal so as to reduce the internal resistance of the power amplifier and hence, to drive it to give power amplification.

Power Amplifier It amplifies the power of the audio signal so as to drive the record head.

Filter It is a trap circuit which does not allow the bias oscillator's signal to go to the amplifier as it will unnecessarily get overloaded.

Heads There is one erase head before the record head, pressing on the tape. It erases all previous recordings.

The next to erase head is the record head (which is also generally the playback head as shown in the block diagram. For professional hi-fi recording, a separate playback head is used after the record head). The record head records audio current in the form of a varying magnetic field in the magnetic material of the tape.

Record/Play Switch It connects the recording power amplifier to the record/play head for recording through record (R) terminal of the record/play switch. When this switch is connected the terminal marked (P), the head is connected to the pre-amplifier of the playback section.

Pre-amplifier, Driver Amplifier and Power Amplifier in the playback path have the same functions as described above for the recording path. The processor does just the opposite of what has been done by the processor in the recording chain. It de-emphasises the high audio frequency signals and emphasises the low-frequency signals to restore the original proportionalities of sound. It also contains volume control, and treble and bass controls.

5.10 CIRCUIT DIAGRAM

A cassette recorder uses the same amplifiers for recording as well as playback with the help of ganged record/play switches. Although modern tape recorders use one or two ICs for complete circuit, a typical circuit diagram using discrete transistors is shown in Fig. 5.9 for better understandability.

Explanation of the Circuit Ganged record-play switches connect the chain of amplifiers to recording or to playback depending upon the position of the switches SI and S2. A record-level control potentiometer is connected to the record level meter in the record mode of the switch S3. A volume-control potentiometer is connected to the next amplifier in playback mode of the switch S4. Switches S5

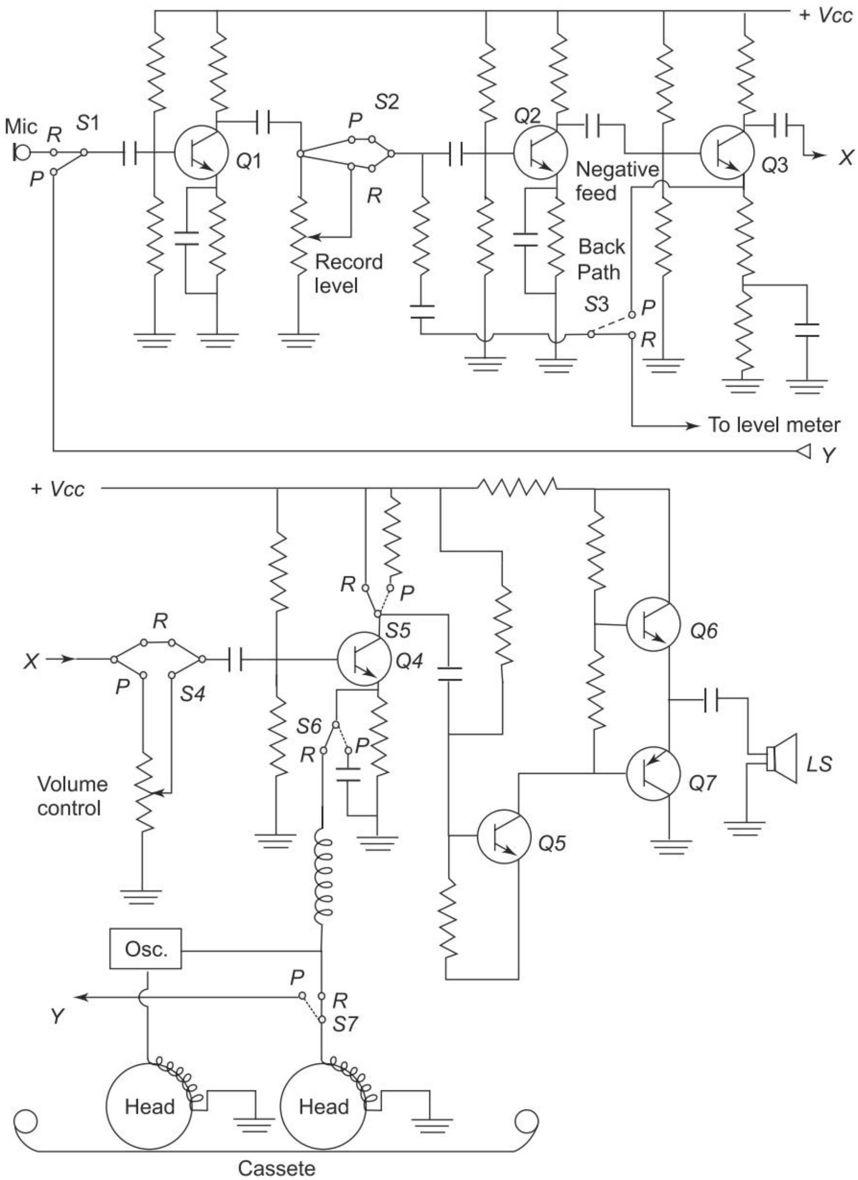


Fig. 5.9 *Typical circuit diagram for audio cassette recorder*

and S6 connect the transistor Q4 in a common-emitter configuration for play-back position and as an emitter-follower in record position. Switch S7 feeds the signal to the head in the record position, and takes it for playback in the play position. A high-frequency oscillator stage feeds an RF signal to the erase head. It also provides ac bias to the record head. Transistors, Ql, Q2 and Q3 are R-C coupled voltage amplifiers and provide desired voltage level to get sufficient power from Q4 for recording. Q4 acts as a driver amplifier in the playback position and drives the complementary symmetry transistors Q6 and Q7. Q5 provides the desired bias to the pair. The output of complementary symmetry amplifier is fed to the loudspeaker for reproducing sound.

5.11 'WOW' AND 'FLUTTER' DISTORTIONS

If the tape speed is not steady, but varies even slightly, distortions called 'wow' and 'flutter' are produced.

When the variation in speed is low, 0.5 Hz to 6 Hz (the frequency range is as per IEEE dictionary of electrical and electronic terms) the playback in case of a tape or disc will result in the sound having a slow up and down variation in pitch. This variation is similar to that of a siren sound, and is called 'wow'. (Although a frequency of 6 Hz in itself is not audible to the ears, but ears can detect very small changes, even less than 1 Hz, in the pitch of notes when a reference is present. This is why even slightly mistuned individual instruments or voices in a musical programme will sound dissonant.) Wow is generally caused by a buckled tape spool (or an off-centre gramophone record). Wow can be easily detected on the sustained note of piano or organ instruments.

When the variation in speed is more than 6 Hz (and up to 100 Hz), the variation in pitch is called 'flutter'. It resembles a quiver. Flutter is generally caused by a damaged capstan or pressure roller or the disc being physically deformed or warped. Flutter is most easily detected on wood–wind instruments.

The minimum perceptible pitch change is 3% at the bass end and only 0.3% at high audio frequencies. Wow and flutter are kept minimum by using a synchronous motor which has steady speed. Wow and flutter distortions are expressed in percentage. A typical value is 0.3% but with good design it can be reduced to less than 0.1%.

5.12 RUMBLE

It is a low-frequency distortion caused by mechanical vibrations of a motor body or capstan. Motor vibrations are prevented by using mechanical filters (vibration absorbers).

Rumble is expressed in dB. The machine can be said to be almost silent if rumble is –45 dB with respect to sound-pressure level (SPL) of 1 μbar.

5.13 HISSING NOISE

A hissing noise is inherent in all tape recordings due to tiny irregularities in the tape coating. The hissing noise becomes comparable to the soft notes and hence is most perceptible during soft silent passages in music. The advent of multitrack recording techniques has further highlighted the noise problem because the noise on tracks is additive.

5.14 NOISE-REDUCTION TECHNIQUES

Pre-emphasis and De-emphasis

As noise is more perceptible during the quiet passages of music, it is desirable to emphasise the low-power notes before recording, so that these are at much higher level than noise. At the receiver, it is essential that the reproduced sound possesses the same proportions of intensities for low and high notes as were present in the original sound (before pre-emphasis). De-emphasis can bring back the originality. Emphasing low-intensity sounds before recording is called *Pre-emphasis* and the process of de-emphasising in the playback circuits to bring originality is called *Equalisation*. Pre-emphasis and equalisation essentially are needed to improve signal-to-noise ratio and at the same time to maintain high fidelity in the reproduced sound. In playback stages, the high-frequency signals are de-emphasised or reduced in intensity, and in the process, noise is also reduced. Thus, signal-to-noise ratio achieved by the pre-emphasis, is maintained.

As high frequencies have low energy content, the simplest technique is to emphasise high audio frequencies above 1 kHz at 6 dB per octave before recording. At the receiver, the intensity of these frequencies can be proportionately de-emphasised. The curve showing such pre-emphasis/ de-emphasis is given in Fig. 5.10. The simple pre-emphasis and de-emphasis circuits are shown in

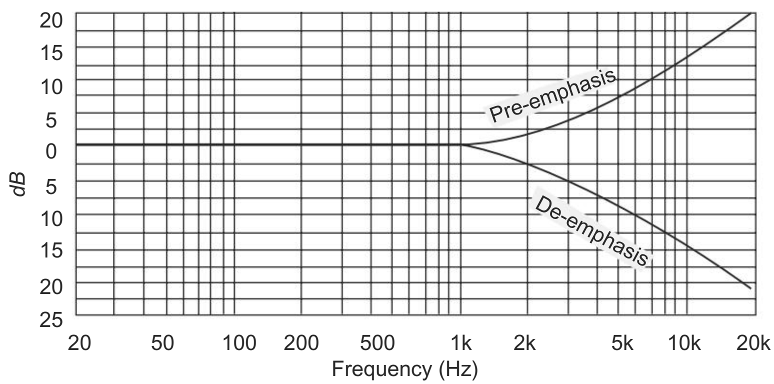


Fig. 5.10 *Pre-emphasis–de-emphasis curve*

Figs 5.11(a) and 5.11(b). In the pre-emphasis circuit of Fig. 5.11 (a), a capacitor is in series. The higher the frequency, less is its reactance and hence less is the attenuation. In the de-emphasis circuit of Fig. 5.11(b), output develops across a capacitor and hence higher the frequency, less is the output.

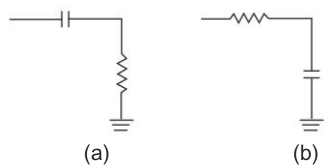


Fig. 5.11 (a) *Pre-emphasis circuit*
(b) *De-emphasis circuit*

Dolby's Method

Dr Ray Dolby introduced a novel system for providing 10–15 dB improvement in recording and playback tapes. In normal pre-emphasis, it is presumed that weak intensity is present only in high frequencies. This is not always the case. All weak signals, irrespective of frequencies need to be emphasised. This difficulty was solved by Dolby as explained below:

When the strength of signals falls below a pre-determined level (say, 40 dB over the noise level), the circuits boost the strength before recording. All signals which are 40 dB or higher pass through the Dolby system directly without any change. The lower level signals pass through the boosting stages which boost these signals by 10–15 dB. Boosting is done before recording. A signal in the absence of boosting is shown in Fig. 5.12 (a). After boosting, the recording noise remains unchanged but the signal is boosted as shown in Fig. 5.12 (b). During

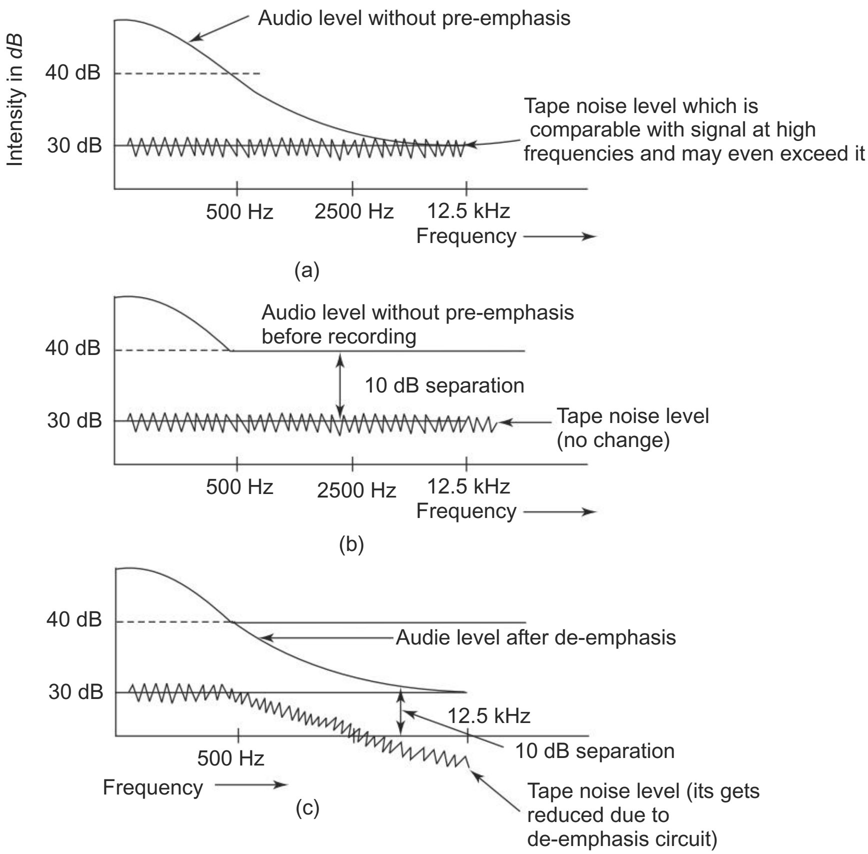


Fig. 5.12 *Reduction of noise by 10 dB in Dolby system*
(a) Position without pre-emphasis
(b) Position after pre-emphasis
(c) Position after de-emphasis

playback, signal and noise both are reduced as shown in Fig. 5.12 (c). Thus, Fig. 5.12 (c) indicates that signal-to-noise ratio is finally improved.

Dolby A System Boosting is done in 4 bands:

I. below 80 Hz
II. 80 Hz to 2999 Hz
III. 3000 Hz and above
IV. 9000 Hz and above

Bands III and IV overlap above 9 kHz, so that the high-frequency noise (hissing sound) is substantially reduced. Each band is processed separately by using low-pass, band-pass and high-pass filters and limiters. The 16 Hz to 80 Hz signal goes to a low-pass filter which causes improvement in signal-to-noise ratio with respect to hum and rumble. The 80 Hz to 2999 Hz signal goes to a band-pass filter which deals with the mid-band noise. Most of the sound energy for music is concentrated in this band. The 3000 Hz and 9000 Hz high pass filters improve signal-to-noise ratio with respect to hiss and modulation noise. The output of the four separate units is added. All this is done in a side branch, and this branch is known as the differential network. The output of the differential network goes to the adder of the main branch as shown in Fig. 5.13. The output of the adder is the Dolby processed signal (generally called Dolbysed signal).

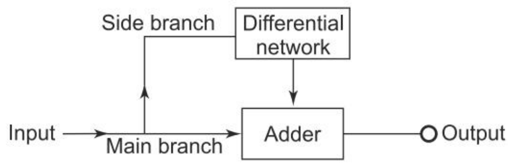


Fig. 5.13 *Coding of signal in Dolby method*

In playback, the differential network separates out the boosted signals in the side branch and subtracts them from the input signal as shown in Fig. 5.14.

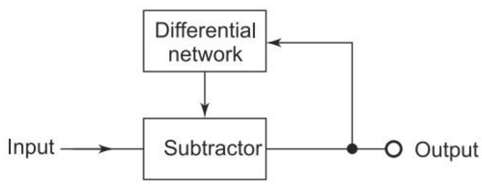


Fig. 5.14 *Decoding of Dolby signal*

The decoded output signal consists of the original signal with noise supressed as shown in Fig. 5.15.

The Dolby A system gives an improvement of 10 dB in the signal-to-noise ratio by up to about 5 kHz, increasing it as the frequency increases until it becomes 15 dB at 15 kHz. It then remains at 15 dB for frequencies higher than 15 kHz. This is shown by the curve *A* (full-line curve) in Fig. 5.15.

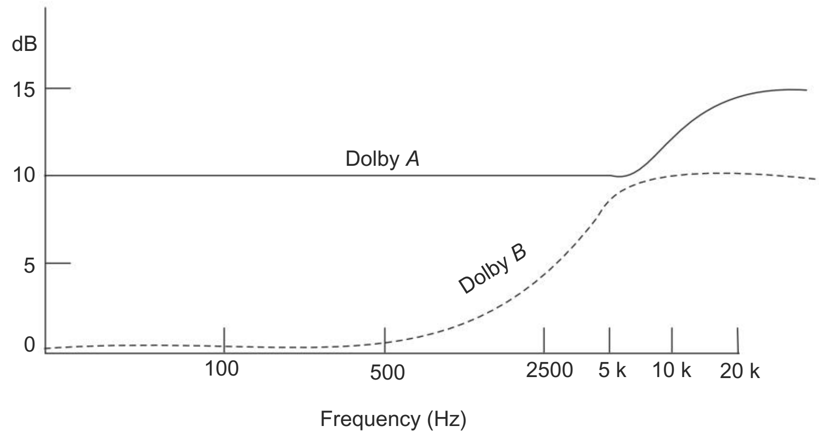


Fig. 5.15 *Signal-to-noise ratio improvement in Dolby A and Dolby B systems*

Dolby-B System A more simplified system than the Dolby A system has been developed for home use and is known as the Dolby-B system. In the Dolby-B system, the encoded signal covers a single frequency band from 500 Hz upwards. Signals below a specified threshold value in level are passed through a variable filter which increases the level by about 1 dB at 500 Hz and increases it progressively as the frequency increases until the increase becomes 10 dB at 10 kHz and then remains at 10 dB as shown by the dotted curve *B* in Fig. 5.15. Thus, the low-level signals in high-frequency range, where hiss and modulation noises are more prominent, are boosted by 10 dB.

The processed signal from the variable filter and the direct input signal are added. The adder gives the Dolby output. For playback, the signal is inverted and fed to the variable filter. The output of the filter goes to the adder to give a decoded output. Basic blocks of encoder/decoder are shown in Fig. 5.16.

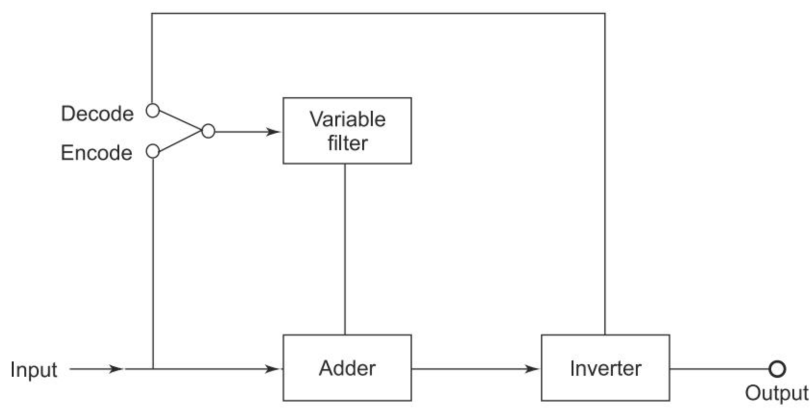


Fig. 5.16 *Basic Dolby-B system*

5.15 QUALITY OF SOUND ON TAPE

The performance of modern record/playback heads and of tapes is so good that magnetic recording can be made to reproduce sound with very high fidelity. The only precaution to achieve the goal of hi-fi recording is to allow for the limitations of the tape as mentioned below:

1. The recorder must have a well-designed tape transport system to avoid wow, flutter and rumble.
2. The tape guides, idlers and pinch rollers must be kept clean with methylated sprit and a fine dry cloth so that friction is minimum.
3. The bearings of motors, idlers and capstan must be kept lightly oiled. After oiling the bearings, all excess oil must be removed. The oil should not be allowed to creep up to the tape. Oiling may be done every six months.
4. Heads should be cleaned by spirit and cotton bud (It is a small wad of cotton wool fixed to the end of a small light stick and is available at chemists' shops for cleaning the nose and ears of babies).
5. Tape noise, a soft hiss, is essentially present in tape recordings. It can be minimised by
   (a) Using a good quality tape
   (b) Periodic demagnetisation of heads
   (c) Periodic demagnetisation of steel components such as guides
   (d) Using Dolby's method of amplifying soft notes to a pre-determined level before recording.

Digital Audio tape

Audio signals, be it speech, dialogues or music, are originally in analog form. For recoding these signals into digital format, the following process steps are followed.

First, the analog signals are sampled at more than then twice the highest frequency present in the audio signal. These samples, although discrete in nature, are still analog because the discrete levels are continuously varying in accordance with the audio waveform.

Second, the sampled levels are quantized into binary bits of constant level (1 or 0) for all samples and the sequence of binary bits, 1 and 0, changes from level to level. For example, if the sampled level is 5 and we use 8 bits to represent it then the sequence of binary bits would be 00000101. Similarly, for level 10, the sequence would be 00001010, and so on.

Third, 1 or 0 may be added for stop/start control. This code is then called *source encoding*. For transmission through a channel, some bits, called *parity bits*, may also be added. The final code is called *channel encoding*.

All the above steps of digitizing an audio signal are a part of analog to digital converter (ADC.)

The bits are in the form of electrical pulses, say high pulse for logic 1 (3.8 V typical for TTL) and low pulse for logic 0 (less than 0.8 V for TTL). Sometimes logic 0 is also represented by a negative pulse.

The electrical pulses in the sequence of bits are used to magnetize the head core. When a magnetic tape passes through the gap of a head, it is magnetized in the same sequence as the input bits in the digital signal.

The basic elements of a digital audio tape are illustrated in Figs 5.17 (a) and 5.17 (b).

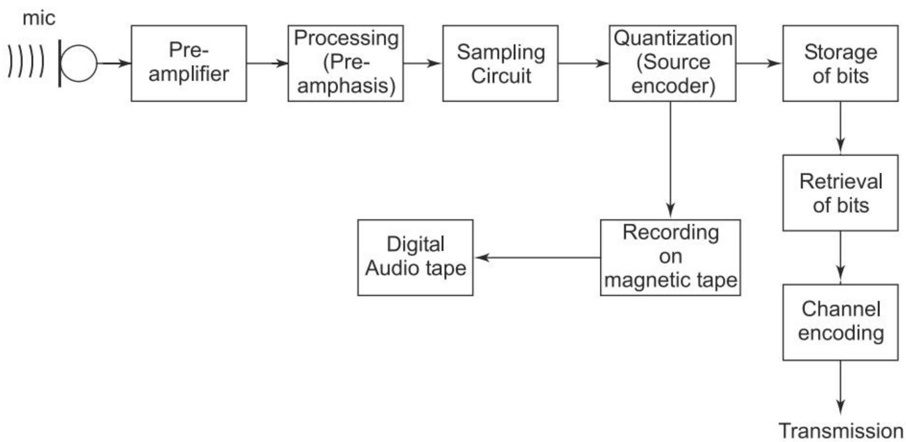


Fig. 5.17 (a) *Recording for digital audio tape*

(Three boxes on the rightmost side are for transmission)

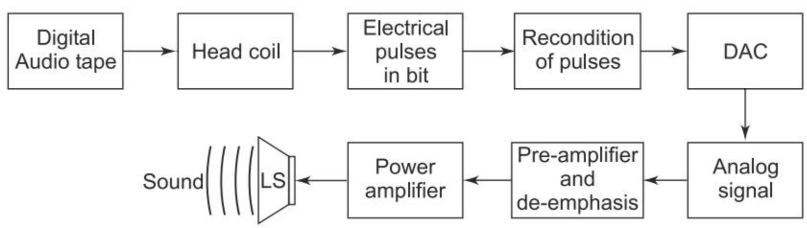


Fig. 5.17 (b) *Reproduction of audio signal from digital audio tape*

Advantages

1. Sound quality is superb as noise and cross-talk are eliminated.
2. All the processing functions can be software controlled, and these functions can be achieved by a remote control unit.
3. Any type of editing and dubbing can be easily done to create special features by processing the signals through a micro-controller chip.

4. The signals can be easily stored, delayed and multiplexed by a microprocessor chip, integrated in the system. Bits can be time multiplexed, or interleaved for other signals to get special effects.

Disadvantages

1. High bit rate requires high bandwidth for transmission of the recorded signal.
2. In an analog system, the quality degrades gradually, but in a digital system, quality remains excellent till a certain C/N ratio, but below that it degrades very rapidly.

The disadvantage of high bit-rate (needing high bandwidth) has been overcome by compression before transmission and decompression in the receiver. These techniques, known as JPEG (for images) and MPEG (for audio and pictures) have been described in detail in Appendix I.

SUMMARY

☞ Magnetic recording is based on the principle that iron oxide becomes magnetised on being brought in a magnetic field. Sound is converted into electrical signals (audio) by a microphone. These are amplified and fed to the coil of an electromagnet. The electromagnet, called 'head', has a small gap through which magnetic lines of force cannot pass. When a tape coated with iron oxide is made to pass across the gap, the lines of force get an easy path through the iron oxide which gets magnetised in accordance with the audio current through the electromagnet. Thus, sound is recorded on the tape in the form of a varying magnetic field.

☞ For reproducing, the tape is again made to pass through a similar gap (or even the same gap) and the varying magnetic field on the tape causes change of flux through the coil of the head, causing induced emf which is amplified and converted into sound.

☞ The highest usable frequency, f_m, of the recorded signal is given by $f_m = S/2G$, where S is the speed of the tape and G is the gap-width. The output decreases rapidly as f exceeds the value given by the above formula, and becomes zero if a frequency, $f_n = S/G$, where f_n is the frequency of null or zero output point.

☞ Due to non-linearity of magnetisation curve, ac biasing is used for the head coil in the recording mode. This removes distortion due to non-linearity.

- ☞ Tape needs to be moved very steadily through a tape transport mechanism. Tape transport mechanism uses ac synchronous motor, properly positioned guides, capstan with press roller and flywheel.
- ☞ Variation in speed of motor up to 6 Hz gives rise to a distortion called 'wow' and variation of more than 6 Hz and up to about 100 Hz is heard as flutter. Motor vibrations are reproduced as rumble sound. Tiny irregularities in tape cause a hissing noise.
- ☞ Pre-emphasis of high notes and de-emphasis of low notes before recording takes care of noise and saturation. The process is reversed during playback to restore the originality of sound. This is called equalisation.
- ☞ Tape recording has many advantages over disc recording, viz., there are no vibrating parts, editing and dubbing is easy, recording can be monitored simultaneously, can be used for immediate playback and the same tape can be used again and again for recording different programmes. These advantages make tape recording and playing very popular.
- ☞ Dr Ray Dolby introduced a novel system for providing 10–15 dB improvement in signal-to-noise ratio. Weak signals, which are less than 40 dB were boosted before recording in 4 bands (16–80 Hz, 80–3000 Hz, 3 kHz and above and above 9 kHz) to bring them at 40-dB level. During playback these were reduced to the original level. Noise was also reduced. This gives good signal-to-noise ratio.
- ☞ Sound can be recorded in digital form on tapes, giving all the advantages of a digitizing baseband signal, like reduction in noise and addition of value-added features

Review Questions

1. Explain the principle of magnetic recording and reproduction with the help of neat sketches.
2. Draw a block diagram of a magnetic recording and reproducing system. Describe the function of each block.
3. What is the relationship between gap-size, tape-speed and frequency of the audio signal? Explain the

importance of relationship to get optimum output for audio bandwidth.

4. Mention standard tape speeds and gap-sizes.
5. Explain the necessity of ac biasing. How is it achieved?
6. Write short notes on
   (1) Tape and tape materials
   (2) Recording, playback and erase heads
7. Explain the following:
   (1) Dolby system of noise reduction
   (2) Equalisation
8. Describe with the help of a neat sketch The tape transport mechanism. What part is played by capstan, press roller, flywheel and guides? What is done to keep the speed of tape transport steady?
9. Explain the cause of wow, flutter, rumble and hissing noises in a tape system. How are these reduced?
10. What are the advantages and disadvantages of tape recording over disc recording?
11. Explain with the help of a block diagram how sound is recorded in digital format on a magnetic tape. What are its advantages and disadvantages over analog system?
12. Explain the need of compression (Appendix I)
13. Describe briefly MPEG-1 techniques (Appendix I)
14. Describe briefly with the help of a block diagram have audio signal is reproduced from a digital tape.

Short-Answer Questions

1. Write down basics of the principle of reproduction of magnetic recording of sound.
2. Write down basics of the principle of reproduction of sound from magnetic tapes
3. Why is soft iron used for the head core?
4. Why is there a small gap in the head core?
5. Why is iron oxide, and not soft iron, used on the tape?
6. Why is the output in playback optimum when the gap width is equal to half wavelength of the sound frequency
7. What will happen if the speed of tape is (i) too high (ii) too low?
8. What is the difficulty in providing dc bias for recording?
9. What is the advantages of ac biasing?
10. What is the role of capstan in tape transport mechanism?
11. Explain briefly the role of pre-emphasis and de-emphasis
12. What is the advantage of Dolby system?
13. What is the difference between a sampled signal and a digitized one?
14. Why are error detecting and correcting bits are not required in digital recording of sound?
15. Why does head gap allow the magnetic lines of force pass though the tape but not through the air inside gap?

Multiple-Choice Questions

1. What is the name given to the length of tape magnetized by one cycle of audio signal?
   (a) Bar magnet (b) wave length
   (c) tape length (d) head gap
2. What will affect the optimum output if gap-size is increased?
   (a) Frequency of the audio
   (b) Length of tape (wavelength)
   (c) Magnetism
   (d) Signal voltage of the tape
3. How does the duration of programme recorded on a tape depend on tape speed?
   (a) Higher the tape speed lower the duration.
   (b) It does not depend on tape speed.
   (c) Lower the tape speed, greater is the duration.
   (d) It depends on gap-size alone.
4. What is the standard speed of tape for domestic cassette?
   (a) 38 cm/s (b) 10 cm/s
   (c) 8.76 cm/s (d) 4.75 cm/s
5. Why is ac biasing required in magnetic tape recording?
   (a) To increase output without distortion
   (b) To decrease output
   (c) To increase duration of programme recording
   (d) To conserve fidelity
6. Which one of the following is not a part of the tape-transport mechanism?
   (a) Capstan (b) Pinch roller
   (c) flywheel (d) pre-emphasis
7. Which one of the following is not a part of the Head?
   (a) Core of soft iron
   (b) Coil
   (c) Iron oxide
   (d) Gap
8. Wow distortion occurs when
   (a) Speed of tape varies from 0.5 Hz to 6 Hz
   (b) Speed of tape varies to 50 Hz
   (c) Motor body vibrates
   (d) Minute irregularities in tape coating
9. When is pre-emphasis done?
   (a) Before recording
   (b) After recording
   (c) Before detection
   (d) After detection
10. When is de-emphasis done?
    (a) Before recording
    (b) After recording
    (c) Before detection
    (d) After detection
11. In a Dolby system, where is pre-emphasis done?
    (a) At high frequencies only
    (b) At low frequencies only
    (c) Throughout the audible range
    (d) Never done
12. For an audio signal varying from 20 Hz to 15000 Hz, sampling will be done at
    (a) 20 Hz (b) 40 Hz
    (c) 15000 Hz (d) 30000 Hz
13. Channel encoding is done
    (a) in tape-transport mechanism
    (b) at the time of playback

(c) before transmission

(d) after transmission

14. Which of the following noises is not concerned with tape-transport mechanism?

(a) Wow (b) Flutter

(c) Rumble (d) Transit

15. In digitized recording and reproduction of audio signals, when are the binary pulses produced?

(a) In sampling

(b) In quantization

(c) In playback

(d) In channel

16. When is the erase head in operation?

(a) After recording

(b) Before recording

(c) In the playback section

(d) Nowhere

Numerical Problems

1. Calculate the recorded wavelength on tape, if the speed of the tape is 4.75 cm/s and the highest audio frequency to be recoded is 9500 Hz.
2. Find the gap width for maximum output in Q.1.
3. Find the highest frequency for maximum output if gap-width is 6 μm and speed is 9.5 cm/s.
4. If tape speed is increased to 19 cm/s, what would be to the highest frequency in Q.3.
5. What would be the tape speed in Q. 3 above if the highest frequency to be reproduced is 5 MHz? What length of tape will be required for recording a programme of 1 hour? Is it feasible?

Answers

Short-Answer Questions

1. In magnetic recording, sound-pressure variations are recorded in the form of elementary magnets whose length and strength depend on the audio signal. The audio signal is passed through a coil wound on a core of soft iron. The core called head becomes the electromagnet whose magnetic strength varies in proportion to the audio signal. The core has a minute gap. When a tape, coated with iron oxide, is made to pass across the gap, the lines of force get an easy path through the iron oxide, which then is transformed into permanent elementary magnets. Thus, sound is recorded on the tape.
2. The recorded tape is made to pass through the minute gap of head. Magnetism on the tape produces a varying magnetic field in the head's

coil. An emf is induced in the coil due to em induction. The value of the induced emf is proportional to the rate of change of magnetism on the tape which, in turn, is proportional to the audio signal. Thus, the audio signal is reproduced.

3. Soft iron produces temporary magnetism, which varies with variations in the input audio signal.
4. The small gap contains air which has high reluctance for magnetic field. When the gap passes over an iron oxide material coated on the surface of a tape, the magnetic field gets an easy path through the iron oxide which becomes a permanent magnet whose strength depends on the magnetic field in the soft iron. Thus, tape gets magnetised as per audio signal.
5. Iron oxide becomes a permanent magnet according to the audio signal, while soft iron achieves varying magnetism, which changes instant to instant in accordance with the audio current flowing through it. Thus, the audio current flowing through the soft iron is recorded permanently in the iron oxide in the form of elementary magnets. Had we used soft iron on the tape, the magnetism would have vanished after the tape cames out of the gap.
6. When the gap-width is equal to λ, it allows full two magnets of equal and opposite polarity on the tape to be impressed through the coil. Two magnets of equal and opposite polarity cancel the magnetic effect through the coil, resulting in 'null' output. When the gap-width is equal to $\lambda/2$, it is fully utilized to produce current though the coil.
7. If the speed of tape is too high, a programme of only small duration will be recorded because a single cycle of audio wave will occupy a long space on the tape. If the speed is too low, magnetic cycles will occupy small space, reducing resolution between adjacent cycles. The high audio frequency cycles will merge with each other, and will cause severe distortion.
8. If no bias is applied, small signals will not be recorded due to knee in the current magnetisation curve. This effect can be removed if a dc bias is applied. But a dc bias causes saturation for large values of signal. This problem is solved by providing ac bias.
9. Direct current bias saturates magnetisation and hence high value signals cannot be recorded without severe distortion. This problem is solved if, instead of dc bias, ac biasing is used Alternating current of constant amplitude and constant frequency (100 kHz) is passed through the head coil. The audio signal is superimposed over ac pulses. The output is obtained on both sides of the magnetising curve. This way high audio levels can be recorded without distortion. Effect of knee voltage will be on the ac bias pulse and not on the audio. Output will be higher.
10. Capstan is a spindle, machined accurately, and pulls the tape past the heads. The tape is pressed against the capstan by means of a rubber-covered pinch roller.
11. At high audio frequencies, intensity is low and noise is high, so much so that signal-to-noise ratio becomes less than 1. If low intensi-

ty of higher frequencies is pre-emphasised before induction of noise in the recording, signal-to-noise ratio will improve up to 10 dB.

In playback, pre-emphasis has to be undone (neutralized) by de-emphasis. But, signal and noise separated by 10 dB, both are deemphasised and hence signal-to-noise ratio obtained by pre-emphasis is retained, and at the same time fidelity (low energy at higher frequencies) is preserved.

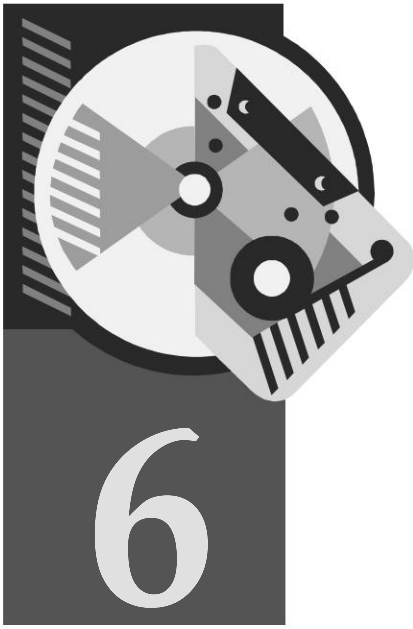
12. In normal pre-emphasis, it is presumed that weak intensity is present only in high frequencies. This is not always the case in music. Even low audio frequencies may be so soft as to make their intensity comparable with noise. Conventional pre-emphasis and de-emphasis will not solve this difficulty. It is solved by the Dolby system, in which signals which are 40 dB or higher pass direct without any emphasis. Lower level signals are passed through the boosting stages which emphasise such signals by 10–15 dB before recording (i.e., before the induction of noise). Boosting is done in four bands (below 80 Hz, 80 Hz to 2999 Hz, 3000 Hz and above and 9000 Hz and above). These bands pass through appropriate filters. All these outputs are added in a side branch, called differential network. The output of the differential network goes to another adder in the main branch where direct signals and processed signal are added. The final output is free of noise. In playback, the output is passed again though a differential network. The deprocessed signal then passes though a subtractor, removing the effect of differential network which was used while recording.
13. A sampled signal has discrete amplitudes varying continuously and so it is an analog signal. When these analog variations are converted into binary bits by ADC, the signal becomes digitised. The digitised signal consists of a sequence of binary bits having only two levels of amplitude. (logic 1 or logic 0).
14. Errors in binary pulses are produced during transmission and hence error detecting and correcting codes are required only when transmission of the recorded signal has to be done.
15. Reluctance of air is high, while reluctance of the magnetic martial coating on the tape is low.

Multiple-Choice Questions

1. (b)	2. (a)	3. (a)
4. (d)	5. (a)	6. (d)
7. (c)	8. (a)	9. (a)
10. (d)	11. (c)	12. (d)
13. (c)	14. (d)	15. (b)
16. (b)		

Numerical Questions

1. (5 um) 2. (2.5 um)
3. (791 Hz) 4. (15833 Hz)
5. (60 m/s, 216 km, No (this pertains to video signals.)



Optical Recording

6.1 TYPES OF OPTICAL RECORDING OF SOUND

Optical recording of sound is of two types:

1. Recording on Photographic Films This is done by converting audio signals into variations of light intensity falling on the film. Such recording of sound appears in the form of a sound track, 2.5-mm wide near one edge of the movie film. Intensity of light from a slit is made to vary in accordance with the sound pressure variations. When this varying light falls on the edge of the main film, a photograph of the varying light intensity is recorded in the same way as variations of light from a picture are recorded.

2. Recording on Compact Discs This is done with the help of laser beams, made ON and OFF by digitised audio signals. These beams fall on a photoresist material on a rotating disc and cause pits of varying width and fixed depth and thus record signals in binary form, flats and pits making 1s and 0s, respectively. The disc is known as *compact audio digital disc* or simply *compact disc*.

The film recording of sound is described in Sections. 6.2 and 6.3, and the compact disc system in Sections 6.4 to 6.7.

6.2 METHODS OF OPTICAL RECORDING OF SOUND ON FILM

There are two methods for varying the intensity of light in accordance with the sound pressure variations—1. variable-density method, and 2. variable-area method. These two methods are discussed below.

Variable-Density Method

In this method, sound is picked up by a microphone, and converted into electrical signals which are amplified. Audio output of the amplifier is fed to the anode of a special type of vacuum tube, called an AEO lamp. The lamp contains a little quantity of helium gas. The anode gets high dc voltage (called HT) in series with the audio voltage as shown in Fig. 6.1. The filament of the lamp is connected to a low dc voltage (called IT).

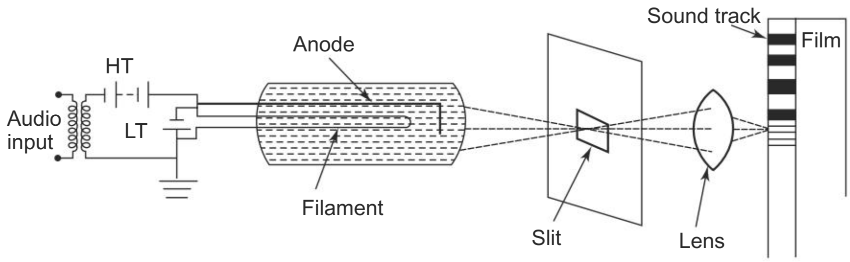


Fig. 6.1 *Variable-density method of optical recording*

The intensity of light coming out from the lamp varies in accordance with the audio signal. This varying light passes through a slit and a focussing lens. The focussed light falls on a moving photographic film where the image is recorded in the form of bars of varying density and distance on the film.

Variable Area Method

In this method, light of constant intensity falls on a slit. The area of the slit opened for this light varies in accordance with the variation of sound pressure. Hence, the light falls on the variable area on the soundtrack edge of the film. Thus, the area which is bright to light varies. The area of the slit is made variable with the help of a mirror or galvanometer as illustrated in Fig. 6.2.

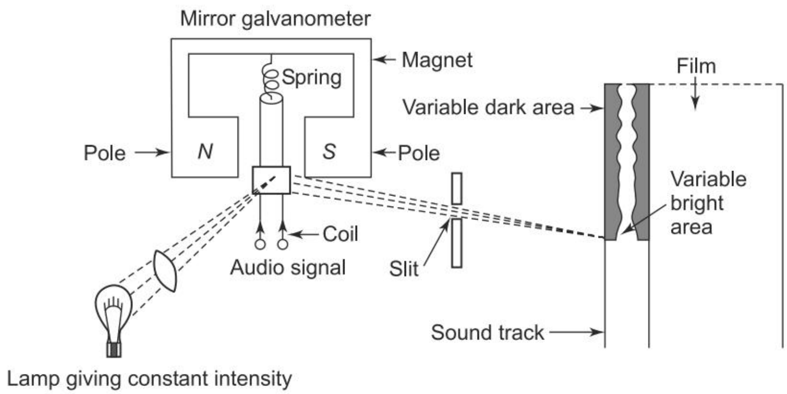


Fig. 6.2 *Variable area method of optical recording*

Sound is first converted into electrical (audio) signals by a microphone. The audio signals are amplified and reach the coil of a mirror galvanometer. The current-carrying coil is placed in a magnetic field and hence, deflects in accordance with amplitude of the audio signal. A mirror is attached to the coil assembly. The mirror also deflects. Light from a lamp, duly focussed by a lens system, is made to fall on the mirror. The light reflected from the mirror goes to a narrow slit. When the mirror deflects, the slit area exposed to the light changes, i.e., the slit is partially illuminated. The extent to which the slit area is illuminated depends on the extent of the deflection of mirror and hence on the strength of audio current. The light from the variable area of the slit falls on the soundtrack edge of the film and is recorded in the form of a photograph of variable area.

6.3 REPRODUCTION OF SOUND FROM FILMS

A sharply focused narrow beam of light is made to fall on the soundtrack of the film. As the film moves, light passing through bright and grey shaded portions in case of a variable-density record and through bright portions of variable area in case of a variable-area record, falls on a photocell which converts this light into electrical signals. In both types of recording (variable density as well as variable area), the quantity of light falling on the photocell will depend on the strength of the recorded audio signal. The output of the photocell will, therefore, be an audio voltage which can be amplified and fed to a loudspeaker which finally converts it into sound. The principle of reproduction is illustrated in Fig. 6.3.

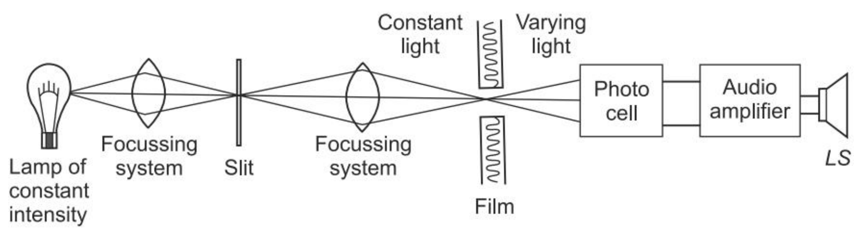


Fig. 6.3 *Reproduction of sound from films*

The motion of the film is illustrated in Fig. 6.4.

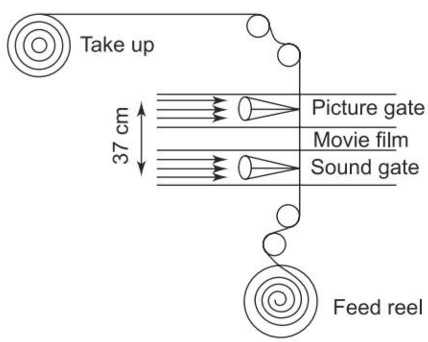


Fig. 6.4 *Picture and sound gates for focusing light on the movie film*

The picture record and its corresponding sound are not placed side by side on the film, but the sound record is in advance by about 37 cm of the corresponding picture record. The reason for this is that each picture is projected twice on the screen to eliminate flicker effect and hence the picture part of the film has to move in jerks, whereas the sound record must move at a uniform speed.

6.4 MODERN METHOD OF RECORDING OF SOUND FOR MOVIE FILMS

In commercial film production, sound may be recorded on a quarter-inch portable magnetic tape. The tape may be played and the picture may be projected as usual on the screen through a projector, as discussed in Section 6.3. For this, synchronisation between the film and tape would be necessary. Hence, a synchronising signal is also recorded on the magnetic tape, related to the speed of the film camera. The synchronisation will allow the locking of sound to picture in post-production. Sound tracks from field recordings are copied on to sprocket-hold magnetic tapes. Sprocketed tracks can be moved relatively in time with the help of mechanical links between multiple sprockets.

More recently digital compact discs (based on laser beam) have been used for recording sound, and synchronising the sound tracks on discs results in audio along with separately projected picture simultaneously. These optical discs have been described in detail in Sections 6.5 through 6.7. Earlier, infrared light was used in these discs, but modern discs use red light and even blue light. These discs are called *digital video discs*. The disc used for audio alone is called *DVD-audio*. The DVDs are described in detail in Chapter 14.

6.5 COMPACT DISC

As digital circuits offer almost complete immunity to noise, efforts were made to record sound in digital form. These efforts resulted in laser discs. Philips and Sony started working on a development project to record digitised sound and their efforts gave birth to the compact disc system.

6.6 OPTICAL RECORDING ON DISC

A compact disc contains a record of digital audio signals in the form of pits of 1 μm depth and 0.5 μm width but of variable length (varying from 1 μm to 3 μm. The pitch of the tracks (separation between adjacent tracks) is 1.6 μm. Dimensions are shown in Fig. 6.5. Width has not been shown in the two-dimensional figure.

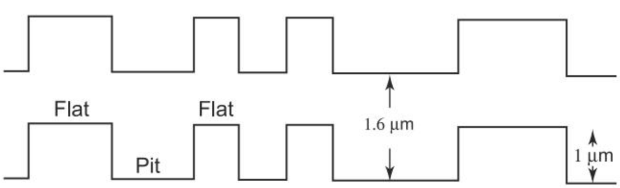


Fig. 6.5 *Dimension of pits on CD tracks*

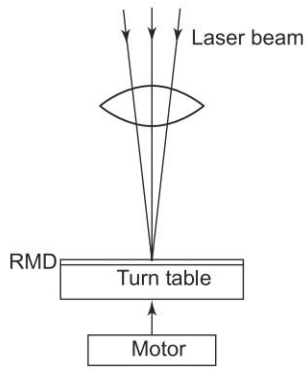


Fig. 6.6 *Laser beam incident on the recording disc*

Recording is done on a Resist Master Disc (RMD) with the help of a powerful laser beam as shown in Fig. 6.6. The laser beam is modulated by a digitised audio signal. The audio signal is sampled at the rate of 44.1 kHz. The quantum levels pertain to 16 bits. Thus the bits per second are 705,600. To these bits are added bits for correction, controlling and modulating the signals. The digital input signal is so encoded that whenever a 1 appears in the signal, it causes transition from OFF to ON. These pulses are then made to modulate the laser beam in the ON-OFF mode. The block diagram of the recording system is shown in Fig. 6.7. The laser beam is focused, and the sharply focused beam is incident on the master disc.

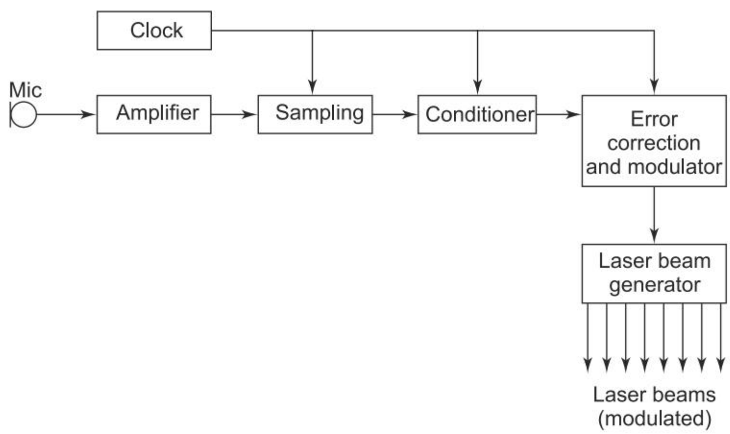


Fig. 6.7 *Block diagram of recording*

Master Disc The master disc, shown in Fig.6.8, is the original disc on which audio signal is first recorded.

The master disc is made of an optically ground glass disc. The glass is polished and is spotlessly clean. It is coated with a photo-resist compound. The coating is

0.12 µm thick and is distributed uniformly. This is known as *Resist Master Disc* (RMD) or, simply *master disc*.

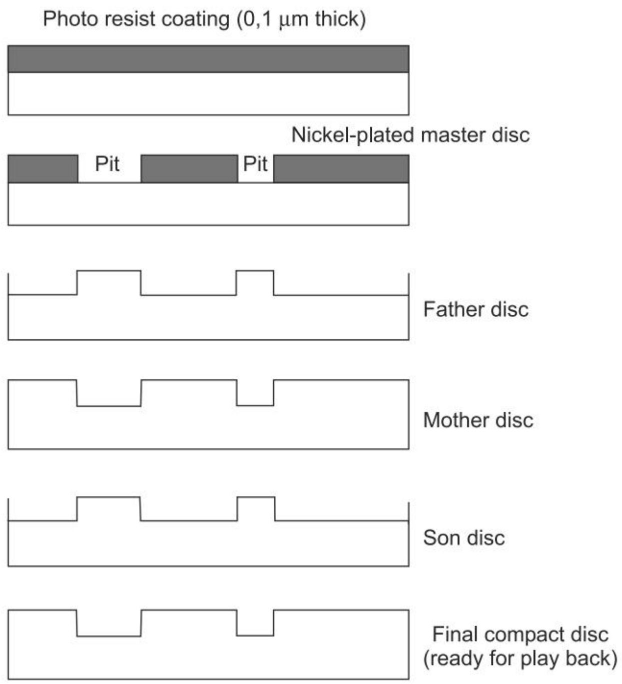


Fig. 6.8 *Preparation of compact disc*

When the modulated laser beam strikes the master disc, it reacts with the photo-resist. The disc is now developed by a process akin to photography. This results in a microscopic-sized pattern of *pits* and *flats*. The developed master disc is coated with silver to make it electrically conductive. Flats are also called lands.

Father Disc The next step is nickel plating. After plating, the nickel is peeled off the master disc, and then it is called 'father disc'. It is a negative replica of the glass master disc, shown in Fig.6.8.

Mother Disc The father disc is again plated and removal of the plating produces a mother disc which is identical in form with the master disc. Generally, ten mother discs are obtained from a single master disc. Mothers are inverted and cannot be used for producing final discs.

Son Disc or Stamper The mother discs are plated (the third plating in the process) and the plating when removed gives a son disc or stamper which is identical with the 'father disc'. Several sons can be obtained from a single mother. A son disc is also called a negative nickel-plated stamper. The father, the mother and the son (stamper) discs are all produced in the same nickel bath.

Consumer Disc or Final Compact Disc Consumer discs for playback are obtained by pressing on the stamper son disc. About 10000 discs can be moulded from one stamper. These discs are positive discs. A consumer disc is made of polycarbonate. A thin layer of aluminium is added to the disc to make it reflective. The consumer disc is protected by adding a transparent layer of lacquer. Recording is done from the centre towards the edges. A hole is punctured in the centre of the disc. It is then packed in a plastic case. Details of a compact disc are given in Table 6.1.

Table 6.1 *Details of a compact disc*

Diameter	12 cm
Thickness	1.2 mm
Rotation	Anti-clockwise
Motion of pick up	Linear from centre towards edges
Velocity of scanning	1.2 to 1.4 m/s
Speed	200 – 500 rpm
Playing time	60 minutes + error correction time 20 minutes
Pit width	0.5 μm
Pit depth	1 μm
Pit length	0.8 to 3.6 μm (varying)
Beam spot	1 μm diameter
Tracks pitch	1.6 μm
Pick up	Solid state laser using Aluminium gallium arsenide (a semiconductor)
Transducer	Photo-diode
Frequency of sampling	44.1 kHz
Quantisation	16 bit
Number of levels	65,536 per channel
Bit rate in M bits/second	4.3218
Error correction bits	3548

6.7 PLAYBACK PROCESS

A laser beam, produced by a solid state laser of semiconductor aluminium gallium arsenide (780 μm wavelength) is made incident on the compact disc through a half-silvered mirror. The mirror allows the beam to pass through itself but does not allow the returning beam to pass. The returning beam is reflected from the aluminium flat surface and represents the digit one. There is only a little reflection from a pit, and it represents 0. Thus, the returning laser beam is the replica of the original laser beam modulated by binary digits of audio signals. (The optics of the reflection process is shown in Fig. 6.9.)

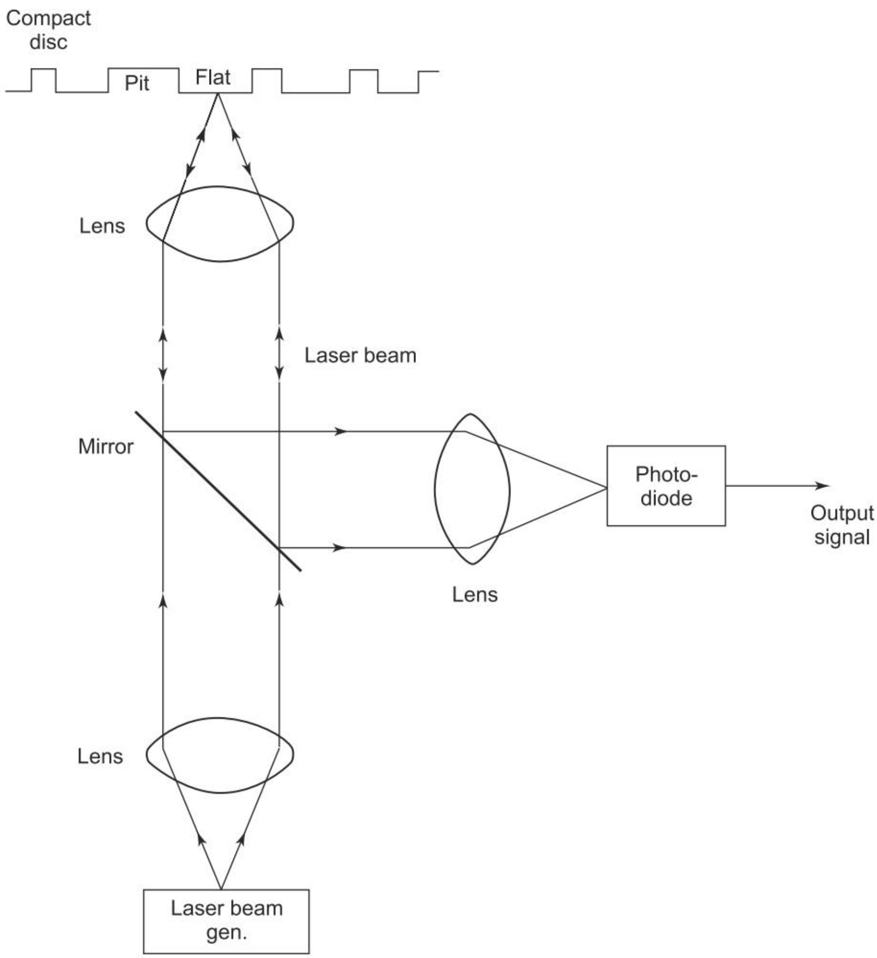


Fig. 6.9 *Laser beam reflected from compact disc*

Light is not reflected from the pit and is fully reflected from the flat surface. Thus, binary digits are reproduced when this ON-OFF reflected light falls on a photosensitive diode. The digital output of the diode is processed and converted into the original analogue signal by using a digital to analog converter. The block diagram of detection or decoding circuit is shown in Fig. 6.10.

Control signals allow any combination of tracks to be played in any sequence with the help of a keyboard. Also, a display of the text is provided to monitor the music being played.

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

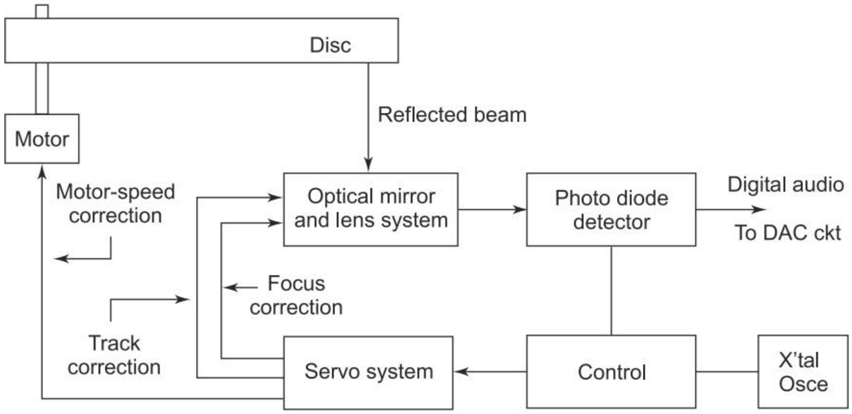


Fig. 6.10 *Basic block diagram of detection circuit*

As it is a very high-fidelity system, it incorporates stereo sound. Stereo signals are multiplexed before modulation of the laser beam. After detection, these signals are demultiplexed to give two separate channels of the stereo system.

Scanning of the tracks by a laser beam is done from the centre proceeding towards the edge. For this purpose, the disc is rotated and the laser is moved from centre to edge. As the circumference of the outer spirals is larger than of the inner spirals, the track speed is made constant (constant linear velocity) by varying the rotational speed of the disc from 500 rpm at the centre to 200 rpm at the outermost edge. Thus, at the centre, one rotation is accomplished in 1/500 min, while at the edge, in 1/200 min.

The scanning speed is about 1.2 m/s. and the total track length is 6 km. This gives a playing time of 60 minutes plus about 20 minutes time for error correction. Frequency response of a compact disc is from 20 Hz to 20 kHz and signal to noise ratio is 90 dB.

6.8 COMPARISON OF COMPACT DISCS AND CONVENTIONAL (GRAMOPHONE) DISCS

A comparison of the performance characteristics of a compact disc and the conventional analogue disc (gramophone) is shown in Table 6.2.

Table 6.2

Item	*Compact disc*	*Conventional hi-fi disc gramophone disc*
Channels	2 or 4	2
Frequency response	20 to 20000 Hz ±0.5 dB	40 to 15000 Hz ± 1 dB

Contd.

Contd.

Dynamic range	90 dB	55 dB
Signal to noise ratio	90 dB	60 dB
Channel separation	better than 80 dB	30 dB
Harmonic distortion	0.05%	0.2%
Wow and flutter	Absent	0.03%
Stylus	Laser beam	Mechanical stylus

Table 6.2 shows that a compact disc is very superior to a conventional disc, and approaches very near to ideal fidelity and is much better than the conventional Hi-Fi Vinyl discs (records).

Advantages of compact discs

1. As it is covered by transparent plastic or transparent lacquer, the tracks and recordings remain safe and are not affected by dust, grease and scratches. A compact disc is immune to surface contamination.
2. Signal-to-noise ratio is high, as high as 90 dB, an improvement of 30 dB over a high- fidelity gramophone disc.
3. Dynamic range is high, as high as 90 dB, an improvement of 35 dB over a high-fidelity gramophone disc.
4. Channel separation is high, as high as 80 dB, an improvement of 50 dB over gramophone discs.
5. Wow does not exist.
6. Flutter does not exist.
7. Total distortion is low.
8. Frequency response is excellent and covers complete audio range from 20 Hz to 20 kHz within only ±0.5 dB.
9. Size is quite small.
10. Drop-outs up to 2.5 nm of disc (4000 bits) are not felt due to error-correction codes, and distortions due to scratches are also automatically corrected.
11. As the audio signals are converted into binary digits, the system has all the advantages of digital systems over analog ones, for example,
    (i) pulses can be regenerated and hence, any noise introduced is automatically eliminated, and
    (ii) no equalisation is required as pulses are free from noise.

Disadvantage Earlier, the cost of a compact disc was more than the high-fidelity analogue disc. Recently, the cost has been substantially reduced due to popularity of the new video-audio discs. Compared to tapes, it has another disadvantage; the recording cannot be erased, and hence fresh recording cannot be done on the same disc without involving a complex and costly process.

SUMMARY

- The optical method is used to record sound on films and on compact disc.
- Recording of sound on motion picture films is done by two optical methods: (1) variable-density method, and (2) variable-area method.
- **In the variable-density method**, audio signals change current in a helium-filled radio-valve (AEO lamp). Helium glows and the intensity of light emitted is in accordance with the audio signals. It is made to focus on the soundtrack edge of the motion-picture film through a slit and is recorded there as a variable density of light.
- **In the variable-area method,** audio current deflects the mirror of a mirror galvanometer. Light of constant intensity reflected from the mirror passes through a narrow slit. The area of the slit exposed to light depends on the position of the mirror and hence, on the audio current. Thus, light coming out from the slit varies in area and is recorded on the soundtrack of the motion-picture film as photographs of dark and bright variable areas.
- For reproduction, the light, duly focused, is passed through the soundtrack part of the film. The light of varying intensity comes out from the soundtrack and falls on a photocell which converts the varying light intensity into audio signals. These are then amplified and fed to the loudspeaker which finally converts them into original sound.
- A compact disc is prepared by making a laser beam of the infrared light incident on a photo-resist material coated on the disc. The audio signal is sampled and coded in digital form. The digital signal is used to make the laser beam ON and OFF. It forms flats and pits on the photo-resist material. Thus, the disc contains patterns of flats and pits. For reproduction, the laser beam is reflected in full from the flats and is not reflected from the pits. This beam, modulated by flats and pits, produces an audio digital signal with the help of a photo-diode. This signal is processed and is converted into an analog signal to be reproduced by a loudspeaker.

Review Questions

1. Explain how sound can be recorded on a film and reproduced.
2. Describe with the help of a neat diagram the working of the variable-density method of optical recording of sound on films.
3. Explain the variable-area method of optical recoding of sound. Draw a neat sketch of the apparatus used.
4. With the help of a neat sketch, explain the principle of reproduction of sound from the soundtrack of motion-films. Explain why the soundtrack runs in advance by about 37 cm of the corresponding picture record.
5. Explain why each picture frame is projected twice through a projector.
6. Explain the recording of sound on a disc by a laser beam.
7. Describe the preparation of a compact disc.
8. Explain how sound is reproduced from a compact disc.
9. Compare the performance characteristics of compact disc with conventional hi-fi disc.
10. Discuss the advantages of the compact disc.

Short-Answer Questions

1. How is recording of sound on photographic films done?
2. How is sound reproduced from films?
3. How is recording on compact discs done?
4. Why does a pit represent logic zero?
5. How do pits and flats allow recovery of baseband signal in a compact disc?
6. How is the reflecting property of an optical disc increased?
7. How is an optical disc protected from dust, grease and scratches?
8. What are the further developments on CD?

Multiple-Choice Questions

1. How much wide is the width of soundtrack on a movie film ?
   (a) 1 mm (b) 1.5 mm
   (c) 2 mm (d) 2.5 mm
2. Why is a picture frame projected twice on the screen ?
   (a) To strengthen the image
   (b) To have good focussing
   (c) To eliminate flicker
   (d) To have good resolution
3. From where is the laser beam not reflected in video discs ?
   (a) From a flat
   (b) From a pit

(c) From a land
(d) From the glass

4. What is the rotational speed (rpm) of the compact discs at the centre?
   (a) 200 (b) 500
   (c) 800 (d) 1000
5. Why does the surface of a compact disc not wear out ?
   (a) Because the sensor is electrical
   (b) Because the sensor is mechanical
   (c) Because the sensor is electronic
   (d) Because the sensor is optical
6. Highest capacity in an optical disc can be obtained by using a laser beam of
   (a) green light (b) red light
   (c) blue light (d) infrared light

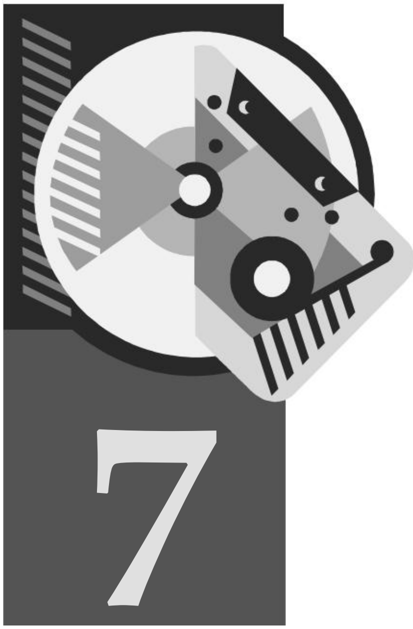
Answers

Short-Answer Questions

1. Recording of sound on photographic films is done by converting audio signals into variations of light intensity falling on the film. Such recording of sound appears in the form of a soundtrack, 2.5 mm wide near one edge of the movie film.
2. A sharply focused narrow beam of light is made to fall on the soundtrack of the film. As the film moves, light passing through the film falls on a photocell which converts the varying intensity of light into electrical signals. The output of the photocell will be audio voltage.
3. Recording of sound on compact discs is done with the help of a laser beam of infrared light. The beam is incident on a photo-resist material on a rotating disc and form pits of varying length and fixed depth. Thus, the signal is recorded in binary form, flats and pits making logic 1 and logic 0, respectively.
4. A pit does not reflect the light and hence it represents logic zero.
5. Light incident on the disc is reflected by flats and is not reflected by the pits. Thus, the reflected light represents the sequence of 0s and 1s. Sound in digital form is thus reproduced which is converted into analog form by a digital to analog converter.
6. The reflecting property of the disc is enhanced by adding a thin layer of aluminium on to the disc.
7. The disc is covered by a transparent plastic or lacquer, the tracks and recordings remain safe and are not affected by dust, grease or scratches.
8. Video was also recorded on CD, and it was called VCD. Later, wavelength of the laser beam was reduced from infrared light to red light, increasing capacity. It was then called DVD.

Multiple Choice Questions

1. (d) 2. (c) 3. (b)
4. (b) 5. (d) 6. (c)



Audio Amplifiers

7.1 TYPES OF AUDIO AMPLIFIERS

An audio amplifier is a device used to amplify audio signals of frequency range from 16 Hz to 20 kHz.

There are two types of audio amplifiers:

1. Voltage amplifier
2. Power amplifier

Voltage amplifiers are used as pre-amplifiers, buffer amplifiers (or intermediate amplifiers) and driver amplifier. Their main function is to amplify the audio signal voltage in stages, so that finally the driver amplifier gives an output voltage sufficient to reduce the resistance of the power amplifier and hence, to drive it to give power amplification.

The final amplification stage always involves a *power amplifier* which feeds audio power to loudspeakers for conversion of the electrical signals into sound waves. For low power outputs, it can be a single transistor stage, but for high power outputs, two transistors are used generally in a push-pull configuration.

Amplifiers are also classified as Class A, Class B and Class C amplifiers. Class A amplifier is so biased in a way that it gives an output for the whole cycle of the input signal (+ve side as well as –ve side). Its efficiency is low (less than 25% for RC coupling), but distortion is negligible. In audio voltage amplifiers Class A biasing is used, because distortion is the dominating consideration for designing audio amplifiers. Claas A amplifiers work on the linear portion of the characteristic curve to maintain good fidelity of the output with the input signal. The audio power amplifiers when used as single transistor have also to be used as Class A amplifiers. Class B amplifiers are biased in a way that they give output for half the cycle only. Their efficiency is high (78%) but they produce severe distortion as only half the cycle is reproduced. However, when used as push-pull amplifiers, using two transistors, the full cycle can be reproduced, and so they can be used as audio power amplifiers.

Class C amplifiers are biased to give output for less than half the cycle. They are most efficient (about 90%), but they produce maximum distortion. They are not used as audio amplifiers, but are used as radio frequency amplifiers and oscillators, where the load consists of a tuned circuit. A tuned circuit can absorb the distortion by flywheel action. The shunt capacitor between base and emitter does not affect the radio frequencies, because the capacitor becomes a part of the resonant load (for example, if a tuned circuit requires 100 pF capacitor in parallel with the coil, and if the value of the shunt capacitor is 20 pF then we can use an external capacitor of 80 pF across the coil which would give 100 pF including base-emitter shunt capacitance value of 20 pF).

For distortion-free amplifier circuit, generally *RC* coupling is used, which is shown in Fig. 7.1. The frequency response of such an amplifier has been discussed in detail in Chapter 8 (Section 8.6). There are two roll-offs; one at lower frequencies due to series coupling capacitor, and the other at higher frequencies due to shunt capacitance. Thus, practically, the whole audio frequency range of 16 to 20000 Hz is well covered by the *RC* coupled circuit. Examples 7.1 and 7.2 clarify this aspect.

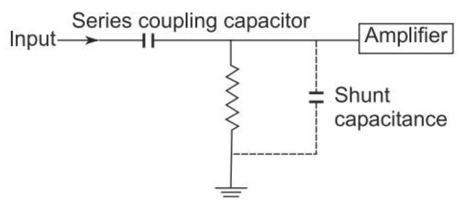


Fig. 7.1 *Series and shunt capacitor to attenuate low and high audio frequencies, respectively*

Example 7.1 *If load resistance is 5 kΩ, calculate the value of coupling capacitor so that its resistance at 20 Hz is one-tenth of the load resistor.*

Solution

Load resistance = 5 k Ω

$$X_c = \frac{5000}{10}\,\Omega = 500\,\Omega$$

Hence, $$500 = \frac{1}{2\pi \times 20 \times C}$$

or, $$C = \frac{1}{500 \times 2\pi \times 20}$$

In microfarads,

$$C = \frac{1 \times 10^6}{20000 \times 3.142}\,\mu\text{F} = 16\,\mu\text{F}$$

Example 7.2 *Calculate the frequency at which reactance of the shunt capacitance of 100 pF would be 10 times the load of 5 kΩ.*

Solution

Load = 5 kΩ

Therefore,

$$X_{cs} = 50\text{ k}\Omega$$

$$50000 = \frac{1}{2\pi\, f \times 100 \times 10^{-12}}$$

Therefore,

$$f = \frac{1}{2\pi \times 50000 \times 100 \times 10^{-12}} = \frac{10^6}{10\pi}$$

= 31847 (say 32 kHz)

7.2 CHARACTERISTICS OF AUDIO AMPLIFIERS

An audio amplifier has the following characteristics:

Gain Ratio of output signal to input signal is called gain of an amplifier. It is expressed in decibels (dB). Voltage gain A_v is given by Eq. 7.1 and power gain A_p by Eq. 7.2.

$$A_v = 20\log\frac{V_2}{V_1} \tag{7.1}$$

$$A_p = 10\log\frac{P_2}{P_1} \tag{7.2}$$

where, V_2 and V_1 are output and input voltages, respectively, and P_2 and P_1 are the output and input power, respectively.

The typical gain of a voltage amplifier is about 60 dB. The typical gain of a power amplifier is about 20 dB. Higher the level of input signal, less is the gain. Linear ICs, used as amplifiers, have much higher gain.

Example 7.3 *Calculate power gain in dB if an input power of 100 mW gives an output power of 1 W.*

Solution

$$\text{Gain} = 10\log\frac{P_2}{P_1}$$

$$= 10\log\frac{1000}{100}$$

(as 1 W = 1000 mW)

= 10 dB

Bandwidth An audio amplifier should pass the whole audible frequency range which is from 16 Hz to 20 kHz. While an amplifier can give flat response for the full audio range, there are other elements like microphones and loudspeakers in an audio system which restrict the overall frequency range.

Distortion An amplifier can suffer from the following types of distortions:

- Frequency distortion
- Phase distortion
- Amplitude distortion or non-linear distortion
- Distortion due to self-oscillations (or instability)

Frequency distortion

When all the audio frequencies are not amplified equally well, it causes frequency distortion. It is due to series coupling capacitor (for low notes) and shunt capacitor (for high notes). Although a good amplifier should have flat frequency response from 16 Hz to 20 kHz, a flat response from 40 Hz to 15 kHz is quite acceptable for even hi-fi systems.

Example 7.4 *How much would be the reduction in voltage across the load resistor, if reactance of series capacitance is equal to the load resistance (input signal is 1 volt ac)?*

Solution

Phase relationship of R and X_c is shown in Fig. 7.2.

Given, $R = X_c$

Resultant impedance,

$$Z = \left(R^2 + X_c^2\right)^{1/2}$$

$$= \sqrt{2}\ R \text{ ohms}$$

Cosine of angle of

$$Z \text{ with } R = \frac{R}{\sqrt{2}R} = \frac{1}{\sqrt{2}} \simeq 0.7$$

Hence, voltage across R

$= 0.7 \times$ voltage across Z

$= 0.7$ V

Therefore, reduction in voltage

$=$ voltage drop in $X_c = 0.7$ V

$= 3$ dB (as 0.7 V = 3 dB)

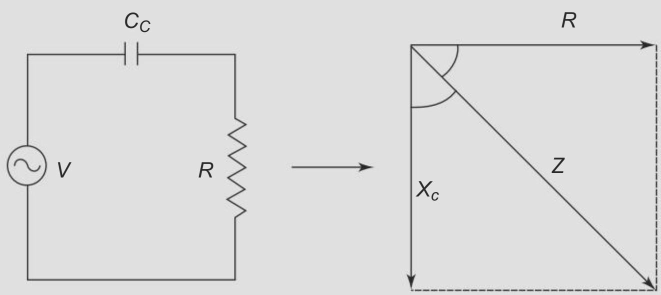


Fig. 7.2 *Phase relationship between R and X_c*

Phase distortion When the relative phase relationship as in the input signal is not maintained in the output signal, it causes phase distortion. It is due to reactive components which cause phase change in the signal. Its effect is to change the frequency because the state of phase and frequency depend on each other. Human ears are not sensitive to phase change in the same sound, and hence phase distortion is not of much significance in audio systems.

Amplitude distortion It is caused due to passage of signal through non-linear portion of the characteristic curve of transistors. The positive and negative peaks of large signals are clipped due to saturation at positive peaks and cut off at the negative peaks. This deformation of signal results in the production of new harmonic frequencies (Fourier's series) which were not present in the input signal.

Typically, non-linear distortion of a good audio amplifier is less than 1%.

An amplifier stage may go into self-oscillations if there is positive feedback due to undesired coupling of output of one stage to input of some earlier stage. These self-oscillations appear as noise along with the signal, and hence cannot be called distortion of the signal in the real sense but these may sometimes overload a stage to cause severe distortion of signal. These can be prevented by decoupling the load from the common power supply line.

Power Output As an amplifier system finally gives power to some device, (loudspeakers in case of audio amplifiers), output power which can be taken out from the power amplifier is an important parameter. In fact, the number of voltage amplifiers preceding the power amplifier will depend on how much power output is required of the final power amplifier.

The requirement of output power varies from a few watts to several hundred watts. As the output power increases, dissipation inside the transistors increases and adequate heat sinks are used to radiate out the heat generated by dissipation of power.

Impedance The source impedance or input impedance of a transistor amplifier is an important parameter. For maximum transfer of power from the power amplifier to the load (say loudspeakers), the impedance of the amplifier (called source impedance) must match with the load impedance. Even in intermediate stages, the output impedance of a stage should match with the input impedance of the next stage as far as possible. When we are interested in voltage in the output, the load impedance should be as high as possible. and hence exact matching is not needed.

Impedance of the loudspeakers is low, and of the source (amplifier), it is relatively high. A step down transformer with turns ratio of primary to secondary as given by Eq. 7.3, is used

$$\frac{n_p}{n_s} = \left(\frac{Z_p}{Z_s}\right)^{1/2} \tag{7.3}$$

where, n_p = No. of turns in the primary
n_s = No. of turns in the secondary
Z_p = Impedance of primary (i.e., source) side
Z_s = Impedance of secondary, (i.e. load) side

7.3 AMPLIFIER CIRCUITS

Voltage Amplifier

Pre-amplifier and other amplifying stages preceding the final power amplifier stage are all voltage amplifiers. It is an RC coupled amplifier operating on the linear portion of the characteristic curve as a Class A amplifier. Controls like loudness control and bass/treble controls, normally used in the output of a pre-amplifier are shown in Fig. 7.3.

Q is an *npn* transistor. Base biasing is provided by R_1 and R_2. R_e stabilizes the bias. C_e bypasses the audio frequencies to ground, so that there is only dc voltage across R_e. (In the absence of C_e, audio voltage will also develop across R_e which would reduce the gain by causing negative feedback.) The signal is fed between base and emitter through the coupling capacitor C_c. The output develops across the load resistance R_c. The output is fed to the processing circuits through C_1, and is processed by loudness control, bass control and treble control. The processed output goes to the next stage.

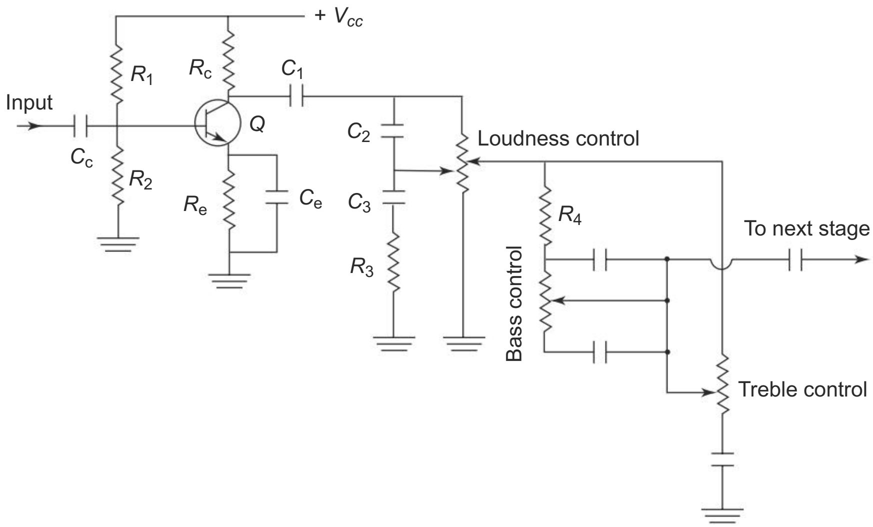


Fig. 7.3 *Class A pre-amplifier*

Power Amplifier

Power amplifiers used in audio systems are of three types:

1. Single-transistor amplifier
2. Push-pull amplifier using similar transistor
3. Push-pull amplifier using complimentary symmetry pair of transistors.

All power amplifiers are preceded by voltage amplifiers. The final voltage amplifier which is just before the power amplifier is called *driver* because the large voltage signal in its output reduces the dynamic resistance of the power amplifier stage, and hence, drives it to give more power. The transistor chosen for power amplification must have low internal resistance and should be able to withstand heavy dissipation of power. A heat sink of calculated size is used to radiate out heat generated by dissipation of power. Another requirement of a power amplifier is a matching transformer in its output to match the low impedance of the loudspeaker with the impedance of the power amplifier. The circuit diagram of a single-transistor power amplifier is given in Fig. 7.4.

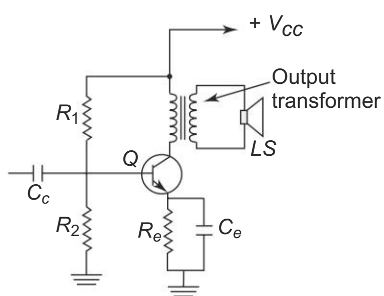


Fig. 7.4 *Power amplifier using a single transistor*

The functions of R_1, R_2, C_c, C_c and R_c are the same as explained for Fig.7.3. Load in the collector of Q transistor is the output transformer. The power developed in the output is transferred to the loudspeaker (LS). The step-down output transformer matches the low impedance of the loudspeaker with the high impedance of the amplifier. The loudspeaker converts electrical energy into sound energy.

Push-Pull Power Amplifier Figure 7.5 shows a push-pull amplifier. It uses two similar *npn* transistors. Input signals from the transformer T_1 are fed to the base of two transistors (Q_1 and Q_2) and are opposite in phase to each other. Direct current to the two collectors flows through the primary of the transformer T_2 in opposite direction as shown by arrows. This saves the transformer from becoming saturated by dc, and hence eliminates non-linear distortion which would have been produced due to saturation of the core.

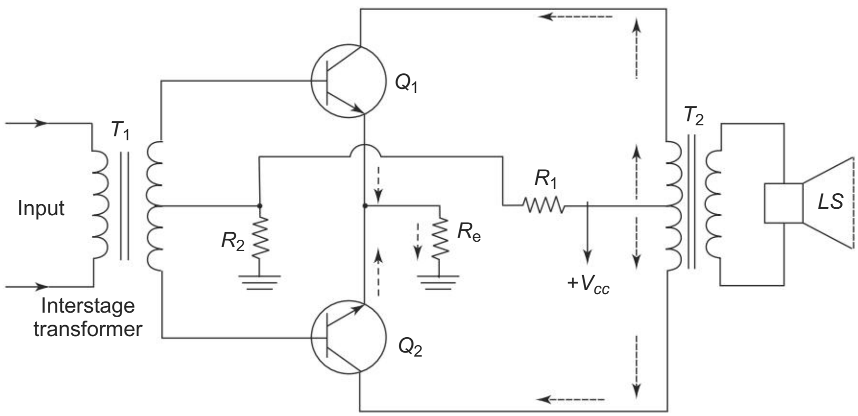


Fig. 7.5 *A push-pull amplifier*

The audio current will flow through the output transformer in the same direction from both collectors, due to the two outputs being in the opposite phase with respect to each other, as illustrated in Fig. 7.6.

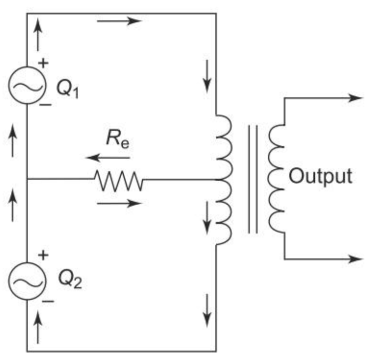


Fig. 7.6 *Flow of audio current*

When the collector of Q_1 is a positive going audio, that of Q_2 will be a negative-going audio and hence the audio currents in the primary will add up. Thus, the audio will develop fully without any saturation of the core by dc.

The advantages of a push-pull amplifier over a single-transistor amplifier are given below:

- Even harmonics are cancelled.
- Iron core of the output transformer does not get saturated.
- Direct currents cancel in the primary and so heat dissipation is low. Hence, more power can be derived.

Complementary Symmetry Push-pull Amplifier

The circuit for a complementary symmetry push-pull amplifier is shown in Fig. 7.7. It requires the same polarity at the input of two transistors.

The circuit uses two transistors, one of *npn* type and the other of *pnp* type. Input signals to the two transistors are in the same phase. (Inter-stage transformer for input is not required.) The *npn* collector gets positive dc voltage and the *pnp* collector, negative dc voltage. Direct current, through the primary of the transformer will be in the opposite directions. The audio currents from the two transistors will add in the primary and then will give all the advantages of push-pull configuration.

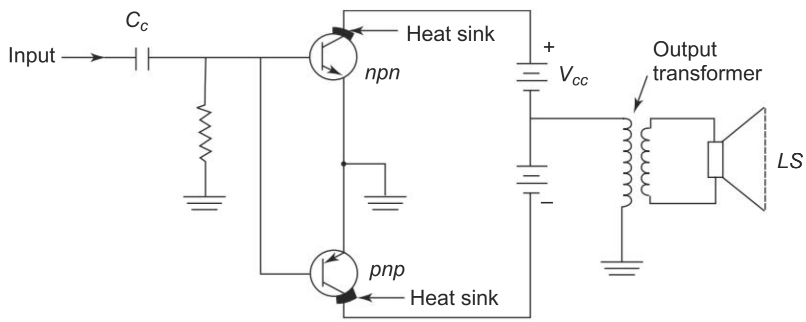


Fig. 7.7 *Complementary symmetry push-pull amplifier circuit with output transformer*

Complementary Symmetry Push-pull Amplifier without Transformer

It can dispense with the output transformer as shown in Fig.7.8. Despite use of two *npn* and *pnp* type transistors, it dispenses with the requirement of two power supplies. It feeds the audio signal to the loudspeaker through the capacitor *C*.

In the circuit of Fig.7.8, two transistors, *npn* and *pnp* form a matched pair when properly biased, The point *P* will be at +20 V for V_{cc} equal to + 40 V. Thus, the collector emitter voltage of *npn* (between *A* and *P*) will be equal to that of *pnp* (between *P* and *B*). Capacitor *C* will charge to 20 V through the loudspeaker.

When the input signal is positive-going, Q_3 conducts, causing its collector voltage to be low. This will increase the forward bias between base and emitter of *pnp* transistor Q_2, and hence will reduce its internal resistance. Due to conduction of Q_3, the voltage at the base of Q_1 will decrease, making it less conductive and finally cut off. Due to non-conduction of Q_1 and conduction of Q_2, the condenser *C* will discharge through the loudspeaker. Voltage at the point *P* may go down to zero if the positive going input signal is very strong.

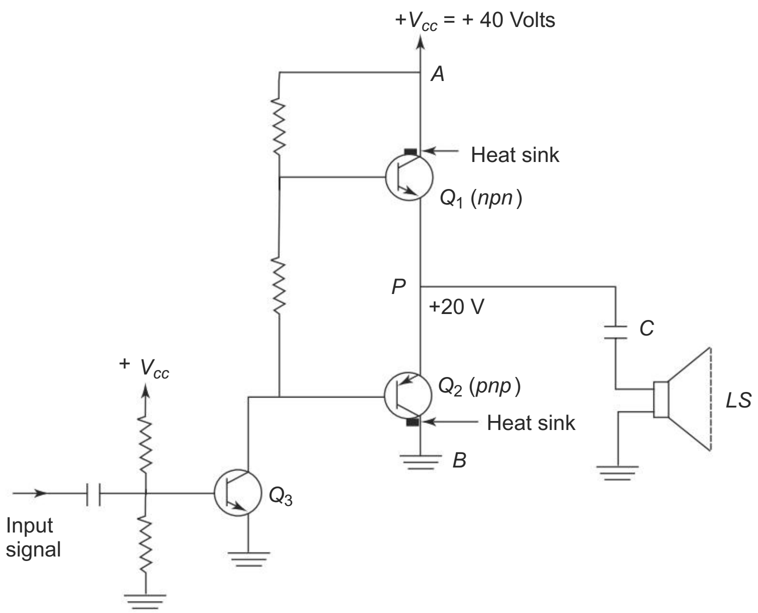


Fig. 7.8 *Complementary symmetry push-pull amplifier without transformer*

Let us now examine what would happen when the input signal reverses its polarity. Q_2 will be driven towards cut off and Q_1 towards conduction. The capacitor C will now start charging through Q_1 and the load (loudspeaker) and the voltage at the point P will increase. Thus, the current through the loudspeaker will be in accordance with the audio signal, duly amplified to +20 volt (peak to peak).

Necessity of dispensing with the Output Transformer

1. Output transformer is costly, and hence its elimination will be economical.
2. The output transformer is bulky and adds greatly to the weight of the equipment and takes up considerable space.
3. It picks up hum from ac mains.
4. It causes distortion due to losses being high at higher frequencies.
5. It also gives resonance effect at a particular frequency near the high side due to inductance of the coil and its self-capacitance.

All these disadvantages of an output transformer are eliminated by designing the circuit such that the output transformer is dispensed with.

7.4 NEGATIVE FEEDBACK IN AMPLIFIERS

When signal from some amplifier stage is fed back to an earlier stage in such a manner that phase of the feedback signal differs by 180 degrees from the input signal, it is known as 'negative feedback'. A basic negative feedback circuit path is shown in Fig. 7.9, using phase-inverting amplifiers.

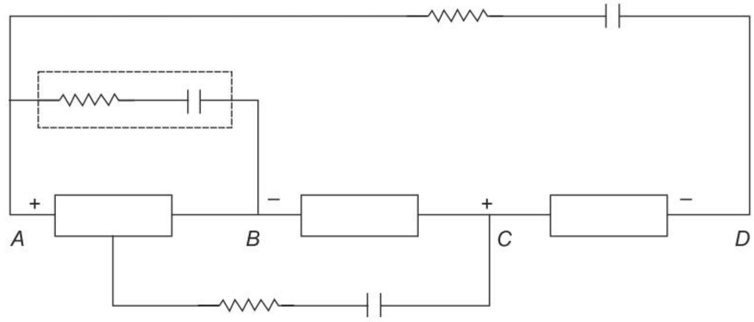


Fig. 7.9 *Negative feedback*

In Fig. 7.9, + and – signs do not represent +ve and –ve dc, but represent +ve and –ve phase, relative to phase at *A*. Negative feedback can be given to the base from the output point *B* or *D* through a simple *RC* network, shown in Fig. 7.9. If feedback is taken from the output point *C*, it will have to be given to the emitter of the first stage. The in-phase signal at the emitter works as –ve phase signal at the base. Thus, negative feedback can be given either to the base or to the emitter of a stage depending upon the phase of the signal from where feedback is being derived.

In operational amplifiers, negative feedback is easily available when feedback signal is fed to the inverting input.

Gain and distortion both are reduced by negative feedback as indicated by Eqs. (7.4) and (7.5), respectively.

$$A_f = \frac{A_0}{1 + BA_0} \tag{7.4}$$

$$D_f = \frac{D}{1 + BA} \tag{7.5}$$

where, A_o = Gain without feedback

A_f = Gain with negative feedback

B = Feedback factor (i.e. fraction of the output signal used as –ve feedback)

D = Distortion without feedback

D_f = Distortion with –ve feedback

Advantages of Negative Feedback

1. It reduces distortion.
2. It stabilizes gain and makes it independent of the characteristics of transistors and circuit components. Hence, replacement of transistor will not affect the gain. Thus, it gives precise control of the gain of an amplifier.
3. Gain becomes independent of temperature variation, variation in mains voltage and aging, so long as *AB* remains > 1.

4. Input signal can swing to larger extent in the presence of negative feedback.
5. Transient response improves. Transients do not overload the circuit.
6. Frequency response improves due to greater flattening of response curve.
7. Phase response also improves. Distortion in phase response curve is reduced which improves phase response.

Disadvantage

1. Gain of the amplifier is reduced.

However, advantages outweigh the only disadvantage of reduced gain, and that is why negative feedback is essentially used in high-fidelity audio systems. Examples 7.5 through 7.7 show the effect of – ve feedback.

Example 7.5 *An amplifier has an output voltage of 20 V. Total harmonic distortion is 10% and gain is 100 without feedback. Calculate B to reduce distortion to 1%.*

Solution

$$D_f = \frac{D}{1 + AB}$$

1% of 20 V = 0.2 and 10% = 2

$$0.2 = \frac{2}{1 + 100 \times B}$$

$$0.2 + 20\,B = 2$$

$$B = \frac{1.8}{20} = 0.09$$

Example 7.6 *Calculate the signal voltage (V_s) required to give 20 V output with negative feedback in the above example.*

Solution

$$\left(V_s - \frac{20 \times 9}{100}\right) \times 100 = 20$$

$$\text{or,} \quad 100V_s = 20 + 180$$

$$\text{or,} \quad V_s = 2V$$

Example 7.7 *Calculate new gain (A_f) in the above example.*

Solution

$$A_f = \frac{100}{1 + 100 \times 0.09} = \frac{10}{100} = 10$$

7.5 PEAK MUSIC POWER OUTPUT (PMPO)

It represents maximum peak power in instrumental music which can be handled by an amplifier. In instrumental music, the intensity of sound varies from loud notes to soft notes. However, to give a special musical effect, a musician may produce an extremely loud note for a moment, generally at the end of the music session. This note has a special significance and should be reproduced faithfully by the amplifier-loudspeaker system. Continuous average power of the music without momentary very loud note is about 8% of the intensity of this note. For example, if the maximum momentary peak represents 600 W then the average continuous power may be only about 50 W. As the reproduction of the momentary loud note is essential to feel the special effect, the amplifier is so designed that while it handles a continuous average power of 50 W only, it should withstand 12 times this power for a moment only. This peak power is called Peak Music Power Output (PMPO). The manufacturers usually quote PMPO for their system. The purchaser should not be led to believe that PMPO is the peak of the continuous average power, but he or she should understand that the practical power coming out of the amplifier would be only 8% of the quoted PMPO.

7.6 CONTROLS IN AUDIO AMPLIFIERS

The following controls are incorporated in the audio amplifiers:

- Microphone gain control
- Volume control
- Tone control

These are described below.

Microphone Gain Control It consists of a potentiometer in the output of a microphone. The function of this control is to adjust the output of the microphone depending upon the speaker's style of speaking. Normally, this control is located in the pre-amplifier or mixer circuit, and is shown in Fig. 7.10. P_1 and P_2 are gain controls and R_1 and R_2, the isolating resistors.

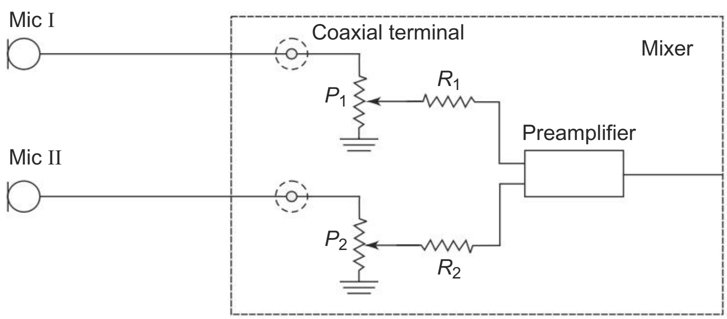


Fig. 7.10 *Microphone control*

Volume Control While gain control sets the output of individual microphones used in the amplifier system, the volume control is used after the mixer stage to control overall volume of the amplifier output. It is also known as *master gain control* or *master volume control*. It uses a potentiometer, not in the base circuit of a transistor (for the reason of keeping base-emitter bias fixed), but in the collector circuit as shown in Fig. 7.11. (In Field effect transistor, it can be had in the gate circuit.)

The value of a potentiometer should be approximately twice the load resistance of the stage ahead. When the sliding contact is towards the collector side, maximum signal will be fed to the next stage. It will decrease as the sliding contact is moved away from the collector.

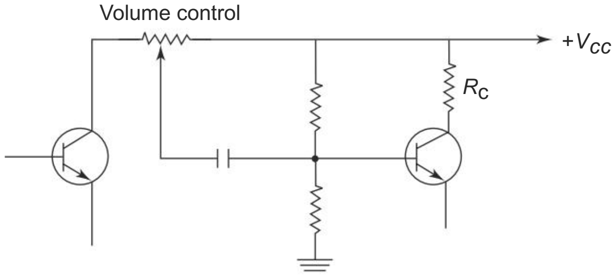


Fig. 7.11 *Volume control*

Volume control potentiometers can be linear or logarithmic. In a linear potentiometer, the increase from zero value is rapid, but in the logarithmic type, the movement of the slider from zero gives only a small change in resistance. A logarithmic type is better as it gives less critical adjustment at low volumes and also matches with the ear's logarithmic response.

Tone Controls

Modern audio frequency amplifiers are designed to give flat frequency response over the whole audio range from 16 Hz to 20 kHz. However, some people like depth in the sound which is given by bass (low notes) and hence for them bass, should be amplified more than treble (high notes). Also, there is greater noise in a high-frequency octave than in a low-frequency octave. For example, noise contained in the frequency range from 8000 to 16000 Hz will be 20 times the noise in the range from 400–800 Hz. Hence to keep signal to noise ratio (or noise limited sensitivity) high, treble is cut. Hum and external noises, converted into electrical signals, are in the low-frequency range and when such noises occur, bass is cut. Some people like a sharp sound and want treble more than bass.

To cater to the individual's taste and also to offset the effect of noise present in the signal, provision of bass and treble controls is made. The combined control is called tone control.

Treble Cut A simple treble cut control is shown in Fig. 7.12.

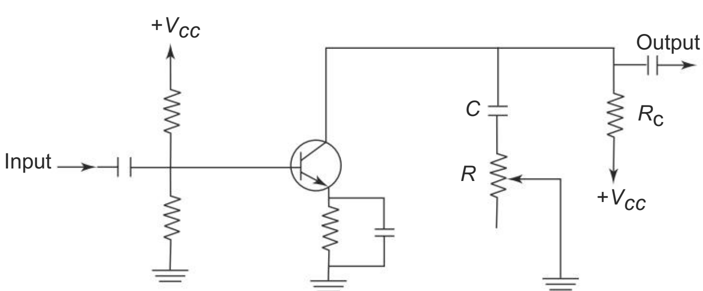


Fig. 7.12 *Simple treble cut*

The potentiometer R in series with the capacitor C forms the treble control. When the slider is at the lower end, maximum signal develops across RC (load). As the slider is moved upwards, less and less resistance of the potentiometer comes in series and hence, there is more and more cut in the high-frequency signals. Cut is maximum when the slider is at the top end, short-circuiting the potentiometer completely. This position is called treble cut. The other position where treble cut is minimum is called *treble boost*. The capacitor alone will have low reactance for high frequencies.

Bass Control Bass would be cut if capacitive reactance in series of signal increases. Lower the capacitance, greater will be the reactance (X_C = ½ πfC). Hence for cut in bass, the value of series capacitance is reduced as illustrated in Fig.7.13.

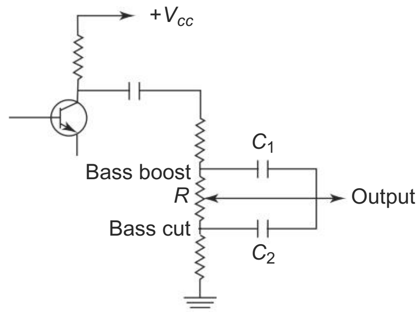


Fig. 7.13 *Basic bass control*

When the slider of the potentiometer R is at the upper end, the capacitor C_1 is shorted and the signal directly goes to the next stage, bypassing the capacitor C_1 and hence, bass has the minimum attenuation. It is called *bass-boost*. When the slider is at the lower end, the capacitor C_1 in parallel with the whole resistance R of the potentiometer comes in series with the signal. In this position, bass will have maximum attenuation. This position is called *bass cut*.

The combined circuit containing both bass and treble control is shown in Fig. 7.14. The potentiometer P_1 acts as bass control, and P_2, as treble control.

Bass boost and cut is effective by +15 dB at 16 Hz compared to the output at 1 kHz. Treble boost and cut is also effective by the same amount at 20 kHz compared to the value at 10 kHz.

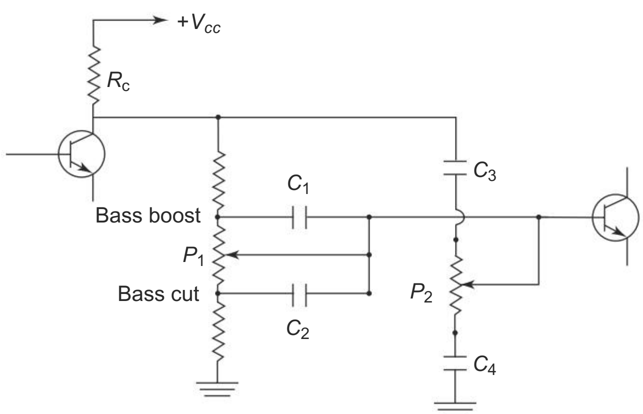


Fig. 7.14 *Bass and treble control*

7.7 SPECIAL TYPES OF TONE CONTROLS

Parametric Equaliser It is designed to provide variable boost or cut up to about 15 dB. The parameters, like frequency and bandwidth, can be varied throughout the audio spectrum of 16 Hz to 20 kHz. The frequency response can be adjusted very precisely and selectively. For example, there is response peak at a particular frequency due to acoustic resonance in the studio. This peak can be cut to the extent desired (from 0 dB to 15 dB), and thus the effect of resonance peak can be neutralised or corrected to produce high-fidelity sound. Alternatively, parametric circuit can be used to give artificial artistic effect to sound. It can also be used to provide special boosting near about 3 to 5 kHz frequencies to give *Presence effect* which means that the vocalist appears to be nearer the audience. Cut at these frequencies makes the vocalist appear farther from the audience.

Three types of variable boost or cut adjustments are shown in Figs 7.15 (a), (b) and (c). Facility of variable adjustments of tone is very useful in a recording studio.

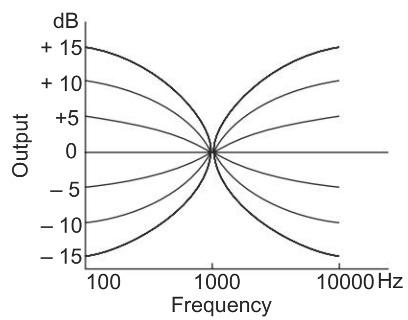


Fig. 7.15 (a) *Variable boost/cut throughout audio spectrum*

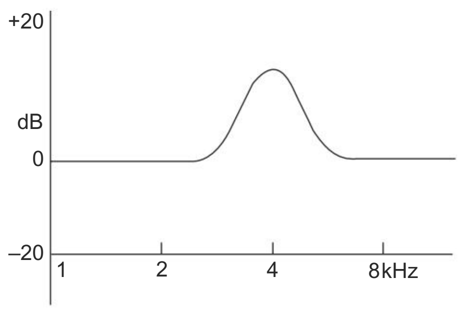


Fig. 7.15 (b) *Boosting at about 4 kHz for "Presence effect"*

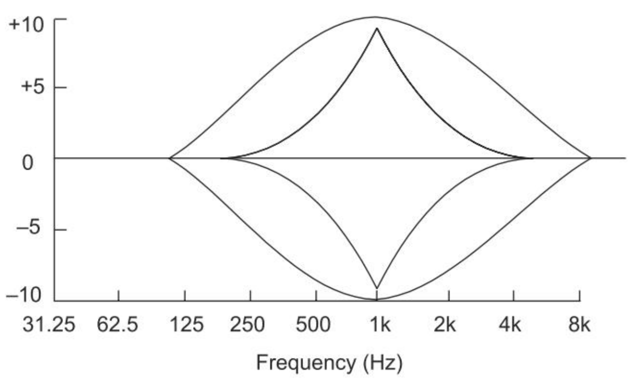


Fig. 7.15 (c) *Boost/cut for specific bands*

Graphic Equaliser In this type of tone control, the audio spectrum is divided into narrow bands. Each band has an individual slider control which can boost or cut the signals in that band from +15 to –15 dB. The centre frequency of each band is generally based on an octave or 3-octave interval.

The shape of the response curve will be given by joining the slider positions mentally by a smooth curve. Hence, the name 'graphic equaliser'. This method provides an excellent equalisation of the frequency response.

Frequency response curves for different bands with centre frequencies at a one-octave interval are shown in Fig. 7.16.

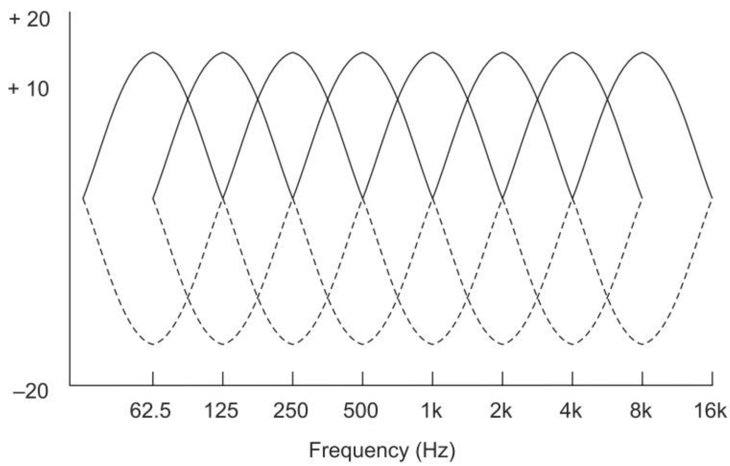


Fig. 7.16 *Boost/cut for bands at frequencies centered one octave wide*

The position of sliders to get a particular response is shown in Fig. 7.17.

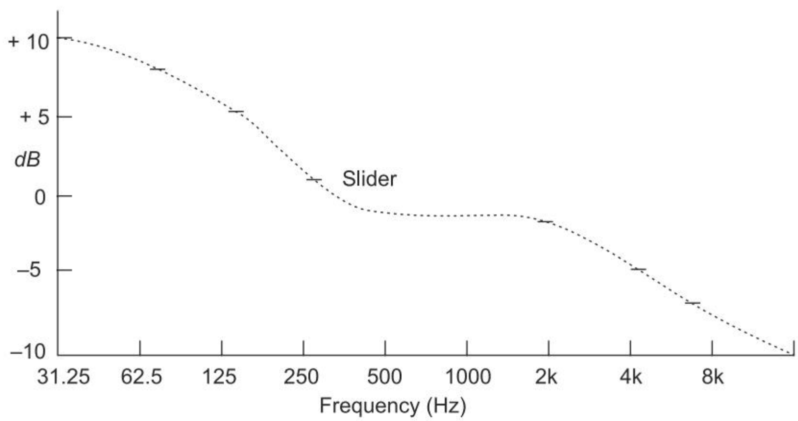


Fig. 7.17 *Position of sliders of graphic equaliser to get a specified response curve (dotted line)*

SUMMARY

- Audio amplifiers are used to amplify audio signals of frequency range from 16 Hz to 20 kHz.
- There are two types of audio amplifiers: (1) voltage amplifiers, and (2) power amplifiers.
- Voltage amplifiers are used as pre-amplifiers, intermediate amplifiers and driver amplifiers.
- The power amplifier is the final amplification stage which feeds power to the loudspeakers.
- Voltage amplifiers are Class–A type.
- Push-pull power amplifiers use Class-B biasing.
- Class-C biasing is used in radio-frequency-tuned amplifiers.
- An audio amplifier has the following parameters: gain, bandwidth, distortion, power output and impedance.
- In voltage amplifier circuits, a series coupling capacitor is kept high so that it provides low reactance to the lowest frequency in use. The emitter resistance is bypassed by a capacitor for audio signals. Stabilised bias is provided. The load is decoupled from the power-supply line to prevent positive feedback in previous stages through common power-supply line.
- The power amplifier circuit is also similar to the voltage amplifier except that (i) its internal resistance is much lower than the voltage amplifier, (ii) output has matching transformer to match its impedance with the low impedance of the loudspeaker, and (iii) adequate heat sinking is essential as heat produced in a power amplifier is much more than in a voltage amplifier. A power amplifier is essentially preceded by a driver amplifier which drives it to give high power by making its internal resistance low.
- Being a high-current device, the power amplifier has higher distortion than a voltage amplifier, and therefore, push-pull circuits are used for power amplification, in which even harmonics cancel out in the output.
- Negative feedback reduces distortion and stabilises gain, making it independent of change in the characteristics of transistor or components. It reduces gain also, but reduction in gain is not much important, and the negative feedback technique is universally used in high-fidelity audio systems.

☞ An audio amplifier is designed to withstand about 12 times the average power for a moment. This is called PMPO. About 8% of PMPO is the continuous average power handled by the audio amplifier.

☞ An audio amplifier has the following controls: microphone gain control, volume control, tone control (treble and bass controls).

☞ Parametric equalisers and graphic equalisers are special types of tone controls in which boosting up to (+) 15 dB and cut up to (–) 15 dB is done throughout the audio spectrum of 16 Hz to 20 kHz precisely and selectively.

Review Questions

1. Explain the basic differences between voltage amplifiers and power amplifiers.
2. What are the basic differences between an audio amplifier and an RF amplifier?
3. Explain the following terms pertaining to parameters of audio amplifiers:
   1. Gain
   2. Bandwidth
   3. Distortion
   4. Impedance
4. Draw a neat circuit diagram of a typical audio voltage amplifier and explain its working.
5. Explain the working of a typical power amplifier with the help of a neat circuit diagram.
6. Draw the circuit diagram of a push-pull amplifier and explain its working. Mention the advantages of a push-pull amplifier over a single-transistor power amplifier.
7. Draw a complementary symmetry push-pull amplifier and explain its working.
8. What is the function of an output transformer in power amplifiers? What are its disadvantages? How can it be dispensed with?
9. Explain negative feedback and write its advantages and disadvantages.
10. What do you understand by bass and treble controls? Explain their inportance.
11. Draw neat circuits, showing bass and treble controls in an amplifier, and explain how do they function.
12. Write a note on parametric equalisers.
13. Describe a graphic equaliser. Why is it called 'graphic'?
14. Explain the Peak Music Power Output (PMPO).

Short-Answer Questions

1. Why is Class-C bias not suitable for audio amplification?
2. How can Class-B bias be used for audio power amplifiers?
3. How is the effect of base-emitter junction capacitance nullified in radio frequency amplifiers?
4. What is the cause of roll-off at low frequencies and at high frequencies in *RC* coupled amplifiers?
5. Why is distortion reduced by negative feedback?
6. Why does an amplifier's gain with negative feedback become independent of the characteristics of the transistor used?
7. What is the function of bass and treble controls in an amplifier?
8. Why is graphic equaliser called 'graphic'?
9. If an audio amplifier is designed to handle 16 Hz to 50 kHz, what are the factors that limit the audio range to about 15 kHz only?
10. What do you understand by presence effect in the design of tone controls?

Multiple-Choice Questions

1. Which type of amplifier has low internal resistance?
   (a) Voltage amplifier
   (b) Power amplifier
   (c) Buffer amplifier
   (d) Amplifier with negative feedback
2. What type of amplifier is used as output amplifier in an audio system?
   (a) Voltage amplifier
   (b) Current amplifier
   (c) Power amplifier
   (d) Buffer amplifier
3. What type of circuit uses a resonant circuit as load?
   (a) Audio amplifier
   (b) Radio frequency amplifier
   (c) Detector circuit
   (d) Push-pull amplifier
4. What type of bias is used in audio voltage amplifiers?
   (a) Class A (b) Class B
   (c) Class C (d) Class D
5. What should be the bandwidth of an ideal-fidelity audio system?
   (a) 10 to 1000 Hz
   (b) 1000 to 10000 Hz
   (c) 50 to 5000 Hz
   (d) 16 to 20000 Hz
6. The output impedance of a power amplifier should match with the impedance of
   (a) microphone
   (b) loudspeaker
   (c) buffer amplifier
   (d) voltage amplifier
7. What is the effect of emitter bypass capacitor on the gain of an amplifier?

(a) It reduces gain substantially.
(b) It enhances gain.
(c) No effect.
(d) Gain is reduced only very little.

8. Which harmonics are eliminated in a push-pull amplifier?
(a) All harmonics
(b) Odd harmonics only
(c) Even harmonics only
(d) None

9. What frequencies are boosted by treble control?
(a) Radio frequencies
(b) Intermediate frequencies
(c) Low audio frequencies
(d) High audio frequencies

10. Which of the following is not the function of negative feedback?
(a) Reduction in gain
(b) Reduction in distortion
(c) Reduction in bandwidth
(d) Increasing input resistance

11. If the PMPO of an amplifier is quoted as 1200 W, what is the average power that the amplifier handles continuously?
(a) About 1200 W
(b) About 600 W
(c) About 300 W
(d) About 100 W

12. What is the function of a decoupling capacitor?
(a) To increase gain
(b) To decrease gain
(c) To decrease distortion
(d) To prevent self-oscillation

Numerical Problems

1. Calculate the value of series capacitor to provide a reactance of 100 Ω for 20 Hz.
2. What will be the reactance of shunt capacitance of 20 pF at 20 kHz? Will it affect the performance if load resistance is 2 kΩ?
3. Calculate gain of an amplifier in dB if output voltage is 1 V and input voltage is 1 millivolt.
4. How much power will be dissipated in a transistor amplifier which gives 100 mA current when V_{CE} is 10 volts?
5. Calculate number of turns of secondary, if load resistance is 4 Ω and source resistance 1600 Ω, turns in primary are 1000.
6. Calculate correctly up to one place of decimal the distortion and the gain after negative feedback, if B = 0.1, gain and distortion without feedback are equal to 1000 and 10%, respectively and output voltage with negative feedback = 10V.
7. Low frequencies give a response of 1 V in the absence of bass control. What will be the response after bass control is adjusted to 6 dB cut.

Answers

Short-Answer Questions

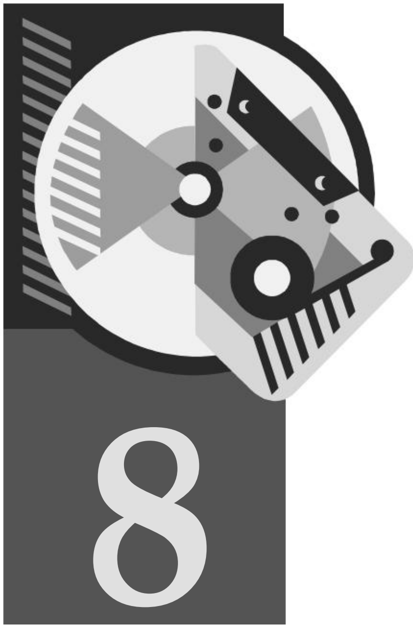
1. In Class-C bias, less than half the cycle of signal voltage is reproduced. This causes severe distortion in the output. In audio systems, distortion is the most undesired thing.
2. By using two transistors in push-pull configuration.
3. Radio frequency amplifiers use tuned circuit as load, and the base-emitter junction capacitance becomes part of the capacitance of the tuned circuit. Thus, there is no adverse effect of the shunt capacitance on the performance of a radio frequency amplifier.
4. In *RC* coupled circuits, roll-off at low frequencies is due to series coupling capacitor, and at high frequencies due to shunt capacitance formed by the base-emitter junction.
5. A portion of the distorted signal is fed back to the input in opposite phase. This results in reduction of the distortion.
6. With negative feedback, gain of an amplifier is practically equal to $1/B$, where B is the feedback factor. Hence, it does not depend on the transistor characteristics.
7. Bass control boosts low audio frequency signals. Treble control boosts high audio frequencies.
8. In a graphic analyser, the shape of the response graph can be obtained by joining the slider positions mentally by a smooth curve. Hence, it is called a graphic equalizer.
9. Microphones and loudspeakers, which are integral parts of an audio amplifier system, limit the overall frequency range of the output.
10. A parametric amplifier provides special boosting from 3 to 5 kHz. This gives presence effect, which means the vocalist is near the audience.

Multiple-Choice Questions

1. (b)	2. (c)	3. (b)
4. (a)	5. (d)	6. (b)
7. (d)	8. (c)	9. (d)
10. (c)	11. (d)	12. (d)

Numerical Questions

1. (80 μF)	**2.** (400 kΩ, No)
3. (60 dB)	**4.** (1 W)
5. (50)	**6.** (0.1%, 9.9)
7. (0.5 V)	



Noise and Distortion

8.1 NOISE

Any unwanted sound present in the environment, or coming out of the loudspeaker in an audio system, is called 'noise'. It is different from music. A musical sound has the following properties.

1. Periodicity
2. Regularity of frequency
3. Regularity of shape
4. Regularity of amplitude
5. Continuity

Contrary to it, noise consists of pressure variations of random nature, without any regularity of frequency, shape and amplitude, and also without continuity. Typical waveforms of noise and sound of musical instruments are shown in Fig. 8.1.

Audible noises are of two types:

1. Environmental noise, and
2. Electrical noise converted into sound waves by loudspeakers.

Environmental Noise

It is an unwanted sound present in the environment. Then, there are sounds which occur suddenly for a short duration, called transients, like thunder, shrieks, explosions. Excluding transients, the general noise level present in the environment is called *ambient noise*. Some typical ambient-noise levels above the threshold of hearing are given below.

- **Recording studio:** 20 dB or less
- **TV studio:** 25 to 30 dB
- **Concert hall:** 30 to 35 dB
- **Quiet home:** 35 to 40 dB
- **Classroom:** 40 to 45 dB
- **Sports stadium:** 60 to 70 dB

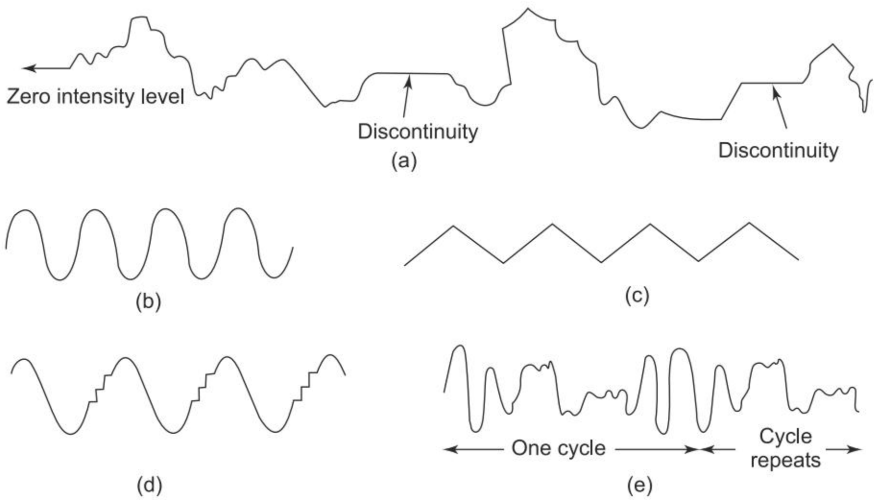


Fig. 8.1 (a) *Waveform of noise* (b) *Tone of a tuning fork* (c) *Vibration of a vowed string* (d) *Vibration of a flute* (e) *Vibration of a clarinet*

A study conducted in UK has concluded that the average noise level of 90 dBA existing for 8 hours a day (99 dBA for 1 hour, 108 dBA for 8 minutes and 117 dBA for 1 minute only) is harmful. Some harmful effects of a too-noisy environment are as mentioned below.

1. It strains the nervous system.
2. It causes mental fatigue.
3. It irritates the workers and lowers their efficiency.
4. Very loud and sudden noise may impair hearing.
5. It retards the normal growth of infants and young children.

It is, therefore, necessary to eliminate noise pollution and to reduce the unwanted sound to the minimum possible. The ratio of intensity of wanted sound to that of unwanted sound (called signal-to-noise ratio) should be more than 50 dB in good quality programmes or sound reproducing systems. (A value of 30 dB is acceptable in telephone conversation.)

Environmental noise in a hall can be reduced by using proper insulation, so that any external airborne noise may not enter the room with full intensity. External noise transmitted through solid structures can be reduced by making the outer walls massive or by breaking the sound path by insulating sections, for example, using double-wall technique with air between the walls.

Environmental noise can enter an audio system through microphones. Hence, proper placement of microphones and use of directive microphones is helpful in preventing environmental noise or other unwanted sound from entering the audio system.

Electrical Noise

It is the noise caused by random motion of electrons in components and reception of unwanted signals. It is also caused by random variation of magnetism (in tape) and groove irregularities in discs. Noise produced by diodes, transistors and resistors (thermal noise) and noise produced by sparks and RF transmissions (external noise) are pertinent to ratio frequency systems and are not of any concern in pure audio systems. The following noises are produced in audio systems.

- Hum noise
- Noise produced by unwanted coupling of different circuits
- Noise produced by recording and playback transport mechanism
- Noise produced by random variation of tape magnetism
- Noise produced by minute irregularities in the grooves of the disc records

All these noises can be considered at the input of an amplifier, and then their sum is called *input noise*. This is amplified and finally fed to a loudspeaker of an audio system. The loudspeaker converts the noise signal into sound waves and these, then, appear as unwanted sound along with the wanted sound. These noises are described below.

Hum Noise It is produced by ripples in the power supply. It can also be produced when ac power lines run close and parallel to the input leads.

Noise Produced by Unwanted Coupling between two Circuits Sometimes, there is coupling between output and input such that feedback is positive. This results in self oscillation of an amplifier stage. These oscillations are reproduced as unwanted sound by the loudspeaker, and give fuzzy appearance in an oscilloscope. The coupling between two different channels (as in a stereo system) gives rise to *cross talk*.

Noise Produced by Transport Mechanism of Sound Recording and Reproducing Devices Slow variations in speed of transport mechanism of disc and tape recorders give rise to noises, called *wow and flutter*. Body vibrations of motors result in noise, called *rumble*.

Noise Produced by Random Variation of Magnetism Head gap may contain magnetic dust which will vary the tape magnetism, or there may be irregularities in the coating of iron oxide on the tape, resulting in hissing noise.

Noise Produced by Irregularities in the Disc Grooves The surface of the grooves may not be perfectly smooth. Minute irregularities in the grooves and friction, faced by the stylus due to roughness, shall cause noise in the playback.

All types of noises, whether converted by a microphone, detected by non-linear characteristics of amplifiers, caused by a motor speed variations/vibrations or due to coupling, or due to irregularities in grooves in discs and magnetisation in tapes, add and are finally fed to the loudspeaker which converts these into unwanted sound.

8.2 METHODS OF REDUCING NOISE

The above types of noises can be reduced by adopting the following techniques:

1. Environmental noise can be reduced by using proper insulation and absorption materials in the hall and also proper acoustic design and placement of microphones and loudspeakers.
2. Electrical noise can be reduced by adopting the following techniques.
   (i) Use of active components, particularly a pre-amplifier, having low noise figure to reduce transistor noise and thermal noise.
   (ii) Proper shielding and grounding for preventing undesired coupling, and hence self oscillation.
   (iii) Adequate decoupling of circuits, particularly from common signal lines like power supply line, to prevent self-oscillation.
   (iv) Use of trap circuits to bypass rf interference.
   (v) Stabilised power supply with little ripple.
   (vi) Mechanical filtering of body vibrations of motors used in recording systems.
   (vii) Use of synchronous and servo-controlled motor for steadiness of speed.
   (viii) Proper adjustment of tension of guides, capstan and roller on the tape. Also, reduction of friction by proper oiling.
   (ix) Proper maintenance of heads to prevent electrical noise in magnetic recording and playback.
   (x) Care in producing recording grooves, so that roughness is minimum.
   (xi) Smooth high quality coating of magnetic material on tape.
   (xii) Use of special circuit techniques like Dolby method for improving signal-to-noise ratio.

Overall noise at the output should be as low as possible. What is important is that the level of signal should be so high and of noise so low that signal-to-noise ratio becomes 50 dB or more in high-quality audio systems.

8.3 DISTORTION

Difference between Noise and Distortion Noise consists of signals outside the wanted signal. Distortion means deformation of the wanted signal itself.

Types of Distortion

There are five types of distortion, named below:

1. Frequency distortion
2. Phase distortion
3. Amplitude distortion
4. Spatial distortion
5. Transient distortion

Frequency Distortion When frequency response is not flat, it results in frequency distortion, or attenuation distortion. The graph of the output signal level for a constant input level should ideally be a straight line for all frequencies within the audible range. Fluctuations of ±1 dB remain undetected by even keen ears. Changes beyond ±1 dB become important and affect the overall tonal balance. (However, practical bandwidth is normally expressed between half power or ±3 dB points, but for high-fidelity systems criteria of ±1 dB is followed.)

Low-frequency response of a system depends upon the capacitive reactance in series with the circuit-handling audio signals (Figs 8.2 a and b). The higher the series capacitance, the lower will be its reactance and hence, lower signal loss. Thus, for good low-frequency response, the series capacitance is kept high (i.e., capacitive reactance is low). Its reactance should be at least one fifth of the load resistance R at the lowest audio frequency.

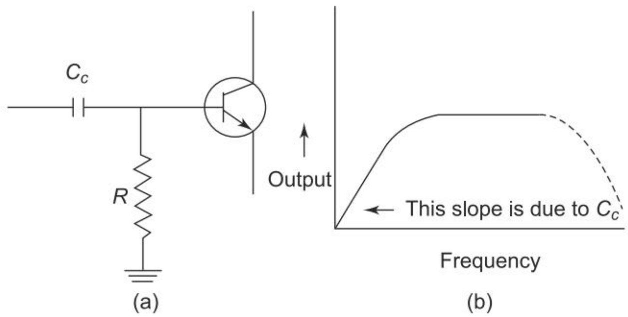


Fig. 8.2 *Effect of series capacitor on frequency response*
(a) *Circuit elements* (b) *Graph showing attenuation at low frequency*

Series capacitance is kept high enough to provide low reactance at 40 Hz in hi-fi systems. All devices which handle audio frequencies do have series capacitance, electrical or mechanical (i.e., compliance as in loudspeakers, microphones, recording units and pick-up cartridges). For keeping the compliance high, loudspeaker cones are made of large size (200 to 300 mm) as in woofers which give desired output at low audio frequencies.

High frequencies are attenuated by series inductance or shunt capacitance or both, as shown in Fig. 8.3(a) and 8.3(b).

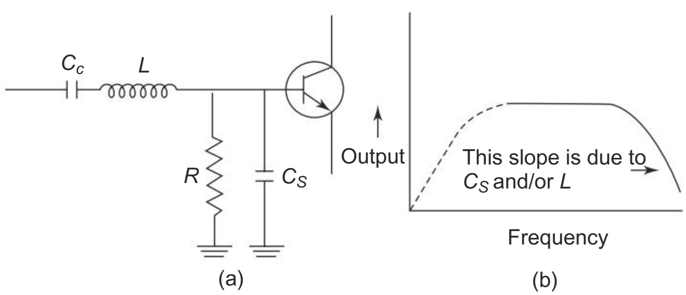


Fig. 8.3 *Effect of series inductance and shunt capacitance on frequency response*
(a) *Circuit elements* (b) *Graph showing attenuation at high-frequency*

The series inductor L will have high reactance in series, and the shunt capacitance, C_s, will have low reactance in parallel with the load and so shall attenuate high-frequency audio signals. Hence, to get good high-frequency response, L should be of low value and C_s should also be small. L and C_s may be electrical as in amplifiers, or may be mechanical as in loudspeakers. In loudspeakers, mass of the mechanical system acts as inductance and it is kept small by using a small-sizes cone (25 to 75 mm) as in a tweeter. Shunt compliance (i.e., shunt capacitive effect) of a tweeter is also made small.

In recording and reproducing systems, there is severe frequency distortion and hence, processing in recording and equalisation in playback stages of the audio signal is done for achieving hi-fi, as already explained in detail in Chapter 5.

Bass/treble controls are used to make the output suited to the individual taste. Frequency distortion is also caused by directional characteristics of moving-coil type microphones and loudspeakers with respect to frequency. For high frequencies, intensity is more in the front than on the sides. This distortion can be reduced by setting the tone control so as to reduce high notes.

Phase Distortion When different frequencies are shifted in phase differently while passing through a network consisting of reactive components, or amplifier circuit, it is called phase distortion. It may happen that while a 400 Hz signal shifts in phase by 180º while passing through an amplifier, a 5000 Hz signal may shift by 160º only. In such a case, phase distortion has occured. While phase distortion is important in TV picture reception and in pulse-handling circuits, it is not important in audio receivers because ears are insensitive to phase distortion. But phase delay becomes important when the same sound reaches through two different paths. If the path difference between two sounds is 180º, these will cancel. If the sound is coming from two loudspeakers and if they are so connected that the signal in one is 180º out of phase of the signal in the other, the person listening to both the sounds will hear nothing or will hear the sound with poor intensity. This is the reason why speakers should all be connected in the same phase, i.e., live wire to the same end of the coil in all speakers.

Amplitude Distortion When the wave shape is distorted in amplitude due to non-linear characteristics of an amplifier it is called amplitude distortion. It is also called *non-linear distortion* and is shown in Fig. 8.4. The operation point for input II and input III curves is on the non-linear portion of the characteristic curve and hence, output is distorted. The operation point for input I is on the linear portion of the curve and gives undistorted output.

(Intermodulation of two signals also takes place due to non-linearity of a diode or a transistor. The intermodulation produces new signals as a result of sum and difference of frequencies. A strong unwanted signal may modulate the desired signal due to non-linearity of the amplifier and will cause cross-talk.)

Saturation of tape magnetism will also cause non-linear distortion. Similarly, strong recording signals in disc-recording will cause overlapping of grooves in adjacent tracks.

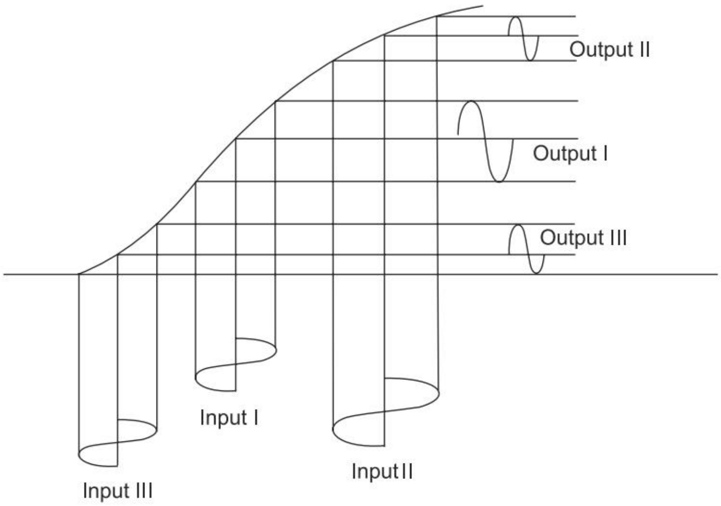


Fig. 8.4 *Non-linear distortion (Input – II and Input III gives rise to distorted outputs. Input – I is on the linear portion on the curve, hence, its output is undistorted.*

Non-linear distortion can be minimised by working on the linear portion of the characteristic curves of transistors and by using negative feedback, adequate decoupling and proper biasing and matching. Transformer core saturation also causes non-linear distortion. Use of push-pull circuits eliminate, such saturation. Reduction of too strong signals before recording will eliminate the problem of saturation in magnetic tapes and of overlapping of grooves in adjacent tracks.

Spatial Distortion The sources of sound in a programme are positioned at different places, in the space available on a stage. Hence, each instrument or singer has a direction with respect to the two ears of the listener. Ears are sensitive to the sense of direction and interpret the directions of instruments and singers correctly. This phenomenon is called *Stereophonic effect*, and this effect should be produced in hi-fi systems. Absence of stereophonic effect is called *spatial distortion*. The spatial distortion is reduced by using at least two independent channels of amplification, recording and reproduction for left and right sides of the stage. The basic stereophonic system is shown in Fig. 8.5. The detailed description of a stereophonic system is given in Chapter 10.

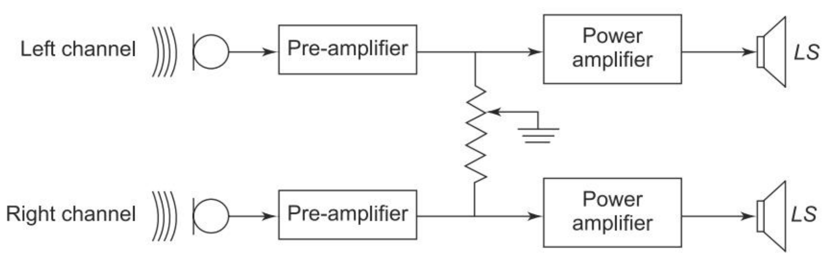


Fig. 8.5 *Basic stereophonic system*

Transient Distortion When a system fails to follow sudden large changes in the sound level, it is called *transient distortion*. Transients can throw a loudspeaker into oscillations at its natural frequency which continue even after the transient has vanished. This effect is known as *hangover*.

8.4 METHODS OF REDUCING DISTORTION

1. Negative feedback reduces distortion.
2. Proper designing of circuits so that series reactance is low and shunt reactance is high. This will improve frequency response.
3. Dynamic range of recorders can be extended by using processing and equalising circuits so that high intensity at lower frequencies is de-emphasised and low intensity at high frequencies is emphasised before recording. During playback, a reverse process called equalisation is adopted which restores the original signal. This reduces overmodulation and undermodulation effects. A compact disc can give an excellent dynamic range without the need of pre-emphasis and equalising circuits.
4. Use of stereophonic method reduces spatial distortion.
5. Use of multiple speaker columns will reduce frequency distortion caused due to mass and compliance of the moving parts of speakers. Similarly, properly designed microphones can give flat frequency response over the audible range of frequencies.

SUMMARY

- ☞ Any unwanted sound present in the environment, or coming out of the loudspeaker in an audio system is called 'noise'.
- ☞ While musical sound has periodicity, regularity of frequency, amplitude and shape, and continuity, noise has neither of these characteristics. It is nonperiodic, irregular, noncontinuous and of random nature.
- ☞ Audible noises are of two types: (1) environmental noise, and (2) electrical noise converted into sound by loudspeakers.
- ☞ Environmental noise consists of conversations of audience, street traffic, noise of fans, type writers, machines, etc., and the transient noise (sudden high intensity sound of short duration like thunder, explosions, etc.).
- ☞ The general noise level present in the environmental, excluding transient noise, is called ambient noise.
- ☞ The ambient noise level of studio is about 20 dB and that of a sports stadium, about 60 dB.

- ☞ Environmental noise can be reduced by using proper insulation.
- ☞ External noise transmitted through solid structures can be reduced by making the outer walls massive or by breaking the sound path.
- ☞ Proper placement of microphones is also helpful in reducing ambient noise.
- ☞ Automatic level limiters in amplifiers help in reducing transient noise.
- ☞ Electrical noise can be white noise produced by active and passive components, noise produced by sparks and rf transmissions, noise produced by record/play motor (wow, flutter and rumble), noise produced by positive feedback, and noise produced by ripple voltages in power supply.
- ☞ These noises can be reduced by proper shielding and grounding, use of trap circuits, use of low-noise pre-amplifiers, decoupling of common lines, mechanical filtering of motor vibrations, use of servo-controlled synchronous motor, use of stabilised power supply, proper maintenance of heads and use of special techniques to improve signal-to-noise ratio (like Dolby method).
- ☞ Noise is harmful as it strains the nervous system. Hence, signal-to-noise ratio should be high. For hi-fi systems, signal-to-noise ratio should not be less than 50 dB.
- ☞ Distortion is different from noise. It is deformation of the wanted signal. Thus, distortion is an internal affair with the signal, while noise is something which comes from outside the wanted signal.
- ☞ Distortion can be of five types: frequency distortion, phase distortion, amplitude (or non-linear) distortion, spatial distortion, and transient distortion.
- ☞ Distortions can be reduced to as low as 1% or even lower by using amplifiers having good dynamic range, proper design of circuits, negative feedback and use of stereophony system.

Review Questions

1. Distinguish between noise and music.
2. What is environmental noise? How can it be reduced?

3. What do you understand by electrical noise? Explain various types of electrical noises that concern audio amplifiers and suggest remedies to reduce the same.
4. Define the term 'signal-to-noise ratio'. How can 'pre-emphasis' before recording and 'de-emphasis in playback' improve signal-to-noise ratio?
5. Explain the terms:
   1. Ambient noise
   2. Noise due to self oscillations
   3. Hum
   4. Transient noise
6. Explain the term 'distortion'. How is it different from noise? Describe different types of distortions, their causes and remedies.

Short-Answer Questions

1. What is noise?
2. What is musical sound?
3. What is ambient noise?
4. Why is typical noise level in sound recording studio kept less than the noise level in a TV studio?
5. What are the harmful effects of noise pollution?
6. What is hum noise?
7. What is hissing noise?
8. How are wow, flutter and hum noises produced in tape recordings?
9. Whether it is noise level or signal-power to noise-power ratio, which is more important.
10. Distinguish between distortion and noise.
11. What is spatial distortion?
12. What is the cause of roll-off at low frequencies in an *RC* coupled circuit?
13. What is the cause of roll-off at high frequencies in an *RC* coupled circuit?
14. Why is phase distortion in the same sound not important, while phase distortion caused due to delay when the same sound travels through two different paths is important?

Multiple-Choice Questions

1. What is the name given to unwanted sound?
   (a) Distortion
   (b) Noise
   (c) Instrumental music
   (d) Vocal music
2. Electrical noise is reproduced as unwanted sound by
   (a) microphone
   (b) amplifier
   (c) cassette
   (d) loudspeaker
3. What type of noise is produced by a resistor?
   (a) white noise (b) black noise
   (c) pink noise (d) red noise

4. Amplitude distortion is caused by
   (a) linear characteristic curve
   (b) non-linear characteristic curve
   (c) inadequate filtering
   (d) *RC* coupling
5. Which of the following will be classified as transient noise?
   (a) Rattle of leaves
   (b) Conversation by audience
   (c) Thunder
   (d) Noise produced by a fan
6. Which of the following phenomena reduces distortion and gain both?
   (a) Positive feedback
   (b) High Vcc
   (c) Negative feedback
   (d) Decoupling circuit
7. Which of the following stages is mainly responsible for the overall noise in the systems input?
   (a) Preamplifier
   (b) Mixer
   (c) Power amplifiers
   (d) Loudspeaker
8. Hum noise is generally caused by
   (a) power supply unit
   (b) audio amplifier
   (c) loudspeaker
   (d) ambient environment
9. Typical ambient noise for a TV studio is
   (a) 30 dB (b) 20 dB or less
   (c) 40 dB (d) 50 dB
10. Radio frequency noise is not relevant to
    (a) mixer stage
    (b) IF stage
    (c) baseband recording system
    (d) preamplifier of radio receiver

Answers

Short-Answer Questions

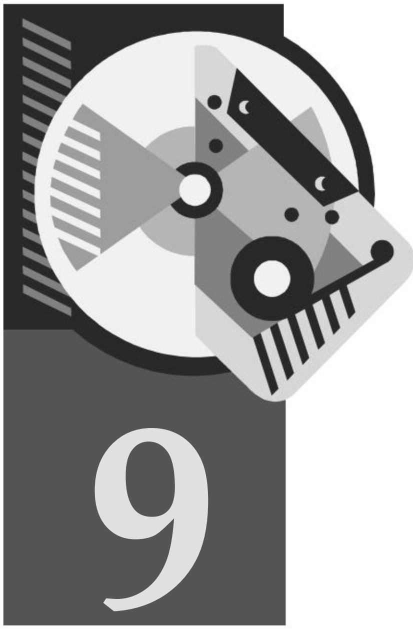
1. Any unwanted sound present in the environment or coming out of the loudspeaker in an audio system is called noise. It consists of pressure variations of random nature without any regularity of frequency, shape and amplitude and also without continuity.
2. Musical sound has periodicity, regularity of frequency, regularity of shape, regularity of amplitude and continuity.
3. The ambient noise is the average noise present in the environment. The examples of environmental noise are rustling of leaves, conversation amongst audience, sound produced by street traffic, sound produced by fans, typewriters, machines, etc. (The sounds which appear suddenly for short duration, like thunder, shrieks, explorious, etc. are not included in the ambient noise.)
4. Noise level in a sound studio is kept lower than in a video studio because a person can visualise a picture from the dialogues, but cannot visualise dialogues from seeing the video.
5. When the noise level increases beyond a certain specified level,

it may strain the nervous system, may cause mental fatigue, can irritate the workers and lower their efficiency, may impair hearing and may also retard the normal growth of infants.

6. Hum noise is the noise of mains frequency. It is caused if ripples are present in the output of a power supply. It may also be picked up by signal leads if these leads run close and parallel to the mains lead.
7. The hissing noise pertains to rf noise detected and amplified and then converted into sound by the loudspeaker.
8. Slow variations in speed of transport mechanism of tape recorders give rise to noises called wow and flutter. Body vibrations of motors result in a noise called rumble.
9. What is important is that the level of signal should be so high, and of noise so low that signal-to-noise ratio becomes 50 dB or more in high-quality audio systems. Noise level may be very low, but if the signal level too is very low then even the low noise will not be tolerable.
10. Noise consists of the signals outside the wanted signal, while distortion means deformation of the wanted signal itself.
11. Spatial distortion is the distortion which is experienced by a listener when he or she is not able to sense the different directions from which the sounds are coming.
12. Series coupling capacitor attenuates low-audio frequencies, but it does not effect high audio frequencies. So roll-off at low frequencies is due to a series coupling capacitor.
13. Shunt capacitance caused by a base-emitter junction causes-roll off at high frequencies. It does not effect low frequencies.
14. Human ears are not sensitive to phase distortion in the same sound and hence phase distortion in sound systems is not important. (It is important in picture as eyes are very sensitive to phase changes.) However, phase delay becomes important when the same sound reaches the ears through two different paths. Echoes and reverberation are examples.

Multiple-Choice Questions

1. (b)	2. (d)	3. (a)
4. (b)	5. (c)	6. (c)
7. (a)	8. (a)	9. (a)
10. (c)		



9 High-Fidelity (or Hi-fi)

9.1 FIDELITY

The word 'fidelity' means faithfulness. In audio systems, it is used to indicate faithful reproduction of sound. The ideal fidelity should have the following characteristics.

1. Complete exclusion of noise from sound. The Signal-to-noise ratio should be infinite.
2. Flat frequency response for complete audio range of frequencies from 16 Hz to 20 kHz. It means that there should be no frequency distortion.
3. Non-linear distortion (or amplitude distortion) should be nil.
4. High dynamic range from 0 dB to 130-dB signal power.
5. Ability to give sense of direction to identify the relative positions from which different sounds are produced in the original programme. It means that there should be no 'spatial distortion'.
6. Environmental conditions should be simulated where the sound is being reproduced.

9.2 HIGH-FIDELITY OR HI-FI

No sound system can be so perfect as to give ideal fidelity. The best fidelity systems are less than perfect but when sound is reproduced with a high degree of similarity to the original or live sound, it can be said to be of high-fidelity, or simply 'hi-fi'. The listening tests carried out all over the world indicated that the reproduced sound will be called hi-fi, if the following requirements are met by the sound system.

1. Signal-to-noise ratio should be better than 50 dB.
2. Frequency response should be flat within ±1 dB over the frequency range of 40 Hz to 15000 Hz.

3. Non-linear distortion should not be more than 1%.
4. The system should possess dynamic range of at least 80 dB (20 dB whisper to 100 dB loud pop-music over threshold of hearing).
5. Stereophonic effect should be provided.
6. Environmental conditions should be such as to eliminate the external noise in the listening room, and to give the desired reverberation time. This may even be better than at the source of sound, and hence may make the reproduced sound more pleasing.

The microphones, recording amplifiers and devices, pick-ups, playback amplifiers and loudspeakers, all contribute to the high-fidelity of the reproduced sound. Reflections, reverberation, ambient noise also affect fidelity. High-fidelity can be achieved by using low noise components and proper design of the circuits and devices.

The various causes affecting fidelity and remedies are discussed below.

9.3 HIGH SIGNAL-TO-NOISE RATIO

The overall noise in the output of the final power amplifier is the combined noise of hum, hiss, cross talk, self-oscillations, detection of sparks and radio frequency signals, irregularities in magnetism for tape recorders, or in grooves for disc recorders, motor vibrations or unsteady speed, and picking up the ambient noise by microphones, etc. For high-fidelity systems, this noise should be at least 50 dB below the signal at the output.

Signal-to-noise ratio can be improved by using pre-amplifiers of low noise figures, proper shielding, grounding, decoupling and filtering circuits, stabilised power supply, directive microphones, filters to check motor vibrations and servo-controlled synchronous motors. Special circuit techniques like pre-emphasis and equalisation and Dolby method improve Signal-to-noise ratio substantially. Many a times, dry solder joints disconnect a filtering or trap circuit and allow undesirable signals to enter the electronic system. Hence, in hi-fi systems, soldering must be perfect. Use of diamond stylus and vinyl disc reduces roughness of the grooves resulting in reduction of noise.

Example 9.1 *The amplified audio signal measured at the loud-speaker load of 4 Ω is 2 V. In the absence of a signal, the electronic voltmeter measures 10 mV. Find Signal-to-noise ratio of the amplifier in dB. Does it meet hi-fi requirement?*

Solution

Output power in the presence of signal $= \dfrac{2\times 2}{4}$ W $= 1$ W

Output power in the absence of signal $= \dfrac{10\times 10}{1000\times 1000\times 4}$ W

$= 0.25 \times 10^{-4}$ W

Signal-to-noise ratio

$$= \frac{1}{0.25 \times 10^{-4}}$$

$$= 4 \times 10^4 = 46 \text{ dB}$$

It does not meet hi-fi requirement as the minimum value of signal-to-noise ratio should be 50 dB in a hi-fi system.

9.4 FLAT FREQUENCY RESPONSE

It is affected by series and shunt reactances. Frequency response can be made flat within ±1 dB for the audible range of frequencies by designing the circuits so that series reactance is low and shunt reactance high. In audio amplifiers, these reactances are provided by coupling capacitor and shunt capacitor, respectively. The coupling capacitor used is such that its reactance at 40 Hz is about one-tenth the load resistance. The shunt capacitance is made of the leads running parallel and the junction capacitance of transistors. Its reactance at the highest audio frequencies should be about 10 times the load resistance. Thus, a transistor of low junction capacitance is used and lay-out of wires is properly designed. In loudspeakers and microphones, mass and compliance act as inductors and capacitors. Columns of woofers and tweeters (or, more preferably woofers, squawkers and tweeters) are used to give flat response for the whole audio range. Among microphones, condenser microphones give excellent frequency response, and are preferred in high-fidelity recording systems. Natural resonant frequencies of mechanical parts of transducers boost up some frequencies and hence, disturb the fidelity.

Sometimes, large changes are sudden and the sound circuits, particularly loudspeakers, fail to follow the same. This type of distortion is called *transient distortion*. This distortion causes oscillations at natural resonant frequencies. This can be prevented by designing the system in such a way that the natural frequencies are below 40 Hz and above 15 kHz. Bass and treble circuits are also incorporated in hi-fi amplifiers to tailor the frequency response to an individual's taste.

Example 9.2 *At 1000 Hz, the output of an amplifier is 10 V. At 100 Hz and 10000 Hz it is reduced to 70.7%. What is the 3-dB frequency response? Is it a hi-fi amplifier?*

Solution

Reduction to 70.7% in output voltage amounts to $\frac{1}{\sqrt{2}}$ times the output voltage.

For voltages, $\frac{1}{\sqrt{2}} = -3$ dB

Thus, 3-dB frequency response is 100–10000 Hz. It is not a high-fidelity amplifier, as the frequency response in the hi-fi system should be at least 40 to 15000 Hz within ±1 dB.

9.5 LOW NON-LINEAR DISTORTION

In a high-fidelity system, non-linear distortion is kept below 1% by using negative feedback in amplifiers, designing bias circuits (so that operating point is in the middle of the linear portion of the characteristic curve), using ac bias for magnetic recording and eliminating core-saturation by use of push-pull circuits.

9.6 LARGE DYNAMIC RANGE

A hi-fi system is required to handle large range (80 dB) of power, from whisper (20 dB or threshold value) or softest passages in an orchestra to loud pop music to loudest passages of orchestra (100 dB).

The ratio, expressed in decibels, of the loudest and quietest sound to which a sound system can respond is called *dynamic range*. Dynamic range of a capacitor microphone is 115 dB; piano, 70 dB; studio amplifier, 85 dB; domestic cassette recorder, 40 dB; and AM radio-transmission, 30 dB. As in all these devices, the softest note is comparable to noise, the dynamic range also represents maximum signal-to-noise ratio.

Dynamic microphones and loudspeakers are capable of withstanding the large change in loudness. The solid-state amplifiers can also be designed to have high dynamic range. The difficulty arises in case of gramophone records. Here, the high intensity may cause overlapping of the signals between adjacent grooves and the low intensity may cause merger of the softest signals with the groove noise or background noise. Groove noise is more apparent at high frequencies. The pre-emphasis of high frequencies before recording will increase signal-to-noise ratio. The de-emphasis in playback will restore the original proportions of signal components and will reduce the noise to give high signal-to-noise ratio. The Dolby method of emphasising weak signals frequency wise is effective in improving signal-to-noise ratio by 10–15 dB. Distance between grooves can be increased by increasing the size of the discs for hi-fi recording and by making grooves of smaller size, called *microgrooves*. Moreover, the intensity of sound is more in low frequencies, and hence to have high dynamic range we can reduce this intensity before recording and increase it later in playback (by equalising circuit) to restore the originality. In magnetic tapes also, high intensity may saturate the core, causing non-linear distortion. Hence, cores of stalloy are used. An ac biasing also increases the dynamic range of the recorded signal. The large dynamic range will eliminate the intermodulation distortion also.

A compact disc, based on digitising audio and then recording it on disc optically by means of laser beams, provides almost ideal fidelity. It neither needs ac biasing nor equalisation and its characteristics are

- *Frequency response* 20 to 20000 Hz within ±0.5 dB
- *Dynamic range* 90 dB

- *Signal-to-noise ratio* 90 dB
- *Non-linear distortion* 0.05%
- *Wow and flutter* Absent
- *Channel separation* 80 dB

9.7 CREATING SENSE OF DIRECTION

A high-fidelity system requires that the reproduced sound should appear coming from various directions as it happens in the original programme. This effect is created by using two independent channels of amplification from microphone to loudspeaker. This is known as *stereophonic system*. In magnetic recording, two independent tracks are recorded for each channel and for disc recording, the technique of 45–45 degree angle recording is used.

9.8 CREATING A PLEASING ENVIRONMENT

Ambient noise in the room, external noise leaking into the room through openings and solid structures, reflections from walls, ceiling, floor and other surfaces make a kind of special environment for sound. When environmental conditions at a different place, where sound is being reproduced, are not identical with the environment where the original sound was produced and recorded, a kind of distrotion, called environmental distortion, is produced.

For example, the new environment may have more ambient noise, or more external noise, or less or more reverberation time, or even distinct echoes. However, the new environment may also sometimes be better than the original environment to give a more pleasing effect than what was present in the original place. To create an environment to give a more pleasing effect, the following action should be taken:

1. External noise should be reduced by using sound-insulation techniques.
2. Reverberation time should be controlled for giving desired richness or liveliness to the music by using absorbents.

To sum up, the high-fidelity systems need very careful design right from microphones to amplifiers, recording mechanism, pick ups, playback amplifiers and loudspeakers, and careful planning for proper stereophonic and environmental conditions. A real high-fidelity system produces dramatic effect, enhances the pleasure of the listeners, particularly music lovers, and hence is worthy of the high cost involved in well engineered and matched equipment.

SUMMARY

- High-fidelity (or hi-fi) means reproduction of sound with high degree of similarity to the original sound. For high-fidelity, the following requirements should be fulfilled by the system.
  1. Signal-to-noise ratio should be better than 50 dB.
  2. Frequency should be flat within ±1 dB over the frequency range of 40 to 15000 Hz.
  3. Non-linear distortion should not be more than 1%.
  4. Dynamic range should be at least 80 dB (from 20 dB whisper to 100 dB pop music over threshold value).
  5. Stereophonic effect should be present.
  6. Environmental condition pertaining to noise and reverberation should not be worse than that at the source of sound.
- Noises in the reproduced sound are hum, hiss, cross-talk, self-oscillations, rf signals detected, irregularities in the recording medium, and low frequency noises due to unsteadiness in motor speed (wow and flutter) and motor-body vibrations (rumble). All these noises reduce signal-to-noise ratio.
- Low frequency response of an audio and mechanical system is affected by series capacitance or mechanical compliance and high-frequency response by shunt capacitance or compliance (or by series inductance and mechanical mass).
- Good design can provide the desired frequency response. Also, natural frequencies of electro-mechanical transducers should be kept below 40 Hz and above 15 kHz as far as possible. Combination of woofers, squawkers and tweeters provide good frequency response in loudspeakers
- About microphones, capacitor microphones give excellent frequency response.
- Selection of low noise transistors for pre-amplifiers, adequate filtering and decoupling, use of servo-controlled synchronous motors, steady supply voltage, perfect solder joints, use of diamond stylus and vinyl disc, high dynamic range of the system, all contribute towards reduction of noise. The Dolby system of amplifying low level signal before recording and reducing it in playback gives 10–15 dB improvement in signal-to-noise ratio.

- ☞ Non-linear distortion can be reduced by properly selecting the operational point, bias and using negative feedback in amplifiers.
- ☞ Disc and magnetic recorders present difficulties about dynamic range. Processing of signals before recording and equalisation in the playback remove this problem. Development of compact disc is the ultimate answer of hi-fi problems.
- ☞ For directional effect, stereophonic system of sound is adopted.
- ☞ Environmental conditions are improved by using suitable absorbers and insulators. The new environment can be made even better than the original environment.

Review Questions

1. What are the characteristics of ideal fidelity?.
2. Define hi-fi. Mention the essential requirements for a hi-fi sound system and discuss how these are achieved in practice.
3. Define 'dynamic range'. Give the value of dynamic range for a normal musical concert. Discuss how a good dynamic range is obtained.
4. Discuss the statement that compact disc is the ultimate solution for the hi-fi problems.

Short-Answer Questions

1. Is high-fidelity an ideal fidelity? Give reasons for your answer.
2. What is ambient noise?
3. How can ambient noise be improved?
4. Can ambient noise be improved in a separate room where the recorded programme is being reproduced?
5. How can non-linear distortion be reduced?
6. What is spatial distortion?

Multiple-Choice Questions

1. What is the typical value of signal-to-noise ratio in a practical high-fidelity system?
   (a) 30 dB (b) 40 dB
   (c) 50 dB (d) 100 dB
2. Which one of the following is not the function of a coupling capacitor ?
   (a) To block dc
   (b) To pass signal
   (c) To cause roll-off at low audio frequencies
   (d) To cause roll-off at high audio frequencies
3. Coupling between two different channels results in
   (a) increased signal
   (b) high sensitivity
   (c) cross-talk
   (d) high stability
4. What is the cause of ripple voltage (hum noise) from loudspeakers?
   (a) Inadequate filtering in power supply
   (b) Oscillator output varying
   (c) Gain of audio amplifier varying
   (d) Input signal too weak
5. What device is used to improve output at high frequencies ?
   (a) Tweeter (b) Woofer
   (c) Squawker (d) Horn speaker
6. A compact disc can produce flat response for a frequency range equal to
   (a) 10 to 1000 Hz
   (b) 20 to 20000 Hz
   (c) 100 to 10000 Hz
   (d) 200 to 20000 Hz
7. Spatial distortion is reduced by
   (a) high quality microphones
   (b) high quality speakers
   (c) stereophonic system
   (d) low noise figure
8. Wow and flutter are produced in the recordings due to
   (a) variations in tape speed
   (b) body vibration of motor
   (c) self oscillations in amplifiers
   (d) coupling between two leads
9. Which device gives the best dynamic range ?
   (a) Gramophone record
   (b) Tape recorder
   (c) Cassette
   (d) Compact disc
10. Which of the following does not contribute to high-fidelity of the reproduced sound?
    (a) Pick-ups
    (b) Wein bridge oscillator
    (c) Playback amplifiers
    (d) Microphones

Numerical Problems

1. Total power output of a hi-fi amplifier measured in a distortion factor meter is 10 watts and the harmonic content included in it was measured to be 0.1 watt. Find percentage distortion of the amplifier.
2. The output voltage of an amplifier was 0 dB at 1000 Hz, 3 dB down at 20 Hz and 20 kHz, 2 dB down at 30 Hz and 17 kHz and 1 dB down at 40 Hz and 15 kHz. What is the best fidelity range for this amplifier?

Answers

Short-Answer Questions

1. No. Ideal fidelity covers 16 Hz to 20000 Hz for output within ±1 dB with reference to output at 1000 Hz. High-fidelity covers 40 Hz to 15000 Hz for the same output.
2. Ambient noise is the noise generally present in the hall, like conversations of people, noise due to fans, noise entered in the hall through windows and doors, etc. It does not include transient noise like thunderstorms.
3. Ambient noise can be improved in the recording room by using acoustic absorbers at appropriate places, proper insulation in the design of Hall's walls (e.g., outer walls made massive), windows and doors, and by breaking the sound path. Proper placement of microphones is also helpful in reducing ambient noise.
4. Yes. If the ambient noise present in the production room is not allowed to reach the microphone then at the time of reproduction in another room, ambient noise would be less than experienced by the audience in the production room, provided that the new room contains less ambient noise.
5. Non-linear distortion can be reduced by properly selecting the operating point and using negative feedback. The dynamic range of the amplifier should also be high.
6. In the original programme, sound comes to the listener from various directions (direct sound, sound reflected from the walls, ceiling and floor). These reflections result in reverberation of sound and make the sound lively. If the same reverberation is not present in the reproduced sound, it is called spatial distortion.

Multiple-Choice Questions

1. (c)	2. (d)	3. (c)	4. (a)
5. (a)	6. (b)	7. (c)	8. (a)
9. (d)	10.(b)		

Numerical Questions

1. (1%)
2. (40 Hz to 15 kHz)

Stereophony

10.1 MEANING OF STEREOPHONY

The word ‘stereophony’ is derived from two Greek words: ‘*stereos*’ and ‘*phone*’, meaning ‘solid’ and ‘sound’, respectively. Thus, stereophony means *solid sound* or ‘three-dimensional sound’. In a programme (say an orchestra), different sources of sound are placed at different positions on the stage. When such a programme is amplified and reproduced (or recorded and played), the originality of sound would be restored if the reproduced sound appears to come from different directions simulating the original programme. This three-dimensional reproduction is called *stereo*. In fact, the term was first used as stereo vision for 3-dimension pictures. The word ‘stereo’ has been borrowed from there. In stereophonic systems, we locate sounds (that is, direction and depth of sound source) as in a real programme.

10.2 STEREOPHONY IN THE HUMAN SYSTEM OF HEARING

There is a minute difference of phase and intensity in the sounds reaching the two ears. This difference is interpreted by the brain in such a way so as to enable the listener judge the directions from which the sounds are coming. Thus, the human system of hearing is stereophonic. The ear–brain combination is so sensitive that if the sounds reaching the two ears differ in time by only 10 μs, the brain will detect the direction correctly.

10.3 DIFFERENCE BETWEEN STEREOPHONY AND MONOPHONY

While the word ‘stereophony’ means ‘solid sound’ (or three-dimensional sound), the word monophony means ‘one sound’, or ‘one direction sound’, or ‘one source sound’.

In monophonic sound, there is only one amplifier. The amplifier combines all the signals and produces one resultant signal. The amplifier output may be fed to a single or several loudspeakers. All loudspeakers shall give the same resultant sound. In such a case, the ears will interpret the reproduced sound to be coming only from one source of sound. The monophonic sound is cheap to produce but lacks naturalness.

In stereophonic sound, the independent amplifiers have their own set of microphones for input and their own set of loudspeakers for output. The recording of stereo-sound requires special recording units, so that the channels may be recorded and reproduced as independent channels.

In stereo amplifiers, the actual sources of sound are virtually transferred to the respective loudspeakers. There are at least two channels, *left channel* and *right channel*, although the more the merrier. Technical complexity and cost have made two channels as the norm. Basic monophonic and stereo systems are shown in Figs. 10.1 and 10.2, respectively.

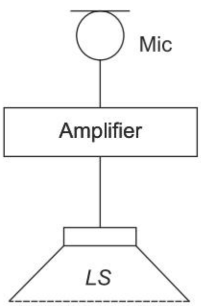


Fig. 10.1 *Basic monophonic system*

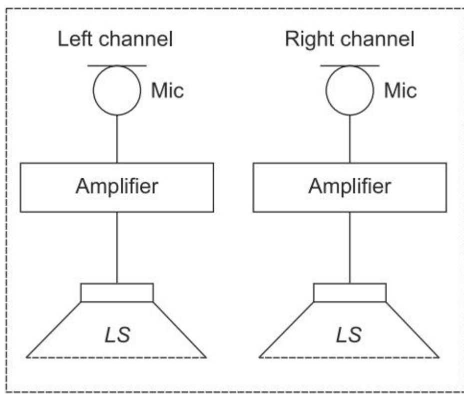


Fig. 10.2 *Basic stereophonic system*

10.4 IDEAL STEREO SYSTEM

An ideal stereo system should be the simulation of the human hearing system. This can be done by placing two microphones about 25 cm apart on a stand, like a pair of ears on the head. Sound from each microphone is separately recorded and amplified. Two earphones (not loudspeakers) can be used such that the sound picked up by the left microphone goes to the left ear only and not to the right ear. Similarly, the sound picked up by the right microphone goes to the right ear only and not to the left. Such a system is shown in Fig.10.3. It, being the simulation of the human hearing system, will be an ideal stereo system. Dots in Fig. 10.3 represent sources of sound.

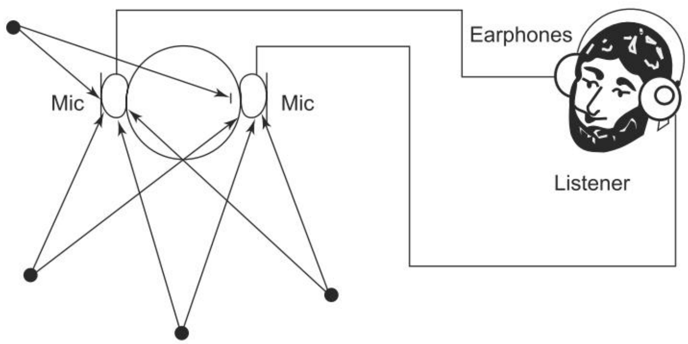


Fig. 10.3 *Ideal stereo system*

10.5 PRACTICAL STEREO SYSTEM

The above theoretical system of using earphones is not at all practical. The moment we use loudspeakers in place of earphones, the two channels of sound do not remain isolated. The sound from the left speaker will reach not only the left ear but the right ear also. Similarly, the sound from the right speaker will reach not only the right ear but the left one also. The ears are smart enough to recognise that the sounds are coming from two separate and distinct sources (loudspeakers) and not from the actual directions of the original programme. However, the ears can be deceived to interpret the sounds from loudspeakers as if these were coming from different directions, when the following procedure is adopted for a practical stereo system.

1. Place the two microphones 3 to 4 metres apart and not 25 cm apart. This will enable the left microphone to pick up sound more strongly from the left than right. Similarly, the right microphone will pick up sound more strongly from the right than from the left. Sound from the middle will be picked up equally well by the two microphones.
2. The output of the two microphones is fed to two separate channels of the amplifier (both amplifiers are in a single cabinet). Each amplifier's output is fed to its column of hi-fi loudspeakers. The two columns of loudspeakers are also placed 3 to 4 metres apart. Thus, each speaker column is placed in the same relative position as a microphone. This way, the sources of sound have been simulated in the two speaker columns. The outputs of the speakers will combine in such a way that the listener gets the sensation of direction and depth comparable to the actual stereophony effect of human ears in the original environment (i.e. without amplifiers and loudspeakers).

A practical symbolic stereophony system using two microphones and the same number of loudspeakers is shown in Fig. 10.4.

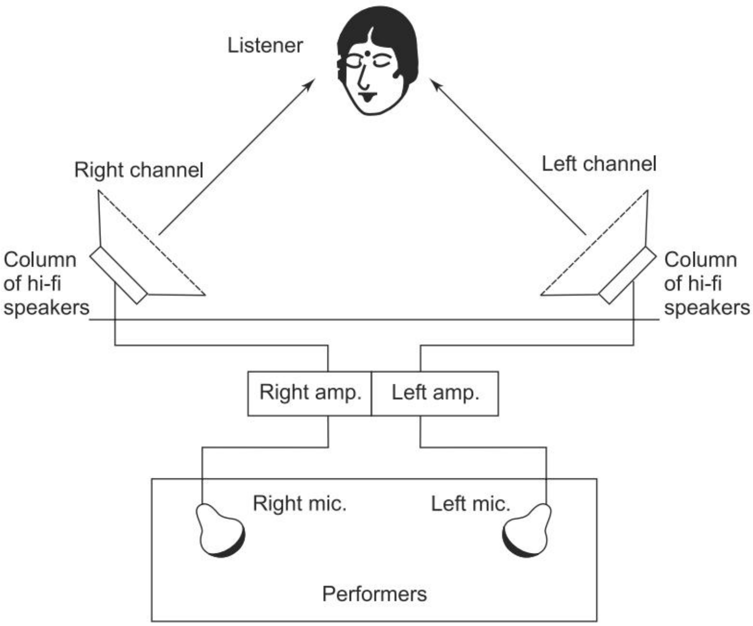


Fig. 10.4 *Practical stereophonic system*

10.6 QUADRAPHONIC AND SURROUND-SOUND SYSTEMS

In a multiple-channel stereo system, if we can include reflections from the ceiling and walls (which occur in a natural environment), the reproduction would be more natural. This can be done by using four channels for recording and reproduction, and this technique is known as *quadraphony*, and the system is called *quadraphonic* (or *quadrophonic*), in short, *Quad system*. While recording, two extra microphones pick up reflection from the rear wall. During reproduction, two extra loudspeakers are placed behind the listener in the corners of the listening room to imitate the actual conditions in the concert hall. The system did not become popular due to lack of compatibility with the existing two-channel stereo system.

After failure of the 4-channel system on account of cost and non-compatibility with the existing 2-channel system, efforts to create the effect of three dimensions did not stop. A new system was developed in UK which was called *ambisonics*. In this system, a special microphone having four diaphragm was used. The diaphragm were so arranged that they sensed the sound from all directions. The four outputs comprised one omnidirectional and three bidirectional signals. The bidirectional signals corresponded to left-right, front-back and up-down sounds. The control circuit provided not only the surround sound, but also the stereo sound , making the machine compatible with the 2 channel stereo system. Although compatible, this was also a costly machine.

Surround Sound Effect To keep the cost within reasonable limits, surround-sound effect was produced in digital systems by introducing artificial delay in the signal produced by two channels of the stereophonic system. The delay so introduced simulated the natural delay in sounds reflected from the walls and ceiling. Thus surround sound effect was achieved at reasonable cost.

10.7 STEREO RECORDING ON TAPE AND REPRODUCTION

For recording the stereophonic sound on a tape, each channel's output goes to a separate head and records a separate track on the magnetic material coating of the tape. Thus, a two-channel system will have basically two tracks as shown in Fig. 10.5. Double the number of tracks can also be recorded, say 2 in one direction and 2 more in the reverse direction. In such a case, the tape can be played in one direction first. It is, then, turned over and is played again in the reverse direction. (For reproduction, audio signal is produced in the two coils of the heads from the respective tracks. These signals are fed to two independent amplifier channels which give stereo sound output.)

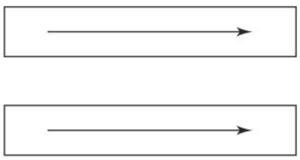


Fig. 10.5 *Basic stereo tracks*

Block Diagram of Stereo Tape Recorder Stereo tape recording is shown in the form of a block diagram in Fig. 10.6. Left-channel and right channel microphone outputs pass through independent amplification circuits, using a pre-amplifier, processor (emphasis of high frequency signals), driver amplifier, power amplifier and magnetic head. The two gaps bear against the tape surface and magnetize it in accordance with the audio signals. Two tracks in the same direction separated from each other are recorded. When the tape is reversed, two more tracks can be recorded on the same tape. A balance control potentiometer after the pre-amplifier adjusts the signals in the two channels such that for equal inputs, the outputs are also equal. (For playback, outputs of the heads will be fed to the normal stereo amplifier.)

10.8 TAPE CARTRIDGE AND CASSETTE TAPE

The open-reel tape system is inconvenient in handling as it needs loading and threading. It is also susceptible to breaking. To solve these difficulties, 'tape cartridge' and 'cassette tape' have been developed. These contain internal spools which hold the tape which is ready to play at any time. The tape remains protected and no loading or threading is required.

Tape Cartridge The cartridge runs continuously and there is no provision of reverse winding. All the tracks are in one direction only. Figure 10.7 shows two-channel stereo tracks. The tracks are 8 in number, all in the same direction. The

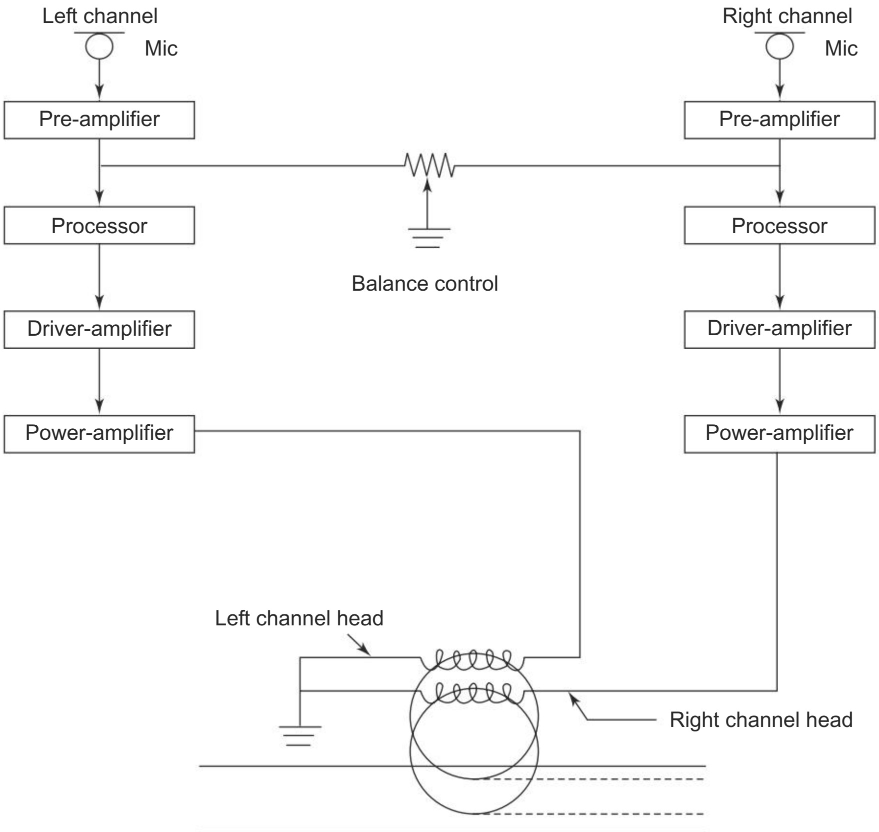


Fig. 10.6 *Block diagram of a basic stereo tape recorder*

first four tracks belong to the left channel and the next four to the right channel of stereo recording. The speed of tape in cartridge is 9.5 cm/s. Tracks 1 and 5 (full lines) are played first. Then, the tape head picks up tracks 2 and 6 (dotted line), then 3 and 7 (dashed lines) and finally 4 and 8 (wavy lines). The heads thereafter come back to play tracks 1 and 5, and so on. Playing will be continuous and will go on repeating until the machine is made 'off'.

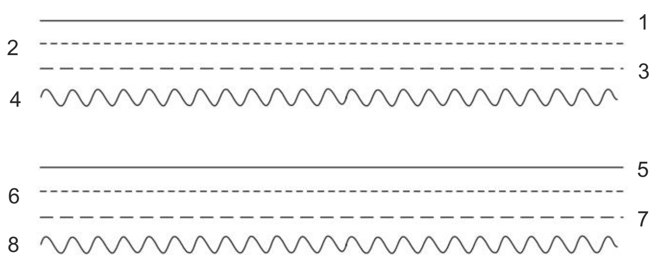


Fig. 10.7 *Stereo tracks in a tape cartridge*

Cassette Tape Unlike a cartridge tape, provision of reverse recording and playback is available in a cassette tape. It is very compact and is most suitable for home recording and playback. Two-channel stereo tracks on cassette are shown in Fig. 10.8. The stereo tracks 1 and 3 are in one direction and 2 and 4 in the reverse direction.

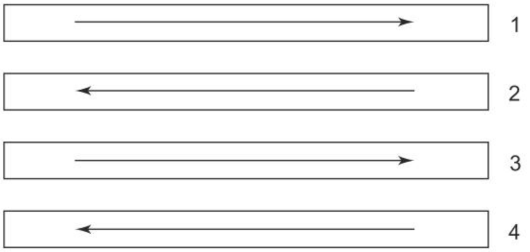


Fig. 10.8 *Two-channel stereo tracks in a cassette tape*

The speed of a cassette tape is 4.75 cm/s. It can hold a programme up to 2 hours (one hour in each direction). Cassettes are designated as C-30, C-60, C-90 and C-120. The digits indicate duration of the programme recorded on both the tracks, for example, C-30 means a cassette which can be played for 15 minutes in one direction and for another 15 minutes in the reverse direction.

10.9 HI-FI STEREO REPRODUCING SYSTEM

Figure 10.9 shows the block diagram of a high-fidelity stereo reproducing system.

High-fidelity sound can be obtained from the recorded stereo tape or in live system from the microphones. (Stereo signal can also be obtained from the record player.).

The stereo signal is fed to two independent amplification channels through a tape-mic switch. The amplifier system consists of a low noise high gain pre-amplifier, equalizer, well designed amplifiers giving flat frequency response and little distortion by using negative feedback circuit and then the matching transformer. (A balancing circuit is incorporated to balance out any imbalance in the characteristics of otherwise identical circuits.) The secondary of the matching transformer of each channel is connected to the respective loudspeaker column. For hi-fi, the loudspeaker columns consisting of woofer, squawker and tweeter are used.

All the blocks are designed so as to get flat frequency response (from 40 to 15000 Hz), little distortion (less than 1%), high signal-to-noise ratio (more than 50 dB) and high dynamic range (100 dB) to achieve the final output of high-fidelity.

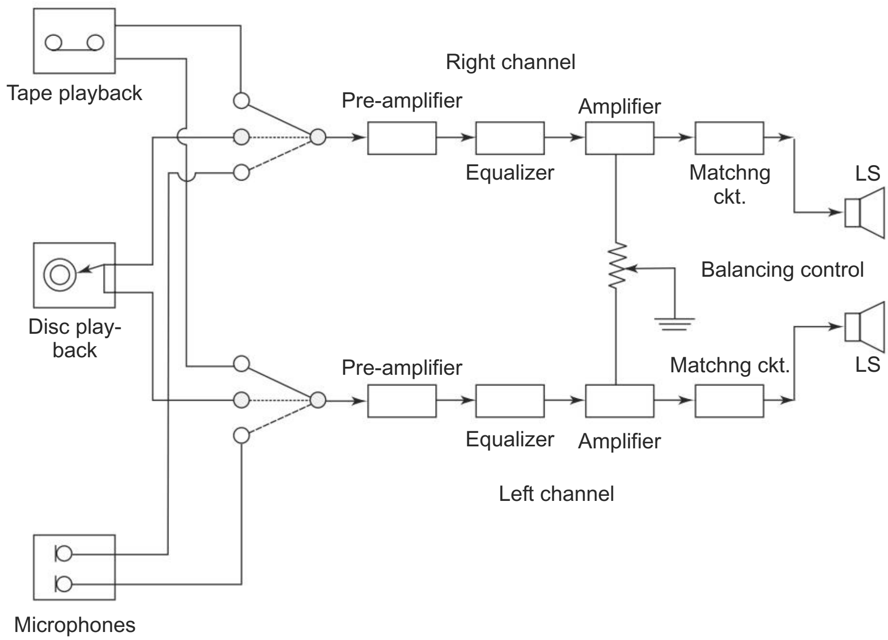


Fig. 10.9 *Block diagram of hi-fi reproducing system*

10.10 STEREO CONTROLS

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 and ICs and positioning of loudspeakers and furnishings with respect to the listener. The circuit used to compensate for such variations is called *Balance Control*.

A simple balance circuit is shown in Fig. 10.10. The balance control is a potentiometer. When it is set in the centre, the current through LED I and LED II should be identical, if the signals in the left and right channels are equal. In that case, both LEDs will be equally bright. In case of any inequality, the two brightnesses will also become unequal. When the balance control is moved down, the output of the left channel will increase while that of the right one will decrease, and vice-versa when moved up.

For adjusting the balance control, a monophonic signal is fed to both the channels, so that input is identical. When the two LEDs are made to brighten equally with the help of balance control, the system is balanced for stereo. Some circuits use a centre zero meter instead of LEDs.

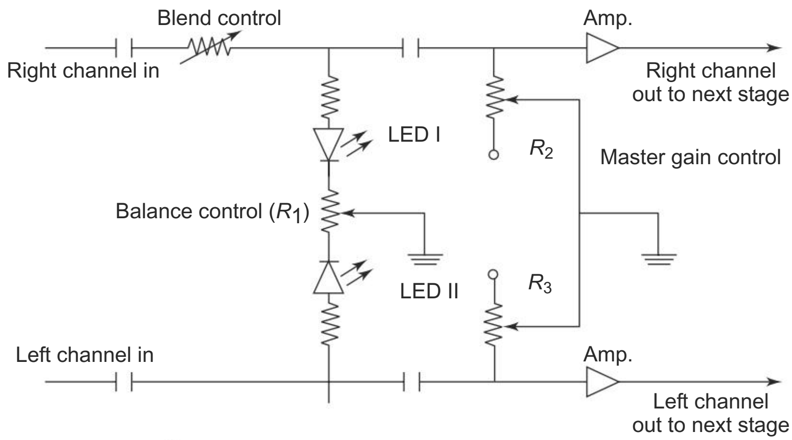


Fig. 10.10 *Balance control, master gain control and blend control*

Loudness Control Sometimes music is at low level of volume (as in case of the background music). 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 big boosting at bass and a little boosting at treble is called *loudness control*. It boosts audio by +12 dB at 50 Hz and +3 dB at 10 kHz. The loudness control should be used only when sound level is low. Sometimes, the volume control is so designed that it raises the level of audio in accordance with ear's logarithmic response. Such a volume control also acts as loudness control and is known as 'Contour Control'. A basic loudness control is shown in Fig. 10.11.

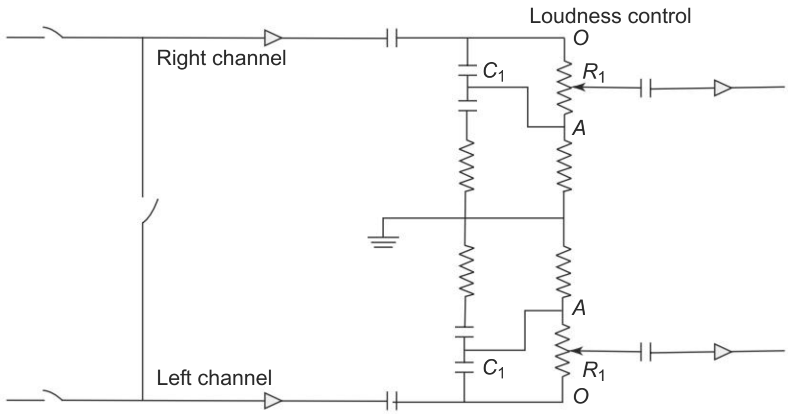


Fig. 10.11 *Loudness control*

At high levels, the R_1 potentiometer is set towards A, so that R_1 C_1 combination comes in series. At low level, R_1 is set at O, and the signal passes direct without attenuation. This provides boosting of +12 dB at 50 Hz and boosting of +3 dB at 10 kHz for adjusting loudness.

Bass and Treble Control It is provided to tailor bass and treble as per personal taste of the listener (Circuit has already been described in detail in Sec.7.6.).

Master Gain Control A master gain control is used for adjusting overall volume without disturbing the balance. This is achieved by using dual concentric shafts, the inner shaft adjusts the balance control and the outer shaft, the overall gain or volume of the amplifier. A typical master gain control circuit is shown in Fig. 10.10. R_1 is adjusted for balancing the two channels and then R_2 and R_3 are adjusted for increasing or decreasing the volume of the channels. R_2 and R_3 are ganged.

Blend Control The stereo effect is diluted by this control when it is too much left-right effect. Diluting is done by disbalancing the two channels. It is shown in Fig. 10.10. 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 and their speakers for a monophonic source of signal. This is doen by a switch called quasi-stereo switch, not shown in the figure.

SUMMARY

- ☞ In a musical programme, instruments are located at different positions on a stage, and the sound reaches the listener from different directions. When such a programme is amplified and reproduced, for good fidelity, the reproduced sound should appear to come from different directions simulating the original programme. Such a system is called *Stereophonic* or simply *Stereo*. In human beings, the two ears provide stereophonic effect.
- ☞ A basic stereophonic system consists of two independent channels of amplification, known as ‘left’ and ‘right’ channels. Left and right amplifiers have their own microphones placed to the left and right of the stage, respectively. Similarly, each amplifier has its own loudspeaker system
- ☞ A multi-channel system can give a more natural solid sound. Quadraphonic (or quadrophonic), or simply quad system, employs 4 channels of amplification for recording and reproduction. Two extra channels take care of reflections from the back wall of a concert hall.

- ☞ Stereo recording on a disc is done by the 45 degree angle method (also known as Westrex method). In this system each channel is recorded along a direction at 45° to the horizontal plane of the disc surface.
- ☞ For recording stereophonic sound on tape, each channel's output goes to a separate head and records a separate track on the tape. Thus, a two-channel stereo system will have basically two tracks on the tape for the same programme. Tape cartridges and cassette tapes have also been developed for stereo, in which loading and threading of tape reel is not required.
- ☞ Balance control, loudness control, bass and treble control, gain control and blend control are used to process the signal for the desired results.

Review Questions

1. What do you understand by stereophony ? How does it differ from a monophonic system?
2. Explain how the human system of hearing is stereophonic. Draw sketches for 2 channel stereo system.
3. What do you understand by quad system ?
4. Explain with neat sketches the stereophonic recording on discs. Draw a block diagram of stereo recording on a disc and write down the function of each block.
5. Explain with neat sketches the stereophonic recording on magnetic tapes. Draw a block diagram of stereo recording on tape and write down the function of each block.
6. What do you understand by tape cartridge and cassette tape ? Give their advantages over an open-reel tape system.
7. Draw block diagram of hi-fi stereo reproducing system and write down the function of each block.
8. Explain the function of the following:
    1. Loudness control
    2. Bass and treble control
    3. Master gain control
    4. Blend control
    5. Quasi stereo switch
9. What is quadraphonic sound? Explain.
10. What do you understand by surround-sound system? How is it achieved in modern digital systems?

Short-Answer Questions

1. What meaning does the word 'stereophony' convey?
2. How is the human hearing system basically stereophonic?
3. What is an ideal stereo system? Explain briefly.
4. What is the difficulty in achieving an ideal stereo system in practice?
5. Why did the quadraphonic system not become popular?
6. What is 45-degree recording on disc?
7. What is the function of balance control in a stereo recording system?
8. What are the difficulties in open reel tape system ? What is the alternative?
9. What is the difference between a cassette and a cartridge?
10. How is stereo recorded in DVD ?

Multiple-Choice Questions

1. Which phenomenon is sensed by listeners in stereophony ?
   (a) Loudness (b) Timbre
   (c) Pitch (d) Direction
2. What is the role of the human brain in creating a stereophonic effect?
   (a) Interpreting the difference in phase of two sounds
   (b) Psychological perception only
   (c) Human brain has nothing to do with sound, and hence no role
   (d) Stereophonic effect is created by ears alone
3. How many independent channels (minimum) are required for a stereophonic sound system?
   (a) 4 (b) 3
   (c) 2 (d) 1
4. In what system is the 45-degree recording used ?
   (a) Monophonic
   (b) Stereophonic
   (c) Quadraphonic
   (d) DVD
5. The least number of tracks for recording stereophonic sound on tape or on DVD are
   (a) one (b) two
   (c) three (d) five
6. What device does not have provision of reverse winding ?
   (a) Tape cartridge
   (b) Cassette tape
   (c) Reel tape
   (d) Stereophonic disc
7. Lack of sense of direction in the reproduced sound is called
   (a) amplitude distortion
   (b) frequency distortion
   (c) phase distortion
   (d) spatial distortion
8. Speed of stereo cassette tape is
   (a) 38 cm/s (b) 19 cm/s
   (c) 9.5 cm/s (d) 4.75 cm/s

9. A two-track monophonic cassette tape type C 120 means it contain a programme of
   (a) 30 minutes/track
   (b) 60 minutes/track
   (c) 120 minutes/track
   (d) 240 minutes/track

10. The surround sound system gives more solid sound than stereophonic system. This statement is
   (a) true
   (b) false
   (c) partially true
   (d) uncertain

Answers

Short-Answer Questions

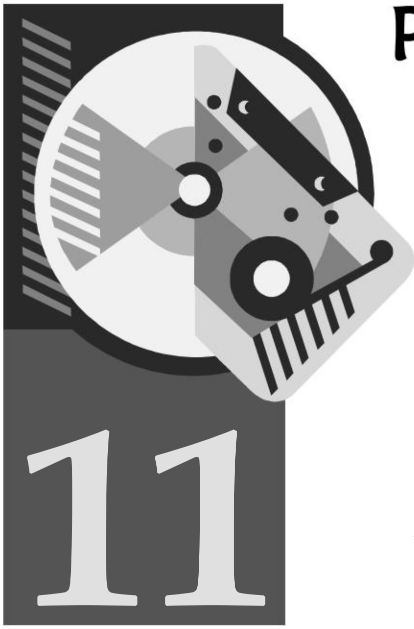
1. The word 'stereophony' is derived from two Greek words: '*stereos*' and '*phone*' meaning solid and sound, respectively. Thus, stereophony means 'solid sound' or 'three-dimensional sound'.
2. There is a minute difference of phase and intensity in the sounds reaching the two ears. This difference is interpreted by the brain in such a way so as to enable the listener judge the directions from which the sounds are coming.
3. When the human hearing system is simulated by a physical system of microphones and headphones, it becomes an ideal stereophonic system. In this system, a stand (like a human head) is used with two microphones in place of ears. The listener uses two headphones, one on the left ear and the other on the right ear. The headphones are connected to the microphones by wires. This way, the ears receive the sound exactly in the same fashion as they receive in the auditorium. Thus, this arrangement works as an ideal stereo system.
4. The ideal system using headphones is not at all practical. It can at the most be arranged for one person only. When listeners are more than one, cabling would prohibit the use of headphones.
5. The quadraphonic systems, using four channels, approached better to the ideal stereophonic system. But it was not compatible to the two channel stereophonic system which was being used widely. Hence, it did not become popular.
6. The stereo recording on disc is done by the 45-degree method. In this system, each channel is recorded along a direction of 45 degrees to the horizontal line of the disc surface. The two directions, one for the left channel and the other for the right channel are at 90 degrees to each other. Being at right angles, the motions of two channels do not interfere with each other.
7. Two amplifiers of a stereo system, although independent of each other, are built as a matched pair to give equal output for the same input. Despite two amplifiers being

identical, there may be variations in the output of each channel due to variations in the characteristics of transistors. The balance control compensates such variations and gives equal outputs in the two channels for equal inputs.

8. The open-reel tape system needs loading and threading. It is also susceptible to breaking. The alternative is a cassette tape, which is ready to play at any time.
9. A cartridge runs continuously. There is no provision of reverse winding. All the tracks are in one direction only. Unlike a cartridge, in a cassette, provision of reverse recording and playback is available.
10. Instead of an electromagnetic 'head' in tape recording, a laser beam of red light is used in DVD. Principles of stereophony remain the same in DVD as in tape. Two tracks of pits and lands are recorded on the digital video disc to take care of left and right channels.

Multiple-Choice Questions

1. (d)	2. (a)	3. (c)
4. (b)	5. (b)	6. (a)
7. (d)	8. (d)	9. (b)
10.(a)		



11 Public Address System

11.1 NEED AND USE

The intensity of sound decreases with distance. Hence when a large gathering is to be addressed, sound needs to be amplified so that people at a distance from the rostrum or stage may receive good intensity of sound for comfortable listening. The system which fulfils this function is called *public address system* (or simply *PA* system.) It is used in sports meets, public meetings, auditoriums, concerts and functions. It is also used to convey information to isolated locations as at railway stations, airports, hospitals, factories, etc.

11.2 BLOCK DIAGRAM

It is an electroacoustic system in which sound is first converted into electrical signals by a microphone. The electrical audio signals are amplified, processed and fed to another transducer, the loudspeaker, which converts the audio signals into sound waves. A block diagram of a basic PA system is shown in Fig. 11.1. Description function of each block of PA system follows:

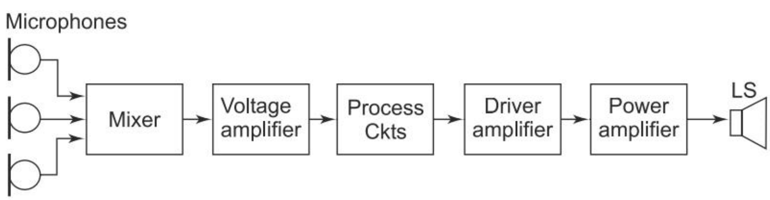


Fig. 11.1 *Block diagram of a basic PA system*

Microphone It picks up sound waves and converts them into electrical variations, called audio signals. Generally, amplifiers have provision of two or more microphones and in addition, an auxillary input for tape/record player.

Mixer The output of microphones is fed to a mixer stage. The function of the mixer stage is to effectively isolate different channels from each other before feeding to the main amplifier. It can be either a built-in unit or a separate plugable unit.

There are three types of mixers. The simplest one does not use pre-amplifiers and amplifiers, but uses only gain controls (also called faders) and isolating series resistors. A more sophisticated one uses common pre-amplifier after isolating resistors. It is shown in Fig. 11.2. P_1 to P_5 are gain controls and R_1 to R_5 are isolating resistors. The most sophisticated one has separate pre-amplifiers, for separate channels and then after the gain control potentiometers and isolated resistors, there is a common amplifier and an emitter follower. Low impedance of emitter follower matches the input impedance of the voltage amplifier of the PA system. The function of the separate pre-amplifiers and common amplifier in the mixer stage is to amplify the weak signals.

Voltage Amplifier It further amplifies the output of the mixer.

Processing Circuits These circuits have 'master gain control' and tone-controls (bass/treble controls).

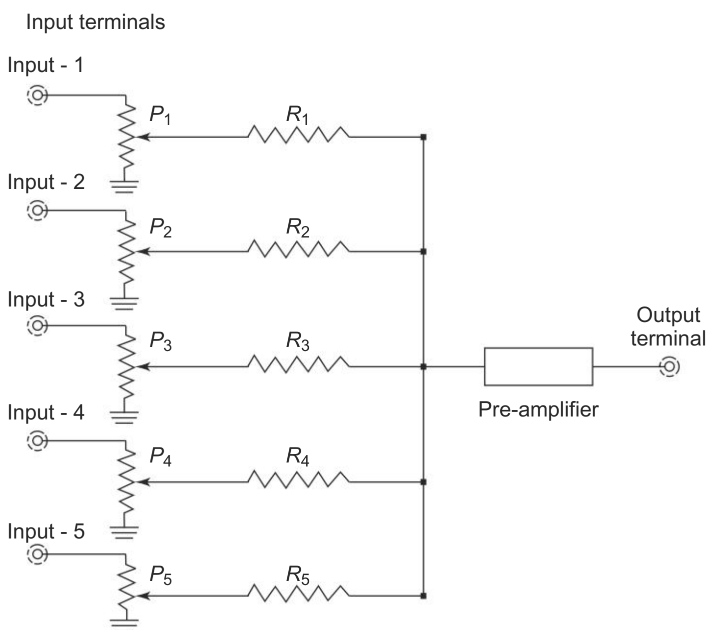


Fig. 11.2 *Mixer circuit for PA system*

Driver Amplifier It gives voltage amplification to the signal to such an extent that when fed to the next stage (power amplifier stage), the internal resistance of that stage is reduced. Thus, it drives the power amplifier to give more power.

Power Amplifier It gives desired power amplification to the signal. It uses push-pull type of circuit in general, so that the even harmonics are eliminated from the output, and the transformer core does not get saturated. The output of the power amplifier is connected to the loudspeaker through a matching transformer to match the low impedance of the loudspeaker for maximum transfer of power. In some circuits, the design is such that a separate matching transformer is not needed.

Loudspeaker It converts electrical audio signals into pressure variations resulting in sound.

11.3 REQUIREMENTS OF A PUBLIC ADDRESS SYSTEM

The following precautions should be taken while installing a PA system.

Acoustic Feedback Sound from the loudspeaker should not reach the microphone, as the acoustic feedback will result in a loud howling sound.

Distribution of Sound Intensity Loudness of sound is contained in low notes, and the intelligibility in high notes. As high notes suffer greater attenuation with distance than the low notes, intelligibility suffers at farther distances. The PA system should take cognizance of this fact, and hence sound should be uniformly distributed amongst the audience. This means that instead of one or two powerful loudspeakers near the stage alone, audio power should be divided between several loudspeakers to spread it right up to the farthest point so that each one covers a specified area. It should be ensured that the number and wattage of loudspeakers is sufficient to handle the maximum power of the amplifier.

Reverberation In a reverberating medium, the intelligibility is poor. This is due to overlapping of successive sound waves. The PA system should throw additional power in those areas where the direct sound gets submerged in the echoes. The problem of reverberating halls can be solved by locating several small power loudspeakers at various points of the auditorium rather than using a single high-power unit.

Orientation of Loudspeakers To make the best use of the available power of the PA system, loudspeakers should be so oriented as to direct the sound towards the audience and not towards the walls. Loudspeakers should preferably be placed a metre off the floor, so that their axes are about the height of the ears of the seated listeners. Also, distinct echoes due to reflections from distant buildings should be eliminated.

Ambient Noise When ambient noise is high as in a sports event, or a market place, the PA system should boost the high frequencies (treble boost) to restore the intelligibility. It is the high frequency part of the noise spectrum which affects intelligibility. Also, in a high noise environment, amplification is kept at high levels. At high-amplification level, low notes are heard more strongly than high notes which make the sound appear unnatural. Hence the PA system should attenuate the lower notes to keep natural timbre of the sound in a high-noise environment. Noise-cancelling microphones are also helpful.

Dynamic Range Limitation The amplifier of a good PA system is equipped with a level limiter which keeps the output level constant when the input level exceeds a certain predetermined value. The provision takes care of the speaker's drawbacks (mouth too near the microphone, exaggerated speech level, i.e., shouting, etc.).

Selection of Microphones Microphones for a PA system should preferably be of cardioid type, so that they neither pick up reflected sound nor the sound from loudspeakers. For dramas where speakers may have to speak from a distance from the microphone, sharply directive microphone, for example, a vertical column of unidirectional microphones should be used to pick up more sound power.

Sense of Direction of the Source of Sound Loudspeakers should be so placed that sound appears to be coming from the direction of the source. Human ears perceive the direction from the first sound received. Hence, small speakers may be so placed as to give the correct direction to the listeners, and for the volume, large speakers can be used at farther distances. Alternatively, a delay of 10–20 milliseconds can be caused artificially in the signal reaching the loudspeakers in digital systems, and hence loudspeakers will be heard later than the sound direct from the speaker. This will give the desired loudness and the correct direction irrespective of position of the loudspeakers.

Phase Delay Sound from the nearest loudspeaker may be heard along with the sound from other loudspeakers with time difference. A delayed sound impairs the intelligibility when delay is 45 ms or more. This delay corresponds to about 16 metres. Hence, loudspeakers should not be located beyond 16 metres apart. A 10-metre separation is considered quite good.

Matching of the total loudspeaker impedance with the output impedance of the amplifier is necessary for maximum transfer of energy from amplifier to loudspeaker. Hence, series–parallel combination of loudspeakers should be such as to ensure maximum power transfer.

When loudspeaker leads have to be long as in large public meetings, a 100-V system should be used. In this system, signal from the amplifier will be fed to the cables at 100 V, and the matching transformers will be at the loudspeakers.

It may be pointed out here that very expensive cables for loudspeakers have no advantage over the conventional flexible cables of adequate current rating. The only consideration here is that the impedance of cables should be much less than the impedance of the loudspeaker. This criteria is fully satisfied by using cheap flexible cables, as the resistance of such cables is only 0.02 Ω per metre.

Grounding Chassis and shields of equipment and coaxial cables should be properly earthed through water pipes (not through earth of mains plug).

Example 11.1 *Two loudspeakers of 8 Ω each and 4 loudspeakers of 4 Ω each are to be connected to an amplifier of 8 Ω output. How will you connect?*

Solution Loudspeakers should be so connected that the resistance of loudspeakers becomes 8 Ω. Hence, we can connect 8 Ω speakers in series giving 16 Ω, and 4 Ω speakers in series giving 16 Ω. Then the two combinations of 16 Ω each should be connected in parallel giving a resistance of 8 Ω, as shown in Fig. 11.3.

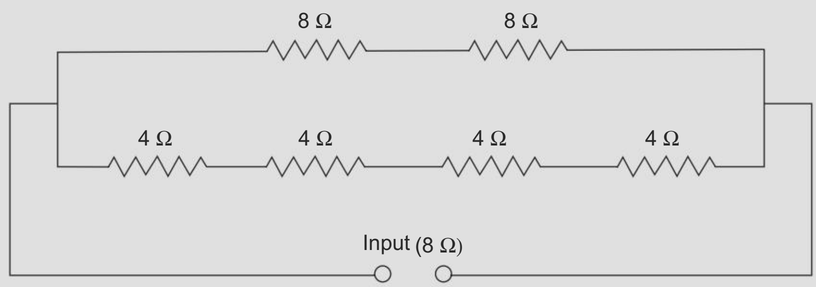


Fig. 11.3 *Connection of loudspeakers for Example 11.1*

Example 11.2 *What will be the distribution of power, if the amplifier gives 8 W output in the above example?*

Solution Power divides equally in two branches.

Therefore, power in 8 Ω loud speakers = 2 W each.

Power in 4 Ω loudspeakers = 1 W each.

Power in the upper arm will be 4 W and also 4 W in the lower arm. So, power will be distributed uniformly.

Amplifier Power A PA system gives out amplified sound, so that it is comfortably audible to the audience at a distance. The output power of the amplifier may be a few watts for class lectures or a small gathering to a few hundred watts for large public meetings or a sports meet. The total output power of the amplifier required should be calculated on the basis that sound intensity equal to 80 dB over the threshold of hearing should be available to the audience at the farthest point. As the threshold of hearing is 10^{-12} W/m^2, the absolute intensity required for audience is $10^{-12} \times 10^8 = 10^{-4}$ W/m^2. To obtain this much intensity at R metres distance, we should have an amplifier of P watts output given by Eq. 11.1, using loudspeakers of 'E' per cent efficiency.

$$10^{-4} = \frac{P}{4\pi R^2} \times \frac{E}{100}$$

or,
$$P = \frac{4\pi R^2 \times 10^{-2}}{E} \tag{11.1}$$

This power of the amplifier can be distributed uniformly by placing several loudspeakers.

The above analysis does not take into account the absorption by the audience in open air meetings, or absorption by furnishings in an auditorium. The factor of absorption should also be considered by estimating the same from the absorption coefficients of different absorbing materials.

Example 11.3 *If the distance R is 100 metres, efficiency of the horn-type loudspeaker is 25%, calculate minimum power of the amplifier.*

Solution
$$P = \frac{4\pi \times 100 \times 100 \times 10^{-2}}{25}$$
$$= 16\pi = 50 \text{ W}$$

Example 11.4 *If normal conversation level is 25 μW and we need at least 3 dB more for comfortable listening in a meeting, calculate the intensity of sound required in dB over threshold of hearing, presuming 3 dB of power is absorbed by the audience.*

Solution Threshold of hearing is 1 pW.

$$\text{dB for } 25\,\mu\text{W} = 10 \log \frac{25 \times 10^{-6}}{1 \times 10^{-12}}$$
$$= 10 \log \frac{100}{4} \times 10^6$$
$$= 10\,(2 - 0.6 + 6)$$
$$= 20 - 6 + 60 = 74 \text{ dB}$$

Intensity for conversation level = 74 dB

Lowest intensity for comfortable listening = 74 + 3

Absorption by audience = 3 dB

Hence, intensity required = 80 dB

Choice of Loudspeakers The loudspeaker chosen should be able to withstand the output power of the amplifier. Suppose we have a 100 W amplifier, and the loudspeakers are 25 W each, then we must have at least 4 loudspeakers or preferably more (to take care of transients of power).

When high fidelity is required, speaker columns containing woofers, squawkers and tweeters may be used.

For high efficiency, horn-type speakers are more suitable for PA system than cone type.

Closed-ring Connection of Loudspeakers For better reliability, loudspeaker leads should form a closed ring as shown in Fig. 11.4. If the lead is broken at any point, it will not make any loudspeaker inoperative. If the lead is broken, say at *B*, the speakers *P* and *Q* shall receive audio power from the amplifier through left hand side leads, and *R*, *S*, *T* and *U*, speakers from right hand side leads.

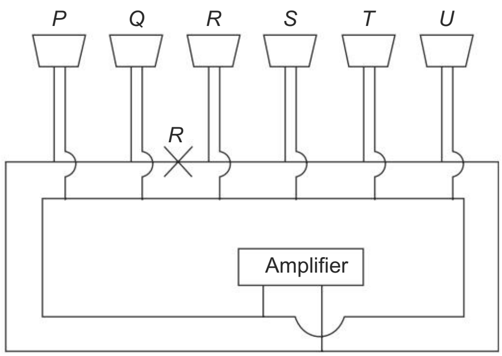


Fig. 11.4 *Closed ring connection of loudspeakers*

AC Hum Microphone leads can pick up hum from an ac mains wire. Hence, microphone leads should be as short as possible. If long leads are unavoidable, precaution should be taken not to run the microphone leads parallel to the ac leads. Impedances of microphone, leads, and amplifier input should all be matched to each other to get the best results.

Placement of Microphones Placement of microphones should be made in such a way that they give total coverage for all sources of programme sound, and at the same time not respond to unwanted sound. Also, if path difference between microphones is $\lambda/2$ for sound from a particular source, it will cause cancellation of signal in the amplifier at least at one particular frequency and will affect fidelity. Hence, if more than one microphone are to be used, these should be judiciously placed, generally several wavelengths apart for high frequencies (wavelength for 10 kHz is 3.3 cm).

RF Pick-up Due to poor grounding, cold or dry solder joints or defective RF by-pass capacitors, local radio broadcast stations and RF transmissions are picked up and detected by the amplifier. The quick remedy for this problem is to connect an RF by-pass capacitor at input terminals of the amplifier.

Presence Not to be Felt An ideal PA system is one in which everyone among the audience can hear the programme comfortably without becoming aware that the amplifiers are in use. This is not always possible but the PA equipment should be as unobtrusive as possible without sacrificing intelligibility. This aspect should be kept in mind when permanent installation is being done at the time of construction of a multipurpose auditorium.

11.4 TYPICAL PA INSTALLATION PLANNING

Installation of a PA system needs careful thought and logical planning. A poorly installed PA system is quite irritating and spoils the otherwise good programme. The planning has to take into account the situation where the PA system is to be installed. Planning for the following typical situations is discussed below:

1. Public meeting (large gathering)
2. Auditorium having the seating capacity of about 1000 persons
3. Debating chamber
4. Football stadium
5. College sports

11.5 PA SYSTEM FOR A PUBLIC MEETING (LARGE GATHERING)

A typical PA installation plan for a public meeting in open area, say 120 m × 120 m is shown in Fig. 11.5.

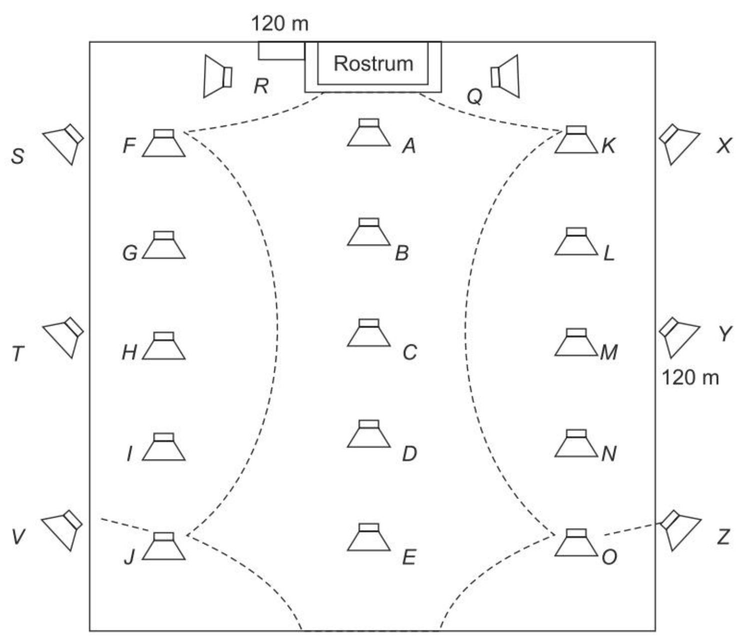


Fig. 11.5 *PA installation plan for public meeting*

Its salient features are the following

1. The loudspeakers *A*, *B*, *C*, *D* and *E* in the centre line will give the sense of direction to most of the audience and can be mounted on poles.
2. Loudspeakers, *F*, *G*, *H* and *I* on one side and *K*, *L*, *M* and *N* on the other side will give full coverage to meeting ground on both sides of the central area.
3. To cover the remote semicircular side and corner areas, loudspeakers *J*, and *O* are used. These will throw sound power towards corners.
4. The loudspeakers *Q* and *R* will cover the left and right sides, respectively near the rostrum.
5. There may be some loudspeakers (*S*, *T*, *V*, *X*, *Y*, *Z*) to give coverage to audience standing outside the meeting park. These may be slightly inclined, as shown in the figure.
6. Microphones should be of cardioid type and the loudspeakers may be of horn type.
7. The output audio power of the amplifier may be calculated by using the formula given in Eq.11.1.
8. It is preferable to use HOT standby amplifiers with batteries.

11.6 PA SYSTEM FOR AN AUDITORIUM HAVING LARGE CAPACITY

An auditorium may be used for wide range of activities like public meetings, conferences, cultural programmes, etc. Hence, the loudspeaker system should have a wide dynamic range (40 dB to 120 dB) and good frequency coverage (from 20 Hz to 16 kHz). Columns of loudspeakers (having good bass and treble response) should be mounted facing the front on either side of the stage. If the hall is wide, a small column may also be mounted in the centre of the front line. Another pair of small columns, slightly inclined, may be placed at about one third, and then two third way down the hall from the front. The 4^{th} pair, placed last, need not be inclined.

A separate versatile mixer unit is desirable. It may have tape and disc inputs and several microphone inputs. The amplifier should be of 50 to 100 W. There should be a standby amplifier also. Microphones should be of moving-coil type so that they may withstand likely misuse in handling and may withstand Rock 'n' Roll music levels. Ribbon types may be used for drama. A typical plan for an auditorium is shown in Fig. 11.6. If the hall has a line of pillars (not shown in the figure), small loudspeakers may be mounted on the pillars facing the audience (i.e., back wall). The back wall should be furnished with a good sound-absorbing material.

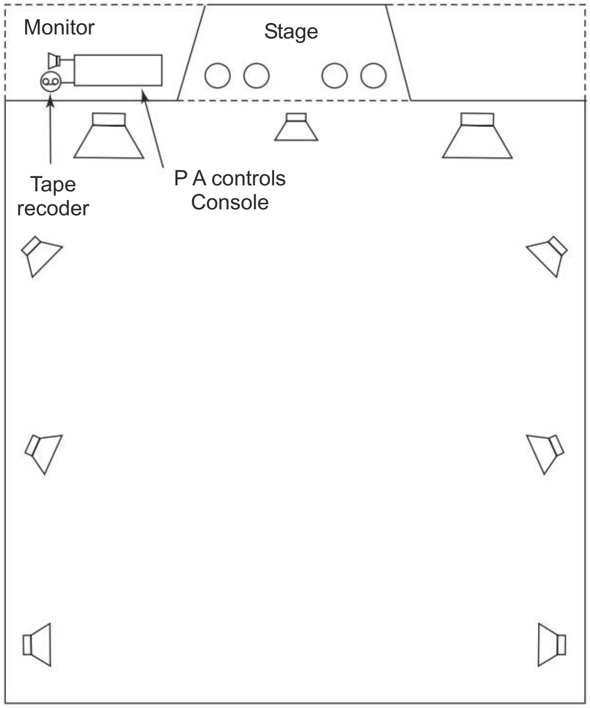


Fig. 11.6 *PA plan for an auditorium*

The height of the side speakers should be such that their axes point towards the ears of the audience, and not towards the floor or ceiling.

11.7 PA SYSTEM FOR A DEBATING HALL

Here, microphones are to be fitted for each speaker on his/her desk. The audience may be present in the public gallery and they should be able to hear the debate. Loudspeakers shall mainly be concentrated on the audience to avoid acoustic feedback. However, some sound reinforcement is desirable in the debating chamber and this has to be done without causing feedback. Auto-voice control, which automatically cuts off the pre-amplifier of the concerned microphone in the absence of voice, is essentially required to keep the channels dead until a microphone is actually used. In some low-cast systems, a switch is used to make the microphone on when the speaker wants to use it.

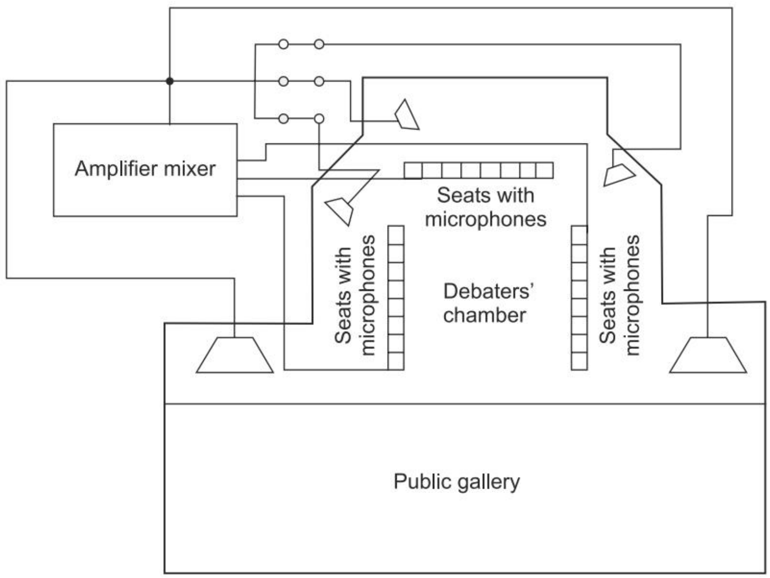


Fig. 11.7 *PA system plan for a debating hall*

A typical system is shown in Fig.11.7 for 24 persons participating in the debate and about 150 persons listening to the deliberations.

11.8 PA SYSTEM FOR A FOOTBALL STADIUM

A very exhaustive plan is to be made so as to give coverage to all sides for spectators, players, preparation tents, canteen, etc. It should also be kept in mind that ambient noise will be quite high.

The 'spectators' stands can be covered by a row of columns in front of each group. Columns may be tapped at 10 W. Alternate columns should be connected to separate amplifiers to save the sound system if one amplifier fails. There should be several amplifiers of 60 to 100 W each. Microphones should be of cardioid type. A sketch of the plan is shown in Fig. 11.8.

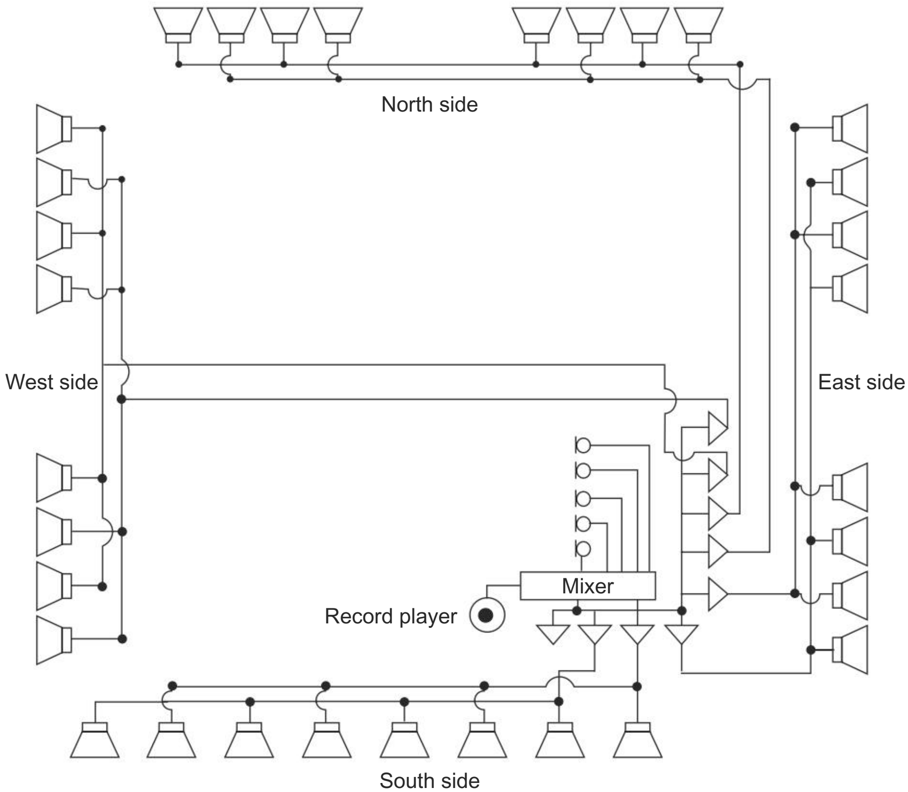


Fig. 11.8 *PA system plan for a football stadium*

11.9 PA SYSTEM FOR COLLEGE SPORTS

The annual event of college sports requires coverage to the teachers and guests on the entrance side, to the spectator students on the opposite side of the field and to the participants in the field. Sometimes a marquee is erected nearby for serving tea. This requires coverage so that people working there may not be deprived of the entertainment and thrill which the sports event generates.

The field and students enclosure can be covered by four columns, two on each side. Enclosure for teachers and guests can be covered by two columns. One column is adequate for the marquee. The plan is shown in Fig. 11.9. The columns should be mounted vertically without any tilt.

A 60 W amplifier will be adequate to give 10 W to each column for field and students enclosure and 5 W to each column for teachers and guests and one 5 W column in the marquee.

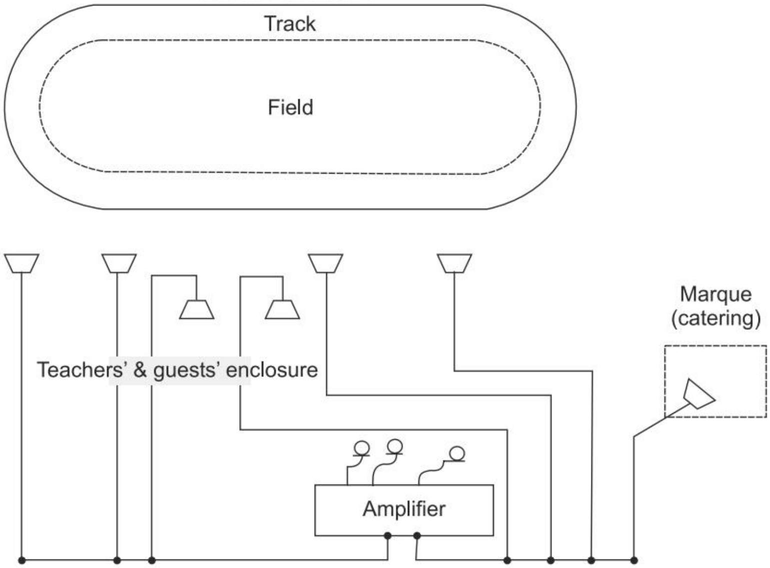


Fig. 11.9 *PA system plan for college sports*

SUMMARY

- Public Address (or PA) system is an electroacoustic system, in which sound is first converted into electrical signals (audio) by microphones, and then audio signals are processed and amplified and finally are fed to loudspeakers which convert them back to sound. The amplified sound provides comfortable listening to large gatherings as in public meetings, sports, concerts, functions or to isolated locations as at railway stations, airports, etc.
- Some PA amplifiers have mixer stages to allow several microphones and other devices to be connected to a single PA amplifier.
- For minimum distortion and uniform and sufficient intensity of sound all around the concerned place, the following precautions need to be taken while installing a PA system.
  - Acoustic feedback should not occur.
  - The amplifier's audio output power should be sufficient and should be uniformly divided by using several loudspeakers. Loudspeakers should be able to withstand the power being fed to them.

- Excessive reverberation should be reduced by appropriate furnishing and proper placement of loudspeakers.
- Loudspeakers should be properly oriented to direct the sound to the audience.
- Noise-cancelling microphones and treble boost in amplifiers should be used where ambient noise is high as in sports.
- The amplifier should have automatic level limiter to reduce distortion due to overloading.
- Microphones should be carefully selected to suit the nature of the programme.
- Matching of impedance between microphone output and amplifier input and between amplifier output and loudspeakers is essential.
- Microphone leads should be isolated from ac leads to eliminate chance of picking up ac hum.
- Radio frequency pick-up should not occur.

☞ PA installation plans for public meeting, auditorium, debating hall, football stadium and college sports have been illustrated in Figs 11.5 through 11.9.

Review Questions

1. Explain the working of the public address system with the help of a neat block diagram.
2. What are the essential requirements for good PA installation? How can these be met in practice?
3. Explain the importance of the following terms for a PA system.
   1. Acoustic feedback
   2. Reverberation
   3. Ambient noise
   4. Level limiter
   5. Noise-cancelling microphone
   6. Phase delay
   7. Matching
   8. AC hum
   9. RF pick-up
   10. Amplifier power
4. Write a short note on the mixer stage in a PA system.
5. Give the typical installation plan sketch for
   (a) a public meeting,
   (b) an auditorium
6. Describe the PA installation plan for a debating chamber.

7. Describe the PA installation plan for a football stadium
8. Describe the PA installation plan for annual sports meet of your college.

Short-Answer Questions

1. What is a public address system?
2. What are the various controls in a PA system?
3. What is acoustic feedback?
4. How should loudspeakers be oriented to optimize the output?
5. Why are cardioid-type microphones preferred for large public meetings?
6. Why are horn-type speakers more suitable than cone types in public meetings?
7. What is the advantage of closed-ring connection of loudspeakers?
8. If power supply to a PA system is ripple-free, what could be the cause of hum generation in the PA system?
9. What is RF pick-up?
10. Why is PA system made unobtrusive?

Multiple-Choice Questions

1. What is the full form of PA system?

   (a) Power Amplifier
   (b) Public Address
   (c) Personal Amplifier
   (d) Public Account

2. What is the remedy in an amplifier for those speakers who keep their mouth very near the microphone?

   (a) Noise canceller
   (b) Level canceller
   (c) Level limiter
   (d) Level amplifier

3. The most popular microphone for use in PA system is

   (a) carbon microphone
   (b) capacitor microphone
   (c) ribbon microphone
   (d) moving coil microphone

4. Output impedance of the power amplifier is matched with that of the loudspeaker by the

   (a) leads (b) transformer
   (c) resistor (d) inductor

5. What type of noise can be picked up by a microphone when its leads run parallel and close to the mains leads?

   (a) RF noise (b) Hum noise
   (c) Hiss noise (d) Rumble

6. The intensity required by audience for comfortable hearing is

   (a) 1 pW/m^2 (b) 100 nW/m^2
   (c) 100 mW/m^2 (d) 10 W/m^2

7. Calculate the intensity of sound for 40 dB above 1 pW, the threshold of hearing
   (a) 40 pW (b) 10 nW
   (c) 2 mW (d) 1 W
8. What is the optimum separation of loudspeakers to give good intelligibility?
   (a) 10 m (b) 20 m
   (c) 30 m (d) 40 m
9. In a monophonic amplifier system, which of the following controls is not needed?
   (a) Bass
   (b) Treble
   (c) Master gain
   (d) Balance
10. What is the function of mixers in a PA system?
   (a) To isolate different channels
   (b) To produce intermodulation frequencies
   (c) To produce intermediate frequency (IF)
   (d) To prevent RF noise pick up

Numerical Problems

1. Calculate intensity of sound in microwatts for 60 dB above 1 pW, threshold of hearing, and also 40 dB below 10 W, threshold of pain.
2. An amplifier gives 100 watts audio power, and this power is to be distributed in 5 loudspeakers of 10 watts each and 10 loudspeakers of 5 watts each. How will you connect 15 loudspeakers?
3. The output impedance of an amplifier is 2 Ω. Connect 4 loudspeakers of 4 Ω each and 2 loudspeakers of 8 Ω each such that the net impedance of the loudspeakers is matched with that of the amplifier.
4. Find out the intensity of sound in μW at a distance of 50 metres from a loudspeaker whose efficiency is 20% and the PA system feeds 10 watts of audio power to the loudspeaker.
5. Calculate the turns ratio of matching output transformer required to match the loudspeaker impedance of 16 Ω with the amplifier impedance of 1600 Ω.

Answers

Short-Answer Questions

1. When a large gathering is to be addressed, sound needs to be amplified and transmitted through loudspeakers so that the people at distance from the rostrum may receive sound of adequate intensity for comfortable listening. This system consisting of microphones, an amplifier and loudspeakers is called public address system.

2. A basic PA system has the following controls:
   1. Microphone's gain control (or level control)
   2. Master gain control
   3. Bass control
   4. Treble control
3. When sound from a loudspeaker reaches the microphone, it causes a loud howling sound, called *acoustic feedback*.
4. Loudspeakers should be so oriented as to direct the sound towards the audience and not towards the walls. Also, any distant structure should not reflect the sound to cause echoes.
5. Cardioid-type microphones have heart-shaped directivity pattern, sending maximum sound in the front. Sound at +90 degree from the axis is reduced and there is a null towards the back. This pattern is suitable for large gatherings and also reduces chance of acoustic feedback.
6. Efficiency of horn-type speakers is about 40% while that of the cone types, only about 5%.
7. When loudspeaker leads form a closed ring, reliability of the system becomes better. In such a system, if a lead is broken at any point, it will not make any speaker imoperative.
8. When mains leads run parallel and close to the microphone leads, ac mains frequency signal can be picked up, which would be amplified by the amplifier and produced as hum by the loudspeakers.
9. Due to poor grounding or cold and dry solder joints or defective RF bypass capacitors, programmes of local radio broadcast stations are picked up and these can be detected by the nonlinear characteristics of the amplifier.
10. An audience wants to hear the programme comfortably without becoming aware that the amplifiers are in use. The audience is not interested in seeing the amplifiers, but are interested only in enjoying the programme. Hence, a PA system is made as unobtrusive as possible.

Multiple-Choice Questions

1. (b) 2. (c) 3. (d)
4. (b) 5. (b) 6. (c)
7. (b) 8. (a) 9. (d)
10. (a)

Numerical Questions

1. (1 μW, 1 mW)
2. Connect five loudspeakers of 10 W each in series and 10 speakers of 5 W each in another series. Then the two series should be connected in parallel.
3. Connect 4 loudspeakers of 4 Ω each in two series, each of 2 loudspeakerss, and connect these two series in parallel, resulting in 4 Ω. Then connect 2 loudspeakers of 8 Ω each in parallel with the earlier combination. The result would be 2 Ω.
4. (64 μW/m^2)
5. (10: 1)

12 Acoustic Reverberation

12.1 REVERBERATION

Sound, being a wave motion, suffers reflection, refraction, diffraction and absorption in accordance with the specified rules as for any wave motion. These effects change the original sound substantially and should be given a careful consideration while designing auditoriums and studios and even living rooms.

A person receives sound directly from the source as well as sound reflected from the walls, ceiling, floor, etc. The reflected sound is heard as a distinct echo if the time gap between the original (direct) wave and the reflected wave is more than 60 ms (corresponding to path difference of 21 metre). Reflections over shorter distances shall simply prolong the sound due to multiple reflections in halls, shown in Fig. 12.1, in which Loudspeaker (LS) is the source of the sound, and *A*, the listener who receives the direct sound as well as the reflected sound. The sound persists even after the source of sound has stopped sounding. It fades away only gradually. The gradual fading of the continuing echo is called 'reverberation'.

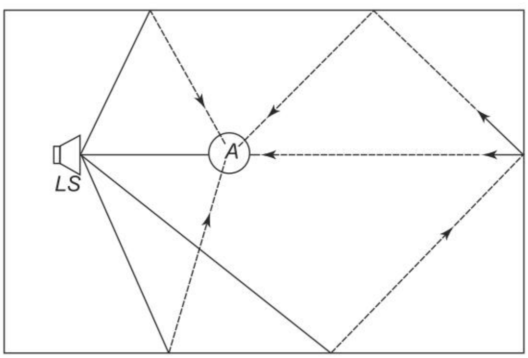


Fig. 12.1 *Multiple reflections in a hall*

Reverberation time is defined as the time taken for sound energy in a room to drop to 10^{-6} times (i.e., one millionth) of its initial value, or 60 dB below its initial value.

12.2 NECESSITY OF REVERBERATION

If all the echoes are eliminated by fixing sound-absorbent materials on the ceiling, floor and all the four walls, the result would be a sound which could only be described as *Lifeless* and unnatural. All natural sound in a hall includes a proportion of continuing echoes. Variations in this proportion give sound a quality of liveliness or richness. (Reverberation is different from the distinct echoes. In case of distinct echoes, there is a gap between the finish of the original sound and arrival of the echo. But reverberation has a continuity of sound. The distinct echoes are irritant and undesirable and hence, must be eliminated, while reverberation to some extent is pleasing and should be incorporated in the design of rooms.)

12.3 TYPICAL REVERBERATION PERIODS

Reverberation is very much desirable in music concerts to make the programme appear natural and lively. But in lecture halls, speech-recording studios, factories and workshops, reverberation is kept rather low. The typical reverberation periods for wavelength of about 70 cm (frequency of about 500 Hz) are given below:

Church (medium size)	: 2.5 seconds
Big concert hall	: 2.0 seconds
Medium concert hall or small orchestra	: 1.5 seconds
Dance hall	: 1.0 second
Pop music programme	: 0.8 second
Conference room	: 0.5 second
TV studio	: 0.4 second
Lecture halls and speech recording studios	: 0.3 second

A small reverberation time tends to minimise the ambient noise level produced by the outside sounds which do penetrate to some extent into the room.

12.4 FACTORS ON WHICH REVERBERATION TIME DEPENDS

Reverberation time depends on: (1) volume of the room, (2) surface area, (3) absorption coefficient of the surface area, and (4) velocity of sound (and hence wavelength).

WC Sabine of Harvard University studied the phenomenon of reverberation in detail with a view to improve the acoustics of a defective lecture hall of the university. He established Eq. 12.1 for reverberation time, T, in seconds:

$$T = 55.3\frac{V}{ca} \tag{12.1}$$

where, c = the velocity of sound = 344 m/s or 1120 ft/s

V = Volume of room

a = Total absorption

In FPS units, the formula becomes

$$T = \frac{0.049\,V}{a} \tag{12.2}$$

In MKS units, the formula would be

$$T = \frac{0.161\,V}{a} \tag{12.3}$$

'a' in Eqs. 12.1, 12.2 and 12.3 depends on the surface area of each surface and its absorption coefficient, as defined in Eq. 12.4.

$$\begin{aligned} a &= \Sigma \alpha S \\ &= \alpha_1 S_1 + \alpha_2 S_2 + \cdots \end{aligned} \tag{12.4}$$

where, α_1 = absorption coefficient of surface area S_1

α_2 = absorption coefficient of surface area S_2

and so on.

The unit of $\alpha\, S$ is sabine when S is in square metres.

12.5 ABSORPTION COEFFICIENTS

Absorption coefficient, α, is given by Eq. 12.5 as

$$\frac{\text{energy absorbed by unit surface area}}{\text{total energy received by unit surface area}} \tag{12.5}$$

Being a ratio between similar quantities, it has no unit.

Typical values of the absorption coefficients of some materials are given in Table 12.1.

The audience also absorbs sound energy. On an average, each person's absorption (surface area × 0.84) is 4.5 in FPS units and 0.45 in MKS units. Hence, total absorption for the audience can be calculated. These values are for winter clothings. For summer, absorption for each person may be taken to be 4 in FPS units and 0.4 in MKS units. When absorption for the audience is taken into account, absorption for chairs occupied by them should not be considered. Total absorption for a seated person may be taken to be equal to 0.5 in MKS units (sabine).

The absorption coefficients indicated in Table 12.1 are not specific, and actually depend on the method of mounting and also on the frequency of the incident sound, being somewhat higher at higher frequencies. The values given in Table 12.1 are for sound of 500-Hz frequency.

Table 12.1 *Absorption coefficients of materials*

	Name of the absorbent material	Absorption coefficient
1.	Open window	1
2.	Hair felt and woolen Galeecha	0.58
3.	Carpets (1 cm thick)	0.25
4.	Curtain	0.15
5.	Thick drapes	0.57
6.	Wooden chair	0.17
7.	Upholstered chair	0.30
8.	Cushion	0.20
9.	Celotex	0.36
10.	Unpainted brick or plastered wall	0.03
11.	Painted brick wall	0.02
12.	Special acoustic plaster	0.25
13.	Acoustic tiles	0.55
14.	Door-wood	0.05
15.	Wooden floor	0.09
16.	Glass panes	0.25
17.	Audience	0.84

12.6 GROWTH AND DECAY OF SOUND IN AN ENCLOSURE

When a source of sound starts producing sound in a room, reflections from walls, ceilings and the floor make sound energy at a point in the enclosure grow and become more and more uniform with increasing time until the sound source stops sounding. After that, the energy starts decaying and in a short time becomes inaudible. It is somewhat similar to the growth and decay of electric current in an inductor.

Figure 12.2 shows growth and decay of sound at a point in an enclosure. *X*-axis indicates time. At time '0' the sound at the source starts. At *a*, the direct sound from the source reaches the listener and there is no reflected sound till *b*. After that, the reflected sound begins to reinforce the direct sound, and hence, the energy density increases up to the time c. The sound from the source stops arriving and then, the reflected sound from nearby surfaces decays to the same extent as the reflected sound from the distant surfaces increases, so that the net reflected sound becomes more or less constant. The constant energy is between c and d times. Then this reflected sound decays rapidly.

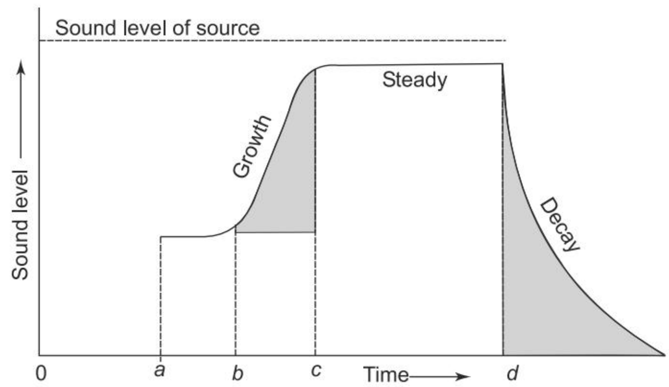


Fig. 12.2 *Growth and decay of sound in an enclosure*

Mathematical analysis shows that for a plain room, growth of sound intensity is given by Eq. 12.6

$$I = I_m (1 - e^{-act/4V}) \tag{12.6}$$

where, I = Intensity at any instant

I_m = Maximum intensity

a = Total absorption

$= \alpha_1 S_1 + \alpha_2 S_2 + - - -$

c = Velocity of sound

t = Time in seconds

V = Volume of room

α = absorption coefficient of surface S

Time constant of the process is 4 V/ac. If a is small, longer time is needed for the intensity to reach maximum.

Decay of sound is given by Eq. 12.7

$$I = I_0 \, e^{-act/4V} \tag{12.7}$$

where, I_0 = maximum intensity of direct sound.

Sabine's Equation The reverberation time can be derived directly from Eq. 12.7:

$$\frac{I}{I_0} = 10^{-6} = e^{-act/4V} \quad \text{or,} \quad \frac{I_0}{I} = 10^{6} = e^{act/4V}$$

$$t = \frac{4V}{ac} \log_e 10^6 = 55.3 \text{ V/ac} \tag{12.8}$$

In FPS units, c = 1120 feet per second

therefore, t = 0.049 V/a

$$t = \frac{0.049\,V}{\alpha_1 S_1 + \alpha_2 S_2 + \cdots} \tag{12.9}$$

In MKS units, c = 344 metres per second,

therefore,
$$t = \frac{0.161\,V}{\alpha_1 S_1 + \alpha_2 S_2 + \cdots} \tag{12.10}$$

The reverberation time can be controlled precisely by addition or reduction of the absorbents and hence, changing the total absorption in the enclosure.

Eyring modified the Equations of reverberation time and gave Eq. 12.11 in FPS units and Eq.12.12 in MKS units for nearly dead rooms.

In FPS units,

$$t = \frac{0.049 \text{ V}}{5 \log_e (1-\alpha)^{-1}} \tag{12.11}$$

In MKS units,

$$t = \frac{0.161 \text{ V}}{5 \log_e (1-\alpha)^{-1}} \tag{12.12}$$

where, α is the average absorption coeffieient

Example 12.1 *There is a hall of 20 m × 15 m × 5 m having the following surfaces:*

Open windows	*2 m × 1 m*	*8*
Thin carpet	*15 m × 10 m*	*1*
Woolen galeecha	*10 m × 5 m*	*1*
Curtains	*2.5 m × 1 m*	*8*
Audience	*410*	

Calculate reverberation time, using absorption coefficients given in Table 12.1.

Solution

Absorption by windows	$= 2 \times 1 \times 8 \times 1$	=	16
by thin carpet	$= 15 \times 10 \times 0.25$	=	37.5
by woolen galeecha	$= 10 \times 5 \times 0.58$	=	29.0
by curtains	$= 2.5 \times 1 \times 8 \times 0.15$	=	3.0
by audience	$= 410 \times 0.45 = 184.5$		

Total absorption = 270 sabines

Volume = 20 × 15 × 5

= 1500 cu m

$$T = \frac{0.161 \times \text{volume}}{\text{Total absorption}}$$

= 0.161 × 1500/270

= 8.050/9 = 0.89 second

12.7 INSULATION

The property of reducing sound intensity (W/m^2) is expressed in terms of *sound reduction index* (SRI). It is defined as the ratio of the intensity of incidental sound to the intensity of sound coming out of the insulating material. It is usually indicated in decibels.

The reduction of sound intensity by material or a structure is caused by the combination of its mass and stiffness and hence its natural resonance. All good absorbers are good insulators. SRI takes into account not only the absorption, but loss due to resonance and scattering also. It depends on thickness of the material, while the absorption coefficient depends on the surface area only.

It is linearly related to mass per metre square (kg/m^2). If SRI for 6 mm thick plywood is 20 dB for 5 kg/m^2, then for plywood of 20 kg/m^2, it would be 4 times, that is 26 dB.

12.8 ACOUSTICS OF AUDITORIUMS

The following conditions should be fulfilled to get good acoustical properties in an auditorium.

1. Each separate syllable spoken should produce energy in every part of the hall, enough for comfortable listening.
2. Multiple reflections in the absence of suitable absorbents cause the successive sounds to blend with the earlier ones resulting in lack of intelligibility. This is known as *excessive reverberation* which should be checked and confined to the desired limit. It means that the reflecting surfaces and absorbents should be so planned that the sound of each syllable should decay before the arrival of the next syllable. This will improve intelligibility and at the same time will provide some reverberation (about 0.5 second) to minimise the ambient noise and to make sound appear natural.

3. Sound should be fairly uniformly distributed throughout the auditorium. Concentration of sound due to focussing in any specific area of the hall should be eliminated. There should be no zone of silence or no region of poor audibility anywhere in the hall. (Focussing is caused by concave surface).
4. The tonal quality of sound should not be adversely affected by any unpleasant reinforcement of any of the overtones of a complex wave.
5. To achieve low reverberation time for an auditorium, the absorption should be adequate. The absorbing material should be distributed randomly to eliminate distinct echoes. Absorbents should not be located near the performers. Absorption by absorbents is low at lower audio frequencies and this upsets the tonal balance. Low frequency resonators, using fibrous substances increase absorption at low frequencies. Absorption by the seats and audience should also be considered while calculating the reverberation time.

The following considerations at the time of constructing an auditorium are necessary.

Reverberation time for auditoriums The acoustical design of an auditorium depends on the purpose for which the hall would be used. A studio designed for musical concerts may have a reverberation time of about 1 second, while one for speech may have reverberation time of less than 0.5 second.

Resonance Any resonance in the wall, floor or ceiling is harmful as it emphasizes specific frequencies and upsets tonal balance. Resonance cavities are used to absorb such frequencies. Concave surfaces such as domes, curved arches and paralleled ceilings should be avoided.

Openings There should be an opening of about 0.5 square metre per person in the room. For air-conditioned halls which do not have windows, good absorbents save the purpose of windows.

Insulation The outer walls of the auditorium should be thick enough (massive and rigid) to insulate the extraneous sound coming through solid structures.

SUMMARY

- Due to multiple reflections from walls, ceiling, floor, etc., the sound in an enclosure fades away only gradually after the source of sound stops. This continuing echo is called reverberation.
- Reverberation time is defined as the time taken for sound energy in a room to drop to 10^{-6} times (i.e., 60 dB below its initial value). Reverberation gives liveliness or richness to sound. However, excessive reverberation will make the sound unintelligible. Typical reverberation time for a concert hall is 2 seconds, while for a lecture hall it is only 0.4 second.

  Sabine's formula Sabine found out the equation for reverberation time, T, which in MKS units is given below:

  $$T = 0.161 \frac{V}{a}$$

  where, V = Volume of the room,

  a = Total absorption of sound by materials

  Absorption coefficients of some typical absorbing materials are given in Table 12.1.
- Growth and decay of sound in an enclosure

  Direct sound reaches a point in a room first and thereafter reflected sounds start arriving at that point. So, the intensity of sound grows with time and becomes uniform. After some time when the sound source stops, the sound energy decays rapidly.
- Sound intensity can be reduced by material or a structure. Reduction depends on density multiplied by thickness (i.e. $\frac{\text{mass}}{\text{vol.}} \times$ thickness = kg/m^2) This property is expressed in terms sound reduction index (SRI)
- **Acoustic requirements of auditoriums**

  The important requirement of an auditorium is high intelligibility and comfortable listening, for which absorbent materials opening and resonant cavities should be planned meticulously.

Review Questions

1. Define reverberation time. Explain the importance of reverberation.
2. What are the factors on which reverberation time depends? Give typical values of reverberation time for a church or temple, concert hall, auditorium, lecture hall, studio.
3. Write down Sabine's equation for reverberation time, and define the terms used. Give absorption coefficients (typical values) for open window, heavy curtains, celotex, unpainted brick walls, glass panes, audience.
4. Explain with the help of a neat figure, how sound energy grows and decays at a point in a reverberating room.
5. What are the requirements for a good auditorium for pleasant listening? Give salient features of acoustical design for an auditorium.
6. Write a short note on 'Insulation for an auditorium'.

Short-Answer Questions

1. Distinguish between echo and reverberation.
2. Define reverberation time.
3. What type of sound is called lifeless?
4. What are the factors on which reverberation time depends?
5. Define absorption coefficient.
6. Why does the sound level remains constant for some time in a room after the sound from the source stops arriving?
7. Why is the reverberation time for a sound studio lower than that for a TV studio?
8. Why are diverging walls desirable in an auditorium?
9. Define sound reduction index.
10. Why should large walls and ceilings not be continuous?

Multiple-Choice Questions

1. A distinct echo is produced when the minimum time gap between the original sound and the reflected sound is
   (a) 10 ms (b) 20 ms
   (c) 30 ms (d) 60 ms
2. What is the typical value of reverberation period for a TV studio?
   (a) 2 second (b) 1 second
   (c) 0.4 second (d) 0.1 second
3. The unit of absorption coefficient is

(a) Sabine (b) Erg
(c) Newton (d) none

4. The absorption coefficient of an open window is
(a) 0 (b) 0.25
(c) 0.50 (d) 1

5. What is the typical open area (on an average) per person in a hall to give good acoustic quality?
(a) Zero
(b) 0.25 sq. metre
(c) 0.5 sq. metre
(d) 1 sq. metre

6. What is the typical value of SRI for 6-mm thick glass for 10 kg/m^2 mass?
(a) 40 dB (b) 24 dB
(c) 19 dB (d) 10 dB

7. The reverberation time is the time taken by sound energy to drop to some value below its initial intensity. What is this value?
(a) 0 dB (b) 20 dB
(c) 60 dB (d) 100 dB

8. Reverberation is caused by
(a) reflection (b) absorption
(c) diffraction (d) refraction

9. What is the typical value of total absorption of sound by a seated person?
(a) 0.1 sabine (b) 0.3 sabine
(c) 0.5 sabine (d) 0.7 sabine

10. Which one of the following is not an absorbing material for sound?
(a) Mirror
(b) Perforated board
(c) Asbestos
(d) Painting.

Numerical Problems

1. Calculate the reverberation time for an auditorium of 1000 cu m, having total absorption equal to 230 sabine units.
2. If the SRI of a substance is 23 dB and if 500 mW/m^2 of sound intensity was incident, calculate intensity of reflected sound.
3. Calculate the reverberation time for a hall whose length, width and height are 50 m, 30 m and 5 m, respectively, and the average absorption coefficient is 0.161.

   (Hint: Calculate total surface area of all walls, floor and ceiling)
4. Calculate the total absorption by the audience in a hall if there are 1000 persons and if the average absorption coefficient per person is 0.84 and the average surface area per person is 0.54 sq. metres.
5. If SRI for glass of 5 kg/m^2 mass is 18 dB, calculate SRI of glass of 20 kg/m^2 mass

Answers

Short-Answer Questions

1. When the reflected sound is heard distinctly, it is called echo. When the reflected sound is not heard distinctly, but causes gradual fading of the continuing reflections, it is called reverberation.
2. The reverberation time is defined as the time taken for sound energy in a room to drop to 10^{-6} times (that is, one millionth) of its initial value or 60 dB below its initial value.
3. When all the reflections are eliminated by fixing sound-absorbing materials on ceiling, floor and all the four walls, the result is a lifeless sound. All natural sound in a hall includes a proportion of continuing reflections. Variations in this proportion give sound a quality of liveliness or richness.
4. Reverberation time depends on

1. Volume of the room
2. Surface area of the absorbing material
3. Absorption coefficient of the surface area
4. Velocity of sound and hence wavelength

The formula for reverberation time T is T = 55.3 V/(ca) where V is volume of room, c the velocity of sound (344 m/s at room temperature) and a is the total absorption ($\Sigma\ \alpha\ S$), α being the absorption coefficient and S, the surface area of the absorbing material.

5. The absorption coefficient is defined as the ratio of energy absorbed by unit surface area to the total energy received by that area.
6. When the sound stops coming from the source, the reflected sound from the nearby surface decays to some extent, while the reflected sound from distant surfaces increases. The net result is that the reflected sound remains more or less constant for some time. Afterwards the reflected sound decays rapidly.
7. In a sound studio, impression of action is given by sound alone, that is, one can visualise the scene from the dialogues. But dialogues cannot be visualised by seeing the picture.
8. The side walls should be diverging to ensure higher sound level at the rear of the hall.
9. Sound reduction index of a material is defined as ratio of the intensity of incident sound to the intensity of sound coming out of the material.
10. Continuity should be broken to scatter and diffuse the sound.

Multiple-Choice Questions

1. (d)	2. (c)	3. (d)
4. (d)	5. (c)	6. (b)
7. (c)	8. (a)	9. (c)
10. (a)		

Numerical Questions

1. (0.7 second)
2. (2.5 mW/m^2)
3. (1.97 seconds)
4. (454 sabines)
5. (24 dB)

13 Television Fundamentals

13.1 ELEMENTS OF TV COMMUNICATION SYSTEM

The word ‘television’ consists of two Greek words, ‘*tele*’ and ‘*vision*’. ‘*Tele*’ means ‘at distance’ and ‘*vision*’ means ‘seeing’. Thus, television (or simply, TV) stands for ‘seeing at a distance’. It became possible when variations in light intensity (brightness) and colour could be converted into electrical variations (video signal) by photosensitive materials (camera tube) at the transmitter. This conversion required scanning of a photosensitive target. The electrical signals were reconverted into brightness and colour at the receiver by specially designed cathode ray tube (called *picture tube*) with the help of synchronized scanning of the screen. The electrical signals pertaining to brightness and colour can be transmitted to long distances with the help of a radio frequency carrier (in VHF and higher bands) by using amplification and modulation techniques. At the receiver end, the video signals can be recovered from the modulated VHF/UHF wave by adopting the superheterodyne techniques and detection and amplification. Thus man’s dream of seeing at distance became a reality. To sum up, the basic elements of a TV system are:

1. TV camera tube with provision of scanning the photosensitive target
2. TV transmitter (modulator and antenna)
3. Receiving antenna and superheterodyne receiver
4. Video detector and amplifier
5. Picture tube with provision of synchronized scanning.

13.2 SCANNING

The light intensity and colour variations in pictures are spread over the length and width of the picture. Such signals, varying in space, will need thousands of channels for transmission. An electrical variation can be transmitted through a single channel, if it varies with time alone. So, the space-dependent light (brightness and colour) is to be converted into a time-dependent phenomenon. This is accomplished by the process of *scanning* and using the property of *persistence of vision* of the eye (i.e. the ability of the eye to continue to see the light for 60 ms after the source of light is removed).

The process of exploring a video picture image (called a picture frame, or simply, a frame) point by point, in quick succession, is called scanning. The scanning starts from left, proceeds to the right up to the rightmost element of an image, and then returns (retraces or flies back) very quickly to the left on the next lower line, and so on, until the image is fully scanned. This is like reading a book line by line. This is made possible by applying sawtooth signals to the horizontal and vertical deflection circuits of the camera tube. These circuits deflect an electron beam horizontally and vertically. A similar process is adopted at the receiver to reproduce the picture. A sharply focused electron beam scans the screen of the picture tube from left to right and from top to bottom to convert the signal varying with time into brightness varying with space. Complete scanning of a line from the lefttmost point to the right most is called a *scanning line*, or simply a line.

Fundamentals have been discussed in the following sections, using values of TV system developed in Germany (known as CCIR-B system) and adopted in India, Values of other TV systems have been discussed in chapter 14.

Picture Frame In shooting of a cinema film, the image of a scene from within the field of view of the camera lens is formed on a photographic film. By successive opening and closing of the shutter, 24 photographs are obtained every second. Each photograph is called a picture frame. The same term 'picture frame' was adopted by TV engineers for one complete sequence of scanning from left to right and top to bottom and then again to the top (blanking). In Indian TV system, 25 frames are scanned every second. There are 625 scanning lines per frame. Thus, there are 625 × 25 (or 15625) scanning lines per second.

Flicker and Interlaced Scanning Although 25 to 30 frames per second are sufficient to give continuity of motion in a movie scene, the dark interruptions between bright pictures become visible as flicker (because persistence of vision increases as brightness decreases). Hence, a frame is scanned twice, using the interlaced scanning technique, shown in Fig. 13.1. In this system, the scanning lines of one sequence fall in between the lines of the previous sequence. This gives 50 interruptions per second and eliminates flicker.

Fields The two sequences of scanning of a frame are called *Field*s. The scanning sequences, 1st, 3rd, 5th etc., (shown by solid lines in the figure) are called *odd* Field, and 2nd, 4th, 6th etc., (shown by dotted lines), even Field . Thus, there are 50 fields per second, each field having 312½ lines. *The interlaced scanning, on the one hand, eliminates flicker and on the other, it conserves bandwidth.*

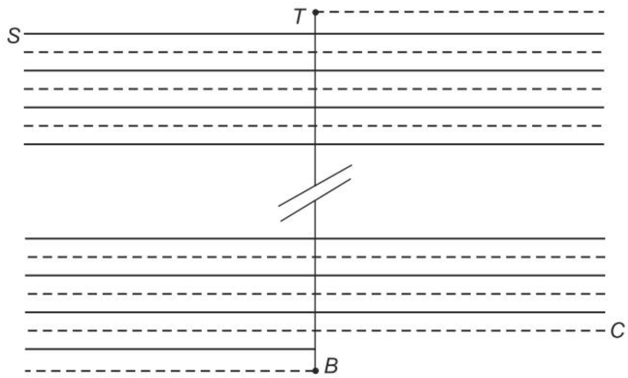


Fig. 13.1 *Interlaced scanning. Full horizontal lines show scanning in one sequence (odd field). Dotted lines show scanning in the next sequence (even field). Line BT (bottom to top) shows net vertical retrace at the end of odd field. AT the end of even field, scanning ends at C, the retrace takes the beam to the top at S.*

13.3 SYNCHRONISATION

It is most essential that the scanning process in the receiver is identical with the scanning process in the transmitter, i.e., with respect to the scanning sequences, scanning must start at the same relative point on the screen, should occur at the same speed, should end after the same interval and should repeat in identical fashion. Even the slightest deviation from the identicalness will distort the picture and make it unacceptable. The process of ensuring exact similarity (identicalness) in the scanning process in the receiver with that in the transmitter is called synchronisation. It is achieved by starting scanning signal in the transmitter by triggering the horizontal and vertical scanning oscillators by pulses, called H-sync pulse and V-sync pulse, respectively. These pulses are added to the video signal in the transmitter and are transmitted along with it. In the receiver, these are extracted and used to trigger the scanning oscillators.

In the colour TV transmitter, colour is encoded by amplitude modulating a sub-carrier (4.43 MHz), using the suppressed carrier AM method. (In another system, called SECAM system, colour is encoded using frequency modulation.) For reception of colour, suppressed sub-carrier is required to be regenerated at the receiver to demodulate colour signals. Again to maintain identicalness in the sub-carrier signal generated in the receiver with that produced in the transmitter, about 8 cycles (called colour burst) of sub-carrier signal are sent along with the video signal.

Sync pulses and colour burst signals are added to the blanking pulse which is transmitted to cut off the cathode ray tube during retrace of the scanning beams. This cut-off is essential so that the retrace and the controlling pulses are not visible on the screen as video signals. (The level of blanking pulse is called *pedestal level* and the difference between average brightness and pedestal level is called *pedestal height*.)

13.4 ASPECT RATIO

The width-to-height ratio of a picture frame is called *aspect ratio*. The width is kept longer than the height because of the following facts.

1. Horizontal dimension of a scene is generally more than its vertical dimension.
2. The retina (screen of the eye) has greater width than height.

As a result of intensive subjective tests by the cinema technicians, the aspect ratio of 4:3 was considered most suitable for cinema screens. The same ratio was accepted by television engineers for television, as cinema films formed a major part of TV programmes. Thus, for conventional TV screens, the aspect ratio is 4:3 in all TV systems. However, for wider screens of high definition TV systems, the aspect ratio of 16:9 has been standardised.

13.5 PICTURE ELEMENTS (PIXELS)

Light from a picture is converted into video signal by the scanning process. The tiny portion of a picture that can be covered by the spot of a scanning beam is called a *Picture element*, or, in short, *Pixel*. Its vertical dimension will be equal to the distance between two scanning lines, or centre-to-centre distance between two adjacent spots. Its horizontal dimension will also be the same. As width to height ratio is 4:3, the number of pixels on a horizontal scanning line will be 4/3 times the pixels on a vertical line. The total number of pixels activated on the screen reproduce a picture.

Pixels lost in Vertical Blanking When the scanning beam flips from the bottom of the picture frame to the top, the flipping takes some time. It is equal to the scanning time of 20 lines for each field or 40 lines for each frame. These lines, and hence the corresponding number of pixels, are lost, as the screen remains cut off or blanked during this period.

Active Lines The actual lines which reproduce pixels are called *active lines*. The number of active lines (N_a) is equal to the total scanning lines per frame (N_t) minus the lines lost (N_l)during vertical blanking, as indicated by Eq. 13.1.

$$N_a = N_t - N_l \tag{13.1}$$

Loss of Pixels Due to Kell Effect The pixels are not arranged in an orderly manner in a picture. These are distributed rather randomly. Hence a spot of the scanning beam may cover partly two pixels, or a scanning beam may miss some pixels. It has been estimated by intensive subjective experiments that about 28 to 32 per cent pixels are lost in this way. It means that only 72 to 68 per cent pixels are actually reproduced by the active scanning lines. This effect is called Kell effect. Due to this effect, the number of active lines is to be multiplied by a factor varying from 0.68 to 0.72 (mean value = 0.7) to get the number of lines which are actually effective in reproducing the pixels. This multiplying factor is called 'Kell factor'.

13.6 RESOLUTION

Resolution means ability to distinguish between closely spaced pixels. Greater the number of pixels, smaller would be their size and hence higher would be the resolution. High resolution will, thus, enable us to distinctly see closely spaced small objects, or the fine details of a picture, for example, wrinkles on a face, hair of the eyebrows, veins on leaves. The number of pixels on a vertical line on the screen will be equal to the number of active lines multiplied by the Kell factor (K). This is called *vertical resolution* (R_v)and is given by Eq. 13.2.

$$R_v = (N_t - N_l) \times K = N_a \times K \tag{13.2}$$

As the aspect ratio is 4:3, the number of pixels on a horizontal line would be equal to R_v multiplied by 4/3. This is called *horizontal resolution* (R_h) and is given by Eq. 13.3.

$$R_h = R_v \times A \qquad \text{(where } A \text{ is the aspect ratio)} \tag{13.3}$$

Generally, resolution of a TV system is expressed in terms of the vertical resolution. For the CCIR-B German system, it is about 400 lines and for the CCIR-M American system, about 340 lines. (For a computer monitor, it is expressed as $R_h \times R_v$. Typical resolution for videographic array-type monitor is 1024×768 pixels)

Picture Definition (Tonal Gradation)

Sharpness of the edges of a picture is expressed by the term picture definition. In good, quality picture, edges should be sharply defined, and should not appear smeared. Good definition is obtained when the high frequency response of the video amplifier is good.

This is illustrated in Fig. 13.2(a). Smear is shown in Fig. 13.2 (b).

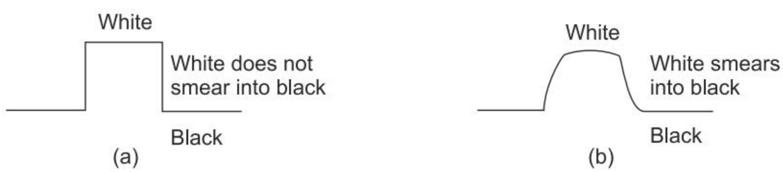


Fig. 13.2 **(a)** *Good Definition;* **(b)** *Poor definition*

13.7 BANDWIDTH

It is the highest video frequency related to the time taken in scanning two adjacent pixels. Figure 13.3(a) shows 4 successive pixels, and Fig. 13.3(b) shows

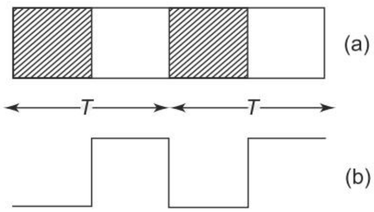


Fig. 13.3 **(a)** *Four adjacent pixels of black and white shades;* *(b) Two cycles of brightness signal corresponding to pixels of Fig. (a)*

the corresponding 2 cycles of brightness. The time period T is the time taken in scanning one cycle or 2 adjacent pixels.

If t is the time in seconds taken in scanning one line (excluding retrace time), then R_h pixels are scanned in t seconds. Therefore, 2 pixels shall be scanned in $2t/R_h$ seconds. Thus, time period, T, is given by Eq. 13.4.

$$T = \frac{2t}{R_h} \tag{13.4}$$

Video bandwidth will be given by Eq. 13.5.

$$\text{Bandwidth } = \frac{1}{T} \tag{13.5}$$

Relationship between Resolution and Bandwidth

Smaller the size of pixels that can be reproduced, better would be the resolution. At the same time,smaller the size of pixels, less would be the time taken in scanning two adjacent pixels, as shown in Fig. 13.4.

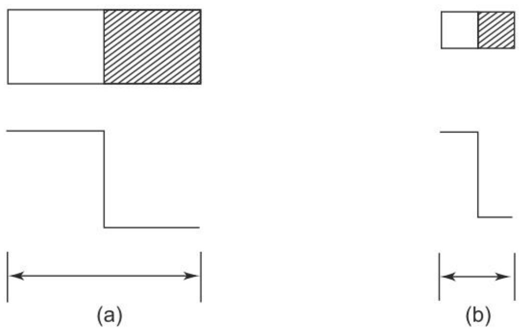


Fig. 13.4 **(a)** *Big pixels (low resolution) and hence long time period (low bandwidth);* **(b)** *Small pixels (high resolution) and hence low time period (high bandwidth)*

Thus, the resolution and bandwidth depend on each other. Higher bandwidth gives better resolution.

The resolution and bandwidth both depend upon the number of active lines of scanning per frame (greater the number of active lines, greater is the bandwidth and higher is the resolution.)

The maximum number of scanning lines per frame will depend upon the size of the focused spot of the scanning beam. Smaller the size of the spot, greater would be the number of lines that can be accommodated on a picture frame without overlapping. So, for the high-resolution systems, the electron beam has to be sharp.

Bandwidth will also depend upon the frequency of the frames, i.e., upon frames per second. Greater the frame frequency, less will be the time taken in scanning one line and hence higher would be the bandwidth.

13.8 COMPOSITE VIDEO SIGNAL

Video signal, before final modulation, consists of the following components:

1. Luminance (brightness) information.
2. colour information (chroma signal).
3. H and V-blanking pulses
4. H and V-sync pulses
5. Colour burst signal (about 8 cycles of colour subcarrier signal)

As all the components of a composite video signal are to be received on one single channel (7 MHz wide), so that they may be tuned by a single tuner), it is necessary that all these should be located properly within the channel.

The blanking pulses, sync pulses, colour-burst signal and actual video signal (picture brightness and colour information) all form one composite signal called *Composite Video Signal* (or CVS) as shown in Fig. 13.5 for one H-scanning line. Percentages indicated in the figure, represent amplitude in terms of percentage of carrier level. The width of the pulses have also been shown.

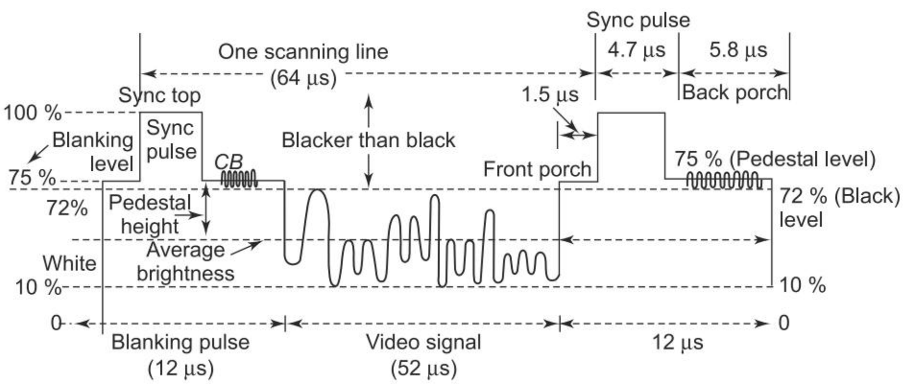


Fig. 13.5 *Composite video signals (CVS)*

The vertical blanking pulse (shown in Fig. 13.6) is of much longer duration (1280 μs than the H-blanking pulse, and appears 50 times per second. Five pre-equalising pulses are superimposed on the blanking pulse before and an equal number post-equalising pulses after V-sync pulse to ensure correct interlacing. The V-sync pulse (2.5 H-width in India) is serrated after every H/2 (32 μs) interval to ensure continuation of the H-synchronisation during the long time interval (160 μs) of V-sync pulse. Vertical blanking also forms part of the composite video signal in the same way as the horizontal blanking pulse. Vertical blanking pulse is shown in Fig. 13.6 (a) at the end of odd field and in Fig. 13.6(b), at the end of even field, Vertical sync is slotted so that H sync is not lost during the long period.

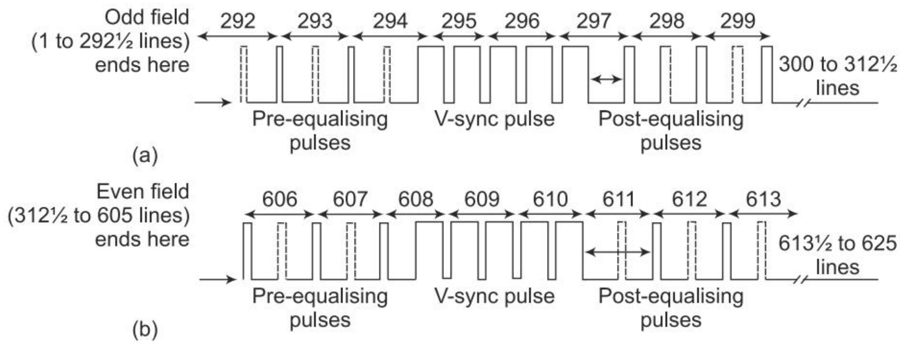


Fig. 13.6 *(a) V-blanking pulse at the end of odd field (20 lines, starting at 292 ½ and ending at 312 ½ line); (b) V-blanking pulse at the end of even field (20 lines, starting at 606th line and ending at 625th line)*

13.9 MODULATION OF VIDEO SIGNALS

The composite video signal modulates a video carrier in VHF/UHF band (say, 62.25 MHz for the 4th channel of TV transmission in India), using amplitude modulation. To conserve bandwidth, a lower side band is attenuated or vestiged. The modulation, thus, is AMVSB. It occupies 5.75 MHz bandwidth (flat) (0.75 MHz on lower sideband side and 5 Hz on the upper sideband side). Colour information (chroma signal) is inserted (interleaved) in between the slots vacant in the distribution of luminance energy within the video bandwidth. Colour signals modulate a subcarrier (4.43 MHz), using amplitude modulation carrier suppressed (AMSC) method with upper sideband vestiged. (The French system uses frequency modulation). This modulated subcarrier becomes an integral part of the composite video signal which finally modulates the video carrier. The complete radio frequency spectrum for 61-68 MHz channel (4th channel) is shown in Fig. 13.7 for TV system used in India.

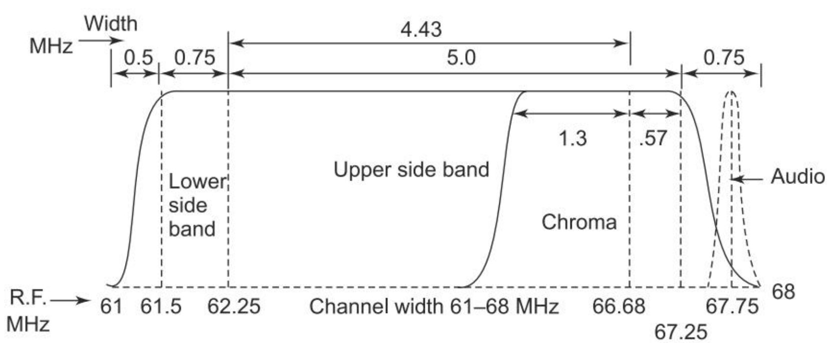


Fig. 13.7 *Radio frequency spectrum for fourth channel of band I*

13.10 MODULATION OF AUDIO SIGNALS

Audio signals, associated with picture, modulate a separate carrier (called audio carrier), using the frequency modulation method. This audio-modulated signal lies within the prescribed TV channel width. The difference between a video carrier and audio carrier is constant. This difference is called *inter-carrier frequency*. Its value is 5.5 MHz in CCIRB system.

13.11 MONOCHROME TV CAMERA TUBES

Information about brightness and colour in a scene or picture is converted into electrical (video) signals by means of a photosensitive target. The device which produces video signals is called ‘camera tube’. Vidicon and Plumbicon are most widely used camera tubes which are described below.

Vidicon Camera Tube

Construction The Vidicon camera tube, developed by RCA in 1951, has a photoconductive target plate and an electron-beam focusing and scanning system. The photoconductive material is antimony-trisulphide whose resistance at any point depends on the intensity of light falling on that point. Construction of the Vidicon camera tube is illustrated in Fig. 13.8.

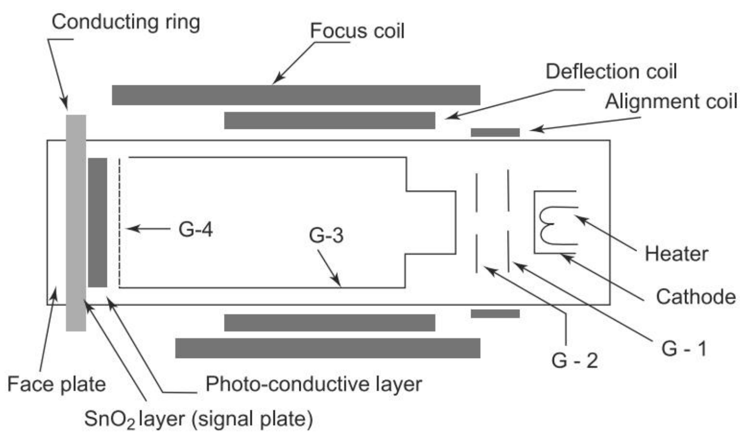


Fig. 13.8 *Vidicon camera tube*

The input light from a scene passes through a lens system and is incident on the face plate made of optically flat glass. The light from the face plate falls on a target plate which has two layers. Facing the light is a thin coating of tinoxide which is transparent to light and is a good conductor of electricity. This layer is called *signal plate*. Electrical connection of load is made to the signal plate through a terminal which is a conducting metal ring surrounding the tube. The back of the target plate (facing the electron gun) is coated with antimony trisulphide, a semiconductor. When light falls on it, free electrons are produced in the material. In the absence of light, the material is almost an insulator, having very few free electrons. Its resistance is as high as 20 MΩ at normal temperatures. Bright light reduces the resistance to 2 MΩ. The resistance will lie between 2 MΩ and 20 MΩ for various shades of grey, increasing as the shade becomes darker.

The target plate is scanned by a focused electron beam produced by an electron gun consisting of indirectly heated cathode, a control grid G-1 (at –50 V variable) and an accelerator grid G-2 (at 300 V). A focus grid G-3 is at a lower potential (260 V) than G-2 and hence slows down the electrons, converging them to the centre of the beam. The wire-mesh G-4 (at 400 V) works as a muzzle of the electron gun. The target is at much lower potential than the mesh and so slows down the electrons to such an extent that they land at the target with almost zero velocity and are also perpendicular to the target. Deflection coils, mounted outside the tube, allow the beam to scan the target horizontally and vertically. A focus coil, also mounted outside the tube, sharpens the beam further. Alignment coils align the beam to the axis of the tube in the absence of deflection.

Principle of Working The signal plate is kept positive by an external source of supply. When light produces free electrons in the material, these migrate to the positive signal plate, leaving deficiency of electrons on the semiconductor material towards gun side. This deficiency of electrons (or +ve charge) will be proportional to the number of free electrons produced by light and migrated to the signal plate. Thus the deficiency of electrons for any element on the gun side of the semiconductor material (or the +ve charge on the material) will be proportional to the light incident on that element. This means that a charge-image proportional to the optical image has been formed on the gun side of antimony trisulphide. This is illustrated in Fig. 13.9

When the scanning beam lands on the target material, it will leave electron in the material just sufficient to neutralise the deficiency of electrons (or +ve charge). (The electrons remaining in the beam will return back under the influence of +ve grids. These returning electrons are not utilised in Vidicon).

Neutralising the deficiency of electrons will result in an electric current through the load (about 50 kΩ). The electric circuit will be complete through the electron beam and cathode as shown in Fig. 13.10, the current at any spot of scanning will be proportional to the electrons deposited by the beam, which, in turn, will be proportional to the intensity of the brightness at that spot.

As any element of the target is scanned every 40 ms (one frame's time) in the CCIR-B system, the free electrons will be produced for 40 ms at that element, and so the net change in resistance of the material, or net deficiency of electrons is a cumulative effect, called *storage action*. This increases sensitivity of the camera tube.

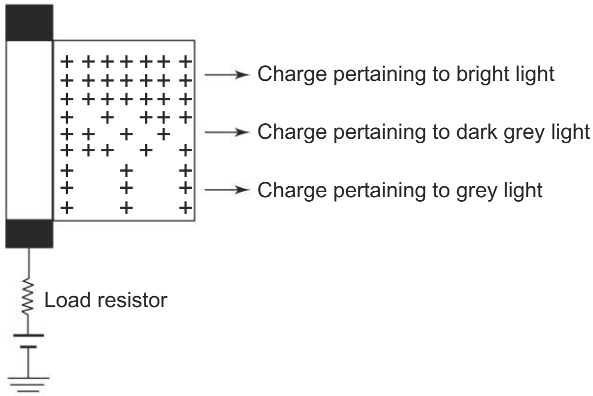


Fig. 13.9 *Formation of charge in Vidicon*

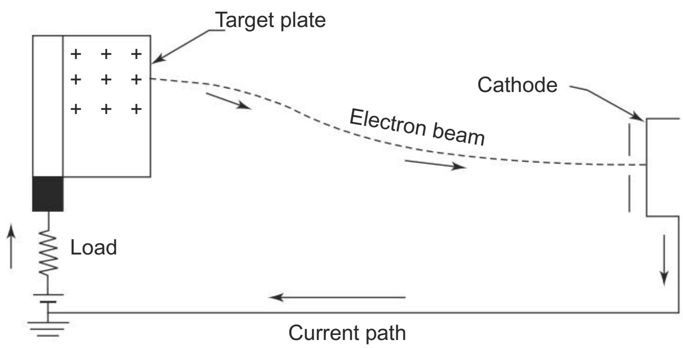


Fig. 13.10 *Electric circuit in Vidicon*

During very bright portions of the picture the number of electrons produced is too high, all of which cannot be withdrawn by the dc voltage and hence a faded image is left behind. This is called 'image lag' and this is a demerit of the Vidicon tube.

Characteristics of Vidicon Camera Tube

Sensitivity	:	125 μA per lumen.
Signal-to-noise ratio	:	45 dB
Image lag	:	Severe (about 20%)
Halo	:	No halo
Ghost	:	No ghost
Dark current	:	20 nA
Resolution	:	55% at 400 lines (5 MHz)
Spectral response	:	Close to eye's response
Infrared sensitivity	:	Moderate

Plumbicon Camera Tube

The Plumbicon camera tube, developed by Philips in 1963 to eliminate the image lag effect of vidicon, has a photoconductive target plate of lead mono-oxide (PbO), and intrinsic material, sandwiched between an *n*-type and *p*-type layer. The tube has been named Plumbicon because lead (plumbum or Pb) has been used in the target material. Its construction is similar to that of Vidicon except the target which is described below

Target of Plumbicon The target consists of a PIN diode as shown in Fig. 13.11. When light falls on the target, electrons are released. The electric field gradient set up in the intrinsic layer is very high. This high gradient makes all released electrons sweep past the target quickly and eliminates image lag. The PIN-diode acts as a capacitor and the free electrons released by light cause a resistance to be formed across the capacitor.

The capacitor charges to the bias voltage in the absence of light, i.e., when resistance is infinite (or very high). When light is incident and the electrons are produced, a resistance is formed across the capacitor, and therefore, the capacitor discharges; the potential difference across the capacitor decreases. This means that if the left-hand side plate of the capacitor is maintained at +40 volts, the right-hand side one will have a voltage varying from 0 to 40 volts depending upon the value of resistance, or quantity of the electrons produced or the intensity of light. As the intensity of light increases, resistance *R* decreases, and voltage on the right-hand side plate increases. Thus, voltage on the plate towards the gun side is the charge-image proportionate to the optical image.

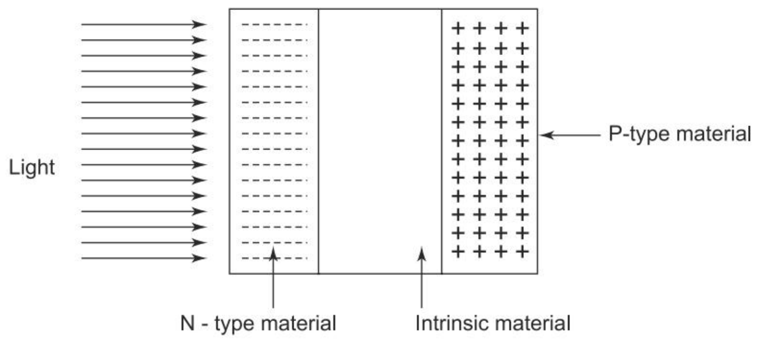


Fig. 13.11 *Target of Plumbicon camera tube*

When the scanning beam lands on an element on the target, it leaves sufficient electrons to neutralise the positive voltage. When the positive voltage is neutralised, it amounts to flow of current through load resistance as shown in Fig. 13.12. The current will depend on the positive voltage on the plate (gunside), and hence will be proportional to the intensity of light . Thus, a video signal is produced in the load.

Use of PIN diode eliminates image-lag because of very high potential gradient across the PIN diode. Sensitivity of Plumbicon is 400 µA per lumen and dark current is 1 nA only

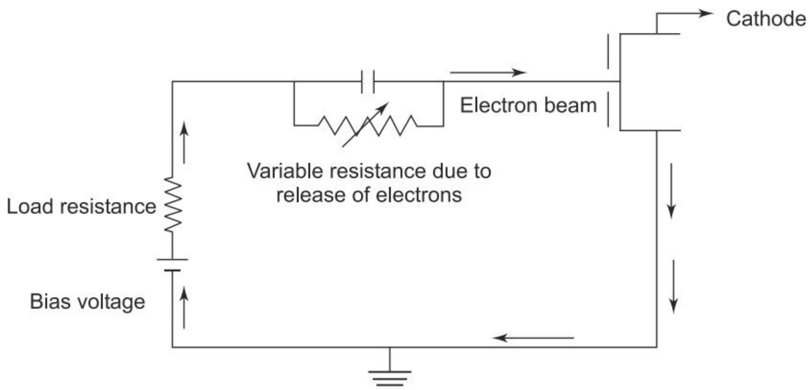


Fig. 13.12 *Discharging current flowing through load*

13.12 COLOUR CAMERAS

Three-tube camera for reproducing colours, the light incident on the camera tube is first broken into three primary colours (red, green and blue) with the help of dichroic mirrors (or better prisms) and colour filters. The three colour beams strike the three camera tubes. The camera tube in which the red beam is incident will give a video signal pertaining to brightness present in the red colour. For example, if the object to be televised was blue, the output of this camera would be zero. Similarly, the second camera will give video signals of green light and the third camera, blue light. This is shown in Fig. 13.13. The terms V_R, V_G and V_B represent signal voltages pertaining to red, green and blue colours, respectively.

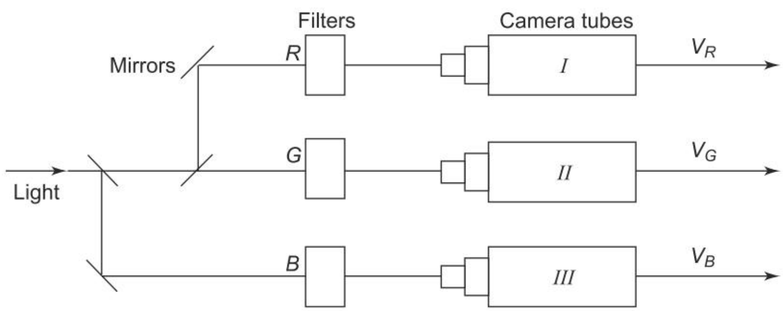


Fig. 13.13 *Production of color signals by tree-tube camera*

Single Tube Camera In this system, the face plate contains very closely spaced stripes of red, green and blue colours which act as colour filters. The target also has corresponding stripes. The target converts colour signals into respective charge signal pattern. The scanning beam produces video signals on the respective load resistors. Hence, we get video signals corresponding to red, green and blue colour. The Y-signal is obtained with the help of a resistive matrix. Thus, we get the same signals as we get in a 3-tube camera.

The single-tube colour camera is quite portable and uses a 17-mm Vidicon tube. It is very suitable as domestic colour camera, and is also suitable for Electronic News Gathering (ENG).

Solid State Video Camera Photo-diodes, using semiconductor technology, are good transducers for converting light intensity into electrical signals. Solid state video cameras have been developed, using photodiodes, MOS capacitors and shift registers (for scanning). These are described in Appendix II.

13.13 COMPATIBILITY

When colour transmission was developed, millions of monochrome (black and white) TV receivers were in use. It was not possible to discard these receivers. The colour transmission and receiving system had to be so designed that the colour transmission might be received as black- and-white pictures in the monochrome TV receivers and the monochrome transmission might be received by the colour TV receiver as black-and-white pictures. This phenomenon is termed *compatibility*. The following conditions should be essentially fulfilled for achieving compatibility.

1. The monochrome luminance signal should be transmitted as a separate component in a colour composite signal, so that a monochrome receiver may receive it without any change in its circuits.
2. The colour signal should be transmitted in such form that it is ignored by a monochrome receiver, but is available in appropriate proportions of three primary colours at the input of the colour picture tube.
3. the overall bandwidth and channel width should remain the same for colour transmission as well as for monochrome transmission.
4. The position of audio carrier in the overall channel width should remain unchanged.

13.14 LUMINANCE SIGNAL

The white signal consists of equal quantities of green, red and blue light, say 1 lumen each. Monochrome camera tube is so adjusted that we get 59% of green, 30% of red and 11% of blue in its output. For colour cameras, luminance signal (represented by letter Y) is obtained by a resistive matrix, shown in Fig. 13.14.

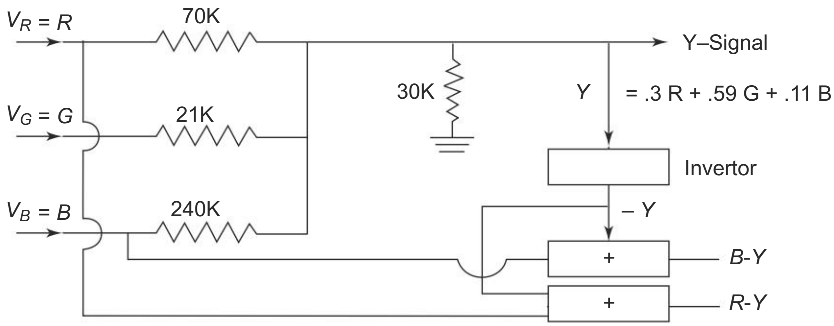


Fig. 13.14 *Production of luminance (Y) signal and colour difference signals*

13.15 CHROMINANCE SIGNAL

In a colour camera, outputs form chrominance signal. In Fig. 13.14, the voltages of colour signals are indicated for simplicity, by the letters R, G and B in place of V_R, V_G and V_B. These signals are further transformed into colour-difference signals, *B–Y* and *R–Y* by inverting the *Y* signal and adding it to *B* and *R* to achieve compatibility. This is also shown in Fig. 13.14. These signals modulate a subcarrier. As two signals, *R–Y* and *B–Y* are required to modulate a single subcarrier, the technique of quadrature modulation is used. In this technique, the subcarrier is modulated by *B–Y* in the same phase, and rotates by 90° to get modulated by *R–Y*. The two modulated signals are added and finally form chrominance or chroma signal. The carrier is known as a subcarrier because the chroma signal further modulates the main video carrier of VHF or higher band.

13.16 PICTURE TUBES

In all TV and video-monitoring systems, a picture tube is used to convert video signals into variations of light.

In monochrome monitors, it consists of an electron gun, comprising of an indirectly heated cathode, a control grid, screen grid or accelerating anode and a focus grid. It produces an electron beam resulting in a sharply focused spot on a fluorescent screen. A large angle deflection of the electron beam is caused by strong magnetic field produced by sawtooth currents through horizontal and vertical deflection coils (yoke) mounted on the neck of the picture tube. There are tiny magnets for centering the beam and for eliminating the pincushion effect. It is illustrated in Figure 13.15.

In colour monitors, 'Precision-In-Line' (PIL) picture tubes are used. Three guns (shown in Fig. 13.16(a)) are used for video signals corresponding to red, green and blue colours, These beams are focused on a shadow mask containing vertical slots (shown in Fig. 13.16(b)). There are three stripes for each slot (one of red phosphors, the other of green phosphors and the third of blue phosphors on the screen). The three beams, diverging from a slot, strike the three stripes causing the red, green and blue phosphors to glow. The eyes integrate the three colours to impart the impression of one resultant colour. The intensity of the glow is proportional to the intensity of the respective electron beams. Hence, the original colour is reproduced on the screen. PIL picture tube is illustrated in Fig. 13.16(c) shown external mounting for adjustments, like convergence and colour purity.

Gamma Effect The intensity of light produced on a fluorescent screen is not linearly proportional to the intensity of a video signal. The curve representing the relationship between the light and video signal follows the square law. If the

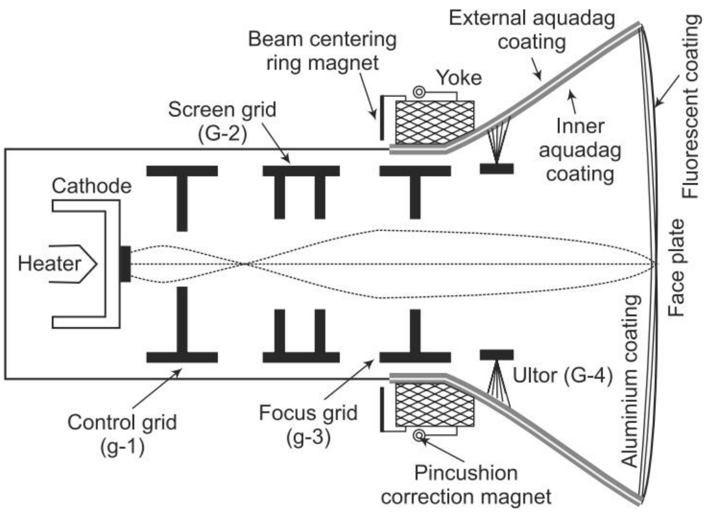


Fig. 13.15 *Monochrome picture tube*

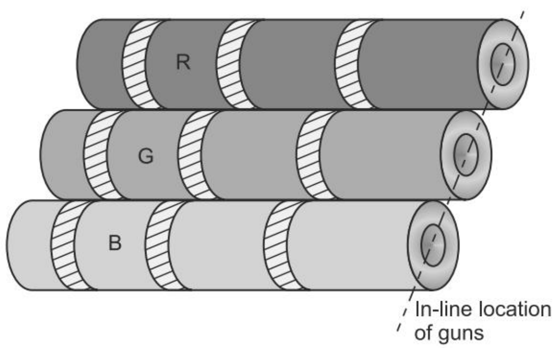


Fig. 13.16 (a) *Guns in in-line configuration in PIL picture tube*

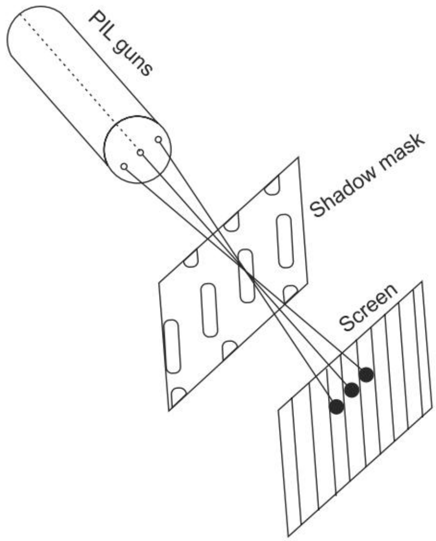


Fig. 13.16 (b) *PIL picture tube's mask*

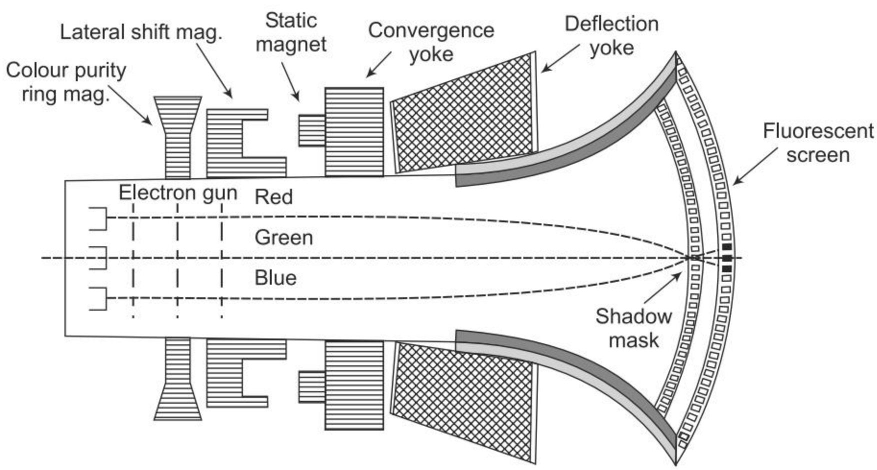


Fig. 13.16 (c) *Layout of different external yokes and components*

intensity of the video signal doubles, the intensity of light becomes four times. This is called 'Gamma effect'. It affects the reproduction of colours adversely. To reduce this effect, a 'gamma correction circuit' is used in the camera tube. This circuit makes light and signal characteristics follow the square-root law. It neutralises the square-law effect in the monitor and hence, a linear relationship between light and video signal is obtained.

13.17 SOLID STATE PICTURE TRANSDUCERS

The conventional picture tubes are thermionic tubes, using heated cathodes to emit electrons in vacuum and control the same by means of electrostatic fields through various grids. These tubes have all the disadvantages of thermionic devices like bulky size, provision of a heater, provision of vacuum, high voltages, etc. It has been a long cherished dream of television scientists to discover suitable solid state transducers and develop a flat panel. Study of the following devices like light emitting diode (LED), gas discharge plasma and liquid crystal display (LCD) has been persued towards this end. These devices are described below:

LED It is a semiconductor made of gallium-arsenide phosphide which can emit bright light of three primary colours (the colour depends on the composition of phosphorus in gallium-arsenide). However, LEDs are costly and consume considerable power. So they were discarded for use as TV display device.

LCD It is cheaper and needs little power and so they hold maximum promise for being used as flat panel display.

Liquid crystals are organic compounds whose physical properties are the same as those of a crystal, but whose behaviour resembles that of a liquid. Hence they are known as liquid crystals. A molecule of a liquid crystal has an elongated rod like shape and its structure is somewhat rigid. Electric conductivity, refractive index and polarization of a light coming out of it depends on the orientation of its crystals. This property of a liquid crystal is used to make the light, passing through it, change its polarization. The liquid crystals adjust themselves parallel or perpendicular to the electric field.

The electric field applied to an LCD cell twists the molecules, the twist being proportional to the signal voltage. The electric field is applied at right angles to the axes of liquid crystals. The LCD element has a thin (about 10 micron thick) liquid crystal sandwitched between two transparent plates. When the voltage applied to the plates varies, the polarization of light passing through the liquid crystal changes. It can change between 0 and 90 degree. The action of an LCD is shown in Figs 13.17(a) and 13.17(b).

The polarization of the incident light is random. The vertical polarizer selects only vertically polarized light. In Fig. 13.17(a), the LCD crystal causes a shift of 90° in the polarization and hence it becomes horizontally polarized light. The next polarizer is the horizontal polarizer which allows this light to pass and so light comes out of the LCD element.

In Fig. 13.17(b) the LCD crystal causes no shift in the polarization. So the light coming out of the liquid remains vertically polarized, which is stopped by the horizontal polarizer. Thus, no light will come out of the LCD element, Consequently the transparency of the LCD pixel element will depend on the degree of shift in polarization. It will have different intensity within two limits of full light to zero light. Relationship between signal-voltage and transparency of the cell is linear.

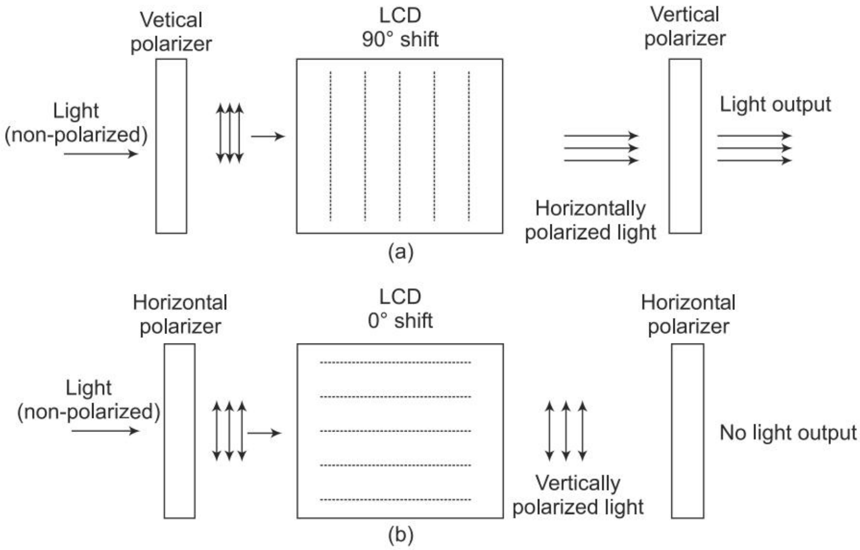


Fig. 13.17 *(a) Orientation of liquid crystal molecules to allow light in the output (b) Orientation of liquid crystal molecules to block light.*

In a practical system, 100 000–400 000 pixels of LCDs are arranged in a rectangular matrix. Each pixel is controlled individually by its built-in thin film transistor. It can be opened or closed or driven to any point between the two extremes to produce the original pattern of light intensity on the screen.

Pocket TV receivers, electronic toys and lap-top computer monitors are widely using LCD display panels because of their obvious advantages of low power, small weight, compactness and high reliability. LCDs have also been developed as large size flat TV panels.

Plasma Screens Plasma Screen was invented at the University of Illinois (USA) in (1964). Many tiny cells hold a mixture of xenon, neon and helium gases (known as noble gases) at low pressure. Each cell represents a pixel, thus, there are hundreds of thousand cells. The noble gases between two panels of glass are ionized and converted into plasma when a suitable electric potential is applied across the panels. The discharge in each cell is controlled by applying the video voltage in correct sequence.

The back of each cell is coated with a phosphor. As the gas-ions rush to the electrodes and strike, ultra-violate light is emitted which excites the phosphors to glow and reproduce the original picture. Plasma displays have been developed as large luminescent flat panels. Their advantages and disadvantates vis-à-vis LCD are as follows.

Advantages

1. Plasma TV is slim (thin screen).
2. Mounting on the wall is feasible

3. Colour reproduction is better
4. Image is brighter
5. Contrast ratio is superior (1000 000 : 1)
6. Viewing angle is wider
7. Refresh-rate is very high and response time is faster. Hence, there is no motion-blur for fast moving objects like aeroplane, cricket-ball.

Disadvantages

1. Plasma display is susceptible to large area flicker
2. Can be used only for large panels (80 cm or more). Cannot work for small screens
3. More ambient light is reflected. There is reflection glare in bright rooms.
4. Although slim, they are heavier than LCD panels, because of use of glass (LCDs use plastic, which is light).
5. Electric power consumption is more
6. They are not able to work at high altitudes due to pressure problem
7. Plasma contributes to global warming, as nitrogen-trifloride (a very potent green house gas) is used during production.

LCD vs Plasma AT present, there is great rivalary between LCD and plasma screens. But because of lower power consumption and lack of green house gases, LCD screens have an edge over the plasma screens. Moreover, plasma screens cannot have smaller size; where smaller sizes are needed as in calculators, computer, mobile, etc. there is no substitute for LCDs.

Solid State Panels Vs CRT Despite impressive progress, the solid state displays are still inferior to the CRTs. Colour-resolution and low cost of a CRT are hard to beat. The cathode ray tube designers continue to improve the performance. A 38 × 48 cm tube (designed by Flamders Research, New Jersy) provides resolution of 3300 × 2560 pixels suitable for imaging applications in medicine and defence. Although with vast funding for research on solid state panels, the flat panel display systems are evolving fast, yet they are nowhere on the horizon to compete with the century old CRT which is still dominating the consumer market of TV receivers at the start of the third millennium.

13.18 TV BROADCASTING SYSTEMS

There are two types of TV broadcasting systems: (1). Monochrome (or black and white) TV, and 2. Colour TV. Their basic elements are described below.

Monochrome TV Transmitter The basic elements of a monochrome TV transmitter are shown in Fig. 13.17. Their functions are described as follows.

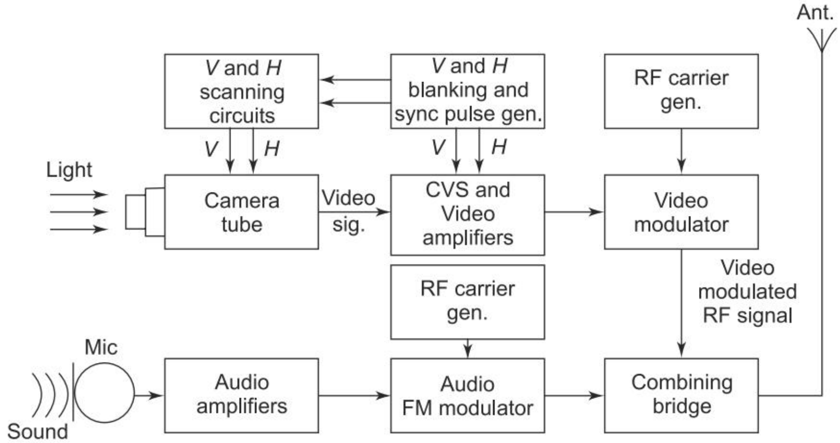


Fig. 13.17 *Basic Elements of a Monochrome TV Transmitting System*

Camera Tube Converts intensity of light from a scene into electrical variations, called video signal by using a photosensitive target plate.

Scanning and Sync Circuits Electrical current is extracted from the photosensitive target of the camera tube with the help of a scanning beam which is produced by sawtooth currents through horizontal and vertical deflection coils.

Blanking and Sync Pulse Generators The start of a sawtooth or sweep current signal is triggered by pulses called sync pulses. The retrace is blanked by blanking pulses and these pulses are periodic and appear for the specified time by using a monostable multivibrator.

Video Amplifiers Video signals along with blanking and sync pulses, called composite video signals are amplified by using wideband RC coupled amplifier circuits.

The RF carrier generator and video modulator are radio frequency modulated by the video signal. Modulation is of Vestigial Sideband (VSB) type AM to save the bandwidth. In this type of modulation, one sideband is vestiged to a small fraction of the whole band. In a TV, the lower sideband is vestiged and the upper sideband and carrier are sent in full.

Microphone It converts sound pressure variations into electrical variations, called audio signals.

Audio Amplifiers These amplify the weak audio signals.

RF Carrier Generator and Audio Modulator A radio frequency carrier is generated and is frequency modulated by the audio signals at low level. The radio frequency is then multiplied and amplified to the full level for transmission.

Combining Bridge Video modulated and audio modulated signals pass through a combining bridge (also called diplexer circuit) to go to a common transmitting antenna. The bridge prevents audio modulated signals from going to video sections and vice versa to avoid overloading.

Transmitting Antenna Video modulated signals and audio modulated signals are fed to the common transmitting antenna which radiates out the modulated signals into space in the form of electromagnetic waves. The antenna is omnidirectional in the horizontal plane.

Monochrome TV Receiver

The basic elements of a monochrome TV receiver are shown in Fig. 13.18.

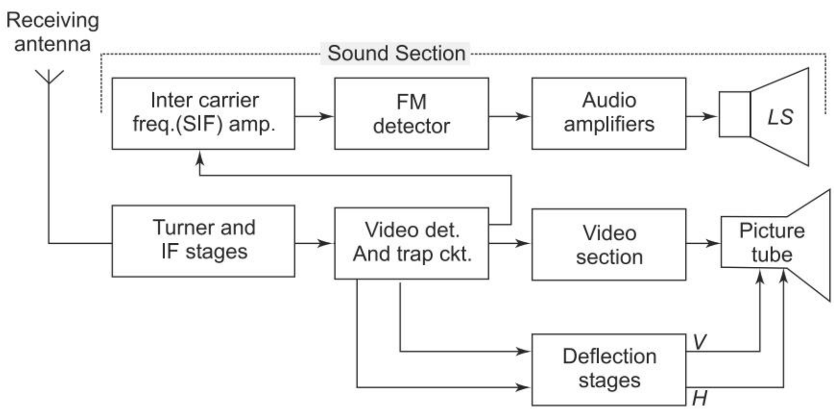


Fig. 13.18 *Basic elements of a TV receiver*

Tuner and IF Stages The receiver is a superheterodyne receiver to achieve high selectivity and high gain. In a superheterodyne receiver, the radio frequency signal, duly amplified by a pre-amplifier, is mixed nonlinearly with the oscillations of higher frequency but of fixed amplitude, generated by a local oscillator. The output of the mixer consists of several intermodulation products (due to non-linear mixing), one of which is a signal having a frequency equal to the difference of frequencies of the two signals. The difference frequency is called intermediate frequency, which is selected and amplified. The advantages of the superheterodyne technique are better selectivity and higher gain.

Video Detector and Trap Circuit The amplified IF goes to the video detector which recovers video signal from the modulated wave and feeds it to the video amplifier for amplification through the trap circuit, which prevents the video signal from entering into audio channel.

Video Amplifiers These are wideband RC coupled amplifiers. The amplified video signal goes to the picture tube.

Picture Tube The video signal varies the strength of the electron beam. This beam strikes the phosphor dots on the fluorescent screen which glow, the intensity of the glow being proportional to the intensity of the video signal.

Deflection Stages The phosphor dots glow in quick succession from left to right and top to bottom with the help of scanning currents in the deflection coils. The synchronising pulses, recovered by the detector, trigger the scanning circuits which produce deflection currents, duly synchronised with the scanning currents used in the transmitter. The deflection currents go to the deflection coils to deflect the electron beam horizontally and vertically on the fluorescent screen to reproduce the picture.

Sound Section It consists of a Sound IF (SIF) amplifier, FM detector, audio amplifier and loudspeaker. Difference of frequency between frequency-modulated IF and video carrier IF is called *intercarrier frequency*, or second IF or sound IF. It is received from the video detector and passes to the SIF amplifier though a trap circuit, which prevents SIF signal from going into video amplifier. The FM detector detects the audio signal which is then amplified by audio amplifiers. The amplified signal goes to the loudspeaker which converts it into sound. Thus, the original sound is reproduced.

Colour Transmitter Transmission and reception of a colour picture is more complex than a monochrome picture because of the requirement of compatibility with black and white pictures. The basic features of a compatible colour transmitter are shown in Fig. 13.19.

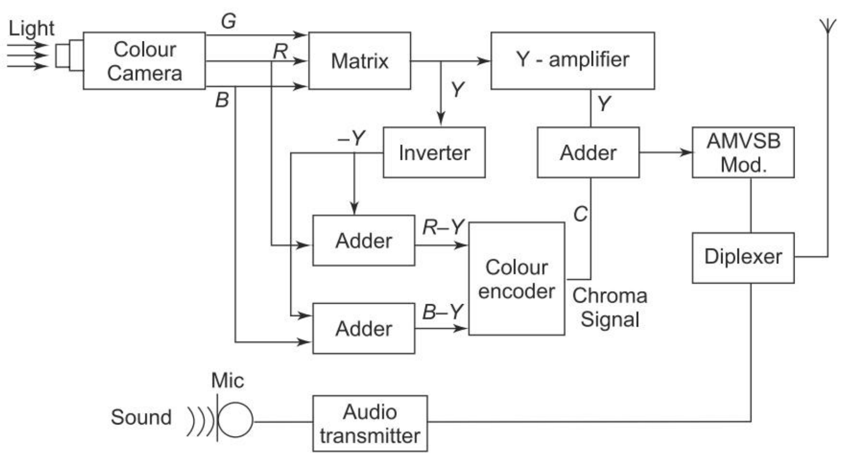


Fig. 13.19 *Basic block of a colour transmitter*

For colour transmission, colours from a picture are separated into three primary colours, red green and blue with the help of prisms or dichroic mirrors and colour filters. The brightness present in each colour is converted by colour camera tubes into three electrical signals.

The three colour signals are passed through a resistive matrix to obtain resultant luminance signal (called Y signal), using Grassman's law. Colour signals are sent in the form of colour-difference signals.

A colour subcarrier (4.43 MHz in European system and 3.58 MHz in American system) is modulated by two weighted colour difference signals (*B–Y* and *R–Y*), using synchronous quadrature modulation. This is called *colour encoder*. The colour difference signals when multiplied by suitable factors to prevent overmodulation, are called weighted colour difference signals. Its output is called *chroma signal* (or *C* signal).

Then luminance signal *Y* and chroma signal *C* are added and modulate the main video carrier signal, using AMVSB technique. This modulated RF carrier is transmitted along with an audio modulated RF carrier through a common transmitting antenna with the help of a diplexer.

Colour Receiver

Basic elements of a colour receiver are shown in Fig. 13.20.

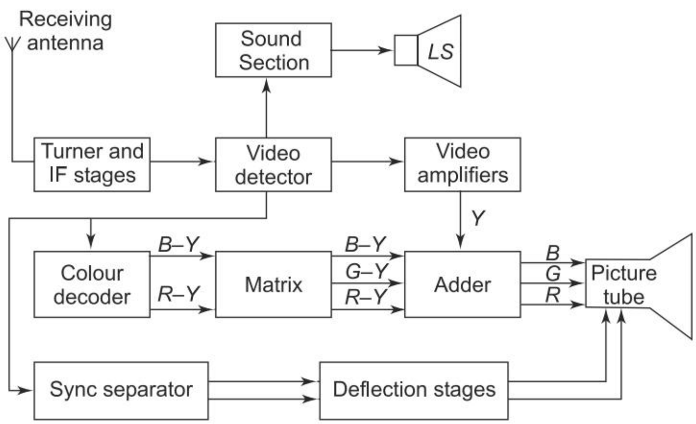


Fig. 13.20 *Basic blocks of a colour receiver*

Colour receiver is a normal superhetrodyne receiver of the monochrome system up to the video detector stage. The detector detects *Y* and *C* signals. In a colour decoder section, the *C* signal is separated by a bandpass filter, (*B–Y*) and (*R–Y*), are decoded, using synchronous quadrature detector. The third signal (*G–Y*) is obtained with the help of a resistive matrix. Then *Y* signal is added to each of the colour-difference signals to get original camera signals pertaining to red, green and blue colours. These three colour signals are converted into coloured lights by making phosphor elements of the colour picture tube glow red, green and blue for every pixel. The eye integrates these primary colours by additive mixing and sees the resultant original colour. All the pixels on the screen are made to glow in quick succession with the help of deflection circuits which produce horizontal and vertical deflection currents for scanning the screen. Thus, the coloured picture is reproduced on the screen of the picture tube.

Sound is reproduced from the loudspeaker in the same manner as in case of a monochrome receiver.

Types of Colour Systems

There are three types colour systems:

1. NTSC (Designed by National Television Systems Committee of USA)
2. PAL (Phase alternation by line, designed in Germany)
3. SECAM (Sequential Colour and Memory, Designed in France)

NTSC In this system, the weighted colour difference signals, called I and Q signals are present on each line simultaneously and maintain a mutual phase difference of +90°. Modulation technique is AMSC. In this system, phase error if any produces a different colour (and for this reason Americans jokingly call their system as 'Never The Same Colour').

PAL In this system, the weighted colour difference signals are called U and V signals. They maintain a phase difference of 90° between each other, but on one scanning line, V is +90° away from U, and on the next line, it is –90° from U. thus, both signals are present simultaneously on each line, but with a different phase difference. This eliminates phase error automatically. Due to requirement of automatic phase changing switch, the PAL system is costlier than NTSC system. Modulation is AMSC type as in NTSC. India uses PAL system.

SECAM In this system, the weighted colour difference signals are called D_B and D_R. They do not appear simultaneously on any line. If D_B appears on one line, then D_R appears on the next line. So, the question of phase difference and hence phase-error does not arise. Modulation technique is FM. It is the cheapest system, but resolution is reduced to 50%. For colours, high resolution is not needed.

13.19 VIDEO MONITORS

A video monitor is a device which receives the baseband video signal at its input amplifier, and displays it on a fluorescent screen. It is different from a TV receiver. A TV receiver gets radio frequency (modulated) signal at its input, processes it in the tuner and IF amplifier stages and recovers the video signal in a detector stage. In the monitor, the video signal is received directly. Thus, a video monitor does not have the tuner, IF and detector stages.

A monitor may either receive the composite video signal from a VCR camera or closed circuit TV (CCTV) camera, or may receive a video signal and sync pulses separately as in a video display unit (VDU) used in computers. A typical video monitor is shown in Fig. 13.21. In VCR or CCTV monitors, video amplifier and deflection stages are the same as in a TV receiver. A computer monitor may have a sound section also (without intercarrier frequency amplifier and FM detector).

A monitor can be designed for high resolution because there is no limit on bandwidth as radio frequency transmission is not needed. Resolution as high as 1024×768 pixels is being used for computer monitors. As there is no constraint about bandwidth, progressive scanning can also be used in monitor systems.

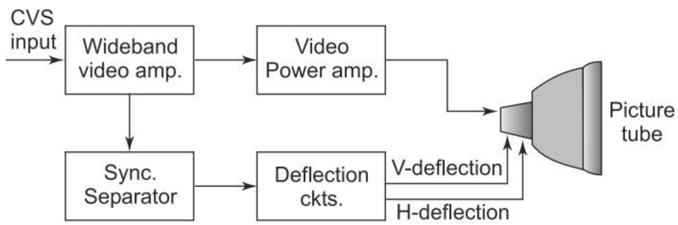


Fig. 13.21 Block diagram of a typical *video monito*

13.20 WIDE-DIMENSION HIGH DEFINITION TV

Wide-screen TV systems in which the size of the screen is 1 metre or more can give real life-size display as on a cinema screen. Such a system is called Wide Dimension Television (WDTV) or High Definition Television (HDTV). The large size gives naturalness to the display and is more enjoyable than the display on the small screen. Wide-screen high-definition TV systems require different techniques, and therefore the present-day 625/525 line systems are designated as conventional TV systems to distinguish them from wide-screen HDTV systems.

Difficulties in Conventional TV Systems The quality of picture appears to be excellent even on the small screen of a conventional TV. However, when the same techniques of scanning and interleaving are used on a wide screen, the defects of the conventional techniques become glaringly apparent. The scanning lines (625 of European system or 525 of American system) will appear distinct, lacking in continuity and blending on a wide dimension screen. The picture will not appear as one whole but will appear as torn into discrete elements and hence annoying. Moreover, the cross-talk between luminance and colour signals which are pictures obtained by using conventional techniques, when presented as a wide dimension display, will appear quite blurred and faded. So, to get high-quality near life-size pictures on a TV screen, the size of the screen should be increased on the one hand, and on the other, different techniques and standards need to be used. The new wide-dimension high definition TV system uses the techniques described in Section 13.20.1.

13.20.1 New Techniques Used For High Definition

The new technique for HDTV, developed in Europe, is based on Time Division Multiplexing (TDM) of two signals, luminance and colour. Unlike a conventional TV, the two signals are not interleaved but are multiplexed on the time axis, as shown in Fig. 13.22.

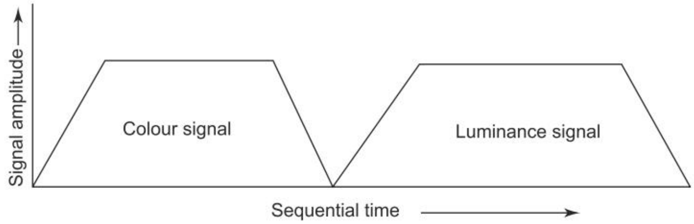


Fig. 13.22 *Time division multiplexing of luminance and colour signals*

The TDM technique eliminates mutual interference between colour and luminance as they appear separate from each other in the time frame. In a conventional TV, they are interleaved and so appear in between each other, i.e., chroma signals lie in between luminance signals and vice versa. Despite favourable energy distribution for two signals for interleaving, a bit of overlapping does occur and causes interference, called *cross-talk*, on the analogy of interference between two sound channels. This cross-talk becomes more apparent when the picture of a conventional TV system is presented on a wide dimension screen.

The TDM technique used in high-definition TV is called Multiplexing Analog Components or MAC technique. A basic MAC encoder is described in Section 13.21.

13.21 MAC ENCODER

The block diagram of a basic MAC encoder is given in Fig. 13.23. Description of its various stages follows:

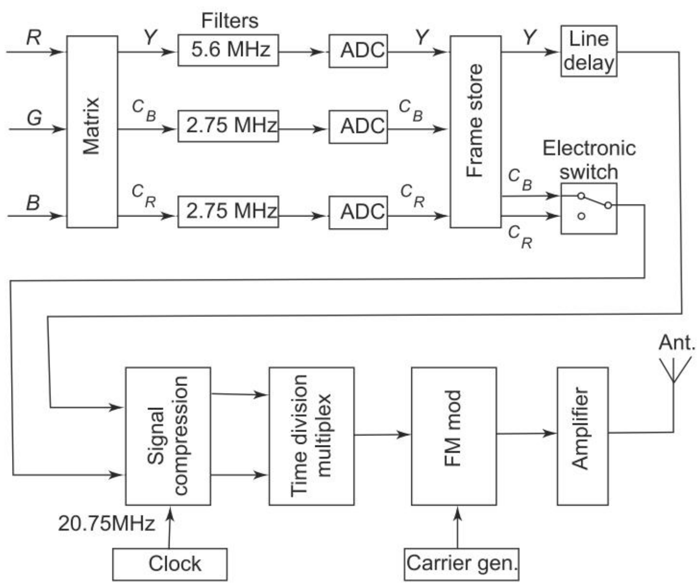


Fig. 13.23 *Block diagram of a basic MAC encoder*

Matrix Video signals, R, G and B, produced by the colour camera tubes represent the intensity of light of three primary colours, red, green and blue, present in each pixel of the picture. These signals are fed to a resistive matrix, incorporating resistor circuits, invertors and adders (as in a conventional TV system) to give luminance signal Y (=0.11 B + 0.30 R + 0.59 G). and duly weighted colour difference signals designated as C_B (for weighed B–Y) and C_R (for weighted R–Y).

Filter These are bandpass filters, allowing bandwidth of 5.6 MHz for Y-signal and 2,75 MHz for C_B and C_R signals. (These bandwidths are different from the bandwidths used on conventional TV system and form part of the new standards for HDTV.)'

Analog to Digital Converter The filtered signals are sampled for digitisation. The minimum sampling rate is equal to twice the maximum bandwidth frequency. The samples are coded as 8-bit codes, producing a word of 8 bits for each sample of the analog waveform taken.

Frame Store Digital pulses are fed to a frame store. The frame store isolates the input and the output and hence synchronisation is not required. Total storage capacity is 20.25 million samples per second (6.75 for colour difference signals and 13.5 for luminance signal.)

Line Delay The luminance signal Y is delayed by one line. This is achieved by using two RAMS, one for storing luminance signal for the current line (the line which is scanning) and the other for the previous line (the line which has just been scanned). This automatically synchronises the sequence of the luminance signal and the chroma signal.

Line Sequential Switch It is an electronic switch which allows C_B signal on odd numbered lines and C_R signal on even numbered lines, as in SECAM system.

Compression Stage The luminance signal, clocked (or sampled) at 13.5 MHz and chrominance signal at 6.75 MHz, are compressed by clocking at a higher frequency of 20.25 MHz. Compression ratio for U and V is 3:1 and for Y, 3:2.

Time Multiplex Switch This is the final stage of a MAC encoder. The Y and C signals are multiplexed so that odd line contains Y and CB signal and the even line, Y and CR signal. As the bits are allowed to pass at 20.25 million bits per second, (13.25 million bits of Y signal and 6.25 million bits of C signals pass in one second). Thus, Y and C bits are divided into time slots.

Frequency Modulator The multiplexed signal modulates a sine-wave carrier, using frequency modulation and thus, we get a modulated signal duly multiplexed for Y and C. FM makes the system almost immune to noise.

Final Power Amplifier It finally amplifies the power and delivers it to the transmitting antenna.

13.22 MAC RECEIVER AND DECODER

In the receiver, the information is received as usual and detected by the FM detector. The detector output consists of words of 8 bits, which are demultiplexed to give Y and C words in separate slots of time. These words are then expanded to offset the effect of compression which was done in the encoder. The result of expansion is that Y + C_B signal occupies one full line and Y + C_R signal another full line. The demultiplexed signals are further combined in a matrix to recover R, G and B signals which are fed to the picture tube to reproduce the original picture. The receiving process is shown in the block diagram given in Fig. 13.24.

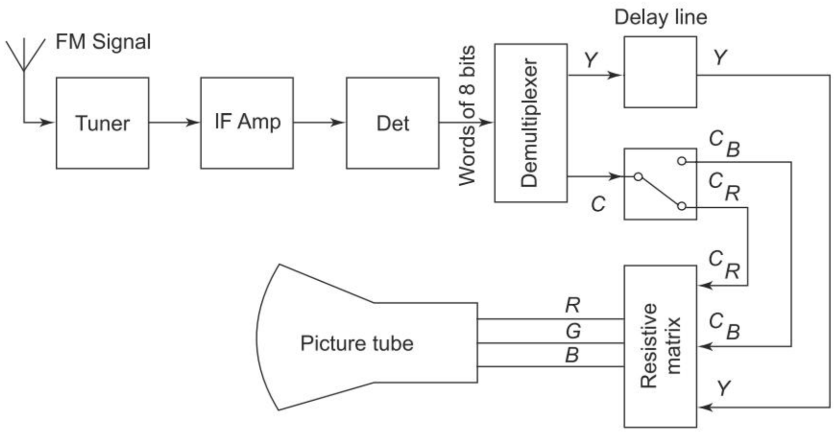


Fig. 13.24 *Block diagram of receiving system*

From the above, it is quite obvious that the main function of a MAC system is to eliminate the need of interleaving of chroma signals in between luminance signals within a common frequency spectrum, and to provide two separate independent locations for the signals. This eliminates cross-talk interference and allows-wide screen to be used.

The European union has proposed 1250 lines/50 fields/progressive scanning for the new HDTV system. Japan developed a somewhat different system of HDTV which has been adopted by USA, Canada and many other countries. This system offers 1125 lines/60 fields/interlaced scanning (2:1 ratio) for the new system. It also uses a technique different from MAC technique which is known as Multiple Sub-nyquist Encoding (MUSE). It is based on compression of the bandwidth. Although everybody wishes a single world standard for HDTV, but the proponents of the two systems consider their respective systems to be better than the other one. It is hoped that CCIR (a wing of UNO) would be able to take a decision to finalise a single world standard for wide dimension HDTV.

Whatever be the final outcome, Federal Communications Commission (FCC) of USA has decided that in view of many advantages offered by digital TV, the television broadcast in USA would be digital only. This will require replacement or modification of existing receivers, and hence a firm decision in this respect taken in advance will be helpful to the people.

Advantages of MAC System

1. No interference of cross-talk type between luminance and chroma signals.
2. Wide screen with 16:9 aspect ratio can be used which gives cinema like effect.
3. Resolution is very good.
4. Picture is very sharp and bright.
5. Fidelity of picture is very good.
6. It does not require synchronisation.
7. It can be used as a single world standard.
8. Final transmitted signal being frequency modulated, the system is immune to noise and signal-to-noise ratio is very high. Whatever noise is introduced, it is further eliminated when we recover the signal by FM detector in the form of binary pulses. These pulses can be easily reconditioned to become clean of any noise.

Principles of high-definition TV have been well established and the minor problems here and there are being looked into. Sooner than later, the people would be able to enjoy near life-size pictures of high resolution and free of distortion. The high-definition WDTV would become a normal item in homes in the near future the world over.

13.23 DIRECT-TO-HOME (DTH) TV

When TV programmes relayed by a satellite can be received direct by domestic TV receivers in homes, the system is called DTH TV. This system has been explained in Appendix III.

13.24 CABLE TELEVISION (CATV)

Instead of Direct to Home TV, cable television service providers receive the TV signals and then send then to the homes. Such a system is called CATV system. This is described in Appendix IV.

13.25 DIGITAL TV SYSTEM

The future TV may become wholly digital in which picture information will be sent as a data stream of bits. What would be sent might be a binary message that the picture element '*x*' has brightness level '*y*' and colour '*z*'. At the receiving end, the binary message would be decoded for each picture element which would be rebuilt by glowing liquid crystal diodes on a flat plastic screen according to the brightness and colour level contained in the transmitted message.

SUMMARY

- ☞ Transmission of TV signals through a single channel requires scanning of the image on the photosensitive target plate of a camera tube from left to right and top to bottom. Reception of these signals require identical scanning and hence the need of transmission of control pulses (sync pulses) along with video signal and their extraction in the receiver to ensure identicalness in the scanning.
- ☞ Interlaced scanning is used to reduce flicker. Two sequences of scanning in interlaced scanning are called fields. Number of fields per second is equal to the frequency of electric mains to eliminate moving hum bar. It results in different scanning lines per frame in different countries. This decides the standards of resolution and bandwidth for a TV system. Hence, every TV transmission and reception system has a set of specifications (called TV standards).
- ☞ For conversion of brightness and colour into electrical signals, photosensitive devices are used.
- ☞ The system using such devices is called a camera tube. Vidicon and Plumbicon camera tubes are most widely used. Solid state cameras have also been developed.
- ☞ For colour, we may use three camera tubes, one for each primary colour. Also, a single-tube camera has been developed for recording colour signals.
- ☞ Video signals obtained from camera tubes modulate a radio frequency carrier which is transmitted. At the receiving end, a superheterodyne receiver receives these signals and recovers video information which reproduces picture on the fluorescent screen of a picture tube.
- ☞ Monitors used in VCR cameras, CCTVs and computers do not require RF, IF and detector stages, but receive video signals directly at the input.
- ☞ High-definition WDTV is capable of giving a cinema like effect in viewing. Here, new techniques of time division multiplexing of the luminance and chrominance signals are used.
- ☞ Direct-to-home TV from satellite and from Cable Service providers (CATV) are widely used these days.

Review Questions

1 What do you understand by scanning? Why is it needed? What is flicker? How is Flicker removed?
2. Write short notes on the following:
   (a) Picture frame
   (b) Aspect ratio
   (c) Pixel
3. What do you understand by (a) active scanning lines, (b) Kell factor, and (c) picture details?
4. Explain resolution and bandwidth and show that both are related to each other. Give their typical values for the CCIR-B system and the CCIR-M system.
5. Explain the principle of working of the Vidicon camera tube.
6. Describe the construction and working principle of the Plumbicon camera tube.
7. Describe the working of a solid state video camera and discuss its advantages (Appendix)
8. Explain how you can obtain a video signal of a coloured picture.
9. Explain the working of a monochrome TV system.
10. Explain the working of a colour TV transmitter and receiver.
11. Explain the differences between NTSC, PAL and SECAM Systems.
12. Write short notes on video monitor.
13. Describe a picture tube for colour monitors. How does it differ from a monochrome monitor?
14. Write a note on 'Plasma TV' and 'LCD TV'.
15. Draw CVS for two lines of scanning and show all pulse widths and amplitude levels as used in India.
16. Draw V-blanking pulse and its constituent after the end of an odd field.
17. Describe MAC encoder and decoder with the help of a block diagram.
18. Write a short note on 'digital TV system'.
19. Explain DTH TV system. (Appendix)
20. Discuss Cable TV (Appendix)
21. What do you understand by Scrambling and Descrambling?

Short-Answer Questions

1. Why is picture frame frequency of 25 frames per second in India and 30 per second in USA?
2. Why is flicker not removed by progressive scanning?
3. Why is the number of scanning lines per frame always odd?
4. Why is the vertical deflection very slow as compared to the horizontal deflection?
5. Why are sync pulses needed in TV systems?
6. Why is retrace blanked?
7. Why is AMVSB preferred to FM for transmission of video signals?
8. Why is SECAM system called sequential?
9. What would happen if the conventional TV transmission using 625

or 525 lines per frame is viewed on a 1-metre wide screen?

10. How is the problem of cross-talk interference removed in HDTV?
11. What is the difference between a TV receiver and a video monitor?
12. Why microwave signals of DTH TV are converted into UHF signals after the LNA? (Appendix)
13. How are pictures distorted to prevent them from being received normally in TV receivers not subscribing to pay-TV? (Appendix)
14. What is done to reduce adjacent channel interference in CATV system when a large number of channels are passing through the same cable? (Appendix)

Multiple-Choice Questions

1. Aspect ratio for width to height for a TV picture frame is
   (a) 1:1 (b) 2:1
   (c) 4:3 (d) 5:4
2. Which one is not a transducer?
   (a) Camera tube
   (b) Microphone
   (c) Transformer
   (d) Picture tube
3. What is the number of fields per second in India?
   (a) 25 (b) 50
   (c) 625 (d) 15625
4. What is the name given to video signal below 10% of the carrier in CCIR-B system?
   (a) Blacker than black
   (b) Black
   (c) White
   (d) Whiter than white
5. How is flicker removed?
   (a) Fast progressive scanning
   (b) Slow scanning
   (c) Interlaced scanning
   (d) increasing aspect ratio
6. How many scanning lines are lost in each vertical scanning in the Indian TV system?
   (a) 50 (b) 30
   (c) 20 (d) 21
7. What is the relation between bandwidth and resolution?
   (a) Lower the bandwidth, higher is the resolution
   (b) Higher the bandwidth, higher is the resolution
   (c) Higher the bandwidth, lower is the resolution
   (d) Bandwidth is not related to resolution.
8. A composite video signal consists of
   (a) sync pulses
   (b) blanking pulses
   (c) picture information
   (d) all given in a, b and c
9. Compatibility means
   (a) monochrome receiver should be able to receive colour transmissions
   (b) colour receiver should be able to receive monochrome transmissions
   (c) both (a) and (b) should be true
   (d) neither (a) nor (b) should occur

10. Which colour system does not have both the colour difference signals on each line?
    (a) SECAM (b) PAL
    (c) NTSC (d) D-PAL

11. The MAC system of WDTV was developed in
    (a) India (b) Japan
    (c) USA (d) UK

Numerical Problems

1. If aspect ratio is 4:3 in a conventional TV and 16:9 in an HDTV, calculate width and height for 100 cm size screen for both TV system.
2. Calculate number of lines (n) per frame to subtend an angle of 1 minute $\left(\text{by a space } S = \frac{n}{h}\right)$ at distance of 4 times the height h of screen.
3. Calculate the bandwidth for a TV system, having the following specifications. Lines per frame = 625, frame frequency = 25 Hz, retrace time for each line = 12.5 µs, time of vertical blanking for each field = 1280 µs and Kell factor = 0.72
4. Calculate vertical resolution of a TV system, having 5.5 MHz bandwidth and 52 µs trace time.
5. Calculate the bandwidth for a system in which R_h = 520 and t = 52 µs (CCIR-B system)
6. Calculate the bandwidth for a system in which R_h = 450 and t = 53.5 µs (CCIR-M system)
7. In CCIR-B system, total number of scanning lines per frame is 625, and the lines lost per field is 20. calculate vertical and horizontal resolutions, presuming Kell factor equal to 0.7.
8. In CCIR-M system, N_T = 525, N_L = 42. Calculate R_V and R_H
9. Calculate the minimum distance between adjacent pixels for the viewing distance equal to 2.5 m
10. Calculate the number of pixels in 50 cm size TV screen for question no. 9 above.

Answers

Short-Answer Questions

1. Picture frame, frequency is one-half of the electric mains frequency. Hence, it is 25 in India and 30 in USA
2. Progressive scanning occurs at 25 frames per second, and hence there are 25 blankings per second. This is inadequate for removing flicker. If we increase the frequency to 50, bandwidth will increase which is not desirable.
3. Number of scanning lines for interlaced scanning is equal to (n + ½) × 2 = 2n + 1 which is always an odd number.

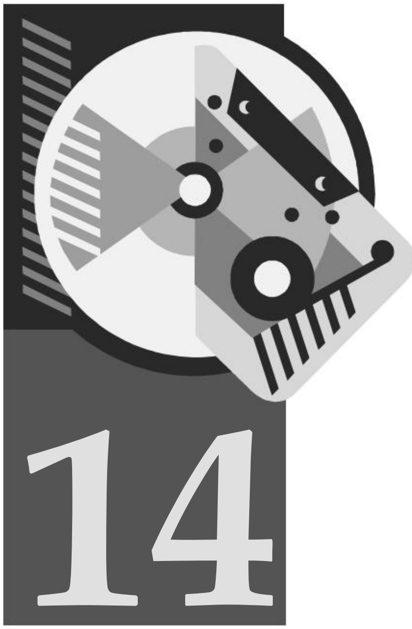
4. To allow maximum number of pixels to be scanned.
5. To get identical scanning sequence in receiver with that used in the transmitter.
6. If retrace is visible on the screen, it will cause distortion in the picture.
7. AMVSB is preferred for video signals because on the one hand it saves bandwidth, and on the other its detection is simple, keeping the receiver's cost low. FM requires very high bandwidth, and moreover FM detector can detect random changes in phase which will cause noise in the picture.
8. SECAM system is called sequential because both colour-difference signals do not appear simultaneously in each line.
9. Scanning lines of conventional TV (625 or 525 per frame) will appear distinct, lacking in continuity and blending when seen on a wide-dimension screen. Thus, the picture will appear as torn into discrete elements, and hence annoying.
10. In HDTV, chroma signal is not interlaced in between luminance signals, but occupies separate portion in time-frame (using TDM).
11. A TV receiver uses an amplifier, mixer, IF amplifier, and detector. In a Monitor, all these stages are not used. The video signal is directly fed to a wideband video amplifier and then to the picture tube.
12. Microwave signals are converted into UHF at the earth's receiving station, so that a cable may be used in the outdoor from LNA to the receiver. This would be cost effective.
13. Pictures in Pay TV system are distorted by removing sync pulses.
14. Guard band is used between channels as in actual radio telecast system. All channels are made of equal strength. TV receiver's tuner and IF tuned circuits reject the adjacent signals. Had the adjacent channels been stronger, adequate rejection would not have been possible.

Multiple-Choice Questions

1. (c)	2. (c)	3. (b)	4. (d)
5. (c)	6. (c)	7. (b)	8. (d)
9. (c)	10. (a)	11. (d)	

Numerical Questions

1. (80, 60 and 87, 49)
2. (860) 3. (5.45 MHz)
4. (429 lines) 5. 15 m Hz
6. 4.2 m Hz 7. (409, 546)
9. (00.073 cm) 8.(338,451)
10. 2, 25, 183)



14 Television Standards

14.1 DEFINITION

Television standards consist of a set of specifications for transmission of TV programmes. Broadly speaking, these specifications pertain to the scanning process, colour encoding system, composite video signal, modulation methods, bandwidth, intercarrier frequency, channel width and radio frequency spectrum utilisation, etc.

14.2 NEED OF A TV STANDARD

A TV receiver cannot receive TV programmes unless it conforms to the specifications adopted for transmission. For example, a TV receiver needs scanning identical and synchronised to the scanning process that was used in the transmitter. Similarly, for detection of colour signals, the same subcarrier is required to be generated in the receiver as was employed in the transmitter. Due to this essential requirement, synchronisation pulses and colour-burst signals are transmitted along with the video signal. These are separated out in the receiver to trigger the oscillations and hence to produce identical scanning currents and colour subcarrier. Further, different colour- encoding processes adopted for transmission require compatible colour-decoding processes in the receiver.

Different countries adopt different standards. Hence, the receivers in a country are designed to receive TV transmissions of the specifications followed by that country. Even a minor difference in the transmitter specifications of a country may not allow receivers in some other countries to receive TV programmes of that country. On account of this, the TV signals have to be specified exactly and hence the need of TV standards.

14.3 REASONS OF TV STANDARDS BEING DIFFERENT IN DIFFERENT COUNTRIES

The different standards of TV transmission are not compatible with each other. No American set can work in Europe and vice versa. Even in Europe, a set used in Germany cannot be used across the border in France. A receiver used in the Indian subcontinent will not work in Myanmar and Japan, and a receiver in England will not work across the English channel in France. Non-compatibility of various standards is the biggest hurdle in international exchange of programmes. A single world standard of TV would have been desirable. Such a world standard exists for sound broadcast as any sound programme can be received anywhere in the world (within the communication range) simply by tuning into the desired station. The need of a single world standard for TV is being felt all the more due to the development of satellite communication. Hence, there must be some compelling reasons for TV standards being different in different countries. These are discussed below:

1. Field frequency in the scanning process is related to the electric mains frequency. These two should be the same so that hum due to power-supply ripples may not cause moving bars (called hum-bars) on the screen. The moving hum-bar is caused by beats between fields frequency and mains frequency.

   Electric mains frequency in American countries is 60 Hz. In the countries of the Indian subcontinent and other continents, it is 50 Hz. Hence, American countries follow a 60 Hz field system; the countries of Asia, Europe, Australia and Africa keep the field frequency equal to 50 Hz. (A few countries are exception to this rule, e.g., Japan and Myanmar, which use American system of 60 fields per second, although their mains frequency is 50 Hz.)
2. Field frequency is also related to brightness. Higher the field frequency, greater can be the brightness of a TV picture on the screen for the same level of reduction in flicker. Greater the brightness, larger can be the viewing distance, and the larger the viewing distance, smaller can be the number of scanning lines per frame. Thus, in the countries where the field frequency is 60 Hz, the number of scanning lines per frame is 525. In Asian and European countries where the field frequency is 50 Hz, the number of scanning lines is 625 per frame.
3. USA, Germany and France developed different-colour encoding and decoding systems, known as NTSC, PAL and SECAM, respectively. NTSC system, developed first, was compatible with the existing monochrome system and was quite acceptable, and therefore millions of TV receivers were sold to the public. However, there was a minor defect (change of original colour due to phase noise) in the system and a hue control had to be provided to overcome that defect. With a view to solve this problem, Germany developed the PAL system, which was

more costly. France developed another system, SECAM, which was of low cost but its resolution was only 50% of NTSC. On account of merits and demerits of these systems and considering the existing monochrome system, countries all over the world chose one of these three systems.

4. Some countries preferred minor changes in channel width, bandwidth, inter-carrier frequency, etc., as their scientists felt that the minor changes would result in improvement of picture quality, but mainly because of the political reasons wherein a government did not want its people to see programmes of other countries to prevent its own culture from getting influenced. The TV system of UK is an example of this. Although UK follows the 625 line PAL system, minor changes were incorporated in their specifications to make programmes of other countries inaccessible to their people.

14.4 DIFFERENT TV STANDARDS

In view of the several existing standards, CCIR (Committee Consultative for International Radio) could not decide on a single world standard, but listed some existing standards, leaving the choice to the countries to choose any of them. TV systems, standardised by CCIR, are designated as A, B, C, D, E, F, G, H, I, K, L, M and N.

Out of the above designated systems, A, C, E and F have not been recommended for future use. A and E systems have already been discontinued by UK and France, respectively. These presented some difficulties. The system 'A' was of 405 scanning lines and hence resolution was poor. Moreover, it used positive modulation for AMVSB in which noise occurred in the white region, which was more spectacular. C, E and F systems also used positive modulation. They had 819 lines per frame, requiring high bandwidth. G, H, I, K and L systems are used in UHF bands IV and V. M and N systems are used in VHF and UHF both. B and D systems are used in VHF bands, I and III. CCIR standard B (developed by Germany) and CCIR M (developed by USA), are popularly known as 625-line European system and 525-line American system, respectively. ***India has adopted CCIR B standard***. Detailed specifications of these two standards are given in Table 14.1.

Table 14.1 *Specifications of CCIR standards B and M*

Item **1**	*CCIR B* 2	*CCIR M* 3
Video scanning standards:		
Number of scanning lines per frame	625	525

(Contd.)

(Contd.)

Trace time per line	52 µs	53.3 µs
Horizontal blanking period	12 µs	10.2 µs
Total scanning time of one line	64 µs	63.5 µs
Number of lines lost in vertical retrace	20 per field	21 per field
Interlace ratio	2:1	2:1
Aspect ratio	4:3	4:3
Horizontal blanking pulse		
Front porch	1.5 µs	1.2 µs
H-sync pulse	4.7 µs	4.75 µs
Back porch	5.8 µs	3.81 µs
Colour-burst signal	8 to 10 cycles of subcarrier frequency, swinging between ± 45° about–(B–Y) signal	8 to 10 cycles of subcarrier frequency without swinging
Time of rise and fall of edges of pulses	0.3 µs	0.3 µs
Vertical blanking pulse		
Total duration	1280 µs	1333.5 µs
Front portion	160 µs	190.5 µs
Back portion	960 µs	952.5 µs
V-sync pulse	160 µs	190.5 µs
Pre-equalising pulses	5 pulses each of 2.35 µs width	6 pulses each of 2.54 µs width
Post-equalising pulses	5 pulses each of 2.35 µs width	6 pulses each of 2.54 µs width
Serrations (slots)	5	6
Width of each slot	4.7 µs	4.4 µs
Width of each serrated pulse	27.3 µs	27.35 µs
Amplitude of baseband components in terms of percentage of carrier amplitude:		
Sync top	100 %	100%
Blanking pedestal	75%	75%
Black level	72–75%	67.5%
White level	10%	12.5%

(Contd.)

(Contd.)

Spectrum:		
Bandwidth of video baseband signal	5 MHz	4.2 MHz
Location of video carrier	After 1.25 MHz of the start of the channel	After 1.25 MHz of the start of the channel
Colour sub-carrier	4.43 MHz	3.58 MHz
Chroma signal bandwidth (with respect to colour sub carrier)	–1.3 MHz to + 0.57 MHz (for both colour-difference signals, U and V)	±0.5 MHzfor q-signal and –1.5 MHz for Q-signal and –1.5 MHz to +0.5 MHz for I-signal
Total channel width in VHF	7 MHz.	6 MHz.
Total channel width in UHF	8 MHz. (in CCIR C)	6 MHz.
Video IF	38.9 MHz	45.75 MHz
Audio IF	33.4 MHz	41.25 MHz
Intercarrier frequency (SIF) Audio carrier	5.5 MHz before 0.25 MHz aof the end of channel.	4.5 MHz before 0.25 MHz of the end of channel.
Modulatation:		
Audio modulation	FM	FM
Frequency deviation for sound	± 25 kHz	± 50 kHz
Video modulation	AMDSB (VSB)	AMDSB (VSB)
Modulation for chroma	AMSC (VSB)	AMSC (VSB)

14.5 SIMILARITIES AND VARIATIONS IN STANDARDS

CCIR G is identical with CCIR B except that G is used in the UHF band while B is used in the VHF band, and that picture-to-sound power ratio in G is 10:1 against 5:1 in B. Similarly, D and K are identical with each other except that D is used in VHF and K in UHF. N is identical with M except that N uses the PAL colour system instead of NTSC system. D, H and I PAL are identical with B for most points except a few given in Table 14.2. L is a 625-lines European system and is similar to CCIR B in many respects but uses SECAM colour transmission and reception. Its distinguishing features have also been shown in Table 14.2.

Table 14.2 *Some variations in B, D, H, I and L standards*

ITEM	*B*	*D*	*H*	*I*	*L*
Video bandwidth in MHz	5	6	5	5.5	6
Lower sideband in MHz attenuated beyond	–0.75	–0.75	–1.25	–1.25	–1.25

(Contd.)

(Contd.)

Audio IF in MHz	33.4	32.4	33.4	32.9	32.4
Intercarrier frequency in MHz	5.5	6.5	5.5	6	6.5
Channel width in MHz	7	8	8	8	8
Video modulation polarity	–ve	–ve	–ve	–ve	+ve
Audio modulation	FM	FM	FM	FM	AM
Picture to-sound power ratio	5:1	10:1	10:1	10:1	10:1

14.6 INTERNATIONAL USE OF TV STANDARDS

The countries world over were free to use any of the three colour systems: NTSC, PAL and SECAM. American countries in general followed 525 lines per frame and 60 fields per second and the NTSC colour system. Japan and Myanmar in Asia also followed the American system. All other countries followed 625 lines per frame, 50 fields per second system, and for colour either PAL system (developed by Germany) or SECAM system (developed by France). PAL system was adopted by all countries of West Europe, UK, Australia and all countries of the Indian subcontinent. The SECAM system was adopted by Russia and countries of East Europe. Names of some countries, using various scanning standards and colour systems are given in Table 14.3.

Table 14.3 *International use of CCIR TV Standards*

CCIR designated Standards	Names of some countries using the standard
B (PAL)	Germany (the inventor), countries of Indian subcontinent, Australia, Italy
B/G (SECAM)	Egypt, Saudi Arabia, Iran, Iraq
D (PAL)	China
D and K (PAL)	North Korea, Romania
D and K (SECAM)	Commonwealth of Independent States (formerly USSR), Afghanistan, Hungary
H (SECAM)	Belgium
I (PAL)	UK, Hong Kong, South Africa
K and L (SECAM)	France (the inventor), Monaco
M (NTSC)	USA (the inventory), Canada, Japan, Myanmar, S. Korea
M (PAL)	Brazil
N (PAL)	Argentina

The aspect ratio of 16:9 has been accepted in the Japanese MUSE system as well as in European MAC system. The other standards for wide dimension HDTV are still under debate but are as follows at present in USA.

* Total lines per frame 1125
* Active lines 1035

*	Field frequency	60
*	Interlace ratio	2:1
*	Line frequency	33 750 Hz
*	Line time	29.63 μs base
*	H-retrace time	3.7 μs
*	V-retrace	45 lines per field
*	Video bandwidth	24.9 MHz
*	Luminance bandwidth	20 MHz
*	Wideband colour signal	7 MHz
*	Narrowband colour signal	5,5 MHz
*	Samples per active line	1920 for luminance signal 960 for colour difference signals
*	Modulation	FM
*	CNR	12 dB. This allows picture quality of grade 4 of CCIR scale.

Principles of high-definition TV have been well established and the minor problems here and there are being looked into. Sooner than later, the people would be able to enjoy near life-size pictures of high resolution and free of distortion. The high-definition WDTV would become a normal item in homes in the near future the world over.

SUMMARY

☞ Television standards consist of specifications for transmission of TV programmes. A television receiver must conform to these specifications to enable it to receive the programmes.

☞ Different countries adopted different standards based on electric mains frequency, colour systems developed in USA (NTSC), Germany (PAL) and France (SECAM) and on some minor changes made by their scientists for improvement, or changes made by government for political reasons to save one's culture from being influenced.

☞ Different TV standards as per CCIR designations are A, B, C, D, E, F, G, H, I, K, L, M, and N. Out of these, A, C, E and F have not been recommended for future use and are being phased out.

☞ Some of these, although differently designated have close resemblance. For example, the system G is identical with the system B except that its frequency range is in the upper UHF band, video–audio power ratio is 10:1, instead of 5:1 and channel width is 8 MHz instead of 7 MHz.

☞ Similar is the case with systems D and K. System I, adopted by UK, has only minor variations with respect to the system B. India has adopted CCIR B/G system; USA the CCIR M system, and France CCIR-L system.

Review Questions

1. What do you understand by TV standards? Why are these standards different in different countries?
2. Compare TV standards used in India, England, France and USA with respect to scanning specifications, video bandwidth, video modulation, audio modulation, intercarrier frequency, colour system, subcarrier and chroma modulation.
3. Give detailed specifications of CCIR-B standards with respect to scanning, blanking pulses, CVS amplitudes, bandwidths and IFs.
4. What do you understand by NTSC, PAL and SECAM systems? Where were they developed? Name three other countries which use each of these systems.
5. What type of modulations are used to get chroma signal in NTSC, PAL and SECAM systems? Write down the chroma-system bandwidth used in CCIR-G and CCIR-M system.
6. Draw complete radio frequency spectrum for NTSC system for the TV channel starting from 174 MHz.
7. Draw complete RF spectrum for 5th channel of CCIR B system used in India.

Short-Answer Questions

1. Why did USA keep frame frequency equal to 30 frames per second?
2. Why are the pictures in USA TVs brighter than the pictures in European TVs?
3. Name two specifications which are common to all TV standards (non-HDTV).
4. Why is the bandwidth of colour signals lower than luminance signals?
5. Which TV standard is followed in UK and in what respects does it differ from the system followed in India?
6. What would happen if there are 3 or 4 fields per frame instead of two?

Multiple-Choice Questions

1. TV system used in India is
   (a) CCIR–B (b) CCIR–I
   (c) CCIR–L (d) CCIR–M
2. Electric mains frequency and field frequency are in general equal. This statement is
   (a) True (b) false
3. American TV pictures are brighter than UK TV pictures because of
   (a) higher mains voltage
   (b) lower mains current

(c) Higher mains frequency

(d) higher price

4. CCIR standards which have not been recommended for future use are CCIR

   (a) A, C, E, F (b) B, D, G, H

   (c) I, K, L, N (d) B, G, M, N

5. Lines per frame used in the European system

   (a) 110 (b) 220

   (c) 525 (d) 625

6. Which TV system is used by Russia?

   (a) NTSC (b) SPAL

   (c) DPAL (d) SECAM

7. Duration of the Vertical blanking pulse used in India is

   (a) 64 µs (b) 100 µs

   (c) 160 µs (d) 1280 µs

8. Intercarrier frequency used in USA is

   (a) 6.5 MHz (b) 5.5 MHz

   (c) 5 MHz (d) 4.5 MHz

9. What is the width of V-sync pulse in India

   (a) 4.7 µs (b) 64 µs

   (c) 160 µs (d) 1280 µs

10. LSB and USB extend up to (i) _______ MHz and (ii) _______ MHz, respectively in chroma modulation in India.

    (a) (i) 1.3, (ii) 0.57

    (b) (i) 1.0, (ii) 0.5

    (c) (i) 0.5, (ii) 1.3

    (d) (i) 1.0 (ii) 0.57

11. In 8 MHz channel width in UHF band, the various carriers occupy the same relative positions as in 7 MHz channel width of VHF band, and there is an extra guard band of 1 MHz at the end of the channel. This statement is.

    (a) True (b) false

Numerical Problems

1. Determine LSB, USB, video carrier, subcarrier and audio carrier frequency for the fourth channel of TV used in India.

2. Determine chroma bandwidth for the 3rd channel of TV in India.

Answers

Short-Answer Questions

1. To match it with half of the electric mains frequency.

2. Field frequency being higher (60 Hz), flicker is removed at higher brightness than in European systems where the field frequency is 50 Hz.

3. Aspect ratio is 4:3 and interlace ratio is 2:1 in all TV systems.

4. Eyes cannot see colour in very small objects and hence resolution for colour is low and therefore bandwidth for colour is also low.

5. The CCIR-I standard is followed in UK. It differs from the CCIR-B system in respect of the parameters given in the following table.

ITEM	CCIR-B	CCIR-I
Video bandwidth	5 MHz	5.5 MHz
LSB attenuated beyond	–0.75 MHz	–1.25 MHz
Intercarrier frequency	5.5 MHz	6 MHz
Channel width	7 MHz	8 MHz
Picture to sound power ratio	5:1	10:1

6. Theoretically, reduced bandwidth would be required to maintain brightness at the level of 2 fields per frame, but the picture will appear to be vibrating up and down.

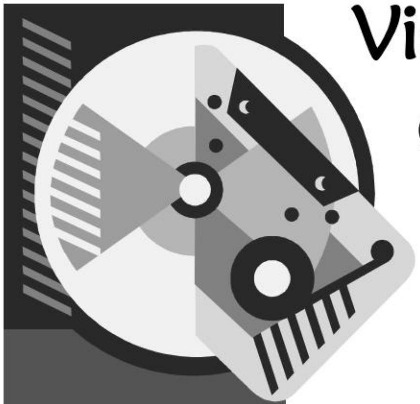
Multiple-Choice Questions

1. (a) 2. (a) 3. (c)
4. (a) 5. (d) 6. (d)
7. (d) 8. (d) 9. (c)
10. (a) 11. (a)

Numerical Questions

1. 4^{th} Channel of TV in India is 61 to 68 MHz. Hence
   LSB = 61.5 to 62.25 MHz
   USB = 62.25 to 67.25 MHz
   Video carrier = 62.25 MHz
   Subcarrier of 4.43 MHz (located at 66.68 MHz)
   Audio carrier = 67.75 MHz

2. 3^{rd} channel of TV in India is 54 to 61 MHz
   Location of colour sub-carrier = 59.68 MHz
   Cut-off frequency of USB of chroma signals = 59.68 + 0.57 = 60.25 MHz
   Cut off frequency of LSB of chroma signals = 59.68 – 1.3 = 58.38 MHz
   Bandwidth of chroma signal
   = 58.38 – 60.25 = 1.87 MHz



15 Video Recording on Magnetic Tape

15.1 INTRODUCTION

A camera tube converts brightness and colour of a picture into electrical signals, called video signals. There are two methods for recording of video signals: (1) recording on magnetic tape, and (2) recording on disc. Recording on optical disc is described in Chapter 16. Recording on tape is done in the form of tiny magnets. Reproduction of signals from the magnetised tape is done using the principle of electromagnetic induction.

15.2 PRINCIPLE OF VIDEO RECORDING

When electrical signals consisting of alternating electric current pass through an electromagnet coil, magnetic flux is produced in the soft-iron core of the electromagnet. The core has a small air gap of high reluctance of the size of 0.5 μm. When a tape with a coating of magnetic material of low reluctance is pressed against the gap, magnetic flux passes through the magnetic material of the tape. The material (iron oxide) gets magnetised. The tape moves past the gap at a uniform speed, and hence the video signal is converted into magnetism on the tape. The variation of magnetism on the tape is in accordance with the variation in the video signals, and the length of tiny magnets formed depends on the speed of the tape as shown in Fig. 15.1.

15.3 RELATIONSHIP BETWEEN TAPE SPEED AND BANDWIDTH

If S is the speed of the tape, and f, the frequency in Hz then the distance λ covered in one time period, T, is given by Eq. 15.1.

$$\lambda = S.T$$

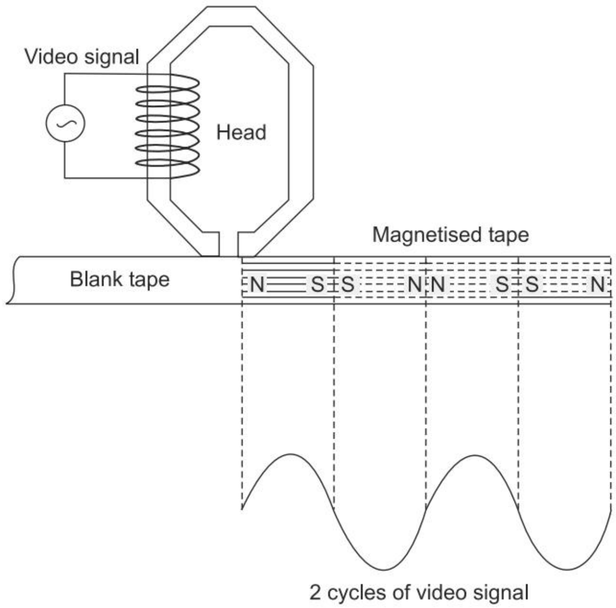


Fig. 15.1 Magnetisation of tape with video signal

as $$T = \frac{1}{f}$$

so $$\lambda = \frac{S}{f} \tag{15.1}$$

Hence, if G is the gap width, it should be equal to $\lambda/2$ for the highest frequency, f_m, to be reproduced. Therefore,

$$G = \frac{\lambda}{2} = \frac{S}{2f_m}$$

or $$f_m = \frac{S}{2G} \tag{15.2}$$

As video signals vary from dc to the highest video frequency, f_m, the difference between the highest frequency (f_m) and the lowest frequency (zero), called the video bandwidth, is equal to f_m. Thus, Eq. 15.2 gives the relationship between tape speed and bandwidth.

Equation (15.2) indicates that for normal speeds of tape, the gap width for video signals works out to be too small, in the range of nanometres as would be clear from Example 15.1.

Example 15.1

Let video bandwidth = 5 MHz

Let speed of tape = 19 cm/s = 0.19 m/s

$$\text{Then, } G = \frac{S}{2f_m}$$

$$= \frac{0.19}{2 \times 5 \times 10^6} \text{ metre}$$

$$= 0.19 \times 10^{-7} \text{ m}$$

$$= 0.19 \times 10^{-7} \times 10^9 \text{ nm} = 19 \text{ nm}$$

(such a small gap size is not feasible)

For gap size to be feasible, say 1 μm, speed of tape required would be

$$= 1 \times 10^{-6} \times 2 \times 5 \times 10^6 = 10 \text{ m/s}$$

Thus, when the gap size is of feasible dimension, the tape speed becomes too high.

15.4 PROBLEMS IN VIDEO RECORDING ON TAPE

Video recording on magnetic tape poses four problems mentioned below:

1. High video frequency of 5 MHz requires too high speed of the tape with the smallest feasible gap size.
2. High bandwidth of luminance signal covers about 19 octaves. No recording system is capable of recording such a wide frequency range.
3. The duration of video programmes is of several hours (particularly, popular programmes of films).
4. Three signals, one of luminance and the other two of colour difference, are to be recorded on the same track.

The above problems have been solved ingeniously as discussed below in Sections 15.5 to 15.8.

15.5 ROTATING HEADS

The practical solution to the problem of gap size and tape speed is to use rotating heads. The absolute speed of the tape is not important. What is important is the relative speed of the tape with respect to the head. Hence the technique of rotating the head will make head-to-tape relative speed quite high even though the longitudinal tape speed may be quite low. Thus, we can have a long recording time for a normal tape length and at the same time, the feasible gap size will be equal to $\lambda/2$ or less for the highest video frequency (of 5 MHz). (The typical value of the gap size is about 0.5 micron for video recording). Although one rotating head could suffice to increase the relative speed (as shown in Fig. 15.2), a considerable part of the tape will remain blank in each revolution. At least two heads will be needed so that no portion of the tape remains blank (i.e. without recording). The signals to and from the rotating video heads is carried with the help of a rotating transformer (shown in Fig. 15.3) whose primary is stationary and the secondary rotates. There is no mechanical wear and electrical noise in the rotating transformer system, which cannot be avoided in an alternative system called slip-ring system. During playback, the secondary acts as the primary and the stationary primary becomes the secondary. Signals for playback are derived from two heads by switching two pre-amplifiers at 25 Hz.

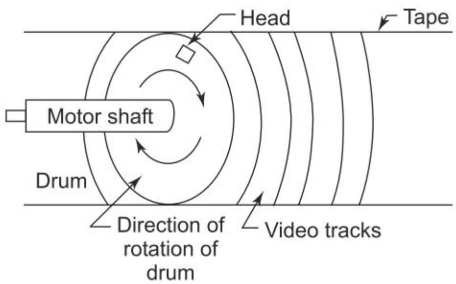


Fig. 15.2 *Rotating head*

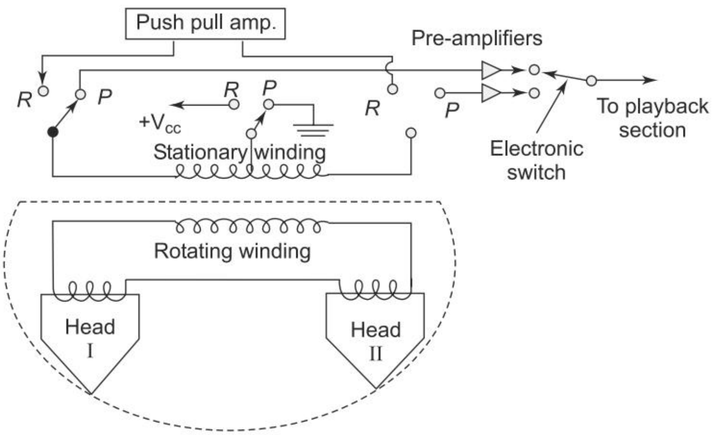


Fig. 15.3 *Rotating transformer*

The audio and control track are recorded by stationary heads (not the rotating heads). A rotating head is not needed for these signals because frequency is low which does not need high speed of tape.

15.6 NEED OF FREQUENCY MODULATION FOR VIDEO RECORDING

Another problem in video recording is high bandwidth of video frequency. It varies from low frequency (say 5 to 10 Hz) to high frequency (say 5 MHz) corresponding to about 19 octaves. Even dc is a part of video signals as it represents average brightness. No recording system can handle such a high bandwidth. The difficulty is solved by modulating a carrier wave by the luminance signal by adopting frequency modulation. The frequency modulation is achieved by changing the frequency of a voltage-controlled oscillator.

FM eliminates the need of ac bias for video recording (in audio recording separate ac bias was needed). FM has been chosen to eliminate random amplitude variations (noise) generated in the recording/playback process. Another advantage of FM for video recording is that a magnetic tape can be driven fully into saturation in both directions as amplitude distortion (in the saturated condition) will not change the FM signal. The recorded FM signal is a square wave. The playback output is high.

The tip of the H-sync pulse in a video signal is clamped to the voltage level that gives 3.8 MHz in the FM oscillator output. The peak white gives 4.8 MHz. Thus, there is a frequency deviation of 1 MHz from the sync top to the peak white signal. Considering the bandwidth of a video signal to be 2.5 MHz, the modulation index will be 1/2.5 = 0.4 only. The modulation index is kept low to get only two prominent side bands and eliminate others to conserve bandwidth.

15.7 HELICAL RECORDING

To make playback time long (at least 3 hours) without using too much tape length, the recording is made diagonally across the tape. Each diagonal track corresponds to video signals for one scanning field. Figure 15.4 shows the diagonal tracks for the video signal on the tape. Slant tracks can be recorded on the tape wrapped around a drum that carries two heads. The heads are 180° apart and the tape wrap angle round the drum is greater than 180° to allow overlap and hence to eliminate any loss of signal due to switching from one head to the other.

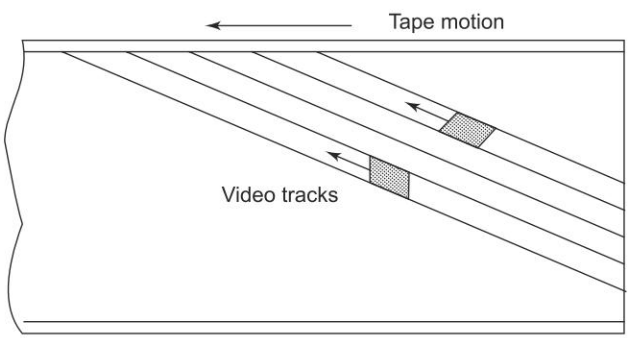


Fig. 15.4 *Helical recording*

15.8 RECORDING OF LUMINANCE AND COLOUR SIGNALS ON THE SAME TRACK

A luminance signal uses frequency modulation to solve the problem of 19-octave wide bandwidth. After conversion, its frequency becomes about 1.5 to 6.5 MHz (less than 3 octaves).

The chrominance signal around 4.43 MHz is down-converted to 562.5 kHz before recording and is up-converted to the original value around 4.43 MHz in the playback section. The purpose of down and up conversions is to eliminate 'time-base error'. The time-base error is produced due to recording being long and the medium being elastic, moving under friction with the help of stationary tape guides. Friction causes stretching and bunching of signals. The time base error may exceed one micro second in domestic video-cassette recorders. This causes severe distortion in colour. Whatever distortion occurs to the signal, the same distortion occurs in the sync-pulse. In the up-conversion, the local oscillator frequency is created by multiplying the sync-pulse frequency and so it also contains the distortion caused due to time-base error. When the distorted signal is heterodyned by subtracting it from the local oscillator signal which is equally distorted, the output would be a pure chroma signal, free of distortion, as clarified in Example 15.2

Example 15.2

(i) Down conversion
= 4.9961 – 4.4336
= 0.5625 MHz

(ii) Add due to time base error
= +distortion

(iii) Resultant down-converted signal available in playback section (i + ii)
= 0.5625 + distortion

(iv) Local oscillator. freq. in playback section derived from signal = 4.9961 + distortion

(v) Resultant up-converted signal (iv – iii) = 4.4336 (without distortion)

The new CVS is formed by combining FM luminance signal with the down-converted chroma signal This new CVS is recorded on a single track on a video magnetic tape. This is called *under colour process*.

The chroma signal is down-converted to quite a low frequency (562.5 kHz) which helps in separating it from the FM luminance signal (1.5 to 6.5 MHz) in the receiver for up-conversion. The up-converted chroma signal and the demodulated luminance signal are again combined to restore original CVS in the playback section.

The broad factures of video recording are shown in the left-hand-side section of Fig. 15.5.

15.9 PRINCIPLE OF REPRODUCTION OF VIDEO SIGNALS IN A VCR

The stored magnetic field can be detected by the playback head (play and record head assembly is common in VCR). In the playback mode, the varying magnetic flux of the tape passes through the coil of the head and hence induces emf in it according to the principle of electromagnetic induction. The signal from two heads derives two low-noise pre-amplifiers (Fig. 15.3). The pre-amplifiers are

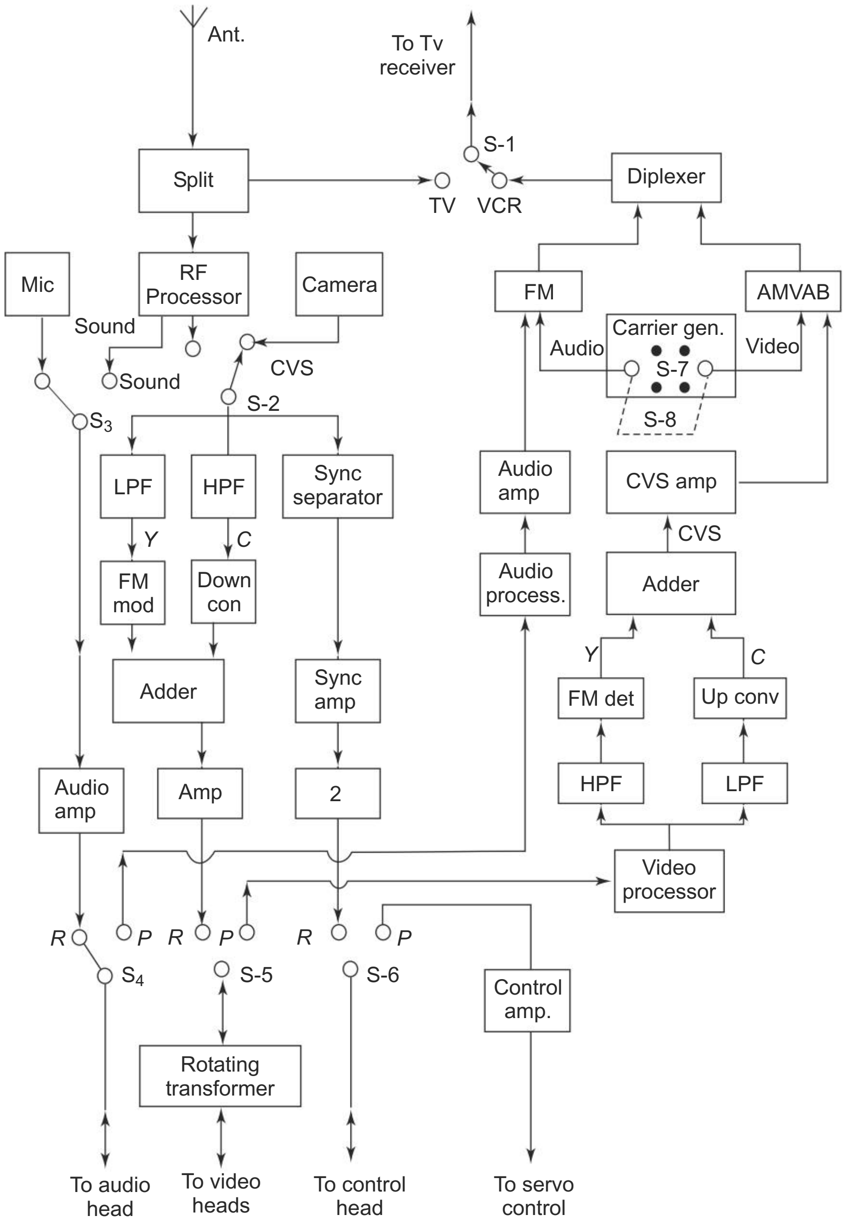


Fig. 15.5 *Block diagram of VCR*

switched on and off by a 25-Hz square wave. The 25-Hz signal turns on the amplifier for the head which is in contact with the tape and turns off the other. This eliminates distortion due to overlapping of two heads for a little while (about 320 µs). The CVS output goes to the filters. The low-pass filter separates out the chroma signal (562.5 kHz) and the high-pass filter separates the luminance signal

(12.5 to 6.5 MHz). The luminance signal is fed to the FM demodulator, which recovers the original luminance signal. The chroma signal is up-converted to its original value of 4.43 MHz. The two signals are then combined to form the original CVS. It, then, amplitude modulates a radio frequency carrier (pertaining to 3rd or 4th channel). The lower sideband is attenuated by a filter to give VSB signal. The signal from the audio head is amplified and made to frequency modulate a separate carrier of 3rd or 4th TV channel. The AMVSB signal and FM audio signal are combined to give radio frequency spectrum of 7-MHz channel width. These signals are then fed to the TV receiver through a TV/VCR switch. The playback process is shown in the right-hand side section of Fig. 15.5.

Audio signals are recovered from the audio head by electromagnetic induction, processed (de-emphasised) and amplified. These signals frequency modulate a separate audio carrier and then are combined with the AMVSB video signal. The combined video and audio modulated carriers are fed to a TV receiver through a TV/VCR switch for reproduction.

15.10 BLOCK DIAGRAM OF A VCR

The detailed block diagram of a VCR for recording and reproduction of video and associated audio signals is given in Fig. 15.5. Function of each block is described below.

Recording Section

Antenna The VCR can record signals picked up by the TV receiver antenna. The function of the antenna is to pick up TV signals traveling through space.

Splitter Besides making impedance match, the two-way splitter provides two parallel paths in the output. One part of the signal goes to a TV receiver through the switch *S*-1. The other part goes to the VCR tuner.

RF Processor The radio frequency signal picked up by the antenna is processed further by stages common to video and audio signals. These stages are similar to the stages in a TV receiver and are described below.

Tuner It selects the wanted telecast, amplifies it, improves signal-to-noise ratio and with the help of a local oscillator and mixer produces an IF signal by the superheterodyning process. IF for video is 38.9 MHz and for audio, 33.4 MHz in the CCIR-B system. In modern VCRs, an electronic tuner is used which uses voltage controller tuned circuit based on varactor diodes.

IF Amplifiers These amplify the IF signal and provide necessary bandwidth and selectivity for the desired signal.

Detector The signal received from the IF amplifier is detected by the envelope detector. The detected video output (CVS) goes to the switch *S*-2. The intercarrier frequency (5.5 MHz in CCIR B system) is separated out from the output of video detector (or better from the output of IF amplifier to prevent beat frequency caused by the presence of inter-carrier frequency and colour subcarrier frequency), is amplified and detected by FM detector. The audio goes to the audio section through switch *S*-3.

Camera Tube and CVS Processor A camera tube is used when programmes are directly recorded. It converts red, green and blue light into corresponding video signals. The function of CVS processor is to produce a CVS signal consisting of chroma signal and luminance signal. The luminance signal also contains blanking pulse, sync pulse and colour-burst signal.

Low-pass and High-pass Filters The composite video signal passes through filters. The low-pass filter allows luminance signal up to 2.5 MHz and prevents chroma signal. The high-pass filter allows chroma signal around 4.43 MHz and prevents luminance signal. Thus luminance and chroma signals are separated for further processing.

Frequency Modulator for Luminance Signal To solve the difficulty of wide bandwidth of about 19 octaves, luminance signal frequency modulates a carrier around 4 MHz. Thus, bandwidth of 2.5 MHz is converted into a bandwidth of 1.5 to 6.5 MHz which is less than 3 octave wide.

Under Colour System for Chrominance The chroma signal around 4.43 MHz is down converted by the superheterodyning process to around 562.5 kHz, so that it is far below the lowest frequency of (1.5 (MHz) FM luminance signal.

Luminance-chroma Combiner The frequency modulated luminance signal and down-converted chroma signal are combined to form a new CVS.

CVS Power Amplifier It amplifies the new CVS signal so as to drive the record heads.

Record Heads The new CVS signal is fed through record/play switch to the rotating head drum with the help of a rotating transformer. The function of rotating heads is to increase the relative speed of the tape with respect to the head. The tracks recorded are slant. (Erase head, not shown in Fig. 15.5, erases all signals before recording).

Control Head Signals The control head records control pulses derived from the vertical (V-) sync signals. The V-sync pulse occurs at 50 Hz (in the CCIR-B system) and are fed to a 'divide by two' multivibrator circuit. The output of the divider is 25-Hz pulses which are fed to the control head for recording. In playback, these pulses are used for deriving control signal for servo control of the motor speed.

Sound Section Audio signals received from FM detector are fed to the audio amplifier. Also, audio signals can be received directly from the microphone through switch *S*-3. The signal from the FM detector or the microphone is processed for pre-emphasis, amplified and goes to the switch *S*-4 for recording. These signals go to the audio head for being recorded longitudinally on one edge (1 mm width) of tape.

Cassette The cassette contains magnetic tape and spools, and it has access to the recording heads through a transport mechanism. (The tape material consists of 25 µm (1 mil) backing of mylar. Coating of magnetic material on the tape is 10 µm (or about 0.5 mil) thick. The magnetic material is either ferric-oxide or chromiumdioxide doped with cobalt. Signal-to-noise ratio is better in chromium dioxide tapes.) Movement of the tape is controlled by guide rollers and capstan. There are two head gaps on the drum and the tape presses against one or the other gap while moving. There are separate heads for audio and control signals.

A servo-controlled motor provides precise speed to the tape and the drum containing heads. In the record mode, the servo-control receives 25-Hz reference pulses from the divider. 25-Hz sampling pulses are also obtained from the head drum-drive motor. The two pulses are compared, and the output of the comparator is fed to the servo-control which controls the speed of the motor. In the playback mode, 25-Hz frequency is derived from 50-Hz mains and is compared with the control pulses obtained from the recording on the tape and their processor (amplifier). The difference, if any, produces a low dc voltage which is used by the servo control to adjust the speed precisely to the desired value. This is illustrated in Fig. 15.6.

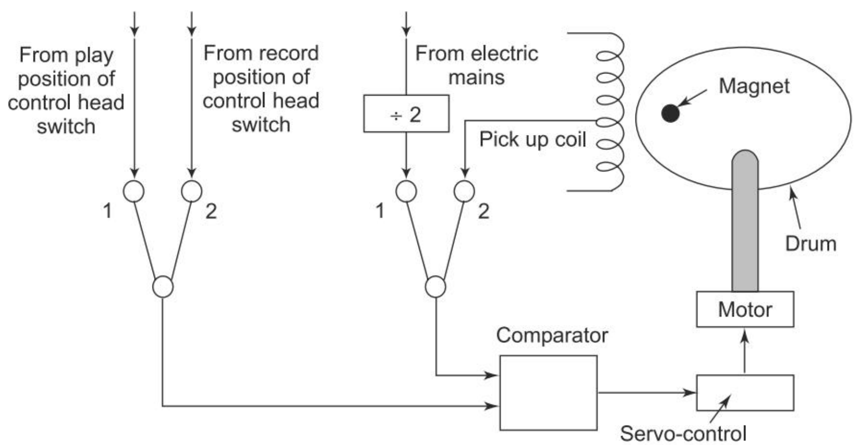


Fig. 15.6 *Servo controlled system*

Playback Section

Playback Heads and Switching The recording heads act as playback heads also in domestic recorders, and are brought into playback mode by three switches *S*-4, *S*-5, and *S*-6 all ganged. When the tape passes over the heads, the varying magnetic field on the tape induces current in the head coils. The video signals are derived from the playback head through the rotating transformer and are fed to two pre-amplifier (as shown in Fig. 15.3). An electronic switch switches the two amplifiers in turn, so that signal from one head only is available to the playback section at a time.

Filters The video signal passes through filters. The low-pass filter allows a chrominance signal of 562.5 kHz to pass and prevents frequency-modulated luminance signal. The high-pass filter passes the luminance signal and prevents the chroma signal. Thus, the two signals are separated.

Up-conversion for Chroma Signal The chroma signal, which was down-converted before recording, is restored to its original frequency of around 4.43 MHz (3.13 to 5 MHz bandwidth) by tup-conversion, using the heterodyning process.

Demodulation of FM Luminance Signal The luminance signal, which was frequency modulated to reduce the octaves before recording, is restored to the original baseband frequencies by FM the demodulator.

Combiner The luminance signal and the chroma signal are combined to give the original CVS.

Video Amplifier The CVS signal is amplified by this stage.

Video Modulator The CVS amplitude modulates a radio frequency carrier of 3rd or 4th channel, generated locally, (channels are switchable by the Switch *S*-7). The VSB filter is used to attenuate lower sideband to conserve bandwidth.

Audio Section The audio signal is amplified, and it frequency modulates a separate carrier (called sound carrier) pertaining to the 3rd or 4th TV channel (selectable by the Switch *S*-8).

Diplexer The amplified and modulated radio-frequency signals pertaining to video and audio are fed to the diplexer circuit where the two signals are combined to be fed to the antenna port of the TV receiver through the Switch *S*-1 for input to the VHF tuner stage of the TV receiver.

(Thereafter, the signals are processed normally in a TV receiver as 3rd or 4th channel signals, depending on the position of *S*-7 and *S*-8 switchs, and finally give picture output on the screen of the picture tube and sound output from the loudspeaker.)

15.11 VIDEO CASSETTE PLAYER (VCP)

VCR contains the facility of recording and playback both, and hence it is a costly equipment. A VCP contains playback facility only and hence is much cheaper than a VCR. A VCP can be used to play the cassette of a desired programme available in the market. Its limitation is that it cannot record camera output, or off-air TV telecast programmes. Its block diagram will be the same as given for the playback section of a VCR (Fig. 15.5).

15.12 CONNECTION OF VCR TO TV RECEIVER

The antenna lead is removed from the TV antenna terminals and is connected to the input terminals of the VCR (input of the splitter stage of VCR). The radio-frequency output of the VCR (output of switch S-1 in Fig. 15.5) is connected to the antenna terminal of the TV receiver. While one channel is being recorded by tuning the VCR's tuner to that channel, the same or another channel can be viewed in 9 TV receiver by keeping the switch *S*-1 in the position marked 'TV' and tuning the TV's tuner to that channel.

15.13 VCR FORMATS

Format is a set of specifications relating to the pattern of tracks produced on a tape for picture and sound recording. These specifications are tape width, tape speed, drum-rotating speed, tape threading path, drum diameter, video track pitch and width, etc. The formats in use for domestic VCRs are known as

- Video Home System (VHS)
- Betamax
- Video-2000 (V-2000)

These formats are not compatible with each other, and hence a cassette relating to one format will not work on any other format without some complex circuits. These three formats are described individually in Sections 15.16, 15.17 and 15.18.

15.14 VHS

Tape path, called M wrap in the VHS system is shown in Fig. 15.7. It is called M-type wrap because the wrap near and around the drum takes the form of the letter 'M'. The drum uses two heads diametrically opposite to each other. The wrap is slightly longer than 180° so that no portion of the tape is missed from being recorded.

When a CVS current flows through heads, flux is developed in the magnetic material of the tape. During playback, the varying magnetic flux induces voltage in the coils of the heads, which is fed to two pre-amplifiers, one for each head. The two pre-amplifiers are switched at 25 Hz to eliminate the problem of overlapping signals in the recorded tape. The Luminance signal is frequency modulated in which a carrier frequency is changing between 3.8 MHz (clamped to the sync) and 4.8 (damped MHz to peak white). This is done to reduce the octaves of the frequency spread of luminance signal. Colour signals are down-converted to 562.5 kHz to enable them to be separated during playback processing.

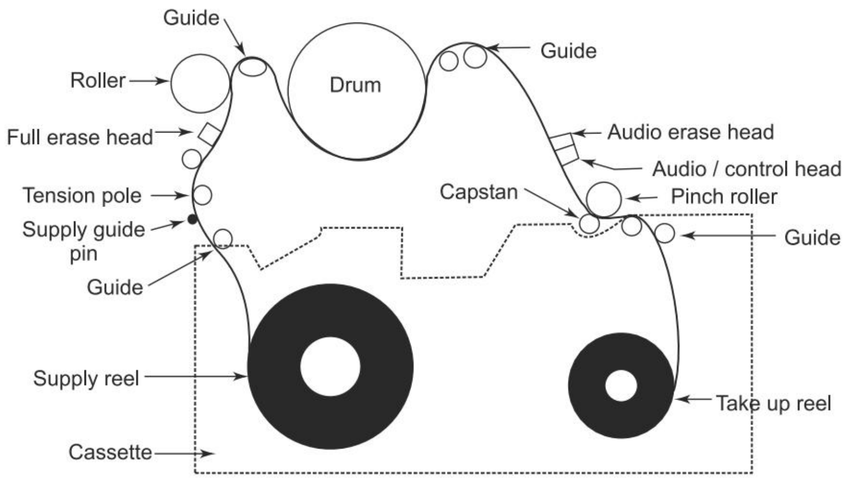


Fig. 15.7 *Tape path of VHS*

The specifications of VHS are given in Table 15.1. The tape-recording format is shown in Fig. 15.8.

The improved version of VHS system provides a programmable timer capable of accepting and recording instructions, tape indexing and timing. Cue and review facility for 'fast winding with picture' was also introduced. It incorporated a soundtrack based on the Dolby method, which reduced the noise. The playback time was increased from 3 hours to 4 hours.

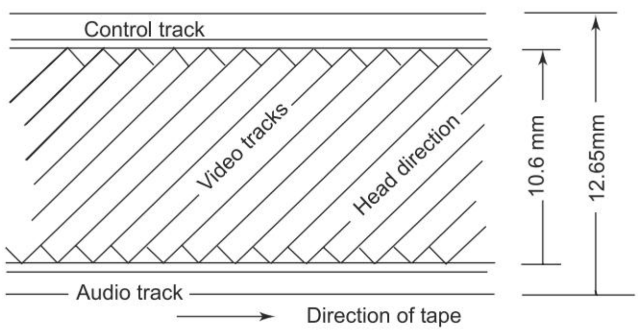


Fig. 15.8 *Format of VHS*

Zero guardband System A vacant space, when provided between individual slant tracks to eliminate cross talk, is called *guardband*. This caused wastage of valuable space on the tape and hence, in a modern home recording system (VHS), the guardband has been completely eliminated, and the system is known as *Zero guardband system*.

Elimination of guardband was made possible by tilting the gaps of two heads with opposite angles. The gaps are tilted (or canted) by 6° (for VHS) from the vertical in opposite directions. Tilting causes azimuth error by which the head-1 ignores high frequency FM video signals recorded on the track of the head-2 and vice-versa.

15.15 BETAMAX

Except minor changes in the tape speed, video head speed, video track width and method of wrapping the tape around the drum, Betamax is no different from VHS. However, there is one important difference. During fast forward and reverse winding, Betamax tape remains laced to the drum, while in a VHS machine, the tape is returned to the cassette during these modes. Thus, wear and tear on a Betamax machine is somewhat more than VHS. (For this reason, VHS became more popular than Betamax and more and more manufacturers took up its manufacture). Cue and review facility was introduced by Betamax later. The cassette in Betamax is smaller than in VHS.

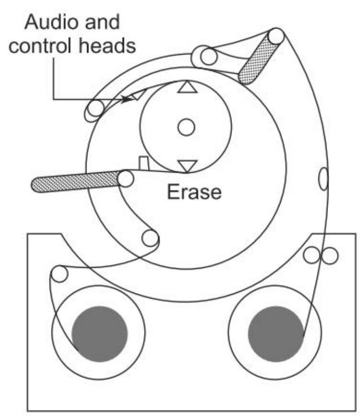


Fig. 15.9 *Tape path of Betamax*

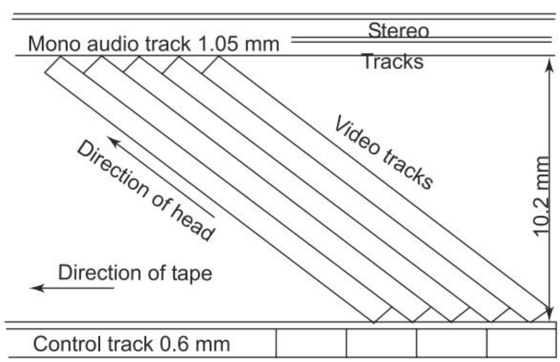


Fig. 15.10 *Format of Betamax.*

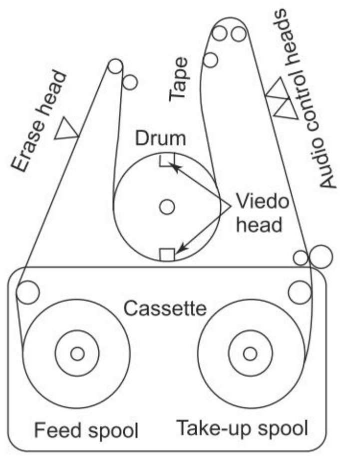


Fig. 15.11 *Tape path of V-2000*

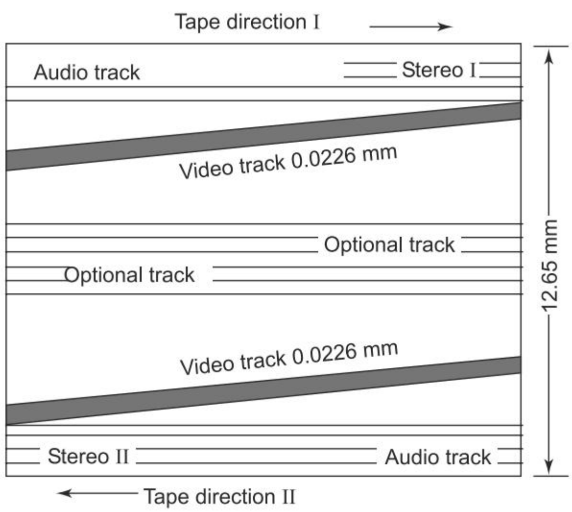


Fig. 15.12 *Format of V-2000*

Specifications of Betamax are given in Table 15.1 and tape path and tape-recording format are shown in Fig. 15.9 and Fig. 15.10, respectively. It has all the special features described for improved version of VHS.

15.16 VIDEO-2000

It is radically different from the other formats in that the tape can be turned over at the end of one recording and the second recording can be started on the other side. It gives a total recording time of 8 hours (4 hours on each side) which is a very spectacular advantage of the V-2000 format. Each side takes up half the width (i.e. 6.3 mm or ¼”) of the tape. This was made possible by a special technique called *dynamic track following* which relies on tape pulse codes during recording. This ensures that the head will read their correct track on the tape.

The V-2000 format does not require control pulse and can serve for editing and tape-code indexing requirements. However, the format has not yet become popular and VHS is still supreme.

Specifications of Video 2000 are given in Table 15.1. Tape path and tape-recording format are shown in Fig. 15.11 and 15.12 respectively.

15.17 U-MATIC

It was introduced by Sony in 1971 and was the first helical format to be used commercially. It uses 19 mm (3/4”) wide tape and the diameter of the drum is 4.33” (110 mm). The track width is 3.35 mil (85 μs), tape speed is 3.75 inch/s (9.5 cm/s) and head-to-tape speed is 410 inch/s (10.4 m/s). The recording time is 1 hour. Although it was aimed at home-user, size and price made it prohibitive for domestic recorders. However, it has become established in industry and also in semi-professional studios for the purpose of training and storage of programmes. Editing is easier in this machine and hence, it became quite popular with studios.

Applications U-Matics are used as industrial and commercial machines of training, sales promotion, educational programmes, and professional studios. TV broadcast services use U-Matics for Electronic News Gathering (ENG) and sponsored programmes.

In high-band U-Matic, luminance carrier frequency is 5.6 MHz, frequency deviation being 1.6 MHz (from 4.8 to 6.4 MHz). Down-converted frequency of chroma signal is also higher which gives better picture quality in colour. It also gives better editing of the recorded picture.

As U-Matic is costly and bigger in size, it will soon be replaced by the improved version of VHS called Super-VHS(or SVHS). S-VHS is superior to U-Matic both in quality and economy.

15.18 8 MM TAPE

It is the latest tape format, very compact, using only an 8 mm wide tape. It may become more popular than VHS when fully developed. Its specifications are

Tape width	=	0.31” or 8 mm
Drum diameter	=	1.57” (40 mm)
No. of heads	=	2
Track width	=	0.81/0.4 mil
(.02 mm/.01 mm)		

Tape speed	=	0.56/0.28 inch/s (14/7 mm/s)
Recording time	=	2 hrs/4 hrs
Head-to-tape speed	=	148 inch/s (3.76 m/s)

15.19 ELECTRONIC NEWS GATHERING (ENG)

The term is used for gathering news by recording public interviews and field events. It requires a portable camera tube and a VCR which meets the minimum requirements of TV broadcast. Due to combination of camera and VCR, it is also called 'camcoder'. The camera is designed to be very compact. It may be single-type camera using the coloured-stripes technique for producing three primary colours. A Charge Coupled Device (CCD) for totally electronic cameras (also called solid camera) is most suitable for ENG because of its small size, low power consumption and immunity to vibrations.

An ENG camera is self-contained. Unlike studio cameras, the sync generator and colour encoder are built-in. Controls are automatic. The battery is of chargeable type and compact and can be fitted into a belt worn by the camera operator.

In the older versions, the VCR was of U-Matic format for ease in editing and reducing copying loss. However, U-Matic has now been replaced by the improved version of VHS, which is more compact that U-Matic.

15.20 VCR NOMENCLATURE: (1/2" VCR, 3/4" VCR AND 1" VCR)

The nomenclature of the VCR is based on the width of the magnetic tape used; ½″ VCR means it uses magnetic tape of ½″ width. Similarly, ¾″ and 1 VCR use tapes of ¾″ and 1 width, respectively. (2″ tape with four heads giving quadruplex ampex format was also used, but it is not used now due to its being bulky and costly).

Applications VHS, Betamax and V-2000 are all 1/2″ VCRs, and are used as domestic video cassette recorders. U-Matic is ¾″ VCR, and is used for industrial and commercial purposes. 1″ tape is of broadcast standard and is used for broadcast of TV programmes by professional and government broadcast agencies. Its format is known as C-type format. In this, the video signal uses the full width of the tape (while others leave separate space for audio and control tracks). Thus, audio signals cut across the video signal. Its specifications are given below.

Tape width M—1″, Drum diameter—5.4″, Track width—50 mils, Tape speed—22 inch per second, Head-to- tape speed 1000—inch per second.

15.21 VCR CONTROLS

A video cassette recorder has quite a large number of operating controls on the front panel and some on the rear side. These controls are described below:

Eject Button It lifts the cassette holder. The cassette is removed or inserted in the raised position. The eject button should not be pressed when the VCR is in operation.

Stop Button It stops the tape.

Rewind Control Its function is to rewind the tape. When this control operated, the motor runs very fast (about five times the normal speed). If rewinding is required when the tape is in playback mode, the stop button should be pressed first.

Fast-forward Control This is operated during playback mode when it is desired to wind the tape at fast speed to reach the desired programme quickly. In the fast forward position, the tape is forwarded to the take-up-spool at 5 times the normal speed.

Play Control It moves the tape in the forward direction at normal speed of play or reproduction of the recorded programme. It starts the playback and all record and play switches come into play position.

Record Control It starts the recording. It is pressed together with the play control button. In the record mode also, the tape moves forward from the feed-spool to the take-up-spool at normal speed. All record-play switches come in the record position with the help of record control.

Pause Button This button, when pressed, stops the cassette for a short period when recording of the unwanted material is avoided.

Reset Button This puts the tape counter to zero position. (The tape counter counts the length of the tape moved.)

Track Control When tapes recorded on another recorder are played, noise and streaks appear in the picture. At that time, this control is rotated to make the picture clear.

Focus Control or Sharpness Control It is operative during playback. It sharpens the outlines (i.e., definition) of the picture.

Timer Control It sets the clock for showing correct time.

OTR (One Time Recording) This control automatically fixes the recording time, say 120 minutes. After the set time, the power is automatically cut off. It is useful when one wants to record a programme telecast at a time when he or she would sleep or would not be available to operate the VCR.

Signal Selector It connects the recorder either to the camera or to the TV antenna.

Tuning Control It tunes in the desired TV station for recording.

VCR/TV Control It is a switch by which playback of the VCR or a signal from the TV antenna goes to the TV receiver for viewing.

Channel Selector It is located in the tuner and selects the desired channel of the TV for recording the programme being telecast on that channel.

Automatic Fine Tuning This button, if available, is kept in the *ON* position. It adjusts the local oscillator frequency to give correct IF.

RF Signal Level Switch If signal strength of the signal received in the VCR tuner is high, this switch is kept in the 'low' position. When the signal strength is too low, it is kept in the 'high' position.

Moisture Indicator Lamp When humidity is high, this lamp lights up automatically and switches off the recorder.

ON-OFF Switch This turns the recorder *on* or *off*. The clock remains *ON* even when this switch is *OFF*.

SUMMARY

- Video recording on a magnetic tape is done by using the principle of electromagnetism, high reluctance of air and low reluctance of a magnetic material.
- When a video current passes through a coil wrapped over a core of soft-iron with a tiny air gap, magnetic flux is produced in the core. When a tape having a coating of magnetic material passes past the air gap, the magnetic flux passes through the tape instead of through air gap due to high reluctance of the air and low reluctance of the magnetic material on the tape.
- Thus, the tape material is magnetised in accordance with the video current.
- *Video recording has to overcome four problems*
  One is high frequency of video signals, in the range of Megahertz, the second is high bandwidth, spreading over about 19 octaves, the third one is to accommodate long-duration (about 3 hours) programmes on the tape and the fourth problem is that of chrominance and FM luminance signals being in the same hf bandwidth. These problems have been solved as mentioned below.
- To accommodate high frequency of video signals, the head gap is reduced to one-tenth the size used in audio recorders and the head is rotated to get high relative speed between the tape and gap. A minimum of two heads are mounted on a rotating drum.

- ☞ To accommodate high bandwidth spreading over 19 octaves, the video signal is made to frequency modulate a 4 MHz carrier. This reduces the number of octaves for the given bandwidth from 19 octaves to about 2 to 3 octaves only.
- ☞ To allow programmes of long duration to be recorded, slant or diagonal recording is made on the tape.
- ☞ The problem of time-base error and problem of FM luminance and chroma signals being in the same band is solved by down-converting the chroma signal.
- ☞ The video recording can be done either by receiving the TV broadcast direct from the receiving antenna and recovering, as in a TV receiver the original CVS, or by using a TV camera and getting CVS.
- ☞ Luminance signal, derived from CVS, frequency modulates a carrier. The colour signal, also derived from CVS, is down-converted (under colour system). FM luminance signal and the down converted colour signal are combined to form new CVS and fed to the recording head.
- ☞ For reproduction, the signal from the tape is extracted by em induction, FM luminance and down-converted colour signals are separated by means of filters, demodulated and up-converted, respectively and then combined to form the original CVS which modulates a video carrier. The modulator give AMVSB output in the 3^{rd} or 4^{th} channel of TV.
- ☞ Audio recording is done by separating out audio signals from the video detector stage, or taking it direct from a microphone. For reproduction, audio signals obtained from the playback position of the head is amplified, processed, and then frequency modulates a carrier. AMVSB picture signal and FM audio signal are combined and then go to the TV/VCR switch of being received in a TV receiver.
- ☞ While a VCR can record as well as play, there is a simpler equipment called a Video Cassette Player which can play the cassettes available in the market. The VCP is a low-priced equipment as compared to a VCR.
- ☞ There are three formats of video recording in use with domestic VCRs, which are not compatible to each other. These are VHS, BETAMAX and VIDEO-2000. All these use ½″ tape width.
- ☞ Another format, U-Matic, using ¾″ tape width, was popular for commercial and industrial recording because of

ease in editing and dubbing, but now VHS has been further improved and U-Matic has become obsolete.

☞ The VCR using 1-inch tape is used for broadcast or professional quality recording. VCR has quite a large number of operating controls on the front panel, like eject button, rewind control, stop button, fast forward control, playback control, record control, pause button, reset button, track control, focus, timer, one time recording control, signal selector, tuning, VCR/TV control, channel selector and automatic fine tuning control, etc.

Review Questions

1. Describe the principle of video recording and reproduction.
2. What are the problems in video recording and how have these been solved?
3. Compare audio cassette recording with video cassette recording.
4. Draw the block diagram of a VCR (video recording and reproduction system) and explain in brief the function of each block.
5. What is the relationship between speed, recorded wavelength and video frequency? Explain why signals are attenuated as the gap size increases beyond $\lambda/2$"
6. Derive the formula relating the highest frequency (which can be reproduced), gap width and tape speed.
7. Compare VHS, BETAMAX and VIDEO-2000.
8. What do you understand by U-Matic format? How does it differ from Betamax and VHS formats?
9. Describe VCR control.
10. What are the disadvantages of tape recording of video signals?

Short-Answer Questions

1. Describe the concept of recording of video on a tape.
2. In magnetic recording, soft iron is used for the core of the head, while ironoxide is used on the tape. Why?
3. Why is a very high-speed tape needed for recording video signals?
4. Why is a very high-speed tape not desirable?
5. How is video recording managed with a low-speed tape.
6. Why are the video tracks recorded slant on tape?
7. Why is a luminance signal made to frequency modulate a carrier frequency in video recording?
8. Why are two heads used instead of one head only in video recording?

9. What do you understand by VCR format?
10. Why is the tape path in VHS called M-type wrap?
11. What is the significance of the name Betamax?
12. What is the function of the eject button in a VCR?

Multiple-Choice Questions

1. Video recording is based on the principle of
   (a) electromagnetism
   (b) piezoelectric effect
   (c) Lenz's law
   (d) Faraday's laws
2. Video reproduction is based on the principle of
   (a) Ohm's law
   (b) Electromagnetic force
   (c) piezoelectric effect
   (d) electromagnetic induction
3. What is the octave width of the frequency range of 10 Hz to 5 MHz
   (a) 5 (b) 10
   (c) 19 (d) 25
4. Relative speed of tape is increased by using
   (a) small gap width
   (b) large gap width
   (c) stationary head
   (d) rotating head
5. For what the letter 'H' stand for in the term 'VHS'?
   (a) Home (b) High
   (c) Hertz (d) Hue
6. Which of the following is not true for video recording being different from audio recording?
   (a) High video frequencies
   (b) Large bandwidth of video signals
   (c) longer duration of programmes
   (d) Magnetisation of the tape
7. For what does the letter 'E' stand for in the term 'ENG'?
   (a) Electronic (b) Emergency
   (c) English (d) Exit
8. Which of the following is not the same for VHS and Betamax?
   (a) Head drum speed
   (b) Tape speed
   (c) Tape width
   (d) Magnetic material
9. The highest frequency which can be reproduced is given by
   (a) $f_m = S/(2G)$ (b) $f_m = S/G$
   (c) $f_m = 2S/G$ (d) $f_m = 2S/3G$
10. Which of the following is not the function of pre-amplifier in a VCR's tuner stage?
    (a) To select the wanted telecast
    (b) To improve signal-to-noise ratio
    (c) To produce IF in the output of the tuner
    (d) To increase image interference

Answers

Short-Answer Questions

1. When video current passes through a coil wrapped over a soft iron with a tiny air gap, magnetic flux is produced in the core, varying in accordance with video current. When a tape having a coating of magnetic material passes past the air gap, the magnetic flux passes through the tape instead of the air gap due to high reluctance of the air and low reluctance of the magnetic material on the tape. Thus, the tape material is magnetised in accordance with the video current.
2. Soft iron does not retain magnetism, and hence magnetism in the core varies as the current in the coil varies. But magnetism on the tape is to be retained permanently and, therefore, iron oxide is used.
3. The highest video frequency is in Megahertz (5 MHz in CCIR B system). Hence, for a physical gap width of 1 micron, the tape speed works out to be 10 m/s which is very high.
4. Very high speed of the tape (in metres per second) will require large length of costly tape and will also cause more wear and tear than the normal speed of only a few centimetres per second.
5. What actually is needed is not the longitudinal speed of the tape, but the relative speed with respect to head. Hence, the head is rotated at high speed by mounting it on a rotating drum, and then the tape speed can be kept low within feasible limits.
6. To increase duration of the programme recorded on the tape, the video tracks are slant.
7. Luminance signal is about 19 octaves wide (from say 10 Hz to about 5 MHz). No recording system can handle such a wide signal. The problem is solved in two steps. First, the luminance signal is reduced to 2.5 MHz. Second, it modulates a carrier of about 4 MHz, using NBFM giving a frequency range of 1.5 MHz to 6.5 MHz (less than 3 octaves wide).
8. A single head rotating on a drum will keep half of the tape blank. Two heads, will record on the full wrap of the tape.
9. A VCR format is a set of specifications relating to the pattern of tracks produced on tape for picture and sound recording. These specifications are tape width, tape speed, drum-rotating speed, tape threading path, drum diameter, video track pitch, width, etc.
10. It is called M-type wrap because the wrap near and around the drum takes the form of the letter 'M'.
11. The name Betamax indicates that the tape's wrap near and around the drum takes the form of the Greek letter β.
12. It lifts the cassette holder. The cassette is removed or inserted in the raised position.

Multiple-Choice Questions

1. (a)	2. (d)	3. (c)	4. (d)
5. (a)	6. (c)	7. (a)	8. (b)
9. (a)	10. (d)		

Video Recording on Discs

16.1 INTRODUCTION

The video disc system was developed to record video and associated audio information on a disc with the help of laser beams to eliminate the disadvantages of magnetic recording on tapes. These disadvantages are as follows.

- The capacity of a tape reel or cassette is inadequate for recording pictures.
- Replication of tape is a time-consuming process.
- Random access of programmes was not possible, and the tape had to be run from the beginning to reach the desired portion of a programme.
- Good-quality picture freeze was not easy in tape recorders.
- A magnetic tape is expensive.
- Wear and tear in a tape is high, hence its life is limited.
- Handling of a tape requires lot of precautions to be observed as the tape surface is prone to scratches, dust, grease, etc.

The above disadvantages are removed in the video disc system which represented a revolutionary breakthrough in audio/video communications, in the same way as the invention of the transistor did in electronics. The advantages of compact disc first designed for audio are discussed in Chapter 6. The disc for video is an extension of the compact disc and is used for recording audio and video.

The disadvantage of a laser compact disc vis-à-vis a video tape was that a tape can be reprogrammed any number of times whereas the disc was usually a 'Read Only Device'. (However, using polarisation techniques, discs which can be written have become available in computers, but being costly have not yet been adopted for entertainment.) Moreover, the preparation of master disc is not as easy as preparation of tape recording for the consumer.

Although video recording on disc was demonstrated as early as 1927 by J.L. Baird himself who had invented mechanical scanning, it died a natural death with the death of the mechanical TV system. Finally, Philips (Holland), Grundig

(Germany) and JVC (Japan) jointly developed a system in 1980, which was named as *Laser Vision*. It was an extension of the compact disc for video. It used a laser beam for recording as well as detection. It is popularly known as Video Compact Disc (VCD) and is described in Section 16.2. Later, in 1994, another version of the disc, known as Digital Video Disc (DVD) was developed which is described in Section 16.3.

16.2 VIDEO COMPACT DISC (VCD)

Original recording is done on a master disc which consists of a glass substrate. A photo-resist material, about 130 µm (or 1300 Å) thick, is coated on the glass substrate. The glass is polished and is spotlessly clean. The glass disc with photo-resist material is called Resist Master Disc (RMD) or simply, the master disc. The video signal is amplified and is then sampled and conditioned to get clean binary pulses free of noise. Error detection and correction pulses are added. A laser beam is then modulated by these pulses (on-off modulation). The block diagram of the whole process of recording is shown in Fig. 16.1. The modulated laser beam reacts with the photo-resist material.

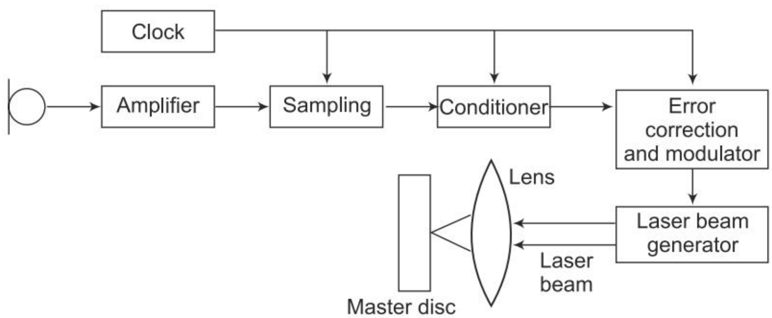


Fig. 16.1 *Block diagram showing digital recording on video disc*

When the disc is developed, a very fine pattern of pits and flats is created. Flats are also called lands. As the disc rotates (1500 rpm for European system and 1800 rpm for American system), the laser beam exposes spirals, and therefore, the spiral tracks of pits and flats are formed on the disc, spiraling out from the centre. The pits are of the fixed depth, but their length and the spacing between them varies in accordance with the signal information.

In some recording processes, DRAW (Direct Reading After Writing) function is included which allows monitoring of the programme as it is being recorded. This is accomplished by beam splitting or by including a second low-power laser beam in the cutter device.

The developed master disc is coated with silver to make it electrically conductive. It undergoes several stages of nickel plating to obtain the final consumer disc, as described in Chapter 6 (Section 6.5)

Playback Process Recovery of a signal from pits by laser beams depends on the principle of diffraction of light as in a narrow slit. Diffraction occurs when a light beam falls on narrow pits of the dimensions of the order of wavelength of light. The effect is very strong if the size of the spot of light is comparable to the wavelength of light, and is focused in a pit whose depth is $\lambda/4$. The diffraction causes nullification of light and hence the light reflected from the pits is too weak. Contrary to it, all the light incident on the flat surface is reflected back. Thus, the reflected light consists of binary signals, 0s from pits and 1s from flats as shown in Fig. 16.2 (a) and (b), respectively.

A photodiode detects the reflected light, as shown in Fig. 16.3.

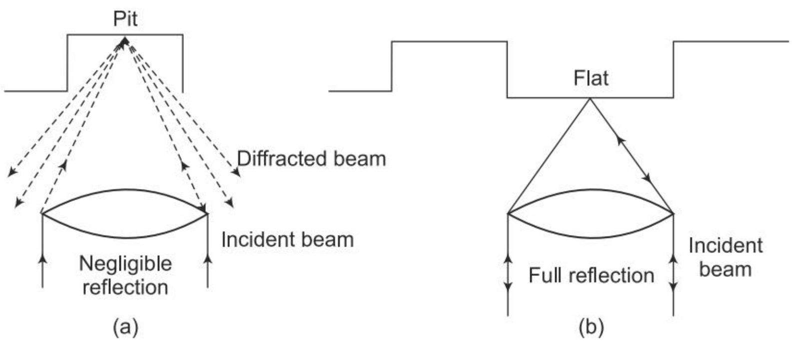


Fig. 16.2 *(a) Light reflected from a pit (b) Light reflected from a flat*

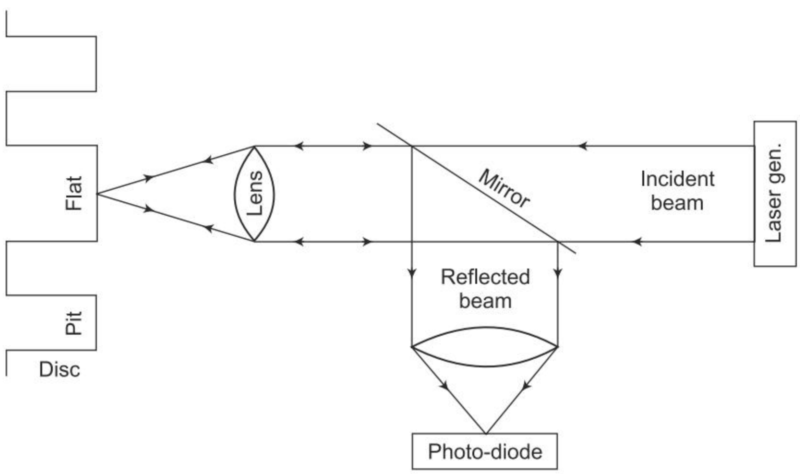


Fig. 16.3 *Reproduction of digital pulses from disc*

16.3 DIGITAL VIDEO DISC (DVD)

The video compact disc did not possess the desired capacity. It could hold pictures of about one hour duration only, which was too inadequate. Despite the success of the VCD, need of a higher capacity disc was felt to meet the requirements of long movies (of about 3 hours duration), high-quality surround sound, computer back-up memory (as portable as floppy disks but capable of recording data of several gigabytes) and multimedia applications. All-round efforts were made by several companies to produce a high-density fast disc.

DVD Forum

Such high-capacity discs suitable for recording video were initially developed in 1994 by two groups separately as two competing formats, known as

1. Super Disc (SD) by Matsushita and Toshiba
2. Multimedia Compact Disc (MMCD) by Sony and Philips

Although both used red light, they were totally incompatible with each other. Under pressure from the computer manufacturers and the movie industries, the two rival camps reluctantly agreed to make joint efforts to develop a single standard to meet the requirements of all concerned, viz. cinema, TV, computers and multimedia. Consequently, the DVD Consortium was formed in 1995 which was renamed as the DVD Forum in 1997 with a steering committee of 10 members. Many companies and organisations joined the forum as members and by the end of the 20th century, there were more than 200 members. The strength of the steering committee also rose to 17.

The forum took upon itself the responsibility of preparing specifications, testing the products and also issuing licenses to the manufacturers and policing the use of DVD logos. The forum constituted several working groups of experts working on different aspects of DVD formats. These working groups approved formats like DVD-video, DVD-audio, DVD-ROM, DVD-R (R stands for recordable) and DVD-RW (RW stands for Re Writable). One of the working groups (WG-11) is studying blue laser format which can increase the data storage capacity of DVD manifolds due to wavelength of blue light being much smaller than red light.

DVD formats which are in use are described below.

DVD Formats

DVD-Video In this, the programme (video and associated sound) is recorded by the production company. The final consumer discs are stamped out in large numbers for distribution en masse (like video cassettes) for playback. It is the most popular of all DVD formats as it can store a full length movie of about 2 ½ hours (or more) on the disc, the same size as a CD. It is rapidly pushing the VCR out of the consumer market because of some very useful features like seamless transitions from movie to movie, panning and scanning, frame freeze, size boosting, etc.

DVD-Audio In this, the music of almost ideal fidelity with surround-sound effect is recorded by the production studio, and discs are stamped out for use by the consumers (like audio cassettes). Like the DVD-video, it is also used for playback only.

DVD-ROM It has been developed as read only memory for computers. New PCs are now provided with DVD drives instead of CD drives. DVD-ROMs are also used by the entertainment industry for modern games consoles. It can store more sophisticated and realistic games.

DVD-R It is of the type 'Record once at user's level and read many times.' (DVD-video, DVD-audio and DVD-ROM were also recorded once and read many times, but the difference is that these were recorded by the programme producers in professional studios, while DVD-R is recorded once by the user himself and then it can be read as many times as one likes.) DVD-R is finding use in video recorders, camcorders, OB vans, product advertisements, etc. Modern Computers using CD writers now may use DVD writers in future. The disc may be used as back-up memory of huge capacity for computers.

It has been recently reported that Philips unveiled an optical disc drive that can play CDs and DVDs in three formats, including the new blue-ray format capable of storing up to 50 GB of data.

Recording and Playback

Recording on DVD is done through the use of a dye which is transformed by a sharply focused red laser beam (of about 10 mW power). The dye is spin-coated onto a clear polycarbonate substrate that forms one side of the disc. The substrate has a microscopic pre-grooved spiral track. This groove is used by the DVD drive to guide the recording laser beam which records data. As the dye layer is heated by the laser beam, it is permanently altered forming microscopic marks to the beam remaining on and off by the signal to be recorded. A thin layer of metal is plated onto the recording layer for the laser beam to get reflected during playback. A protective transparent layer is then applied to the metal surface to save the recording from dust, dirt, grease and scratches. Another substrate is bounded by applying an adhesive layer to give mechanical strength to the disc.

When both sides are needed for recording, the above steps are carried out for each side of the disc. The two recorded sides are bounded together, as illustrated in Fig. 16.4. Also, each side can have two different layers. When only a single side is required for recording, the opposite side can contain labels.

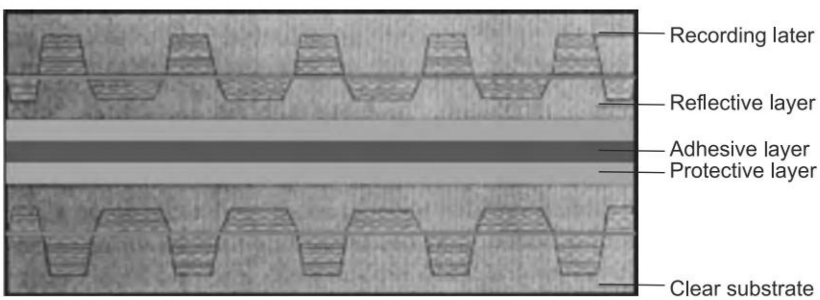


Fig. 16.4 *Layers and sides in DVD*

Playback occurs by using a laser of almost the same wavelength as was used for recording (535 nm or 650 nm). The laser beam is focused onto the disc surface. The flats, also called lands, between pits are reflective while the pits are not. This on-off pattern of the reflected light falls on a photodiode which causes pulses of current, proportional to the pulses of modulated signal which was recorded. Thus, the on-off pattern is reproduced.

No doubt, DVD owes its origin to VCD. But what made it unique is the much smaller size of pits and pitch as compared to VCD. The patterns of pits and pits and flats for VCD and DVD are illustrated in Fig.s 16.5 (a) and (b), respectively.

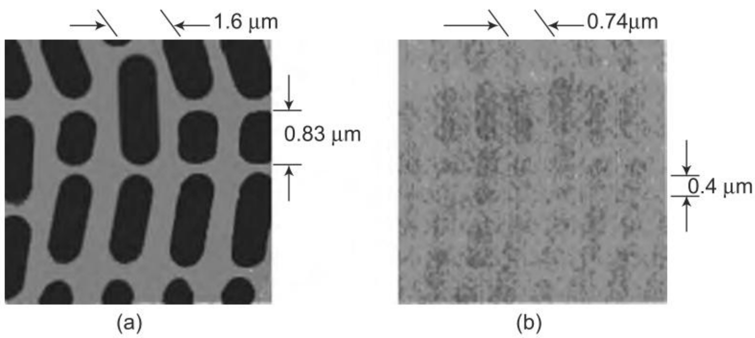


Fig. 16.5 *(a) VCD recording (b) DVD recording*

Some broad parameters of VCD and DVD have been compared in Table 16.1

Table 16.1 *Comparison of VCD and DVD*

PARAMETER	VCD	DVD
1. Material used	Plastic	Plastic
2. Immunity to scratches, greese, etc.	Immune	Immune
3. Diameter	120 mm	120 mm
4. Thickness	1.2 mm	1.2mm
5. Pitch for tracks	1.6 µm	0.74 µm
6. Minimum pit length	0.834 µm	0.4 µm
7. Laser's wavelength	780 nm	635 or 650 nm
8. Technology	Pits and flats formed by laser beam (of infra-red)	Pits and flats formed by laser beam (of visible red)
9. Capacity	700 MB	4.7 GB to 17 GB
10. Resolution		
NTSC	350 × 240	720 × 480

(Contd.)

(Contd.)

PAL	352 × 288	720 × 576
11. Video compression	MPEG-1	MPEG-2
12. Video bit rate	1150 kb/s	9000 kb/s
13. Duration of video recording	About 80 minutes	8 hours, using both sides, with 2 layers on each side.
14. Compatibility	Good	Very good
15. Computer usage	Low	High
16. Quality	Good	Very good

The most distinguishing and meritorious feature of a DVD is its high capacity, as high as about 17 GB which can be used to hold several full-length movies. How such high capacity has been achieved is explained in Section 16.4.

16.4 TECHNIQUES USED TO INCREASE CAPACITY IN DVD

The techniques used in DVD to increase its capacity to hold large information are as follows:

1. The real breakthrough in enhancing the capacity of a laser disc was of red light, 635 nm for professional use and 650 nm for commercial use. This wavelength was a lot smaller than the wavelength of 780 nm (infrared light) used in a CD. The smaller wavelength resulted in a smaller spot. The sharper beam spot increases the capacity in two ways:
   (i) Adjacent tracks became closer, allowing more tracks per disc. The DVD track pitch was reduced to 0.74 mm which is less than half of a CD's (1.6 mm).
   (ii) The pits where data is stored became much smaller than those in a CD. The mminimum pit length in a DVD is 0.4 mm which is less than half of 0.834 mm in CD. This allowed more pits per track.
2. Information can be scanned from more than one layer in a DVD, simply by changing the focus of the laser beam. Instead of using an opaque reflective layer, it is possible to use a translucent layer with an opaque layer behind. While a single layer gives 4.7 GB storage capacity, two layers give 8.5 GB (the second layer cannot be as dense as the first layer and therefore the capacity of two layers is slightly less than two times of a single layer by about 10%). The provision of two layers enables the user to use the higher capacity of a DVD without removing it from the drive and turning over.
3. A DVD allows double-sided discs. A thinner plastic disc was required for the laser beam to focus on the smaller pit depths. This required only a 0.6-mm thick disc, just half the thickness of a CD. Such thin discs were rather too thin to withstand handling. Hence, two discs were bonded back to back, making the whole disc 1.2 mm thick. While bonding was

necessary for rigidity, it doubled the storage capacity as two substrates could be used to record the data. (In a single-sided DVD also, bonding is used for strength, but the data is recorded on one substrate only, the other one remaining blank.)

4. A DVD uses a more efficient error-correction code (ECC). The bits used for error detection consume the space which otherwise could have been used to carry the data. Smaller the number of error detecting and correcting bits, less would be the space required for them and hence more would be the room for real data.
5. A DVD uses the format of MPEG-2 (Moving Picture Experts Group of International Standards Organisation) for compression which gives a higher quality than MPEG-1 used in CD. (MPEG-1, MPEG-2, MPEG-4 and JPEG are the compression standards developed by committees constituted by the International Organisation of Standardisation.; These have been explained in Apendix I)

Layout of Layers in DVD

In a DVD, scanning of information can be done from more than one layer, simply by changing focus of the laser beam. It enables use of a single disc without turning it over. A double-sided dual layer can hold about 8 hours of high-quality video. It has the capability of instant rewind and fast forward and instant search of titles and tracks. Its size is compact and replication is cheaper. It can support special effects like freeze, slow scan and fast scan.

Like a CD, each layer of a DVD contains three areas:

1. Lead-in
2. User's data
3. Lead-out

These are shown in Fig. 16.6.

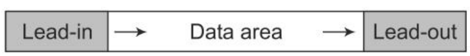


Fig. 16.6 *Areas marked in a single-layer disc*

The lead-in and lead-out indicate to the playback device the inner and outer limits of the recording. They do not contain any information accessible to the user, but are essential for proper functioning of the disc.

The dual layers can have either parallel tracks running in the same direction shown in Fig. 16.7 (a) or opposite tracks, shown in Fig. 16.7 (b). The DVD-ROM uses the pattern of Fig. 16.7(a) for easy access to files on either layer. The DVD-video uses the pattern of Fig. 16.7 (b) to allow a movie on two layers to play seamlessly (i.e., without interruption or necessity of turning over).

Lead-in	⟶ Data Area (layer 1) ⟶	Lead-out
Lead-in	⟶ Data Area (layer 0) ⟶	Lead-out

(a)

Lead-in	⟶ Data Area (layer 1) ⟶	Lead-out
Lead-in	⟶ Data Area (layer 0) ⟶	Lead-out

(b)

Fig. 16.7 *Tracks in dual layers disc (a) Parallel tracks in the same direction, (b) Parallel tracks running in opposite directions*

Blue Laser DVD

Two formats of very high-capacity DVD, about 20 to 25 GB on one layer (against about 5 GB of the current DVDs) have been invented by two Japanese companies, Toshiba and Sony. At the core of both formats are blue lasers which have a shorter wavelength than the red laser used in the current DVDs. Blue laser allows a disc to store data at a higher density needed for high-definition movies. Toshiba has named it as HD DVD (20 GB capacity), while Sony calls their version Blue ray (25 GB capacity). The two versions have different formats which have relative weaknesses and strengths. USA's film studios will probably be the deciding factor in this show-down as had happened in the past in the long drawn-out war between VHS and Betamax formats of video tape recording. However, HD DVD seems to hold better future prospects, because with the same physical structure as a current disc, HD DVD allows manufacturers to use much of their existing DVD equipment, keeping fresh investment minimal and thus keeping its cost low.

16.5 DVD FEATURES

Apparently, a DVD looks similar to a VCD. Both are plastic discs, 120 mm in diameter and 1.2 mm thick. Also, both rely on lasers to read pits and flats on spiral tracks and both are immune to surface contamination. The similarities end here. The digital video disc has surpassed the compact disc in many ways. The following are the value added features which have made the DVD dear to one and all.

1. DVD has a huge storage capacity, with the following options:
   (a) Single Side Single Layer (SSSL), capacity = 4.7 GB
   (b) Single Side Double Layer (SSDL), capacity = 8.5 GB
   (c) Double Side Single Layer each side (DSSL), capacity 9.4 GB
   (d) Double Side Double Layer each side (DSDL), capacity 17.1 GB

A full-length movie can be seen in playback even on a SSSL type DVD. In the DSDL type, 3 to 4 full length movies can be seen.

2. Designed from the outset for video, audio, computer and multimedia, and not just audio, it is very versatile.
3. All formats use a common file system, and hence there is no problem of compatibility.
4. Overall size is quite small and handy, and hence it is portable.
5. Its replication is easy and inexpensive.
6. The strength is same as in a CD, due to the bonding of two substrates.
7. It uses efficient error detection and correcting codes.
8. All special features and advantages of digitised systems are available as DVD recording is digital.
9. CDs and VCDs can be played on a DVD player without any difficulty but not vice versa.

16.6 DVD APPLICATIONS

Applications of DVD technology are so wide that the abbreviation which initially meant Digital Video Disc is now quoted as Digital Versatile Disc. Some of its applications are listed below:

1. Worldwide distribution of full-length movies through DVD video discs with several special and useful features, surpassing VCDs.
2. Worldwide distribution of high-quality music with surround-sound effect through DVD audio discs.
3. DVD-ROMs of huge capacity for computers.
4. DVDs are able to write once on user's level to record field events (camcorders) and to store data as back-up memory for computers and play many times.
5. Getting copies of huge stored data from computers through DVD Writers is easy. It can work as computer back-up memory, and also for exchange of information between different organisations.
6. More sophisticated and realistic games can be played through game consoles, using DVD-ROMs.
7. Re-Writable (RW) DVDs can be used as RAM. These can also be used like VCRs for recording new programmes after erasing the earlier ones.
8. On account of very high storage capacity, fast speed, excellent quality and special features, DVD would eventually replace CDs, CD ROM, video tapes and video game cartridges. It can be rightly called the new generation of optical storage disc which can hold cinemalike video and huge computer data. It can encompass home entertainment and computer information within a single digital format.

SUMMARY

- Video signals have been recorded on disc with the help of laser beam. The compact disc for audio was developed earlier. The Video Compact Disc (VCD) was an extension of the audio compact disc.
- The VCD used infra-red laser to form pits and flats on the photo-resist material coated on the disc's surface. When played back, light is not reflected from pits, but is fully reflected from flats, thus recovering the original digital pulses.
- Use of visible light in place of infrared light resulted in the development of a highly improved version called Digital Video Disc (DVD).
- Pit size was reduced from 0.834 μm (in VCD) to 0.4 μm (in DVD), and track pitch, from 1.6 μm (in VCD) to 0.74 μm (in DVD). In DVD, two substrates, each of 0.6 mm thickness are bonded to give a total thickness of 1.2 mm and to make two sides available for recording.
- Recording can be done on each side in a single layer or two layers per side. Recording on single side only with single layer gives 4.7 GB capacity, while the maximum capacity of recording on double sided double layers per side is about 17 GB.
- Such high capacity makes a DVD very versatile as it can store 8 hours of TV programmes or high-quality music programmes with surround-sound effect. Due to its being versatile in functioning, it is also called a digital versatile disc.

Review Questions

1. How are signals recorded on a video compact disc? Explain with the help of a block diagram.
2. Show that the VCD is just an extension of audio CD.
3. What is the DVD Forum? What are its functions?
4. Draw a block diagram of DVD's playback system and explain its working.
5. Compare VCD and DVD.
6. Explain how the capacity of DVD has been increased.
7. Describe special features incorporated in DVD video.
8. Write a short note on blue laser DVD.

Short-Answer Questions

1. Why are pits not able to reflect light?
2. How do pits and flats allow recovery of baseband signals in a video disc?
3. How is the reflecting property of an optical disc increased?
4. How is an optical disc protected from grease and scratches?
5. Why are two substrates bonded in a DVD?
6. What is the advantage of using laser of red light in DVD over infrared light in VCD?
7. What is the advantage of using blue light laser in DVD?
8. Why is a DVD also called a digital versatile disc?

Multiple-Choice Questions

1. In a DVD, a laser beam is reflected from the
   (a) flat (b) pit
   (c) substrate (d) glass
2. DVD video contains
   (a) video programmes only
   (b) audio programming
   (c) video and audio both
   (d) computer data
3. DVD audio contains
   (a) audio programming
   (b) audio and video both
   (c) audio and computer data
   (d) none
4. SSSL DVD has a capacity of
   (a) 17 GB (b) 8.5 GB
   (c) 4.7 GB (d) 1.5 GB
5. DSDL DVD has a capacity of
   (a) 4.7 GB (b) 9.4 GB
   (c) 17 GB (d) 18.8 GB
6. Tracks pitch and pits density is
   (a) double of VCD
   (b) same as VCD
   (c) half of VCD
   (d) four times of VCD.
7. When are two substrates bonded together in DVD?
   (a) They give same strength as in VCD
   (b) Double the strength compared to VCD
   (c) bonding results in higher pitch
   (d) Video can be recorded in one side and audio on the other
8. Track formation starts
   (a) from the centre to the edge
   (b) from edge to the centre
   (c) from anywhere
   (d) from middle

Answers

Short-Answer Questions

1. Being $\lambda/4$ in depth, pits scatter the light and do not reflect it.
2. As pits do not reflect light, so the output from pits is logiczero. Flats reflect full light and their output is logic, baseband signal is recovered.
3. By coating aluminum.
4. A transparent plastic covers the disc.
5. Primarily, to give strength to the disc. Incidentally, the second disc can also record information and thus doubles the capacity of the disc.
6. Wavelength of red light is shorter than the wavelength of infrared light. This increases pitch of tracks and density of pits.
7. Wavelength of blue light is smaller than that of red light, and so capacity of the disc is increased from 17 MB to 25 MB.
8. Very wide applications of DVD make it versatile.

Multiple-Choice Questions

1. (a)	2. (c)	3. (a)	4. (c)
5. (c)	6. (a)	7. (b)	8. (a)

Troubleshooting in Audio and Video Equipment

17.1 INTRODUCTION

Troubleshooting means to locate the faulty component or connection in the equipment and determining the cause of its becoming faulty. Maintenance includes corrective maintenance (repair), qualitative maintenance (restoring specifications) and preventive maintenance (servicing), Audio and video equipments which use modern electronic devices like PCBs, semiconductor diodes, transistors, ICs, digital circuits, etc., need careful handling not only in operation, but also in fault-diagnosis and repair.

17.2 MODERN ELECTRONIC EQUIPMENT

The salient features of modern electronic equipment are miniaturization (several million components are integrated on a silicon-chip as small as 1 cm^2 size) and high reliability. Digitization (new audio and video systems like DVD, multimedia), computers, satellite (Direct-to-home TV, world-space radio, internet) and optical fibre systems (capable of carrying millions of channels with little error) may be said to be the backbone of modern electronics.

Potential Problems

Modern electronic equipments, although highly reliable, present some potential problems. ICs are sensitive to electrostatic charge and heat. They are also sensitive to dust, vibrations, humidity, variations of temperature and transients in power supply voltages. Aperiodic signals-on multi-line buses in computers need non-conventional test instruments (like logic analysers and signature analysers). Dense packaging in sealed modules or even open PCB cards require special tools, making field repairs prohibitive. These factors call for a sound maintenance policy on the part of the user.

17.3 MAINTENANCE POLICY

It is the strategy to be adopted by the user to keep the equipment available for maximum time possible. Availability depends on two factors:

1. Mean time between failures (MTBF)
2. Mean time to repair (MTR)

MTBF

MTBF is given by Eq. (17.1)

$$NTBF = \frac{t_{u-1} + T_{u-2} + \cdots t_{u-n}}{n}$$

$$= \frac{\sum t_u}{n} \tag{17.1}$$

where, n = total number of failures
t_{u-1} = up-time before the first failure
t_{u-2} = up-time after repair of the first failure
t_{u-n} = up-time after repair of the $(n–1)^{th}$ failure
Σt_u = total up-time

Reverse of MTBF is called failure-rate (FR), that is the number of failures divided by the total up-time. Thus, FR = $n/ \Sigma t_u$ = 1/MTBF.

MTR

Mean time to repair is calculated on the basis of time taken in repair work. It is given by Eq. (17.2).

$$\text{MTR} = \frac{t_{d-1} + T_{d-2} + \cdots t_{d-n}}{n}$$

$$= \frac{\sum t_d}{n} \tag{17.2}$$

where, t_{d-1} = down-time due to the first-fault
t_{d-2} = down-time due to the second fault
t_{d-n} = down time due to n^{th} fault
Σt_d = total down-time for n failures

Availability (A) is defined by Eq. (17.3)

$$\text{A} = \frac{\text{Total up-time}}{\text{Total up-time+Total down-time}}$$

$$= \frac{\text{MTBF} \times n}{n(\text{MTBF} + \text{MTR})} = \frac{\text{MTBF}}{\text{MTBF} + \text{MTR}}$$

$$= \frac{1}{1 + \frac{MTR}{MTBF}} \tag{17.3}$$

Equation(17.3) shows that higher the MTBF and lower the MTR, greater would be the availability. Thus, maintenance policy demands to keep MTBF as high as possible, and MTR as low as possible.

Preventive Maintenance

To keep MTBF high, manufacturers prescribe some periodic routines for the equipment to keep it in perfect health and immune to faults. These routines are as follows.

- Maintenance of physical environment in specified condition (i.e., dust-free atmosphere, controlled temperature and humidity)
- Maintenance of the electrical power free of fluctuations and transients.
- Periodic cleaning and oiling the sliding parts, using oil or grease of the prescribed quality
- Periodic checking of efficiency, alignment, stability and other specifications
- Timely replacement of fast-wearing parts like brushes, relays, switches, heads.

These routines are called preventive maintenance or servicing. It is wiser to nip the trouble in the bud. Preventive maintenance increases MTBF and hence reduces the failure rate, and thus increases availability of the equipment to the user.

Corrective Maintenance

By preventive maintenance, failure rate is reduced, but it cannot be made zero. Faults may occur. Efforts should be that when a fault occurs, it must be rectified quickly. Rectification of fault is called corrective maintenance or repair. Good corrective maintenance demands that after repair, specifications of the set must be checked, and if any deficiency, it must be removed. Specifications must be restored to original specifications within tolerable limits. This part of maintenance is sometimes referred to as quality-maintenance.

Corrective maintenance is related to MTR. Quicker the repair, lower would be the MTR. Low MTR means high availability of the equipment for use. MTR can be made low by

- making spare parts, tools and test instruments readily available.
- Purchasing the equipment in which high *maintainability* (the design aspect by which fault can be diagnosed quickly) has been designed (i.e., easy access to components and provision of well buffered test-points)

In view of the potential problems which modern electronic equipment poses in repair work, it is a wise policy to give annual maintenance contract to the supplier for domestic and office appliances. But where a large number of sets is in use (as in army, police, railways, etc.) it would be desirable to have a well equipped workshop, manned by well-qualified service engineers.

17.4 MAINTENANCE AIDS FOR FAULT DIAGNOSIS

For efficient fault diagnosis, corrective maintenance and preventive maintenance, the following are valuable aids.

- Service manual
- Test and measuring instruments
- Special tools.

These are described below.

Service Manual

The service manual of a set is a document prepared by the manufacturer. A well written and illustrated manual is a very valuable servicing aid. It contains the following information.

1. Detailed specifications and the desired physical and electrical environmental conditions.
2. Block diagrams and circuit diagrams
3. List of spare parts with their values, tolerances and ratings
4. Resistance and dc voltages at each pin point of active components
5. Binary voltage levels and ac signal voltages at special test points.
6. Flow-charts indicating steps for fault diagnosis.
7. Corrective maintenance tables, indicating symptoms of typical faults, likely causes and remedies
8. Preventive maintenance schedules
9. Components lay-out on PCBs and diagram of connection lines
10. Testing and alignment procedures
11. Safety precautions

Test and Measuring Instruments

The following instruments are required for fault diagnosis and maintenance work.

1. Multimetres for measuring voltage, current and resistance.
2. Insulation tester or Megger for checking insulation between two windings or between a winding and the core, or between two terminals or tracks and between the shield and wire of a coaxial cable.
3. Audio power meter for measurement of output power
4. RLC bridge for measuring the values of passive components (resistors, inductors and capacitors)
5. Transistor tester and curve tracer for checking transistors and their characteristic curves.

6. Cathode Ray Oscilloscope (CRO) for checking wave form of input signal and for measuring amplitude, frequency and phase of signals. It can also be used for tracing the signal
7. Distortion Factor meter for measuring distortion produced by the amplifier in the signal.
8. Frequency counter for measuring frequency, period of a pulse, ratio of two frequencies and time-interval between two signals.
9. Q-meter for measuring losses in a coil and self-capacitance of a coil
10. RF Signal Generator for measuring gain of radio frequency stages, sensitivity, selectivity, and image rejection ratio in modern receivers.
11. Audio Signal Generator for measuring gain and frequency response of audio amplifiers.
12. Function Generator for producing sine waves, triangular waves, rectangular pulses and saw-tooth signals. Sine waves can be used to measure gain of an amplifier, triangular waves can test linearity of the circuits, rectangular pulses can test low frequency and high frequency response. The saw-tooth signal can test sweep amplifier in TV and CROs.

The above mentioned instruments are for general applications. However, for special applications for TV measurements, the following instruments are needed.

1. Pattern Generator Generates patterns of cross-hatch, chess-board, circles, dots and bars. The patterns modulate a carrier. The modulated carrier passes through various stages of a TV receiver and finally, the patterns are detected in video detector and are displayed on the screen of the picture tube.
2. Wobulator with marker and CRO is used for aligning the TV receiver.

In digital circuits, voltage level is either low or high, and the pulses are aperiodic, long and complex. Moreover, all the data streams look alike. Hence, conventional methods are of no use for digital circuits and special test instruments described below are used.

1. Signature analyser and logic analyser for recording aperiodic and complex pulses
2. Logic probe for testing presence of logic 1 or logic 0.
3. Logic clip, a multipoint monitor, can give on LEDs a mimic diagram of the status of pins
4. Logic pulser is used for injecting pulses at a node without cutting a track or without removing an IC.
5. Logic current tracer for identifying the device causing short-circuit condition.
6. Logic comparator for comparing the output of a suspected IC with a known good IC
7. Digital IC tester for measuring the minimum level of logic 1 and maximum level of logic 0.

Tools

Tools are the basic requirements of a service engineer, without the tools, one cannot even open a cabinet and have access to the circuits. Some tools, required specially for repairing electronic equipment, are listed in Table 17.1.

Table 17.1 *List of tools and their applications*

Names of Tools		Applications
1.	Set of screw drivers	For tightening or loosening screws. Small screw drivers are used for adjusting trimmers and cores for alignment.
2.	Set of pliers	For gripping. Long nose plier is used as sink while soldering.
3.	Wrenches and box spanners	For tightening and loosening nuts
4.	Tweezer (Pincer)	For picking up small pieces and also to work as sink during soldering
5.	Allen keys	For tightening
6.	Cutters	For cutting wires and pins.
7.	Stripper	For removing insulation on wires
8.	Soldering iron or gun	For soldering and desoldering components.
9.	Desoldering pump or wick	To suck solder for clean desoldering
10.	Cleaning tools	Brushes, knife, blade, alcohol dispenser are used for cleaning leads and connectors
11.	Set of files	For cleaning and smoothening
12.	Head de-magnetiser	For cleaning magnetic heads.
13.	torch	For lighting PCBs
14.	Magnifying lens and inspection mirror	For enlarging the area for better examination
15.	Spring loaded IC extractor	For removing ICs

17.5 PROCEDURE OF SERVICING AND MAINTENANCE

Good servicing and maintenance increases MTBF and hence elongates up-time of the system. Servicing means thorough checking of the equipment periodically as per schedule prescribed in the service manual . It generally includes the following action.

1. Checking the condition of battery/cells, if used
2. Cleaning the equipment thoroughly of dust, dirt, oil, grease, webs, insects, fungi, etc.
3. Cleaning the heads of magnetic tapes
4. Checking the operating knobs for any play or backlash and rectifying it
5. Checking the mechanical parts like connectors, sockets and plugs

6. Oiling the sliding parts to reduce friction (using oil of appropriate grade)
7. Checking the parts which have completed their prescribed life, like relay contacts, switch contacts, brushes, belts, springs, etc.

Maintenance means ensuring correct physical and electrical environment, proper handling by the operating staff and efficient corrective maintenance by well qualified technicians.

For keeping a record of servicing and corrective maintenance, a maintenance logbook must be prepared and kept updated. It serves as a data bank of faults and their remedies. Log-book records can be used to determine the failure rate, MTBF and MTR. Log book can be stored in a computer's hard disk instead of a register.

Circuit Tracing Techniques For the purpose of fault-diagnosis an equipment can be divided into sections, stages and components. First, the faulty section is isolated, then the faulty stage in that section, and finally the faulty component. For determining the faulty section/stage, the test signal may be injected at the input of the set, and then it may be traced through the set. Its presence is detected by the signal tracer at the input and output of each stage. Signal tracing can be done by a simple diode tracer or a CRO. Absence of the signal at the output of a stage will identify the stage as faulty.

For determining the faulty component in the isolated stage, measurements at pin points of the active device will identify the faulty component, which can be traced by a continuity tester (an ohmmeter at the shortest range of resistance). Tracing starts from the pin-point and goes till a component is reached and then to the ground connection.

17.6 SHIELDING AND GROUNDING

The instrument used for measuring the parameters of a component or a device or circuit should be effectively shielded, so that it does not pick up any external noise or interface. A shield is a barrier of conducting material placed in the path of unwanted signals. The interfering field may be either electric field or magnetic field. Electrostatic shields use non-magnetic conductors like aluminium. Magnetic shields use ferromagnetic materials like iron

All circuits finally return to a common point on the chassis, which is called ground point. There is a ground return conductor on the PCB. This ground line is broad and has a very low impedance. The wire carrying signal current should be placed close to the ground return line for reducing the inductive effect. The ground line on the chassis should be connected to an effective earth line (either through a cold-water pipe or through an artificial earth terminal).

17.7 FAULT LOCATION

While troubleshooting in an equipment, the final step is to locate the faulty component proceeding step by step. These steps are as follows.

Identifying the Faulty Section (Functional Area Approach)

Many faults advertise their location by display on the video screen or sound from the loudspeaker. A functional switch gives useful clues to locate the faulty section. For example, if recording in cassette recorder is normal but playback is not working, it means the common amplifier stages are alright and the fault may be in the record/play switch or in the loudspeaker. In a VCR, if the picture is normal, but there is no sound, the sound section could be faulty, In a stereo amplifier, if one channel is working normally, but the other is not, the defect is obviously in the other channel. In a computer, if visual display on the screen is normal but the printer not printing, the fault may be in the connecting cable or in the printer. Such functional area approach is very helpful in isolating the faulty section quickly.

Identifying the Faulty Stage

Stages are connected in four types of circuit configurations:

1. Sequential stages
2. Divergent circuits
3. Convergent circuits
4. Closed loop (feedback) circuits

Methods of identifying faults in these configurations are discussed below.

Sequential Stages

There are three methods of identifying a faulty stage when the stages are in series one after the other (i.e. linear flow of signal).

1. Signal Injection and Tracing In this method, the signal is injected at the input of the system and its presence is detected by a signal tracer at different stages. Signal tracing can be done by a diode tracer, or a Cathode Ray Oscilloscope (CRO).

2. Signal Substitution In this method, a signal is given at the inputs of different stages and the final output is checked.

3. Short circuiting The inputs are shorted to ground terminal one by one for locating the source of noise (like hum, self oscillations, etc.)

In sequential circuits using any of the above techniques, the quickest method of finding out which stage is faulty is the 'split-half method', described below.

Split-half Method

In this method, the output at a stage in the middle of the sequential stages is checked. If it is correct, the first half of the stage is alright and the defect lies in the second half. Then, the output at the middle stage of the second half is checked to find out which portion is defective, and so on, until we reach the correct stage. The split-half method of identifying a faulty stage is illustrated in Fig. 17.1. Output at *A* is checked. If alright, the stages from microphone to stage IV are correct. The output at *B* is checked. If, no output, either the V or the VI stage may be defective. Then, output at *C* is checked which will finally decide whether the stage *V* is defective or VI.

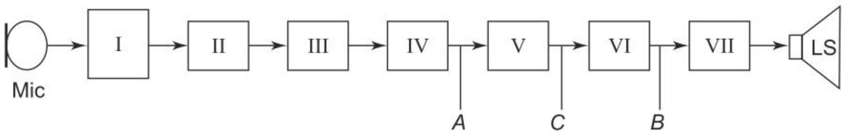


Fig. 17.1 *Split-half method*

Divergent Circuits

The circuit which has one input path, but the output takes several paths is called a divergent circuit. Some more examples of the divergent circuits are.

- Several loudspeakers connected to one output of final power amplifier stage
- Switching circuit with one input and several outputs
- De-multiplexer which receives one input but gives several outputs
- Video detector input is one but there are several outputs, like sound section, luminance, chroma and raster sections.

These are shown in Fig. 17.2. Power supply output is also divergent, as it feeds power to several stages from its output point. This has been described in Section 17.11.

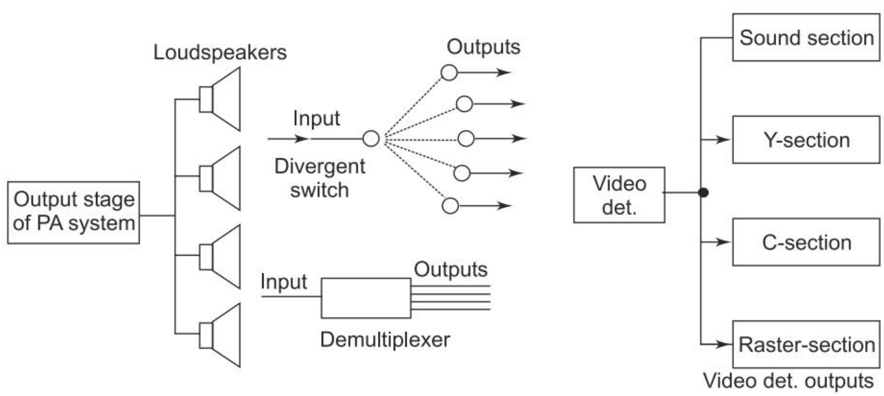


Fig. 17.2 *Examples of divergent circuits*

Troubleshooting in divergent circuits is done by disconnecting (isolating) divergent stages one by one. If the fault is removed, the isolated stage is faulty.

Convergent Circuits

The circuit which has several inputs but only one output is called a 'convergent circuit'. Some examples of the convergent circuits are given below and have been shown in Fig. 17.3.

- Mixer stage for microphones in the PA system in which several microphones are connected in the input and there is only one output for feeding to the amplifier
- Multiplexer which takes several signals in its input and gives out only on one output line.
- Logic gates (except NOT gate) which give one output for two or more inputs
- Switching circuit with several inputs and one output

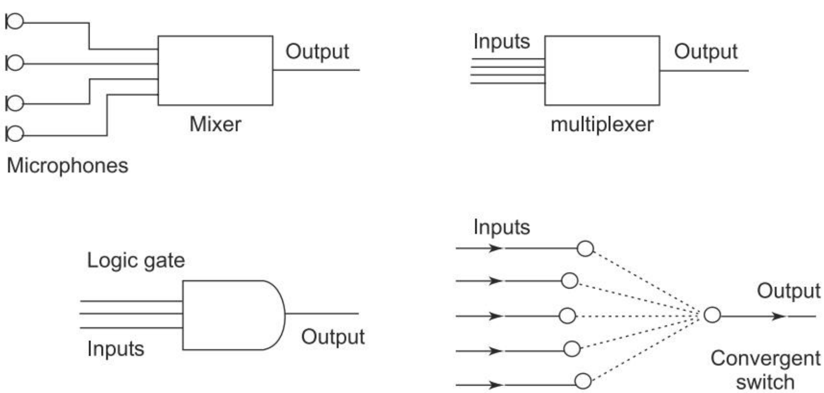


Fig. 17.3 *Examples of convergent circuits*

Fault diagnosis in convergent circuits is done in the same way as in divergent circuits. Each input circuit is isolated to find out whether the isolated stage is faulty or not.

Closed-Loop (or Feedback) Circuits

Feedback circuits form a closed loop. Examples of closed-loop circuits are given below. These are shown in Fig. 17.4.

- Negative feedback
- Servo-controlled devices
- Positive feedback
- Automatic gain control

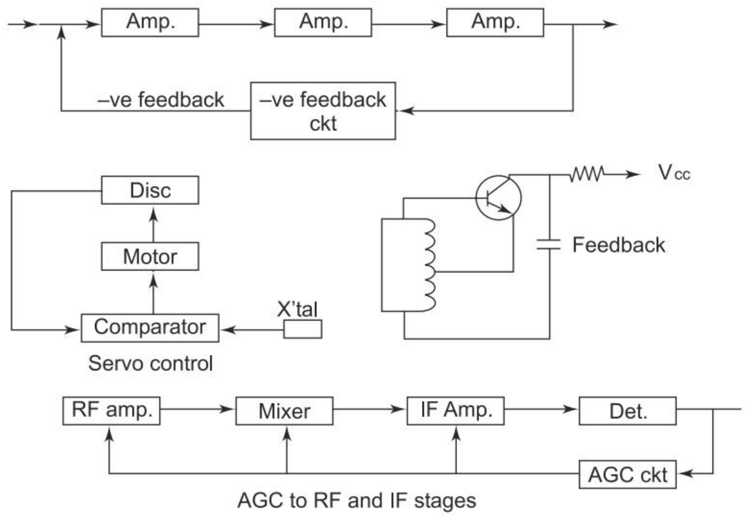


Fig. 17.4 *Examples of closed loop circuits*

As the circuit forms a closed loop, a fault, at any point in the loop appears everywhere in the closed circuit. To locate such a fault, the best strategy would be to isolate the feedback circuit, and then expect a distorted and higher output signal for negative feedback and no signal for positive feedback. If the expected signal is not present, the exact fault can be found out by tracing the signal within the stages other than feedback circuit. If the signal is found as expected, the fault would be in the feedback circuit itself.

17.8 IDENTIFYING THE FAULTY COMPONENT IN THE FAULTY STAGE

A fault within a stage can be checked by the following techniques.

1. Measurement of resistance, dc voltage and ac signal voltage (or gain) at the specified test points
2. Substitution of the suspected component by a good one

17.9 SOME COMMON FAULTS IN COMPONENTS

Passive and active components may become open or leaky or short. Causes for components to become faulty are many. But, voltage fluctuation, transients (sudden surge of voltage or current), temperature variations, moisture, dust, grease between tracks of PCB, dry or cold solder joints, vibrations/shocks, mishandling, expiry of life, etc., are the most common causes for failures. Dust is the biggest enemy of an electronic equipment because it prevents radiation of heat from the ICs and other components. Some common faults are discussed below:

1. Resistor Open Current will be zero. Voltage at the supply end will be full, while voltage at the other end will be zero. (multimeter will show it).

2. Capacitor Open Ac signal will not pass through it (substitution is the quickest method to find an open capacitor).

3. Capacitor Short Dc will pass along with ac (multimeter will show it).

4. Capacitor Leaky Dc will pass along with ac (multimeter will show it).

5. Trimmer Capacitor Disturbed Alignment/gain will be affected adversely (Gain measurement will show it).

6. Inductor Open Neither ac nor dc will pass (multimeter will show it).

7. Inductor Short Ac voltage across it will be zero (measurement of ac signal will show it). Multimeter at the lowest range may also show it, but due to very low resistance of the coil, the result may not be conclusive).

8. Transformer Leaky Between Primary and Secondary or Between a Winding and Core It will show up as loss in power. Transformer may become heated (multimeter at high resistance range or megger will show it).

9. Diode Open No current will pass (multimeter will show it).

10. Diode Short No diode action. It will act as a small resistance, giving rise to heavy current (multimeter connected in reverse bias condition will show it).

11. Transistor Open No current through transistor (voltage measurement by multimeter will show it).

12. Transistor Short No transistor action and heavy current (voltage measurement by multimeter will show it).

13. Transistor Leaky Temperature will rise, current will increase and finally the transistor becomes open (multimeter will show it).

14. Current Gain (β) of a bi-polar junction transistor, or transconductance (g_m) of FET is low (curve tracer will show it).

15. Open or short or leakage in SCRs, UJTs and other semiconductor devices will show up in resistance measurements by a meter. Characteristics can be checked more accurately by curve tracer.

16. Digital IC Output stuck at 1, or at 0, or is unstable (logic probe will show it).

17. Analog or Linear IC (Op-amp.) Output stuck at zero, output stuck at + Vcc, output stuck at –Vcc, IC becomes open or short, CMRR low, gain low, +Vcc or –Vcc grounded, input grounded (all these faults can be shown by a multimeter, CRO, linear IC tester).

18. Computer Bus Defective Logic analyser will show defects in buses.

Common Faults in PCB

1. Solder bridge between adjacent tracks
2. Pinholes
3. Break in plated–through hole (PTH)
4. Loose connection in connectors
5. Dry or cold solder joint

(Multimeter and current tracer will show the above faults)

Example 17.1 *In the circuit shown in Fig. 17.5, determine the likely faults and give the logic behind your answer for the following cases*

i. *Voltage at the collector of Q_1 is equal to Vcc*
ii. *Voltage at the collector of Q_1 is abnormally low*
iii. *Signal voltage at the collector of Q_1 is normal but no signal at the base of the next stage (Q_2). All dc voltages and resistors are OK*
iv. *Voltage at the collector is zero.*

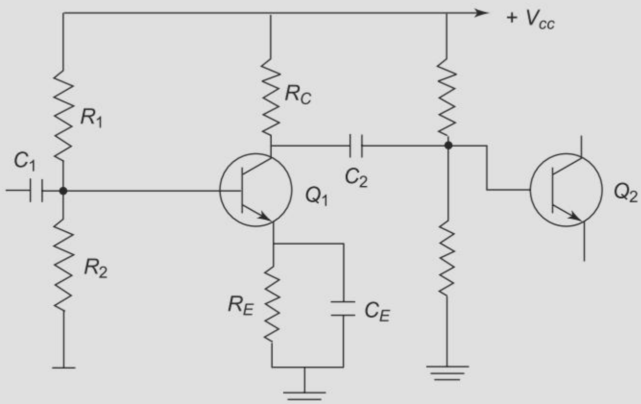


Fig. 17.5 *A circuit diagram for reasoning out the faulty components for Example 17.1*

Solution Likely faults and reasonings are given in Table 17.2.

Table 17.2

Likely Fault		Reasoning
i. (a)	R_1 open	Base voltage will become zero and hence transistor will not conduct. So no current through *Rc*, giving full *Vcc* at the collector
(b)	Transistor open or R_E open	I_C will be zero, giving full *V*cc at the collector.
ii. (a)	R_2 open	High voltage at base of Q_1 sill increase forward bias, and hence too high current through R_c. So more voltage drop in R_c, and hence low voltage at the collector.
(b)	C_E short	Voltage at emitter will be zero, resulting in increase in forward bias of the npn transistor, and so more current through R_c, and hence low voltage at the collector.
iii.	C_2 open	An open coupling capacitor will not pass ac signal. It will not affect dc voltages.
iv.	R_c Open or *V*cc disconnected	Collector has no continuity with *V*cc, and hence voltage at the collector will be zero.

17.10 INTERMITTENT FAULTS

Sometimes a fault might be intermittent. Dry solder joint, heating, loose connectors, etc. may cause intermittent faults. Such faults are most troublesome to locate. When such set is sent to workshop for repair, it can come back with a note "no fault found". Heating by blower may provoke an intermittent fault to become permanent and then the fault diagnosis can be done easily. Storage oscilloscope is helpful in determining the intermittent faults.

17.11 TROUBLESHOOTING IN A POWER SUPPLY UNIT

Of all the stages of an electronic equipment, the power supply unit is the most vulnerable to fault as it has to withstand total current of all the stages. If there is a short circuit in the power-supply line of any stage, the maximum current will flow, not through that stage, but through the components of the power supply unit, i.e., filter, rectifier diodes, transformer, fuse and switch.

Power supply is tested for the fault under the following two conditions:

1. Cold test
2. Hot test

Cold Test

In this condition, power is kept off and resistance at the common supply point (divergent point) is measured. If it shows short, the defect is either in the power supply itself (filter capacitor might be shorting) or in the supply line of any of the stages being fed by the power supply unit. The fault can be detected by the isolation technique explained below.

Isolation Technique Each stage is isolated in turn from the common point. When isolation of any particular stage removes the fault, the isolated stage is faulty and a further check can be made to identify the faulty component causing short.

When a break is made at *X* as shown in Fig. 17.6, the power supply unit gets isolated from the rest of the equipment. If short (measured at *P*) is not removed, the fault is in the power supply unit (rectifier, filter capacitors, bleeder). If short is removed, the fault is in any of the stages (I to IV) of the equipment.

When a break is made at *A*, Stage I gets isolated. If this stage is OK, the resistance at *P* will still show short. Similarly, other stages can be isolated one by one and checked. If measurement at *P* does not show the short when a break is made at *C*, this indicates that the short is in stage III, the driver amplifier. A further check may show that the decoupling capacitor (not shown in the Figure) might be shorting which is quite a common fault.

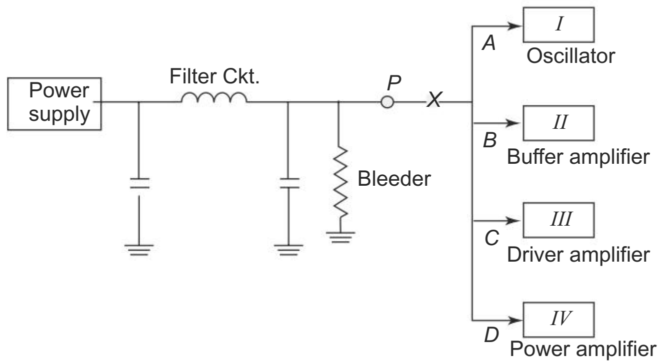


Fig. 17.6 *Identifying the faulty component in the divergent circuit of a power supply unit*

Hot Test

This test is conducted after making the power supply on. A typical basic circuit diagram of a regulated power supply unit is shown in Fig. 17.7. The dc voltage is measured at the output terminal. It can be normal, low or high. If normal, the power supply is alright, else the faulty component is to be identified. Voltage will be low if the input filter capacitor (C_1) is open, or one diode of the full-wave rectifier is open. When a diode is open, the full wave circuit behaves as a half-wave circuit, reducing the output voltage and increasing the ripple. The voltage can become zero, if any series component (like switch, fuse, transformer, choke) is open. The cold solder joint to chassis at the transformer centre tap will also make the circuit open. If the mains indicator (generally LED) is not glowing, it would mean mains voltage is not available. Non-availability of mains voltage will result in zero output voltage. In a regulated power supply, if the series transistor (control transistor) becomes non-conducting, the output voltage will be zero. If the zener diode becomes open, the emitter of Q_2 (error transistor) will become more positive. It will decrease current though Q_2 and hence, voltage at the collector

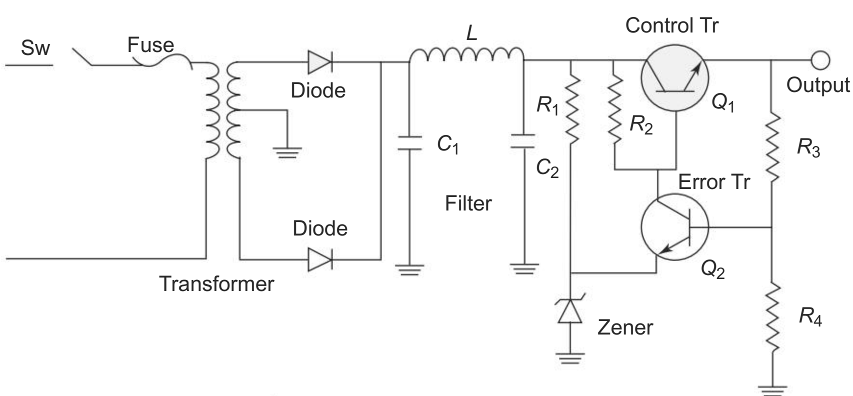


Fig. 17.7 *Basic circuit of regulated power supply*

of Q_2 and therefore, at the base of Q_1 will be high. It will result in heavy current through the series transistor (Q_1). Output voltage will increase temporarily, and finally the heavy current will burn the series transistor and the output voltage will become zero. If the zener diode becomes short or if R_1 becomes open, current through Q_2 will increase, resulting in low voltage at the base of Q_1 and hence low current through the load which would give low voltage at the output..

In a switch-mode power-supply unit, regulation is achieved by using a fast switching transistor, the period of switching being controlled by controlling the duty cycle of rectangular pulses. If the pulse generator circuit gives constant voltage, the switching transistor may be damaged. If duty cycle does not vary, the regulation will not take place. The fault can be diagnosed by checking the duty cycle with the help of a CRO.

Troubleshooting procedure of a power-supply unit is given in the flowchart, Fig. 17.8.

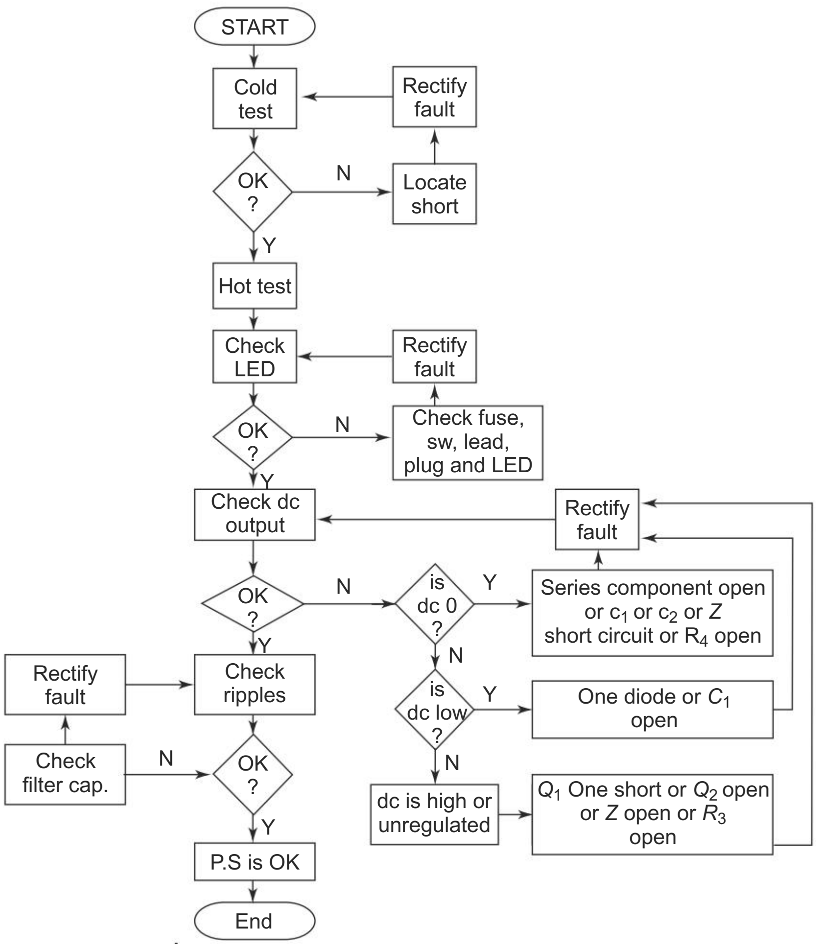


Fig. 17.8 Flowchart for troubleshooting in a power supply unit

17.12 TROUBLESHOOTING IN A PUBLIC ADDRESS SYSTEM

A public Address (or PA) system consists of microphones, mixer stage, voltage amplifiers, power amplifier, loudspeakers, and power-supply unit (shown in Fig. 17.9).

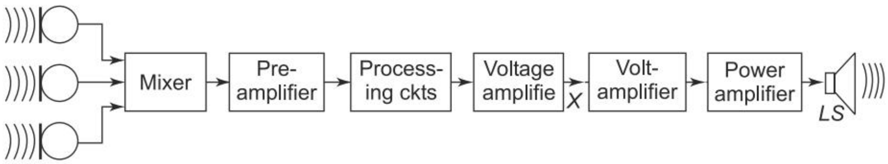


Fig. 17.9 *Block diagram of a PA system*

The flowchart for troubleshooting in a PA equipment is given in Fig. 17.10 and corrective maintenance in Table 17.3.

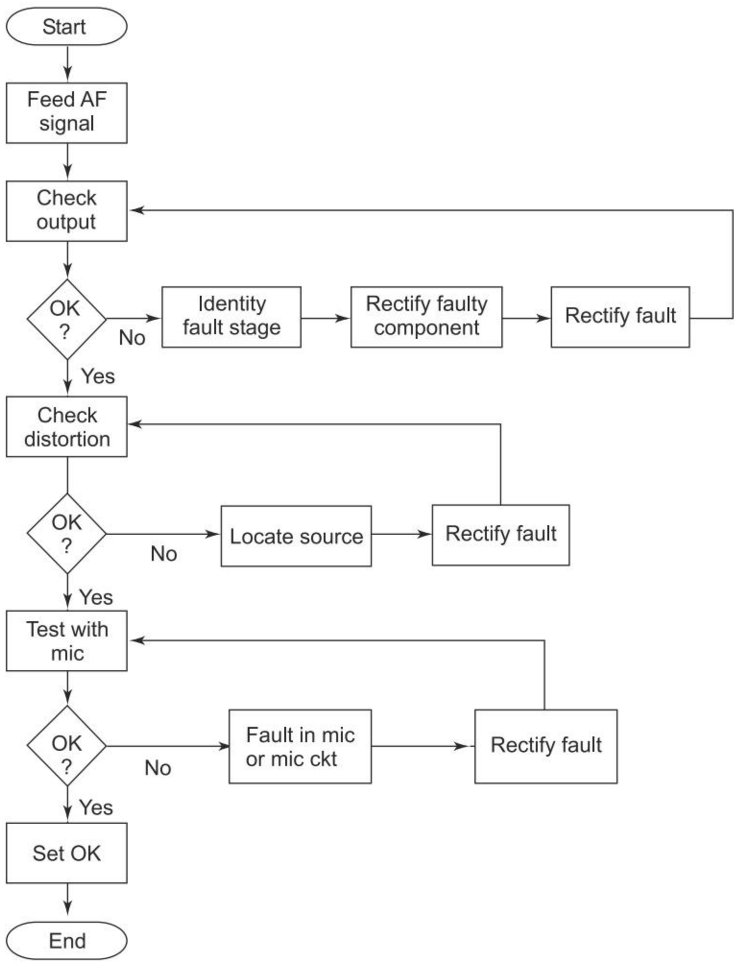


Fig. 17.10 Flowchart for troubleshooting in PA equipment

Table 17.3 *Trouble-shooting or corrective maintenance table for PA system*

Symptom	Likely fault	Remedy
No sound output from loudspeaker, no click, no hum or hissing sound. Set completely dead. No voltage.	Set is not getting power supply. Fuse might have blown off, or mains lead/plug disconnected or power supply unit defective.	Check the mains line and power-supply unit to locate the fault. Rectify the fault.
Resistance and dc voltage measurement inside the set normal but no hissing noise or signal output.	Loudspeaker leads might have become short-circuited or broken or LS coil short.	Check the short by resistance measurement (Remember 16/02 feeder gives 6Ω per 100 m), and check the feeder by capacitance measurement (typical capacitance of feeder is 30 pF per meter).
Noise present, but no signal output. Resistance and dc voltages are alright.	The presence of noise shows that the pre-amplifier and other stages are alright. Hence microphone lead open or shorted to shield.	Check by meter and rectify the fault.
Power supply transformer getting overheated.	Short circuit in the power supply unit, or some stage short circuiting the power supply line.	Check the divergent circuits by isolation method, and locate and remove the fault.
No sound in one LS, normal sound in other speakers.	Speaker defective, or the fault is in loudspeaker lead, or connector	Check the loudspeaker circuit by resistance measurement and remove the fault.
Output low. Resistance and dc voltages are correct.	Emitter bypass capacitor open.	Measure gain of stages by split half method and thus identify the faulty stage. Replace the faulty component.
Excessive hum.	Inadequate filtering in the power supply due to open or leaky capacitor of filter circuit. Ripple picked by base circuit of an amplifier stage. Coupling of mains leads with microphone leads, or the matching transformer near ac loads or near ac transformer.	Put a good capacitor across the filter capacitor. If hum is removed, the filter capacitor is open, replace it. A leaky capacitor can be checked by resistance measurement. If power supply is OK, feed signal from AF generator. If ripple is present, it is being picked up by base circuit. If ripple is absent, it is picked up from mains. Locate the source of ripple and rectify the fault.
Excessive distortion	Positive feedback causing high-pitched whistle. Sometimes, the output may be supersonic. Such supersonic oscillations will not be heard but they will overload the amplifier and cause distortion. Overloading of any amplifier stage changes the characteristics of an active device. Also, negative feedback circuit may be defective.	Check the source of distortion by short-circuiting input of stages one by one and thus identify the defective stage or undesired coupling and rectify it. Check the negative feedback components and replace the faulty component.

17.13 TROUBLESHOOTING IN A STEREO AMPLIFIER

A stereo amplifier consists of two independent channels of amplification from microphone to loudspeaker as shown in Fig. 17.11. Faults in the stereo system may be no output, weak output, distortion, noise, cross-talk, unbalanced output, etc. The general procedure of fault-diagnosis for stereo amplifer is given in Fig. 17.12 (flowchart).

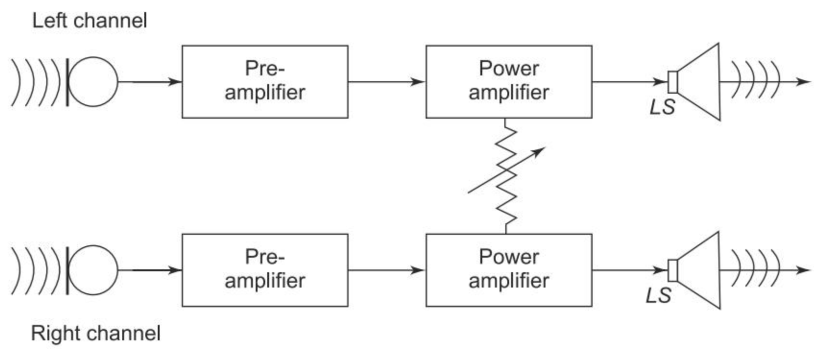


Fig. 17.11 *Basic stereo-amplifier system*

After ensuring that power supply is alright, it has to be found out as to which section is faulty, whether left channel or right channel, and then in that channel whether it is the microphone or loudspeaker or an amplifier stage. Input devices, viz., microphone, tape recorder, disc recorder, etc., are connected to the pre-amplifiers through a functional convergent switch. This functional switch can identify whether any input device is defective or amplifier is defective. If there is no output at any position of the function switch, amplifier is defective. If there is no output for any one input device then that input device would be defective.

If output is present in both the channels but it is not getting balanced, the defect is in the balance control potentiometer. If output is not present in any of the channels, either the power supply is defective or the ground of the balance-control potentiometer is open. In case of only one channel being defective, it can be found out as to which channel it is by keeping one channel off and listening to the other channel.

Identification of Faulty Stage in Amplifiers Faulty stage of the defective amplifier can be identified by signal injection and tracing method, or more easily by making use of the stages of a good channel to replace the stages of the faulty channel by an external test lead as shown in Fig. 17.13.

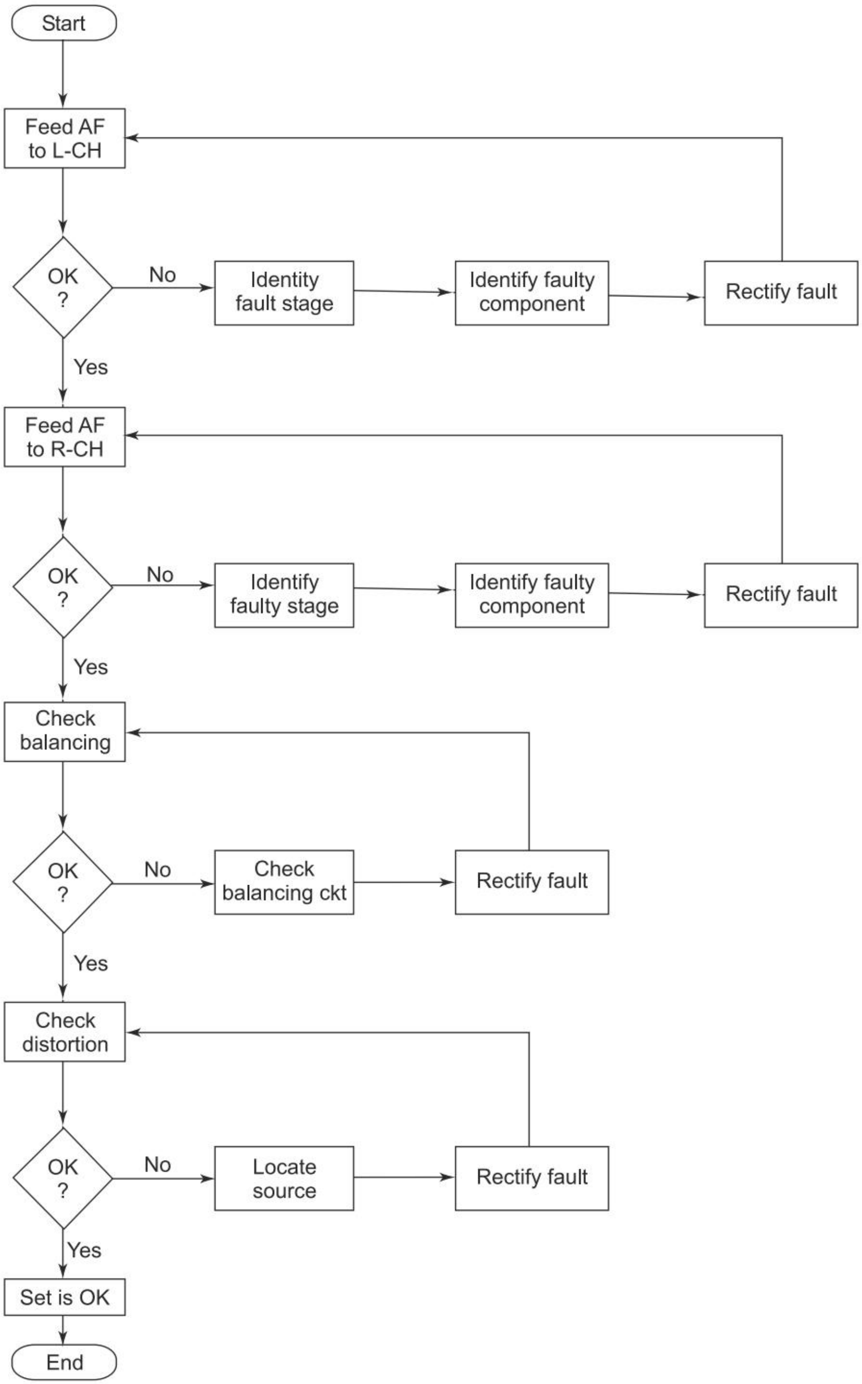


Fig. 17.12 *Flow-chart for troubleshooting in stereo amplifier*

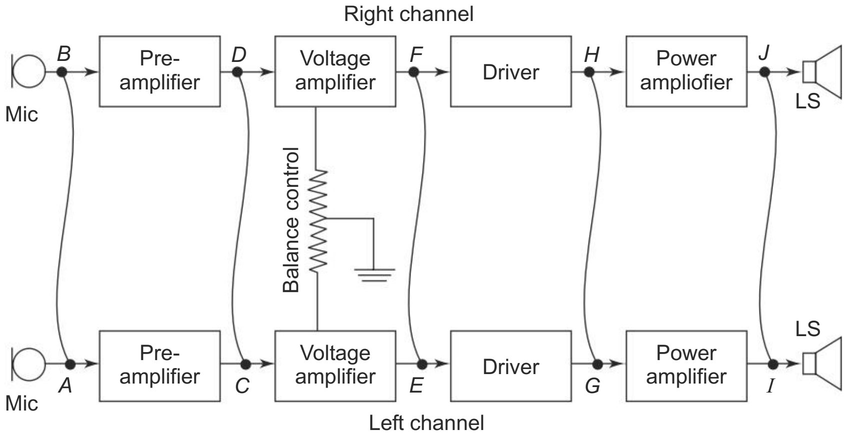


Fig. 17.13 *Testing of defective channel by good channel of the same stereo amplifier*

Let the right channel be defective and the left channel be OK Feed a signal to the right channel. If output in the left channel is normal when the test lead is connected from *A* to *B*, input devices of the right channel are alright. Now the lead is connected from *C* to *D*. If output is correct, the pre-amplifier is alright, otherwise it is defective. The other stages can be checked similarly by connecting the test lead from *E* to *F*, *G* to *H* and *I* to *J* until the defective stage is identified. If the signal up to J is found normal, the fault may be in the loudspeaker or its leads.

Identification of Defective Component Voltage and resistance measurements at pin point of active devices or specified test points will identify the faulty component or connection. The resistor may be open, and the capacitor or inductor or a semiconductor device may be open or short. A solder joint may be dry or cold, or there may be a solder-bridge between closely spaced tracks of PCB. Most of these faults will be indicated by resistance and dc voltage measurement. Some may require measurement of an ac signal, say for open capacitor or shorted inductor. Some typical faults are given in corrective maintenance Table 17.4.

Table 17.4 *Typical faults and remedies in a stereo system: (Corrective maintenance table)*

Fault symptom	*Likely Fault/Cause*	*Remedy*
1. No output in any position of the function switch on one channel. The other channel remains good.	The function switch enables the input devices (microphones, tape recorder, record player, FM tuner) to be connected to the stereo amplifier. All the input devices cannot be faulty. Hence, the fault is in the amplifier of the concerned channel.	Check the amplifier, stage by stage, by signal substitution, or by signal injection and tracing, Identify the defective stage and then proceed to find the faulty component in that stage by voltage/resistance measurements. Replace it.

(Contd.)

(Contd.)

2.	No output in one position of the function switch. Normal output in other positions.	As the output in all positions of function switch except one position is normal, the amplifiers are alright. The input device pertaining to the particular position of the function switch is defective.	Check the faulty input device. The microphone may be faulty. Replace it. If it is a record player, check for worn-out stylus. If the faulty device is the tape recorder, check if the heads are dirty and magnetised. If the faulty device is the FM multiplexer, the decoder should be checked with the help of a stereo-generator. Rectify the fault.
3.	Weak output in both the channels in all positions of function switch.	Low supply voltage.	Check the power supply unit and rectify the fault.
4.	No output on any channel.	No power supply or defective balance control.	Check power supply, the fuse, mains lead, mains plug, or short in power supply line and remove the fault. Ground connection of the balance control potentiometer may be open. Check it and rectify the fault.
5.	Interference from other signals.	Ground connection of rf bypass capacitor open, or the capacitor itself open. Strong local signals of radio transmissions of other services (e.g. Army, Police, Broadcast Station, etc.) may be picked up by an audio amplifier. Signals, being very strong, may overload the amplifier, and hence may get detected due to nonlinear characteristics for overloading conditions.	Check rf bypass capacitor and its ground connection. Rectify the fault.
6.	Distortion in the output.	Negative feedback circuit defective. Self-oscillation in some amplifier stage because decoupling of the capacitor becoming open.	Check the negative feedback circuit, particularly the capacitor and correct the fault. Find out the oscillating stage by short circuiting the stages one by one, and remove the fault.
7.	Hum in the output of both the channels.	Ripple is high.	Check the voltage-supply line for open filter capacitor. Correct the fault. Check proximity between mains power and system leads.
8.	Too much 'noise' in the output.	It is due to irregularity in the grooves or in tape-magnetisation.	Dolby method is used to improve signal-to-noise ratio and hence, to reduce hissing noise. Check these circuits and remove the fault.
9.	Balancing not possible.	The balance circuit may be defective.	Check the balance potentiometer. Clean it if necessary. Replace if not corrected by cleaning.

17.14 TROUBLESHOOTING IN AUDIO CASSETTE TAPE RECORDERS

General troubleshooting procedures apply to the faulty tape recorder equally well. After analysing the user's report, making physical inspection, and ensuring the power supply to be good, the next step in troubleshooting is identification of the faulty section by the functional-area approach technique explained below.

A tape recorder consists of the following sections:

1. Recording section
2. Playback section
3. Tape transport section

The block diagram of a cassette recorder is given in Fig. 17.14.

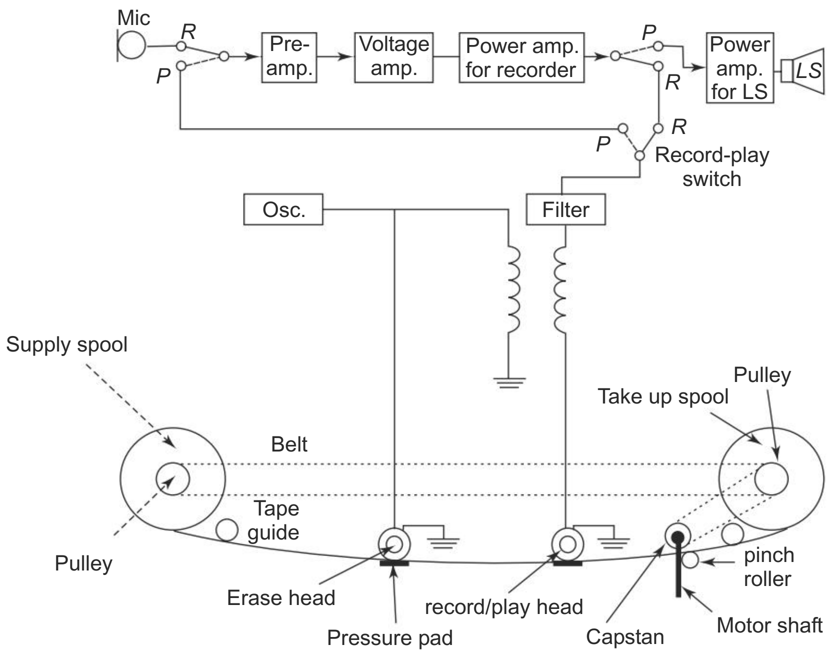


Fig. 17.14 *Block diagram of an audio cassette recorder*

The record/play switch can identify whether the record section is defective or the playback section. The switch is first kept in the 'playback' mode, and a pre-recorded cassette, known to be good, is played. If the output is normal, the playback section is alright. If there is no output or if the output is weak, or distorted, or noisy, troubleshooting is needed in the playback section. If the playback section is normal, the switch is kept on the 'record' position and recording is done on a blank cassette and checked in the playback mode. If OK, the recording section is alright, otherwise it needs troubleshooting. In cassette recorders, amplifiers are common to recording and playback modes, and so if the recording or playback output was normal, the amplifier part is alright, and the defect may be in the record-play switch or in the microphone circuit.

Defect in the tape transport section would be evident if the tape does not move at all, or moves unsteadily or with jerks. Unsteady speed produces wow and flutter and body vibrations of motor produce rumble noise.

Identification of Faulty Stage Once a faulty section has been identified, the stage which is defective in that section should be determined. It can be identified by injecting a signal at the input and checking the output by the split-half method. For the playback section, the source for injecting signal is a pre-recorded tape. The stage prior to the point where output is not available is faulty.

Similarly, in the record section, the source for injecting signal in the input can be an audio signal generator, and the output can be measured at different points of the record position. If the output with signal generator is normal, the microphone or its circuit may be defective.

Identification of Faulty Component After a stage has been identified to be faulty, it has to be checked thoroughly to identify the faulty component. The normal procedure is to measure resistance and dc voltages at pin points of the active devices and the test points, if any, specified by the manufacturer. Most of the faults shall be revealed by voltage measurements alone. Logical analysis of the measured values will identify the faulty components in the record and playback electronic circuits.

Head gaps are also quite troublesome. Most of the faults in tape recorders pertain to iron oxide particles from the tape depositing into the gap. The defects in the head gaps (worn out, dirty, magnetised, lack of proper pressure against the tape) show up in the output of the playback in the form of poor high-frequency response and more noise. Head gaps should be cleaned periodically by pure cotton soaked in alcohol.

Corrective maintenance for a tape recorder is given in Table 17.5 for some typical faults.

Table 17.5 *Corrective maintenance table for cassette recorder*

	Symptoms	Likely fault	Remedy
1.	No output on playback, no recording, no erase.	Power supply output is zero.	Check mains leads, fuse, switch and plug. If OK check power supply and rectify the fault.
2.	High-frequency response is poor in playback.	Dirty or worn-out head.	Check the head. If worn-out, replace it, otherwise clean it.
3.	Erase normal, playback normal, but no recording.	Microphone or its lead or jack defective. Switch at record position not making contact.	Check microphone and its connection to the amplifier. Rectify fault. Check and clean the switch contacts in the record position of switch.

(Contd.)

(Contd.)

4.	Erase is normal, but no playback. Recording OK.	Loudspeaker, or its lead or jack defective, contacts in playback position of switch dirty.	Rectify fault in the loudspeaker or its leads. Clean the switch contact.
5.	Erase is normal, but no playback, no recording.	One of the common amplifier stages defective; ac bias missing.	Trace the faulty stage by signal injection and rectify fault. Check the bias oscillator connection and rectify the fault.
6.	Recording distorted.	The ac bias missing.	Check the bias oscillator connection and rectify the fault.
7.	No erase	Erase-head defective or it is not pressing the tape properly. rf signal for erase head missing.	Check the erase head and clean if necessary. Check rf signal circuit and rectify the fault.
8.	Tape recorded on other machines gives distortion.	Record/play head is tilted.	Correct the head tilting by aligning azimuth of the head.
9.	Hum noise.	Ripple excessive in power supply.	Check input filter capacitor and diode. Rectify the fault.
10.	Fuzzy sound.	Self-oscillation in amplifier.	Check each stage by shorting the input and thus isolate the source and rectify the fault.
11.	Playback output is distorted.	Dirty head, or some stage self-oscillating.	Clean the head. Check the amplifier for distortion and rectify the fault.
12.	Low pitch noise on playback (wow and flutter).	Motor speed not steady.	Check the servo-control circuit and rectify the fault. Capstan or pressure roller may be worn out. Replace.
13.	Rumble noise on playback.	Body vibrations in motor.	Check shock absorbers and rectify the fault.
14.	Tape does not move.	Loose or broken belt. Defective motor. Pressure roller not pressing.	Check motor, belt and pressure roller and rectify the fault.
15.	No rewinding	Driven wheel not pressing on the flywheel, idler wheel not clean.	Check the wheels and rectify the fault.
16.	Tape is not winding on the take-up spool.	Take-up shaft not pressing properly. The shaft may be oily. Transmission belt may be loose or broken, faulty tape.	Check the shaft, belt and tape and rectify the fault.

Troubleshooting steps for cassette recorder are shown in the flowchart in Fig. 17.15.

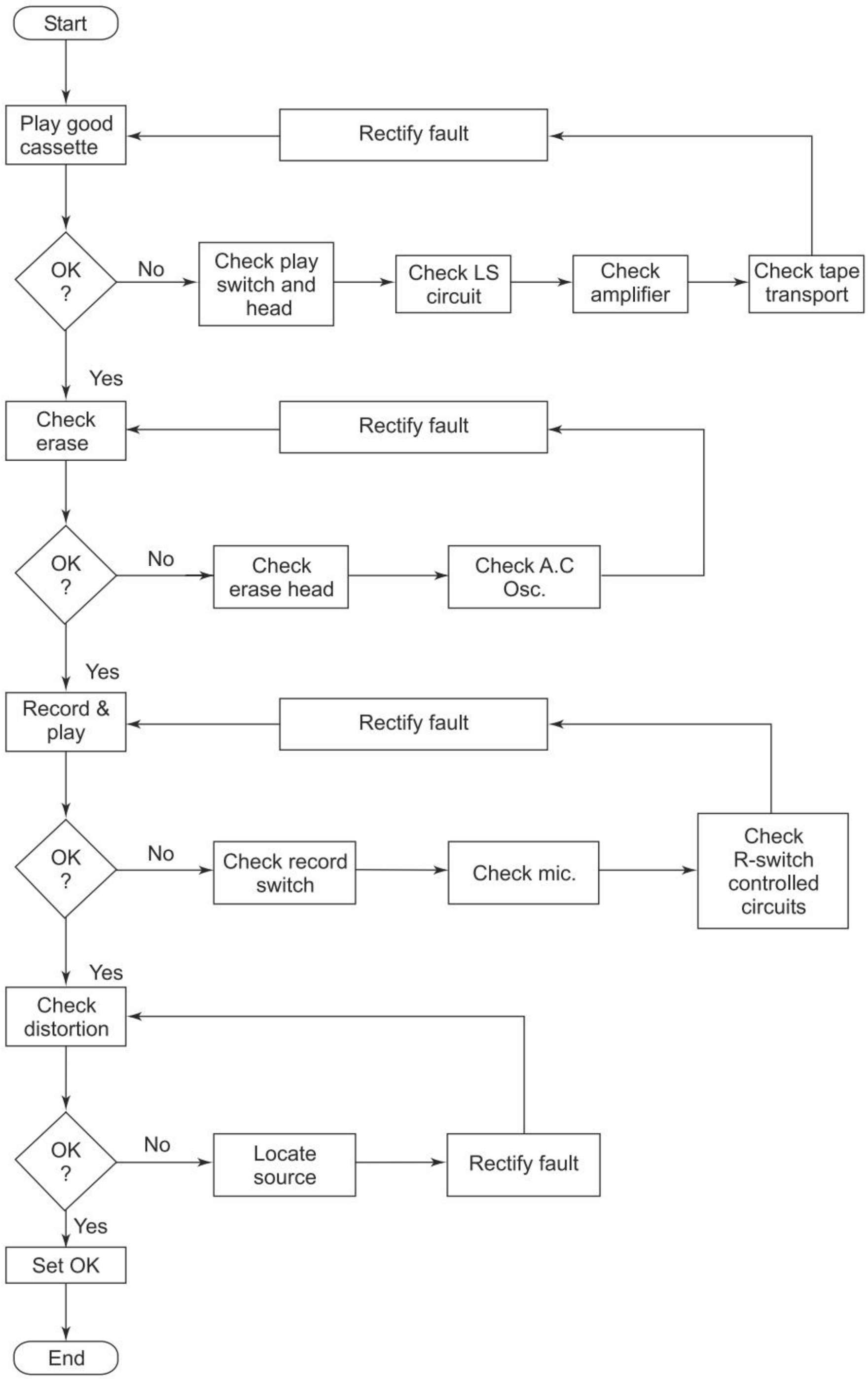


Fig. 17.15 *Flowchart for troubleshooting in a cassette recorder*

17.15 TROUBLESHOOTING IN VCRS

A VCR is a complex equipment, having a large number of controls, jacks and connectors. There are electromechanical parts like rotary and capstan motors, idle wheels, tape-guides, tension controllers, tape heads, etc. However, troubleshooting in a VCR is not as complicated as it appears to be by the complicity of controls. Most of the troubles are due to wrong settings of the controls or dirty and worn-out heads. Trouble inside the circuits can be diagnosed by usual troubleshooting techniques of signal tracing by the split-half method, voltage and resistance measurements and substitution methods. Most of the symptoms appear on the display unit and their causes can be determined logically by applying theoretical knowledge of the circuit functioning.

Test instruments for VCR are similar to those used in servicing of TV receivers. Bandwidth requirement is also the same. Special equipment needed for VCR servicing is the dual-trace delayed sweep oscilloscope. It is used for checking the head-switching. For determining the exact free running frequency of the rotary scanner and capstan motor, a digital frequency counter is needed. Manufacturers generally provide special fixtures to check and adjust mechanical parameters. Manufacturers also provide an alignment cassette. It contains video and audio recordings to allow adjustments for video head switching and playback video level. A video sweep signal in the alignment cassette permits adjustments of the frequency response for the playback pre-amplifier. The alignment tape is recorded on machines with the optimum specifications. If an alignment tape is copied, the copying machine will transfer its deficiencies to the copy-tape, and hence that tape cannot be reliable. Alignment tapes are expensive, and hence should not be used too frequently. It is desirable to have at least one 'work cassette' known to be good for day-to-day general checking. A *pattern generator* is a special type of signal generator with video patterns of the baseband signals. FM audio signals are also available in some pattern generators for testing the audio section.

The block diagram of a VCR is shown in Fig. 17.16.

Broadly, record and playback checks quickly eliminate either the record section or the playback section from the field of suspicion. A test cassette or a working cassette and a blank cassette are helpful for confirming which section is faulty. The inferences will be as mentioned below.

1. If playback is abnormal with own recording of VCR, but normal with a work cassette (pre-recorded tape known to be good), the recording section is defective.
2. If playback on the VCR of the own recorded cassette is abnormal, but playback of the same cassette on another VCR is satisfactory, the playback section is defective.

Many of the faults in a VCR pertain to heads. Noise is caused by dirt in the head gap of one or both of the heads. If half the picture is noisy, only one head needs cleaning. The heads can be cleaned by methanol. A cleaning tip, saturated with methanol, is rubbed across the head gap from side to side (and not vertically). Audio, erase and control heads are rubbed vertically for cleaning.

The picture quality gets degraded after about 500 hours of use because video heads are sufficiently worn out by then. These should be replaced when worn out.

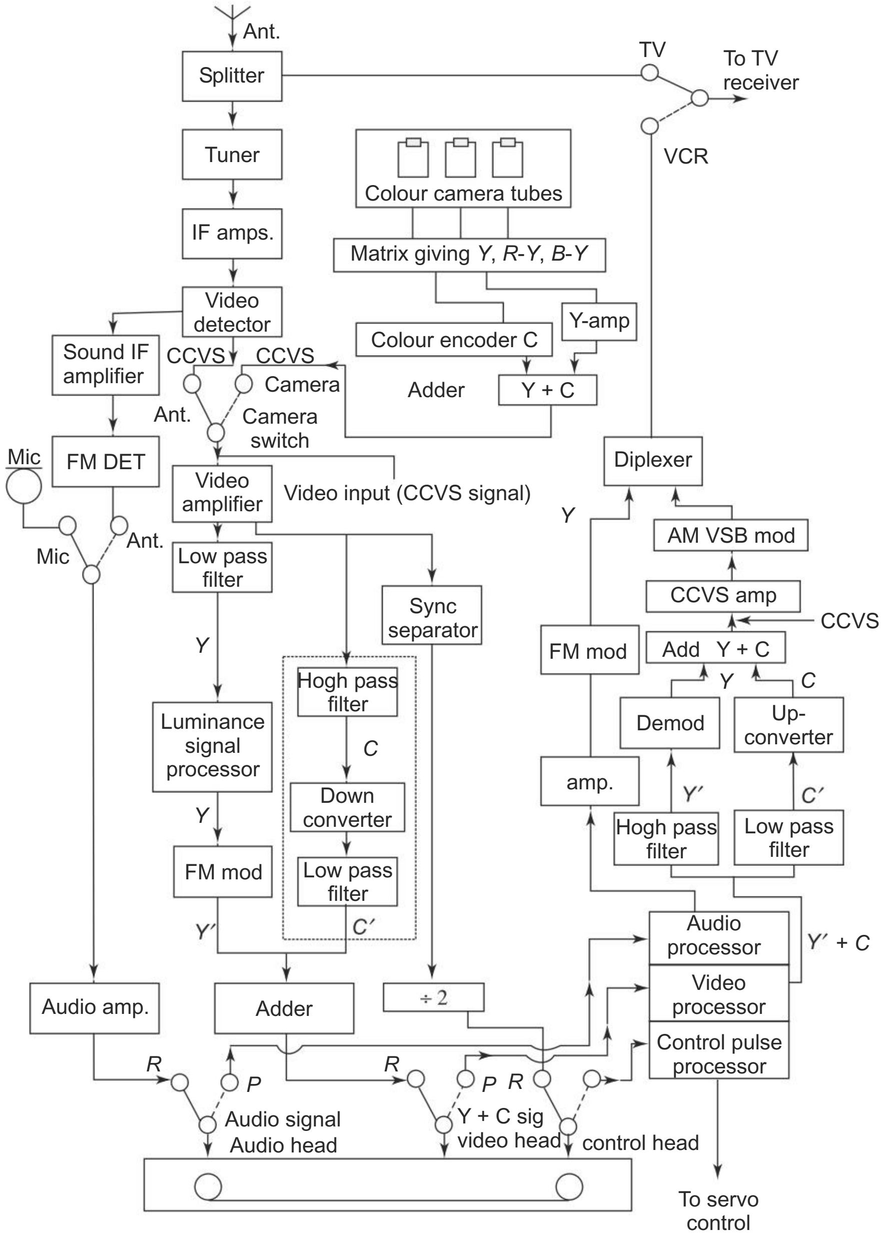


Fig. 17.16 *Block diagram of a VCR*

Some typical faults in a VCR are analysed in Table 17.6. Troubleshooting steps are shown in the flowchart, Fig. 17.17.

Table 17.6 *Common faults in a VCR (corrective maintenance table for VCR)*

Symptom	Probable cause	Remedy
1. VCR does not operate at all.	(i) Power supply is not available.	(i) Check ON-OFF switch. It should be pressed on. IF the fault persists, check fuse, mains lead and plug. (correct the fault so that power becomes available to VCR.
	(ii) Timer recording switch is ON and hence the VCR cannot be operated manually.	(ii) Switch off the timer recording switch.
2. Stops during rewinding.	(i) Tape counter memory switch is *ON*.	(i) Set this switch to *off* position.
	(ii) Tape is damaged	(ii) Cut the damaged portion of tape and splice.
3. TV telecast on desired channel cannot be seen.	(i) TV/VCR switch is set on VCR.	(i) Set the switch to TV.
	(ii) TV receiver defective.	(ii) Check the TV receiver and correct the fault.
4. Telecast cannot be recorded, but field recording is satisfactory.	(i) Antenna cable is defective.	(i) Check antenna feeder and fault if any, may be removed.
	(ii) VCR tuner, IF or detector stages may be defective.	(ii) Identify the faulty stage by signal tracing. Identify the faulty component by resistance/voltage measurements. Replace the faulty component.
5. Recorded picture is normal but no sound.	Tuner, IFs, video detector or further video path OK. Defect in sound path.	Check audio signal path and rectify the fault. Audio head may be dirty; clean it.
6. No sound during recording of telecast.	(i) Sound section is defective. (ii) Mic/RF switch on mic position.	(i) Check sound section stage by stage and identify the faulty component and replace it. (ii) Correct the switch position.
7. No picture on playback. Recording is normal, sound normal.	Video stages in playback section may be defective.	Identify the faulty component by voltage/resistance measurements and replace it by a good one.

(Contd.)

(Contd.)

8.	Noisy picture on playback but display of direct TV telecast is OK.	Video head dirty or worn out.	Clean or replace the video head.
9.	Picture OK. No sound or noise in sound in playback. Telecast sound is OK. Recording is OK.	(ii) Audio section of playback defective.	(ii) Identify the faulty component in audio section of playback and replace it.
10.	No tape movement.	Capstan motor may be defective.	Check it and remove the fault.
11.	No automatic stop when the tape reaches its end.	Photo-transistor may be defective.	Replace the photo-transistor.

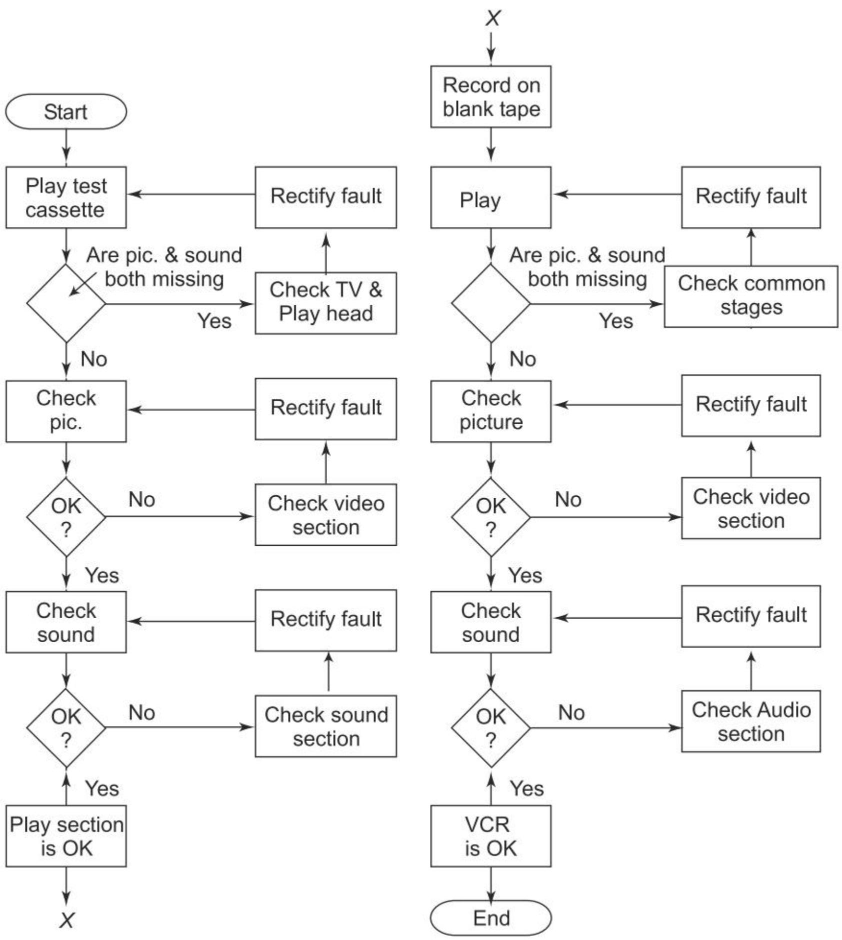


Fig. 17.17 *Flowchart for troubleshooting in a VCR*

As the TV telecast recording section is virtually a TV receiver, the faults in this section have the same symptoms, causes and remedies as in a TV receiver. The playback section consists of baseband amplifiers and RF modulators and its fault diagnosis is similar as in TV transmitter stages.

17.16 TROUBLESHOOTING IN DVD PLAYERS

A DVD player is a complex set of equipment having mechanical, optical and electronic components which are prone to faults. But fortunately the situations are usually simple, although not self-evident. Many glitches in the disc may disappear by a simple process of 'eject, reseat and reinsert' the disc.

Incompatibility due to geographical location, although not a fault, will not allow the DVD to work. TV standards of scanning, colour encoding and decoding, bandwidth, etc. are different for different countries. A player designed for England will not work in France. One designed in America will not work in Europe and in the countries of Indian Subcontinent. Hence, while using a DVD player, one must ensure that the Disc, the player and the TV monitor, all are compatible. Also with wrong setting of the channel in the TV receiver, the player's output would not be displayed.

A DVD may have an aspect ratio of 4 : 3, but with the player, set for 16 : 9 ratio, the images would appear thin and tall in the display. Similarly, DVD using MPEG compression may not be audible in the receiver which cannot decode MPEG code.

When a disc is designed to have provision of 'Digital Theater System' (DTS) and the player is not able to produce the DTS, the common fault would be 'DTS disabled' on the DVD player. Many players have DTS output turned off by default (normal condition by the manufacturer); enabling the player for DTS would remove the fault.

In some players, a plastic screw is fitted on the bottom of the player to save it from transport vibrations. The player will not work if this lock is not opened.

The above points show that the monitor or the player would not recognize the disc if the corrective action is not taken.

Another common fault is that the disc may stop playing half-way through. This may happen in dual layered DVDs. The pause of half-a-second is normal, but if it stops working for more time, the DVD needs replacement. There is no remedy for this; the disc is ruined and is to be thrown out.

Many faults pertain to mechanical connections. Cables and plugs may become loose. Unplug the connection, clean the contacts and refix. In most of the cases, the fault would disappear.

When video picture splits or freezes, DVD or the lens may be dirty. A dirty disc or lens coated with dust, tobacco smoke or grease is the number-one cause of common problems like

- disc being not recognized
- Audible noise

- Erratic tracking
- Sticking or skipping

For this, cleaning of the lens/disc is the first thing to be done. A soft cloth or paper towel moistened with water, is used to clean the disc and lens. The disc should be wiped from centre to edge and dried, using a lint-free cloth. Proper cleaning will fix most of the problems. DVD must be cleaned even if it is brand new and look pristine.

IF DVD player is found totally dead, it may be safely presumed that the electric power is not available to the player. Similarly, if the remote controls unit does not function, its cell might have discharged. Remedy is obvious as in other electronic equipment, that is, check the wall-outlet of power, plug and cable. For remote control unit, change the cell.

Substitution method is the best and quickest to find out what is faulty, the disc or player or the monitor. If a DVD does not work and is suspected to be faulty, another DVD (known to be good) many be inserted. If it works alright, the suspected DVD is faulty. If the good DVD also does not work, then the fault is in the player, not in DVD. Similarly, to find out whether the player is faulty or the monitor, the player may be connected to a monitor, known to be good. If it works, the monitor was at fault; if not, the player was faulty.

These steps are shown in Flow-chart (Fig. 17.18).

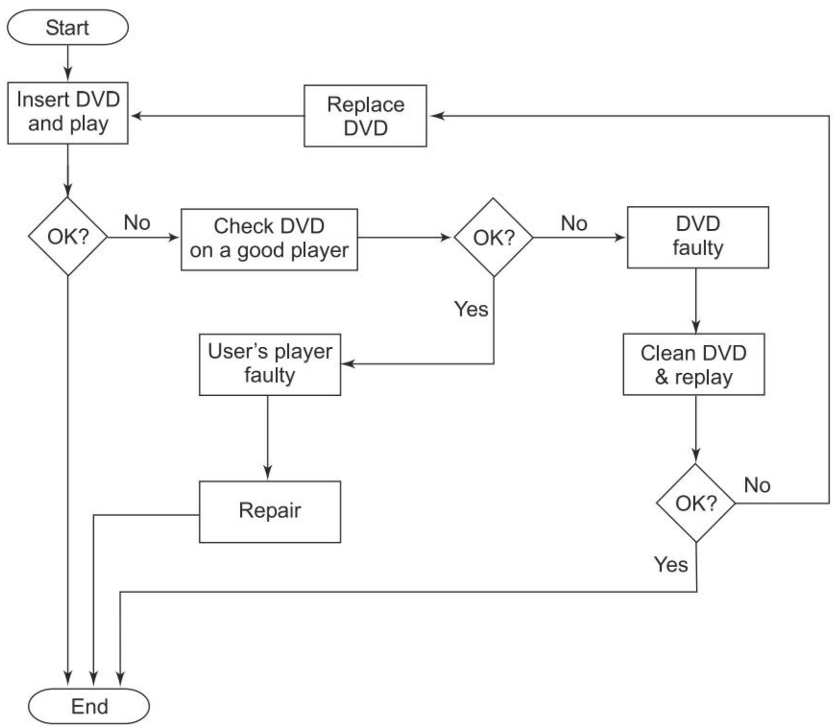


Fig. 17.18 *Flow-chart for troubleshooting a DVD player*

SUMMARY

- Salient features of modern electronic equipment are miniaturization and high reliability.
- Miniaturisation causes some potential problems in diagnosis and corrective maintenance.
- Maintenance policy is so planned that MTBF remains high and MTR low
- Maintenance aids are service manual, test instruments and special tools.
- Systematic troubleshooting requires that after analyzing user's report and making physical inspection, the faulty section should be isolated by seeing the output. Then the faulty stage should be identified by signal injection and tracing by the split-half method and by isolation technique.
- After determining which stage is faulty, the actual component or connection which is defective should be found out. Its cause should be reasoned out.
- The above methods can be adopted using intelligent and logical reasoning of symptoms and of resistance and voltage readings.
- Lastly, servicing of the set may be done by thorough cleaning and oiling (wherever required), replacing parts which are wearing out, aligning and bringing the set to the original specifications within tolerable limits.
- Troubleshooting in a power supply unit, PA system, stereo amplifier, audio cassette recorder, VCR and DVD player has been discussed.

Review Questions

1. What are the salient features of modern electronic equipment.
2. Describe the potential problems to be faced in maintenance. Discuss the correct maintenance policy.
3. Describe the steps which are needed for systematic troubleshooting of electronic equipment.
4. Write short notes on the following:
   (i) Split-half method
   (ii) Isolation technique
   (iii) Functional-area approach
   (iv) Cold test
5. Write down how you will locate faults in a power supply.
6. Write down the troubleshooting procedure for locating faults in PA amplifiers.

7. Explain how you will locate the faults in a stereo amplifier.
8. Describe the troubleshooting procedure in detail for an audio cassette recorder.
9. What are the common faults in a VCR and how will you proceed to repair a faulty VCR?
10. Draw the flowchart for repair of any one audio system.
11. Draw flowchart for repair of a VCR.
12. Prepare typical corrective maintenance tables for (i) tape recorder, and (ii) VCR.
13. Explain the steps for checking a DVD player.
14. Discuss common faults in a DVD player.

Short-Answer Questions

1. What are the likely faults in the following cases?
   (a) Excessive ripple in the output of power supply.
   (b) Output of power supply is low.
   (c) When microphone is used, output of a PA system is normal, but when audio cassette is played, there is no output.
   (d) Output of PA system fuzzy when seen on CRO.
   (e) Power supply is OK but still audio output is low in a PA system
   (f) Left channel of a stereo amplifier is OK. When we connect input of LS of the left channel to the input of LS of the right channel, there is no output.
   (g) Output of two channels could not be balanced.
   (h) There is no output from both the channels in a stereo amplifier.
   (i) Recording in a cassette recorder is normal, but no output in playback.
   (j) Playback is normal but no recording.
   (k) Neither recording nor playback normal. Tape motion is alright.
   (l) Video cassette is OK, picture is OK, but there is no sound when the cassette is played.
   (m) Signal is OK when recorded in a VCR by camera, but there is no recording from a TV antenna.
   (n) TV programmes are received OK, but on playback of a VCR cassette, picture is missing although the sound is normal.
   (o) The TV receiver is neither receiving broadcast nor VCR programmes, although the TV receiver is quite OK.
   (p) DVD does not work on your player, but works alright on some other player.
2. Why is it desirable to perform cold test first in a power-supply unit?
3. What is the difference between troubleshooting, repair, servicing and maintenance.
4. What is the difference between corrective maintenance and quality maintenance?
5. Name four types of circuit configuration from the troubleshooting point of view.

6. What are the common faults in a PCB?
7. How will you determine intermittent faults?
8. Why is it necessary to find out cause of a fault before repairing the fault?
9. What is the difference between a non-maintainable and maintainable equipment?
10. How does the record/play switch in a tape recorder help in locating the faulty section?
11. If a VCR does not operate at all, what are the probable causes?
12. What can happen if disc/lens are dirty in DVD player?

Multiple-Choice Questions

1. What is the most likely fault if a short is found at the output of a power supply?
   (a) Coupling capacitor short
   (b) Series resistance of input filter short
   (c) Decoupling capacitor short
   (d) Emitter bypass capacitor short
2. The output voltage of a full-wave rectifier power supply is low with excessive ripple of 50 Hz. What is the likely fault?
   (a) Filter capacitor short
   (b) One diode open
   (c) Output stage drawing heavy current
   (d) Transformer short
3. The camera tube is OK, broadcast recording is OK but no recording can be done from the camera tube. What is the likely fault?
   (a) Head faulty
   (b) Under-colour system faulty
   (c) Camera switch faulty
   (d) Tuner faulty
4. What is the configuration of a power supply output line?
   (a) Divergent
   (b) Convergent
   (c) Closed loop
   (d) None of the above
5. What is the configuration of a multiplexer?
   (a) Convergent
   (b) Divergent
   (c) Closed loop
   (d) None of the above
6. If the emitter resistor is found open, what will be the voltage at the collector?
   (a) V_{cc} (b) V_{bb}
   (c) Zero (d) 220 V ac
7. An ac signal at the base of the previous transistor is normal, but it is zero at the base of the stage under test. All transistors and resistors are normal, dc voltages are normal. What is the likely fault?
   (a) Coupling capacitor open
   (b) Emitter bypass capacitor open
   (c) Decoupling capacitor open
   (d) Bias is faulty
8. The output of an amplifier is low (not zero). All dc voltages are normal. What is the likely fault?
   (a) Coupling capacitor open
   (b) Negative feedback missing
   (c) Emitter bypass open
   (d) Mains voltage is high

9. In a stereo amplifier, there was no output in any channel for any input. What is the likely fault?
   (a) One microphone faulty
   (b) One LS faulty
   (c) No power supply
   (d) Output transistor of right channel open
10. There is no sound on recording of telecast; field recording is normal. What is the likely fault?
   (a) Mic/Ant switch on mic position
   (b) Audio head dirty
   (c) Audio amplifier defective
   (d) Erase head not getting ac
11. DVD is not recognized by the player. The like fault would be
   (a) Disc and lens dirty
   (b) Incompatibility
   (c) Disc ruined
   (d) All the above

Answers

1. (a) Filter capacitor open (b) Input filter capacitor open (c) Cassette is faulty (d) Amplifier is getting positive feedback (e) Emitter bypass capacitor open (f) LS of the right channel is faulty. (g) Balance potentiometer faulty (h) no power supply. (i) Play terminal open (j) Microphone lead defective (k) Common amplifier defective (l) Loudspeaker cable faulty (m) Antenna lead broken (n) Video stages in playback section may be faulty (o) Splitter stage is open (p) Player is faulty
2. A short in power supply will draw heavy current which may damage the power supply components. Hence, it is necessary to first test the supply for short-circuit without switching it on.
3. Troubleshooting means to locate the faulty component or connection in the set. Repair means action to rectify the fault. Servicing means thorough cleaning, oiling and specification checking. Maintenance includes repair, servicing and preventive maintenance.
4. Corrective maintenance just finds out the faulty component or connection and rectifies the fault. Quality maintenance requires that after the fault is removed, specifications of the set must be checked and if there is any discrepancy in the specifications, it must be removed and specifications should be restored to the original value within tolerable limits.
5. Circuit configurations are
   (a) Divergent circuits
   (b) Convergent circuits
   (c) Closed-loop circuits
   (d) Straight circuit in cascade or sequential stages
6. Solder bridge between adjacent tracks, pinholes, break in a plated-through-hole, loose connection in connectors and dry or cold solder joint are common faults in a PCB.
7. Heating by a blower may provoke an intermittent fault to become permanent and then the fault diagnosis can be done easily.
8. Once a faulty component is located, it is necessary to reason out its

cause. It may happen that there is some other fault in the set which became the cause of the fault detected. For example, if the fuse was found open, there might be a cause (say, a shorted power supply line)which made it blow off. If the cause is not removed, the new part replacing the faulty part will again become faulty. Hence, it is necessary to remove the cause first before rectifying the fault detected.

9. Non-maintainable equipment is that which cannot be repaired, e.g., equipment in a satellite. Components like resistors, capacitors, inductors, transistors, ICs, etc., are also non-maintainable. Sets which can be repaired are maintainable, e.g., a TV receiver, stereo amplifier, a computer, etc.
10. The Record/play switch can identify whether the record section is defective or the playback section. The switch is first kept in the playback mode, and a pre-recorded cassette, known to be good, is played. If the output is normal, the playback section is alright. Keeping the switch on record position, and recording on a blank cassette is done and checked in the playback mode. If OK, the recording section is all right.
11. If a VCR does not operate at all, either power supply is not available to the set, or the timer is ON and hence the VCR cannot be operated.
12. A dirty disc lens of DVD player can cause the following problems:

(i) Disc may not be recognized

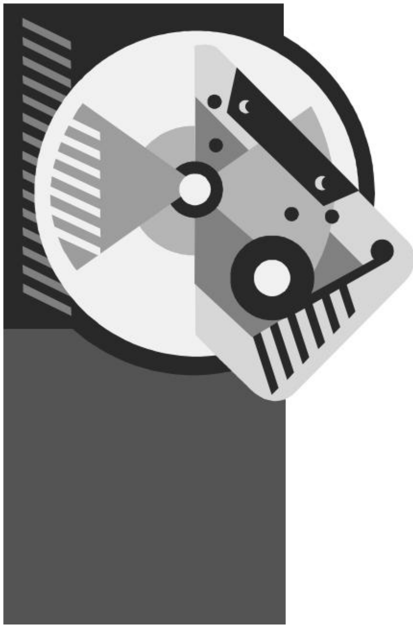
(ii) Audible noise

(iii) Erratic tracking

(iv) Sticking or skipping

Multiple-Choice Questions

1. (c)	2. (b)	3. (c)	4. (a)
5. (a)	6. (a)	7. (a)	8. (c)
9. (c)	10. (a)	11. (d)	



Appendices

APPENDIX I
COMPRESSION STANDARDS

Need of Compression

When analog signal is converted into digital signal, bits produced are in millions. For example, analog video Signal's minimum sampling rate would be 10 million samples per second at Niquist's rate of twice the maximum frequency of the base-band signal. When each sample is quantized by say, 16 bits, the total number of bits generated per second would be 160 million bits per second. HDTV camera generates raw video stream of about 1.5×10^8 bytes per second (one byte is composed of 8 bits) for 25 frames per second system. It is not possible for the transmission channels to handle such high bit rate without degrading the signal to noise ratio. Even DVDs will not be able to accommodate movies with such large number of bytes. A solution to this problem is to compress the digital signal before storage and transmission, and to decompress after detection in the receiver to reproduce the original signal. Compression of signal is called *coding* and decompression, *decoding*. The whole process is called *codec* (an acronym of coding and decoding). The code of compression (or key of the code) should be known to the receiver to enable it to decode or decompress. There cannot be any compromise on this, for, otherwise, the reception would be meaningless. It would be like sending a cipher massage and the receiver does not know the key and so cannot decipher.

Compression ratio It represents the size (number of bits) of the original image divided by the size of the compressed image.

Types of compressions There are two types of compressions:

1. Lossless compression
2. Lossy compression

In lossless compression, the compressed data can be reconstituted (decompressed) without any loss of information or detail. Supression of repeated sequence (like background images) is lossless. Similarly, silence and pauses in conversation can be omitted without loss of data.

In lossy compression, a little information or detail may be lost. An effort is made to achieve the best possible fidelity. In movie videos, one frame after the other is displayed on the screen, but the two consecutive frames differ only a little in information content, as the time interval involved is only 40 ms. Thus, if a frame is transmitted, and the next frame is suppressed, the loss of details would be insignificant. A common and old trick is interlaced scanning which in one way is lossy compression of the data, but the loss is so insignificant that the viewers are not able to perceive it.

Scope of Compression

In colour TV, each picture element (pixel) is represented by one luminance (in short, luma) number and two chrominance (in short, chroma) numbers (B–Y) and (R–Y) to describe colour content of the transmitted picture. Thus, each digitized picture is initially represented by three rectangular arrays of numbers. A common practice of decreasing the data rate is to thin out the two chroma planes. Thinning works because the eye resolves the brightness details better than the chroma details. A proportion of 4 : 4 : 4 is for full chrominance values. When half of the chrominance values are supressed the proportion becomes 4 : 2 : 2. When three-fourth of the chrominance is deleted, the proportion becomes 4 : 2 : 0. Some such suppression is natural in SECAM colour TV system.

Pixels in a frame (still picture) are very close to each other, and the scanning process covers two adjacent pixels in 0.2 μs. It can be easily seen that the change in brightness or colour in such a small period of time would be negligibly small and so adjacent pixels can be conveniently clubbed into 8×8 matrix, which can be compressed into a single coefficient. The data in each matrix can be transformed by discrete cosine transform. (DCT), which converts spatial variations into frequency variations without changing the information in the block. The original block (matrix) can be recreated exactly by applying inverse cosine transform. The advantage is that the image can be simplified by quantizing the coefficients given by DCT. Many of the coefficients pertaining to higher frequency will be zero. This would give good compression ratio.

Background of a scene is generally repetitive and can be coded as one character which can be decoded at the receiver without loss of information. Similarly, blue sky, green grass, ocean, etc. may be stored in compressed form. For audio, when a person is speaking, only his lips move and the rest of the face mostly remains unchanged. The repetitive features of the face (except lips) can be represented by a compressed version.

Thus audio and video compression operates on the premise that much of the information present is not necessary for repeated transmission and can easily be coded in compressed form to get the reduced bit rate (bandwidth). The decoded version at the receiver will be able to produce good perceptual quality. In video, one of the most powerful techniques is to transmit only the inter-frame difference instead of full frames for a specified number of frames. The full frame may be transmitted once a while, say every ninth frame for reference, the reference frame (called I frame; I stands for intracoded or independent reference) does not depend on the data in the preceding or in the following frame. Other frames may be predictive (P frames) or bi-directional (B frames).

Need of Standardization for Compression Technique

Compression cannot be done on adhoc basis. Standards used in compression before transmission must be fully specified so that the receiver may decode the compression correctly to get the original without perceptible loss of fidelity. With this aspect in view, the two international agencies (International Organisation of Standardization, and International Electrotechnical commission) constituted a committee, called JPEG (Joint Photographic Experts Group) in 1986 to develop standards for digital image formats, and another committee, called MPEG (Motion Pictures Experts Group) in 1988 to formulate standards for audio and movie pictures. JPEG covered only still pictures. It neither covered audio nor movie pictures. MPEG covered audio as well as movie pictures. The standards have been specified by ISO numbers, but are generally referred to as JPEG and MPEG standards after the names of the committees who developed these. These standards are described in the following sections.

JPEG

The Joint Photograph Experts Group issued the first JPEG standard in 1992 which was approved by ISO in 1994 as ISO-10918. The standard defines how an image photograph is to be compressed into a stream of bytes before recording and transmission, and decompressed back into the original image at the receiver. These compression and decompression techniques become the umbrella bodies for coordinating formats for others.

JPEG typically achieves 10 : 1 compression ratio with little perceptible loss in quality. It specifies the codec used by digital cameras and other photographic image capturing devices. It is the most common format for storing photographic images on world wide web (internet).

As no motion is involved, it is spatial compression only. The principle behind compression of an image takes note of the fact that the human eye is less sensitive to gradual transitions and also less sensitive to colour variations as compared to brightness variations. It is rightly presumed that neighbouring pixels have almost the same intensity and can therefore be clubbed into a block of information of 8×8 pixels. Thus 64 pixels can be compressed to one coefficient. It is lossy compression, but the loss is very little and insignificant, and it is able to achieve the best possible fidelity with a very substantial reduction in bandwidth by minimizing the bit rate. This finally results in high signal to noise ratio.

Figure A.1 illustrates how JPEG compression can be achieved.

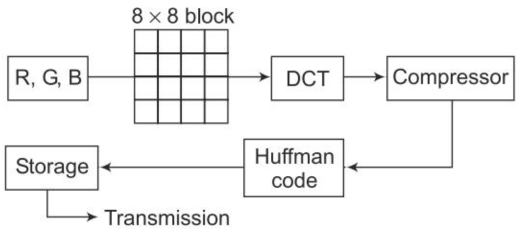


Fig. A.1 *JPEG compressor system*

Colour pixels (Red, Green and Blue) in the image are converted into 8 × 8 matrix blocks. They are processed, using discrete cosine transform (DCT), which like Fourier Transform analyses the frequency components in samples taken at regular intervals. The process discards those frequencies which do not affect the image as the human eye percieives it. The signals accepted in the process are quantized into bits and some redundant bits for error detection and correction are added, using Huffman code (A coding system invented by Huffman, a Ph.D. student at MIT, USA, and based on entropy of probabilities in the information theory. It is also called entropy coding). The compressed data is stored and transmitted.

The decoder would be just reverse of the coder system as illustrated in Fig. A.2.

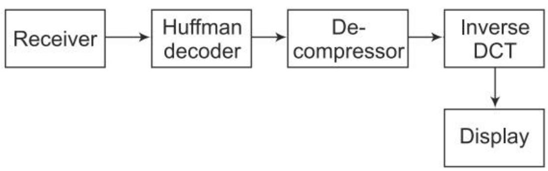


Fig. A.2 *Decoding system*

Limitations of JPEG

1. It caters to the still images only.
2. It does not support audio.
3. It does not support movie pictures.

Appliations

1. It is used in digital cameras
2. It is used in mobile phone cameras
3. It is used in internet for exchange of photographs.

MPEG -1

Motion Pictures Experts Group, constituted by ISO/IEC, started working in 1988 with the aim of defining standards for digital compression of audio and visual signals in movie pictures. For video signals the group first designed compression procedures (called MPEG-1) only for those systems which used progressive scanning . It did not support interlaced scanning.

MPEG-1 compressed video up to 1.5 Mb/s from the minimum of 150 Mb/s of the original. Thus, the compression ratio was 100 : 1.

This standard was first published in 1993 as ISO-11172. It supported resolution of 352 × 288 pixels and also supported audio for two-channel stereo. Techniques used are similar to JPEG except that it took into consideration Inter-frame redundancy also.

MPEG-1 audio layer 3 is also called MP-3 (the term MP-3 applies to audio and should not be confused with MPEG-3).

Weaknesses of MPEG-1

1. Audio compression is limited to 2-channel stereo only. It does not support surround sound.
2. It does not support inter-laced scanning which is used in all TV systems.
3. It is not suitable for high resolution of HDTV
4. It supports only one colour space, that is 4 : 2 : 0 instead of 4 : 2 : 2.

Compression in MPEG-standards consisted of the following components:

1. Discrete Cosine Transform (DCT) of 8 × 8 matrix
2. Predictive and bidirectional coding for inter-frame compression
3. Quantization of compressed data
4. Huffman coding

Thus, DCT, quantization and Huffman coding are of the same nature as in JPEG. Predictive coding and bidirectional coding were added. These considered inter-frame differences as follows.

I is intra-coded (spatial coding) frame; it was independently considered without any reference of the previous or the following frame. *P* frame took the difference between the previous *I* frame or *P* frame, (but not *B* frame). It did not compare itself with any following frame. The *B* frame took the difference between previous and the following *I* frame or *P* frame, but not with *B* frame. *B* frame was not considered reference frame. The picture may be as shown in Fig. A.3.

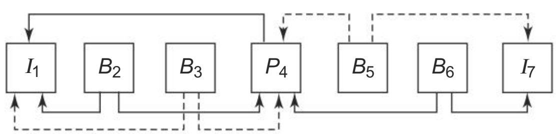


Fig. A.3 *Inter-frame comparison*

The first frame is *I*. It is intra-coded frame and is independent. The second frame is B_2. It compares with previous I_1 and following P_4. The third frame is B_3. It compares with previous I_1 and following P_4. The fourth frame is P_4. It compares with previous I_1. It does not compare with previous *B* frame. It also does not compare with any following frame. The fifth frame is B_5. It compares with previous frame P_4 and following frame I_7. The sixth frame is B_6. It compares with previous frame P_4 and the following frame I_7. The seventh frame is I_7. It is intra-coded and is independent reference.

The difference between P-frame and previous I or P frame and that between a B-Frame and previous or following I and P frame was only a little which resulted in good compression ratio for movie pictures.

The compressed data so obtained undergoes discrete cosine transform for further compression. The compressed data is quantized and stored for transmission.

These processes are shown in Fig. A.4. The compressed data is modulated for transmission

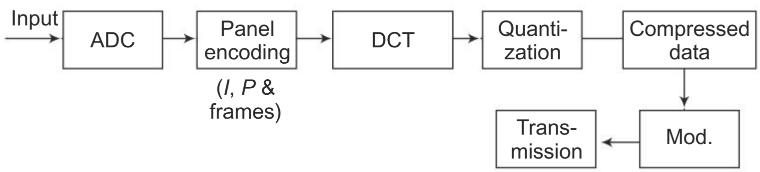


Fig. A.4 *Transmission of MPEG coded signal*

For decoding, the modulated signal is received, demodulated, decompressed, undergoes inverse DCT and digital to analog conversion for reproducing the original signal.

MPEG-2

It evolved out of the shortcomings of MPEG-1. MPEG-2 was designed to support interlaced scanning. Compression up to 3 Mb/s was achieved. Initially, it did not support HDTV, and a new MPEG-3 was designed. But later, with more research, MPEG-2 was extended to support HDTV also, and then MPEG-3 was abandoned. For HDTV, compression went up to 15 Mb/s in MPEG-2, It was also extended to support multi-channel surround system. MPEG-2 decoders can also decode MPEG-1, but not vice versa. ISO finalized MPEG-2 standards in 1994 as ISO-13818. MPEG-2 can compress video of 120 minutes to 4 minutes only (30 : 1 ratio). It gives high quality video. Higher compression will result in too much loss of information and then quality of image would suffer.

MPEG-4

MPEG-2 did not support multimedia. Hence MPEG-4 was developed. In this system, individual objects within a scene are tracked separately and compressed together to create MPEG-4 file. Further MPEG-2 is not optimized for low bit rates, but MPEG-4 caters equally well to very low bit rates to very high bit rates of the original programme (digitized). It also allows developers to control objects independently in a scene and therefore introduces interactivity.

SOLID STATE CAMERA

A solid state camera uses photodiodes as transducers. A photodiode is sensitive to light. When light from a scene strikes it, electrons are generated. Their number is proportional to the intensity of light. Each photodiode represents one pixel. Thus, there are as many photodiodes as there are the number of pixels. Photodiodes are arranged in an array of horizontal rows and vertical columns. There may be about 5×10^5 diodes in 8×10 mm chip (diagonal size 13 mm or ½ inch). When an image of a scene is focused onto the array, the diodes produce a charge image of the whole picture.

Charge from each diode is moved in a precise timing sequence. There is no electronic scanning, but the timing is accurately controlled by a clock and multivibrator circuits. This control produces timing pulses to represent vertical retrace (blanking), horizontal retrace (flyback) and vertical and horizontal traces as explained below.

Scanning of Charge Image in CCD Sensor

A typical arrangement of diodes and other essential elements needed to move charges from the charge-image in correct sequencing is shown in Fig. A.5 on a small scale, 4 diodes in each row and 3 in each column for aspect ratios of 4:3. (In actual set up there may be 600 diodes in each row and 450 diodes in each column in 1/3 inch size sensor).

All gates (marked G in the figure) open during the time interval pertaining to vertical blanking and allow the charge stored in the photodiodes to be transferred to the concerned vertical shift registers (VSRs), arranged in vertical columns.

During horizontal flyback, the charges from the bottommost row of vertical shift registers are transferred to the horizontal shift register, and from the vertical shift registers of other rows to the next lower row.

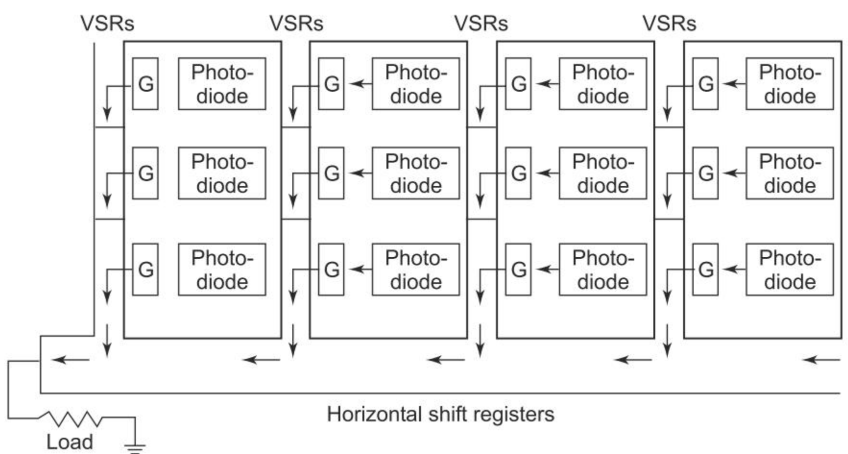


Fig. A.5 *Arrangement of photodiodes, gates and shift registers in a CCD camera.*

During the horizontal trace time, charges move through the horizontal shift registers for each row in sequence and flowing through the load resistor, produce video signal. Thus, charge from each of the photodiodes has moved to the load resistor in correct timing sequence.

The out put of a solid state camera is in essence a digital video signal, popularly known as *digital film*.

Solid state cameras are becoming popular due to the following advantages:

1. Electron beam is not required for scanning and hence no geometric distortion and no non-linearity in the formation of the raster.
2. Can withstand excessive illumination. Image lag quite low. No image burn.
3. High resolution.
4. Sensitivity comparable to the tube type cameras. Illumination of 10 lux can give acceptable picture.
5. Spectral response extends up to infrared region.
6. Being digital, all advantages of digital systems, like computer processing and storage of information, ease in incorporating special features, etc. are available. It is used in mobile phones.
7. Low voltages required. High voltage section is not needed.
8. Low power dissipation (about 100 mW).
9. Reliability is high. They last much longer than tube type cameras.
10. Light weight, small size, portable.

Initially cost was high, but as its use increased, the cost has come down substantially and is less than the tube type cameras.

DIRECT-TO-HOME TV SATELLITE

Direct to Home TV (DTH) Satellite

The system used in individual homes for direct reception of TV from a satellite is illustrated in Fig. A.6. The preferred band for this service is K (12 to 18 GHz or Ka band 18 to 27 GHz). The size of the reflector would be as small as the head light of a car. Description of the blocks shown in Fig. A.6 follows.

Outdoor Unit It consists of a receiving antenna, low noise amplifier and converter. The receiving antenna is a parabolic reflector with a horn as the active element. The horn can be directly in front of the reflector, or it may use an offset feed as shown in the figure. The reflector diameter may be 0.6 m for 11 GHz and still smaller for K and Ka bands.

The low noise block consists of a low noise wideband amplifier followed by a convertor. The output of a convertor consists of a signal of UHF frequency ranging from 950 – 1450 MHz (500 MHz bandwidth). The advantage of using UHF frequency is that a low cost coaxial cable can be used as feeder from the outdoor unit to the indoor unit. LNB cannot be kept indoor because long cable between the horn and the first amplifier (LNA) will cause substantial degradation of the

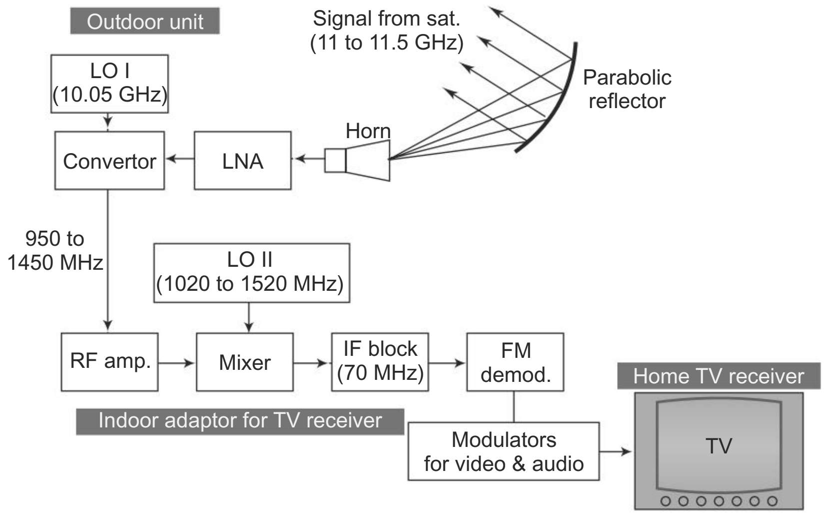


Fig. A.6 *Block diagram of direct of home TV system*

overall noise figure of the set. Cable after the LNA cannot add any significant noise as the dominant noise is due to the first amplifier. The casing of LNB is such that it is able to withstand hostile environmental conditions outside.

Indoor Unit The wideband signal from the LNB is fed to an RF amplifier (tuned to UHF). The amplified signal is fed to a channel selector circuit which selects the wanted channel. The selected channel is down-converted to a fixed IF of 70 MHz by a local oscillator and mixer. IF amplifier amplifies the signal which then, goes to an FM detector. The detector recovers original baseband signal, consisting of a composite video signal and audio signal. The video part modulates an RF carrier, using AMVSB technique. The audio part modulates another RF carrier, using FM. Both carriers pertain to a VHF/UHF channel of the terrestrial TV system. These modulated signals are fed to the normal domestic TV receiver, which after due processing reproduces picture and sound.

(Similar to DTH, there is a direct radio service, known as World-space Radio, through satellites, which was launched by world-Space foundation to broadcast radio programmes direct to homes in African, Asian and European countries for education and entertainment.)

CABLE TV SYSTEM

Cable TV System

A cable TV system (CATV) provides TV programmes to homes through cables. At its centralized location it has antennas to receive several TV channels from satellites besides local TV transmissions. It also has its own studio where special programmes are produced and recorded on tapes or digital video discs for transmission through the cable network.

Input from various antennas and from VCR or digital video discs are combined in a mixer. The mixer is called *Head-End*. The head-end converts all radio frequency signals into VHF-UHF channels.

The outputs from the head-end are fed to a broad-band distribution amplifier from where these are fed to the trunk cables. The trunk cables, in turn, feed the signal to the branch lines from which signal is tapped to be sent to homes. Line amplifiers are used at suitable points on the line to overcome progressive loss of signal. The loss in a cable increases as the frequency of RF signal carrier increases. To rectify this problem, the line amplifiers use equalizers which compensate the frequency dependent loss in the cables. The amplifiers are so designed that the individual TV receivers get the equalized signal of the order of 1.5 mV. This ensures, excellent quality of pictures or sound reproduced by the receiver.

Various elements of a CATV system are shown in Fig. A.7. Description of their functions follows.

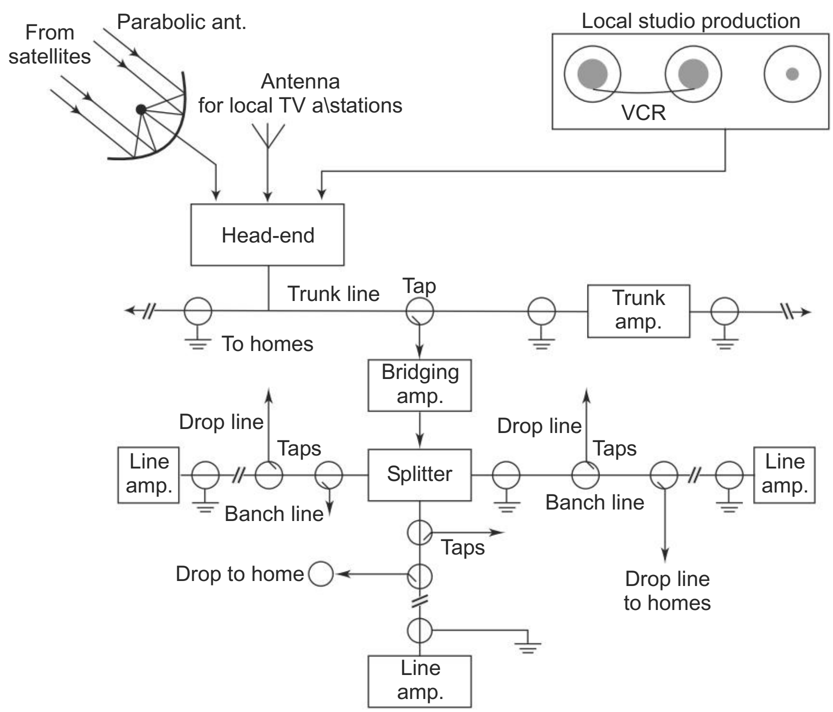


Fig. A.7 *Basic elements of a typical cable TV system*

Head-end The broadcast signals, picked up by the respective antennas are processed. They are amplified by a low noise amplifier and adjusted for correct level. The UHF channels are converted to VHF band. The broadcast signals as well as the signals originated locally from a studio are translated to the frequencies allotted for cable TV. The signal from the head-end then goes to the trunk lines. When optical fibre is used, these signals modulate a laser beam (digital modulation).

Trunk Line Cables Usually it consists of a coaxial cable (called *coax* in short), which is an efficient wideband line. It has the advantage of shielding. A good coaxial cable for trunk line is about 19 mm in diameter. Optical fibres can also be used. However, they are not being used by cable operators in India at present but they have the advantage of very high capacity, low loss, no electromagnetic interference. Hence optical fibres may be used in future when their cost is reduced.

Branch Line Cable Small diameter (about 10 mm) is used because the length is not too long.

Drop Line It is the line running from the branch to the subscriber's home TV. It generally uses coaxial cable of about 6 mm diameter. (All the cables use polyethylene jacket to make them weatherproof. The diameter size indicated for the cables is overall diameter including the jacket.)

Complete system of cable network requires good impedance matching. Usually the cables used have the characteristic impedance of 75 ohm and the line should be correctly terminated to match 75 ohm. Mismatch can make a big difference in strength and hence, quality of the signal.

Trunk Amplifier

There are losses in the cable: DC loss (I^2R loss), skin effect loss and dielectric loss. These losses increase in proportion to the square root of the frequency. At high VHF, the loss may be double of the loss at low VHF. Hence, trunk amplifiers (gain 20 to 30 dB) are placed at suitable intervals along the route. These keep the signal level between 1 and 3 mV. Trunk amplifiers are placed in a weatherproof housing on an electric pole, and power is obtained from a tap on the legally procured electric service line on the same pole.

Bridge Amplifier A branch line comes out of the main trunk line. The branch line feeds a particular neighbourhood. There is a bridge amplifier to act as a bridge between the trunk line and the branch line. It takes care of impedance mismatch caused by the connection with the trunk line and compensates the loss in the trunk line upto the point of connection. Gain of the amplifier is 20 to 30 dB.

Line Amplifiers Branch lines are shorter in length, but they also need amplifier of 20 to 30 dB gain at suitable intervals. When a branch line is to be extended, an amplifier becomes necessary and hence it is also called *line extender*. (All amplifiers in the cable TV system are Class *B* pushpull amplifiers, using transformerless symmetry complementary pair of transistors, to make it compact, efficient and of low harmonic distortion.)

Directional Coupler It is shown in Fig. A.8(a) and its symbol, in Fig. A.8(b). As several branches come out of the trunkline, it is necessary to ensure that each branch line extracts only a small energy from the trunk line. The device called *directional coupler* is used to ensure this. It is a three terminal device.

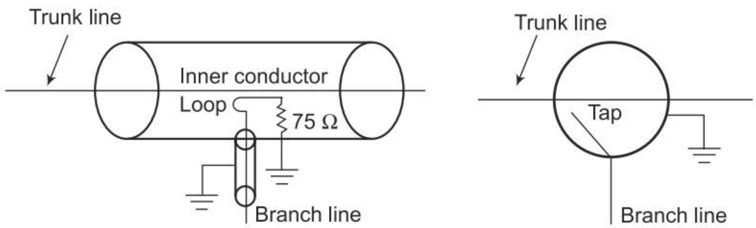


Fig. A.8 *(a) Directional coupler (b) Symbol of directional coupler*

The loop works as a capacitance as well as inductance and is internally terminated in a 75 ohm resistor. The directional coupler ignores the reflected energy and has a very small insertion loss (about 1 dB at 300 MHz). The tap loss is 13 dB which is compensated by the bridge amplifier.

Channels for Cable TV

TV signals picked up by antennas were converted into VHF/UHF frequencies and multiplexed at the head-end before sending them down the trunk lines. At the receiver a convertor, called *set-top convertor*, was provided which selected the desired channel (programme), converted it into the frequency of 3rd or 4th channel of normal TV receiver (54 – 61 MHz or 61– 68 MHz) by superheterodyning process. Tuner of the user's receiver was set at this channel and the programme of the desired cable-channel was received. Thus with the help of a convertor, the subscribers were able to receive all the cable-channels in the receiver, capable of receiving only the 3rd or the 4th channel. The reason for having two channels is that if anyone of these two channels was being employed in local broadcast the other one could be used for the cable.

Cable Ready Receivers

As the development proceeded, cable-ready receivers came into existence which could receive direct (without convertor set all the cable channels which have gone up to about 200. The tuner of this new version of TV receivers was tunable to any channel sent by the CATV system and there was no need of a set-top-convertor. In cable-ready receivers, cable connection could be directly fed to the antenna terminal of the TV receiver. Tuner stage of the receiver selected the desired cable channel.

Pay TV Through Cables

Although the use of set-top-convertor (STC) was not required due to availability of cable-ready receivers, another need arose for a different type of convertor (descrambling unit) for a different purpose. Besides the normal programmes, some premium services like uncut movies, current movies, special sports events, rainbow channel, etc. are offered by the cable TV system on payment of a premium channel fee. To allow access to such programmes to the authorized customer only, the signal is scrambled. The authorized customer will have a descrambler unit supplied by the cable operator. Scrambling and descrambling are described below.

Scrambling The RF signal modulated by the composite video signal is deprived of the sync pulses. Without sync pulses, the receiver cannot lock-in with the sync-supressed signal.

Descrambling For those subscribers who pay extra premium for seeing pay-TV programmes, the scrambled channels have to be descrambled, so that the pictures may appear clear in their sets. The descrambler restores sync pulses to the radio frequency signal. For this purpose, a pilot carrier of frequency lower than the RF channel frequency is sent by the cable operator. The descrambler unit at the subscriber's end consists of a narrow band receiver, which selects the pilot, amplifies it and filters out the spurious components. The filtered pilot carrier passes through an AM detector. The detector's output consists of pulses which pass thorough a pulse-shaping circuit to get clean sync pulses. The arrangement is shown in Fig. A.9.

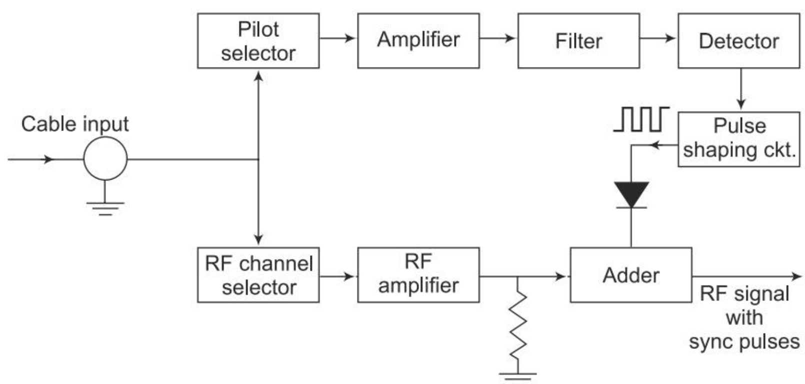


Fig. A.9 *Descrambling with the help of a pilot carrier*

The pulses are correctly timed so that they pass through the diode at the correct time to be added to the sync suppressed signal at the right place, as shown in Fig. A.10.

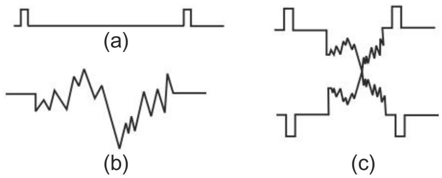


Fig. A.10 *(a) Sync pulses (b) RF signal without sync pulses (c) RF modulated signal with sync pulses*

Two Way Cable Systems

Cable TV system can be used as two-way interactive system. Link from the head-end to the subscribers is called down-stream. An additional link, called up-stream from the subscriber to the head-end is also provided. The downstream signals are in VHF/UHF band, while the upstream signals are in HF band (5 to 30 MHz). The same cable is used for both directions, but separate amplifiers are employed by using HP and LP filters as shown in Fig. A.11.

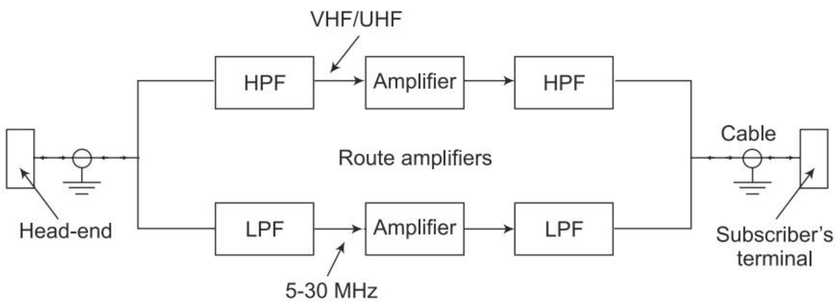


Fig. A.11 *Two-way cable system using single coaxial cable*

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