CHAPTER 1

Basic Concepts: Audio Engineering

1.1. INTRODUCTION

The consumer electronics is the study of electronic devices used in our day to life. Basically these are audio and video systems.

1.2. AUDIO SYSTEMS

An audio system is a system which involves an audio signal. In other words, Audio Systems "process" sound. The examples of audio systems/equipment/components are:

1. Radio
2. Tape Recorder/Player/Stereo
3. Microphone
4. Loud speaker
5. Telephone.

1.3 AUDIO VIDEO SYSTEM

An audio/video system is that which simultaneously process sound as well as picture signals. The examples of AV systems are:

(*a*) *Human AV systems.* Human AV system compresses of two ears and two eyes.

(*b*) *Man made AV systems*

1. Television
2. VCP

3. VCR
4. Cinema Projector
5. Radar
6. Picture telephones.

1.4 IMPORTANCE OF A.V. SYSTEMS

Every body knows the important of Radio, Tape recorder, Telephone, TV, VCR etc. These have been the necessity of every home now.

Now AV Systems have very much entered also in the field of education. Video cassettes have been prepared and these are displayed in "Video Schools". Lectures of eminent professors/ educationists are prepared on all subjects and same can be viewed on TV at home. There is no need to join computer classes. All computer languages can be learnt just sitting in the bed room and watching a TV.

Now it has become a fashion to prepare video films on all occasions. It may be a birth day party, or a funeral procession, we can keep our memories afresh everytime. In old age, we can see our childhood.

AV systems have entered also in the world of crime. Video cameras can be installed in any place or house. All activities of smuggling/killings can be recorded on a video. The same can be presented in the courts as a witness. How a criminal will deny!

Now in medical sciences, say a patient is being operated upon in the operation theatre. A video cassette can be prepared of the same and patient can see at any time later on. Further by using close circuit TV,. The relatives of the patient sitting outside can see what is happening inside the operation theatre.

1.5. AUDIO ENGINEERING

Audio Engineering is the branch of engineering dealing with the study of audio (sound) energy.

Audio (sound) may be defined as generation, transmission and reception of energy in the form of vibrations. *e.g.*, when a spring is stretched, a fluid (air) is compressed, vibrations are produced by their molecules. This results in the oscillatory vibrations and thereby acoutistic waves are generated and transmitted.

1.6. TERMINOLOGY RELATED TO AUDIO ENGINEERING

(1) **Acoustics.** The word "*acoustics*", is a term associated with sound waves, or with individual media, phenomena, apparatus, etc. It is the science of sound waves.

Acoustical. The word "acoustical", is a term related with acoustics.

(2) **Pressure**

(*i*) *Static Pressure* (P_0). This is the pressure in the medium that would exist at a point with no sound waves present. *Microbar* (μ*B*) is the unit of pressure used in acoustic. One m*B* is equal to 0.1 Newtons per m^2.

(*ii*) *Instantaneous pressure*. This is the incremental change from the static pressure at a given instant caused by the presence of sound waves. The unit is μ*B*.

(*iii*) *Sound pressure* (*P*) is the RMS value of the instantaneous sound pressure over a time interval at that point.

(3) **Velocity**

(*i*) *Instantaneous particle velocity*. This is the velocity at a point of a given *infinitesimal* part of the medium at a given instant due to sound waves only. The unit is m/s.

(*ii*) *Effective particle velocity* at a point is the RMS value of instantaneous particle velocity. The unit is meter per second.

(*iii*) *Instantaneous volume velocity*. This is the velocity due to sound waves only and is the rate of flow of the medium perpendicularly through a specified area. This is equal to the product of instantaneous particle velocity and the surface area in consideration. The unit is cubic meters per second.

(4) **Impedence**

(*i*) *Acoustic Impedence* (Z_A). The acoustic impedence at a given surface is the complex ratio (this has the same meaning as the complex ratio of voltages and currents in ac circuits) of average sound pressure (effective) over the surface *to the* effective volume velocity through it. The surface may be either a hypothetical surface of a acoustic medium or the moving surface of a mechanical device. The units are *acoustic ohms*.

(*ii*) *Specific acoustic impedence* (Z_S). This is the complex ratio of the effective sound pressure at a point to the effective particle velocity at that point. The unit is Newton Seconds/m^2 or *ray*l.

(*iii*) *Characteristic impedence* (ρ_c). This is the ratio of the effective sound pressure at a given point *to the* effective particle velocity at that point in a progressive sound wave. This is equal to the product of the density of the medium and the speed of sound in the medium. It is analogous to the characteristic impedence of an infinite long dissipation loss transmission line. The unit is Newton-sec/m^3 or MKS rayl. For air, this value is 407 MKS rayls or 40.7 rayls at 22ºC temperature and 0.76 m mercury pressure.

(5) **Sound Energy Density (D).** This is the ratio of sound energy in a given part of the gas to the volume of that part of the gas. This has units of watt seconds per cubic meter.

(6) **Sound Intensity (I).** The sound intensity measured in a spacific direction at a point is the average rate at which the sound energy is transmitted through a unit area perpendicular to the specific direction at the point considered. The unit is Watt per square meter.

(7) **Acoustic Intensity Level (AIL).** This is the ratio of two sound intensities in a logarithmic form. The unit is decibel.

$$\text{AIL} = 10 \log_{10} \frac{I_1}{I_2}.$$

(8) **Sound Pressure Level (SPL).** Sound pressure level of a sound in decibels is 20 times the logarithmic to the base of 10 of the ratio of measured effective sound pressure (P) of the sound *to a* reference effective sound pressure.

$$\text{SPL} = 20 \log_{10} \frac{P}{P_{ref}}$$

The reference pressure (P_{ref}) is taken as 0.1 newton/m^2 (or 1 μB).

(9) **Acoustic Power Level (APL).** The acoustic power level of a sound source in decibals is 10 times the logarithm to the base 10 of the ratio of the acoustic power radiated by the source to a reference acoustic power.

$$\text{APL} = 10 \log_{10} (W/W_{ref}) \text{ db}.$$

The reference power is taken as 10^{-13} watts. That is, a source radiating one acoustic watt has a power level of 130 db.

Table 1.1

S. No.	Sounds	Pressure		Intensity	
		Pa (*N/m*^{2})	μ *Bars*	*Watt/m*^{2}	*db*
1.	Threshold of hearing	20×10^{-6}	200×10^{-6}	10^{-11}	4
2.	Threshold of pain	65	620	10	128
3.	Thunder	2	20	10^{-2}	98
4.	Heavy traffic	0.2	1.9	10^{-4}	78
5.	Light traffic	18×10^{-3}	200×10^{-3}	10^{-6}	58
6.	Ordinary conversation	65×10^{-4}	650×10^{-4}	10^{-7}	52
7.	Whispering	20×10^{-5}	200×10^{-5}	10^{-9}	18

1.7. TYPES OF VIBRATIONS

The vibrations are of following types :

(*i*) Having a frequency range between 20 Hz to 20 kHz, which have *audible* range for common man, *i.e.* we can hear these vibrations.

(*ii*) Having a frequency range above 20 kHz called *ultrasonics*, which are not audible but have many engineering applications like : drilling, flaw detection in metals, cold welding, ultrasonic mixing and sonar (sound navigation & ranging).

Dogs can hear these high frequency sounds. The birds can hear upto 50,000 Hz. Similarly, bats produce ultrasonics in the darkness to find presence of obstacles in their way.

(*iii*) Having a frequency range below 20 kHz called *Infrasonics*, which are also not audible by man.

We should note that the vibrations (oscillations) may be periodic (sinusoidal) as produced by a tuning fork [Fig. 1.1 (*a*)] or may be *non periodic* (non-sinusoidal) as produced by *noise* [Fig. 1.1 (*b*)].

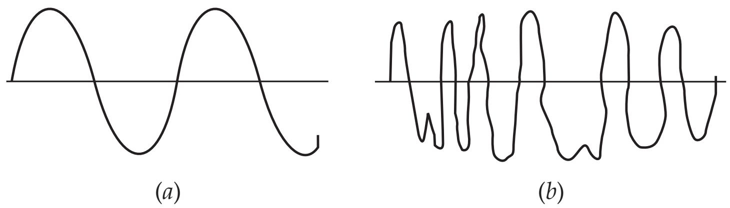


Fig. 1.1

1.8. SIMPLE HARMONIC MOTION (SHM)

The particles of every medium are bound by some elastic forces. When a particle of the medium is disturbed by giving some energy, it is displaced from its mean position. The disturbed particle begins to vibrate with simple harmonic motion. The energy of this particle is transferred to the next particle due to elasticity of the medium. The second particle also starts vibrating about its mean position and hands over the disturbance to the next particle and so on. In this way, the disturbance is transmitted from particle in the medium, whereas the particles donot change their position and continue to vibrate about their mean position.

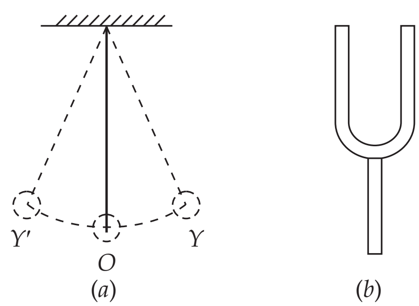


Fig. 1.2

All material objects when struck properly produce waves due to their vibrations. The common examples are a simple pendulum [See Fig. 1.2 (*a*)] and a tuning fork [Fig. 1.2 (*b*)].

The graphic representation of a SHM is given in [Fig. 1.3 (*a*) and 1.3 (*b*)], which clearly show the position (phase) of the vibrating particle at different positions. The displacement of the particle at any time is given by the expression.

$$AM = y = a \sin \theta = a \sin \omega t.$$

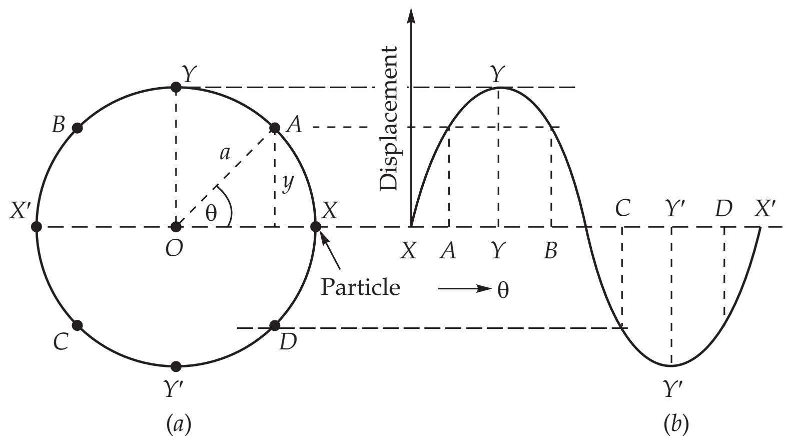


Fig. 1.3

S.H.M. (Definition)

A particular type of periodic motion (which repeats itself over and over again after regular intervals of time) is known as *Simple Harmonic Motion* (SHM). Sometimes their vibration is a combination of two or more than two SHMs.

The SHM may therefore be defined as a motion, which satisfies two conditions :

(1) It accelerates always towards a certain fixed point and

(2) Its acceleration is proportional to its distance from the fixed point.

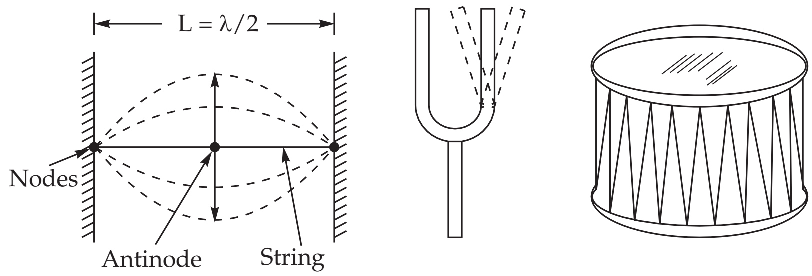
Alternative Definition

The SHM is a motion executed by a particle subjected to a force that is proportional to the displacement of the particle but opposite in sign.

Bodies executing SHM

The various shapes of bodies executing SHM may be classified as : *strings, bars, pipes, plates and diaphragms* etc. Whenever any of them emits sound waves, it is found to be in a state of rapid vibrations.

(*i*) *Vibrations of a string* [Fig. 1.4 (*a*)]. If a string stretched between two fixed points is plucked in the middle, a sound is produced. On touching the string, vibrations disappear and sound ceases.



(*a*) (*b*) (*c*)

Fig. 1.4

(*ii*) *Vibrations of a bar* [Fig. 1.4 (*b*)]. The most common example of bar is a tuning fork. When it is struck at one of its arms, it vibrates and produces sound.

(*iii*) *Vibrations of a diaphragm* [Fig. 1.4 (*c*)]. Sounding of a drum is an example of vibrations of a diaphragm. Similarly a bell starts vibrations, when striked by a hammer.

Note. It should be noted that oscillatory motion is always periodic but every periodic motion is not oscillatory. Examples of periodic motions are :

(1) Change of seasons
(2) Motion of earth around the sun
(3) Motion of planets and comets
(4) Change of phases by moon
(5) Human heart beat.

1.9. VIBRATIONS OF STRETCHED MEMBRANES, DIAPHRAGMS AND THIN PLATES

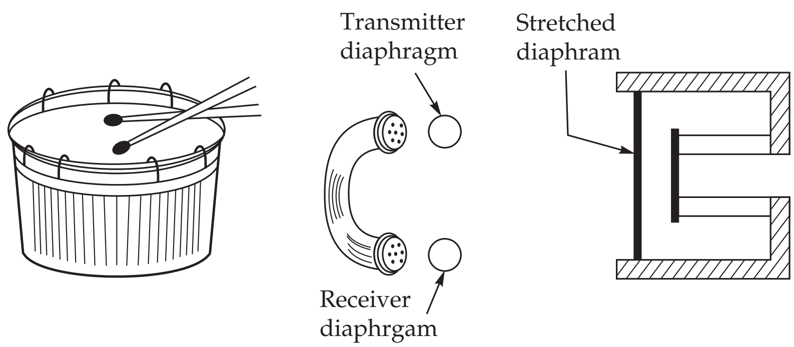
So far, we have considered vibrations of a stretched string which was an example of *one dimensional* system. A membrane is a *two dimensional* system. The analysis of the vibrations of a stretched membrane is helpful in the study of *electro acoustical transducers* (*microphone, loudspeakers etc.*). There are two such cases of vibrations :

(1) Vibrations of stretched circular membrane in which case, restoring force due to stiffness is negligible in comparison to the tension *e.g.,* drum heads, diaphragms of a condenser microphone.

(2) Vibrations of stretched thin circular plates, where stiffness cannot be neglected *e.g.*, diaphragms of telephone receivers (and also transmitters).

The Fig. 1.5 show examples of stretched membranes (diaphragms).

The transverse vibrations of a string (or bar) are sinusoidal, but the transverse vibrations of membranes are not sinusoidal, therefore analysis of membranes is bit complicated. A membrane if regarded as an assembly of parallel strings, it becomes easier to analyse.

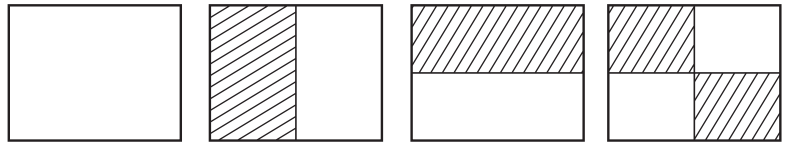


(*a*) Drum (*b*) Capacitor mike (*c*) Telephone

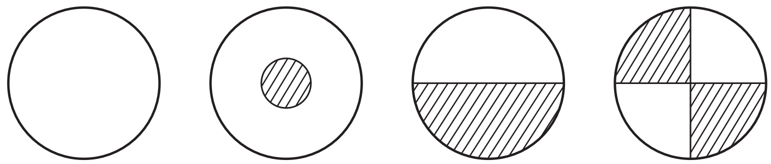
Fig. 1.5

Let us take an example of a drum (or a condenser microphone), which have a membrane stretched tightly over the open ends of a vessel. As the drum head is made to vibrate, the air inside the vessel passes through compression and rarefaction alternately. The velocities of sound waves are lesser in the membrane than in the air but the sound pressure is uniform over entire membrane.

There is a little difference in vibrations in thin plates and in membranes. That is in former, the restoring force is entirely due to the stiffness of the diaphragm, no tension is applied, where as in later the restoring force is entirely due to the tension applied to the membrane.



(*a*)



(*b*)

Fig. 1.6

The Fig. 1.6 (*a*) shows vibration modes of a rectangular and Fig. 1.6 (*b*) shows of a circular membrane.

1.10. TYPES OF WAVES

(A) *Depending upon the direction of propagation of waves and of the medium particles, the waves are of two types* :

(1) **Transverse Waves.** In these waves, the particles of the medium vibrate perpendicular to the direction of propagation of waves. See Fig. 1.7.

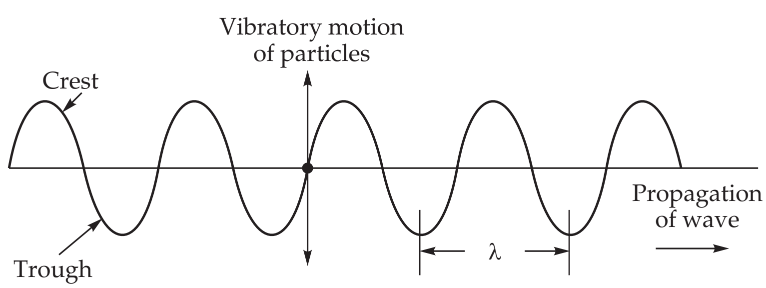
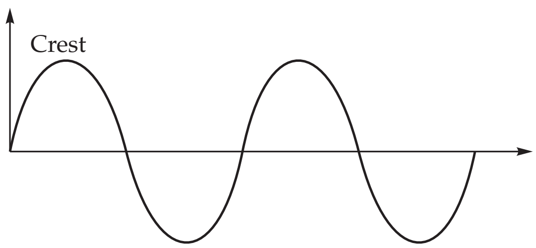


Fig. 1.7

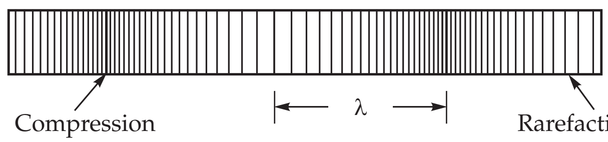
The waves produced in a stretched string are *transverse waves*. These waves travel in the form of *crests* and *toughs*, which are parts of the medium above and below the mean position respectively.

(2) **Longitudinal Waves.** In these waves, the particles of the medium vibrate parallel to the direction of propagation of waves. See Fig. 1.8 (*a*).

The sound waves in air are longitudinal waves. These waves travel in the form of *compressions* and *rarefactions*, which are alternate regions of high and low density of the medium respectively.



(*a*)



(*b*)

Fig. 1.8

The distance between two adjacent crests or troughs (or compressions and rarefaction) is called *wavelength* (λ) [Fig. 1.8 (*b*)].

Table 1.2 Difference between Transverse and Longitudinal Waves

	Transverse Waves	*Longitudinal Waves*
1.	These are the waves in which vibrations occur perpendicular to the direction of propagation of waves.	In these waves, vibrations occur along the direction of propagation of the waves.
2.	These waves travel in the form of crests and troughs.	These waves travel in the form of compressions and rarefactions.
3.	The transverse waves can travel through solids only,	Longitudinal waves can travel through solids, liquids and even gases.
4.	These waves can be polarised.	These waves cannot be polarised.

(B) *On the basis of propagation of energy, the waves can be of two types* :

(1) **Progressive or Travelling Waves.** If there is a continuous transfer of energy in one direction across any cross section of the medium, the waves are called progressive or travelling waves.

(2) **Stationary or Standing Waves.** If the net transfer of energy across any cross section of the medium is zero, the waves are called stationary or standing waves.

These waves are produced due to the superposition of a wave and its reflected wave.

Table 1.3 Difference between Progressive and Stationary Waves

	Progressive waves	*Stationary waves*
1.	The amplitude of each particle is same every where.	The amplitude of the particle varies from zero at *nodes* to maximum at antinodes
2.	The phase of two consecutive particles is not same.	The phase of all particles between two nodes is same and is opposite to that of the particles in the segments on either side.
3.	No particle is permanently at rest.	The particles at the nodes are permanently at rest.
4.	Particles pass through mean position at different instants.	All particles cross the mean position simultaneously.
5.	All particles have same maximum velocity.	The velocity is maximum (largest) for the particles at antinodes and it is zero for the particles at the nodes.
6.	Energy is continuously transferred across every cross-section of the string.	No net transportation of energy takes place across any cross-section of the string.

Notes :

Stationary Waves. When two progressive waves of the same type (transverse or longitudinal), of same wavelength and of same amplitude travel with the same velocity in opposite directions through a medium, they *superpose* on each other to form a stationary wave.

In stationary waves, crests and troughs (or compressions or rarefactions) donot move forward through the medium. They simply appear and disappear alternately at the same positions. No energy transfer takes place in the medium.

Stationary waves are further of two types :

(*i*) *Transverse stationary waves.* These are the waves, which are formed due to the *super-position* of two progressive transverse waves. For example waves produced in a sonometer are transverse stationary waves.

(*ii*) *Longitudinal stationary waves.* These are the waves, which are formed due to super-position of two progressive longitudinal waves. For example, waves produced in organ pipes or in a resonance apparatus are longitudinal stationary waves.

1.11. VELOCITY OF SOUND

Sound waves are produced due to the vibratory motion, which may or sometimes may not be visible. *Sound is a longitudinal wave motion in a material medium.* For propagation of sound, a medium is a must. Sound cannot propagate in vacuum. The material medium required may be solid, liquid or a gas.

The velocity of sound depends upon the

(1) the elasticity of the medium, and

(2) the inertia of the medium (*i.e.,* density/mass of the medium).

The velocity of sound in a medium is given by :

$$V = \sqrt{\frac{\text{Elasticity}}{\text{Density}}}$$

$$V = \sqrt{\frac{E}{\rho}}$$

This is the Newton's formula for velocity of sound. The same formula holds good for all mediums.

(*a*) **Velocity of sound in solids.** For finding velocity of sound in solids, the elasticity (E) is replaced by young modulus of elasticity (Y).

Hence, $$V = \sqrt{\frac{Y}{\rho}}$$

(*b*) **Velocity of sound in liquids.** In case of liquids, E is replaced by bulk modulus of elasticity (K)

$$V = \sqrt{\frac{K}{\rho}}.$$

(*c*) **Velocity of sound in gases (or air).** Newton found that coefficient of elasticity of a gas is equal to its pressure. While providing this, he assumed that sound travelled through air under *isothermal* conditions. He gave the formula for velocity of sound in air as under :

$$V = \sqrt{\frac{P}{\rho}}.$$

where P is the pressure of air at normal temperature and pressure (NTP) condition, and ρ is the density of air.

Pressure of air at NTP, $P = 0.76 \times 13.6 \times 10^3 \times 9.8$ N/m^2

and density $\rho = 1.293$ kg/m^3

Hence velocity of air at NTP

$$V = \sqrt{\frac{0.76 \times 13.6 \times 10^3 \times 9.8}{1.293}} = 280 \text{ m/s}$$

But the velocity of sound in air has been found as 332 m/s, therefore the above formula needs corrections.

Laplace's corrections. The French Mathematician Laplace found that sound travels through air under *adiabatic*, and not isothermal conditions, as newton presumed. According to Laplace :

(1) Temperature rises slightly in the region of compression and it decreases slightly in rarefaction.

(2) Air is a bad conductor of heat and compressions or rarefactions follow each other so rapidly that heat generated in compression or absorbed during rarefaction is not transferred, thus the changes are adiabatic.

He gave the following formula for velocity of sound in air :

$$V = \sqrt{\frac{\gamma P}{\rho}}$$

where γ = Ratio of two specific heats of air (or any gas) *i.e.* at constant pressure and volume = C_P/C_V

P = Pressure of the air or gas

ρ = Density of air or gas.

Using this formula for air, for which, $\gamma = 1.41$

$$V = \sqrt{\frac{1.41 \times (0.76 \times 13.6 \times 10^3 \times 9.8)}{1.293}}$$

$$= 333.5 \text{ m/s}$$

This value agrees with the experimental value of velocity of sound in air at NTP. Hence Laplace correction is valid.

Factors affecting velocity of sound in gases

(1) *Effect of density*. The velocity of sound in two gases at the same pressure, is inversely proportional to the square root of their densities :

$$\frac{v_1}{v_2} = \sqrt{\frac{\rho_2}{\rho_1}}$$

Density of oxygen is 16 times to that of hydrogen *i.e.*

$$\frac{v_h}{v_0} = \sqrt{\frac{16}{1}} = 4$$

or, velocity of sound in hydrogen is 4 times to that in oxygen.

(2) *Effect of pressure*. There is no effect of pressure on the velocity of sound.

(3) *Effect of moisture*. The velocity of sound is more in the moist air than in the dry air.

(4) *Effect of temperature*. The velocity of sound is proportional to the square root of absolute temperature

i.e. $$V \propto \sqrt{T} \quad \text{or} \quad \frac{V_1}{V_2} = \sqrt{\frac{T_1}{T_2}}$$

It can be easily shown that *increase in the velocity per deg centigrate rise in temperature is 0.62 m/s*.

(5) *Effect of wind on velocity of sound*. When the wind blows in the direction of sound, velocity of sound is increased and in the opposite case, it is decreased. If velocity of sound is V_S and of the wind is V_W, the resultant velocity of sound is,

$$V = V_S \pm V_W$$

Note :

Velocity of sound in water. It is more important to find velocity of sound in sea water than in fresh water. The reason being that this has practical importance in all methods of finding distances and depths, locating ships, and submarines under sea water.

For this, sound is recorded by listening it through hydrophone at a distance of 150 to 200 km. A radio signal is also transmitted at the same instant and arrival of both is recorded and compared.

The velocity of sea water is found to increase with temperature and the *salinity* (quantity of salt) of water. The following expression is used for sound in m/s in sea.

$$V = 1410 + 4.21\,\theta - 0.237\,\theta^2 + 1.4S$$

where θ is the temperature of sea water in °C and S its *salinity* in parts per thousand.

Table 1.4. Velocity of sound in different media

S. No.	*Medium*	*Velocity m/s*	*S. No.*	*Medium*	*Velocity m/s*
1.	Steel	5100	6.	Mercury	1470
2.	Cost iron	4700	7.	Water (4ºC)	1432
3.	Brick	3650	8.	Hydrogen (0ºC)	1296
4.	Copper	3586	9.	Air (0ºC)	332
5.	Brass	3500	10.	Oxygen (0ºC)	317

Example 1.1. *Calculate velocity of sound in hydrogen at 363°K. The density of hydrogen at NTP is 0.089 kg/m^3. Assume $\gamma = 1.41$.*

Sol. We know that NTP = 273°K and 0.76 m of mercury pressure (or $P = 0.76 \times 13.6 \times 10^3 \times 9.81 = 1.013 \times 10^5$ N/m^2)

and density $\rho = 0.089$

Now velocity at 273°K, $V_{273} = \sqrt{\gamma \cdot P/\rho}$

$$= \sqrt{1.41 \times 1.013 \times 10^5 / 0.089} = 1267 \text{ m/s}$$

Now $$\frac{V_{363}}{V_{273}} = \sqrt{\frac{363}{273}} \times 1267$$

or $$V_{363} = 1.15 \times 1267 = 1457 \text{ m/s.} \quad \textbf{Ans.}$$

Example 1.2. *At what temperature, the velocity of sound in air is double the velocity of sound in air at 0ºC.*

Sol. Let at temperature t, the velocity of sound is double of that of at 0ºC

$$\frac{V_t}{V_0} = \sqrt{\frac{T}{To}} = \quad \text{or} \quad \sqrt{\frac{273 + t}{273 + 0}} = 2$$

or $273 + t = 4 \times 273 = 1092$ or $t = 819$ºC. **Ans.**

Example 1.3. *An observer sets his watch by the sound of a gun fired from a tower. Find the distance between tower and the observer, if his watch was slow by 5 seconds. Temperature of air is 15ºC.*

Sol. We know that velocity changes at the rate of 0.62 m/s per ºC change in temperature and velocity of sound at 0ºC is 340 m/s.

Now, the velocity of sound at 15ºC.

$$V_{15} = 340 + (0.62 \times 15)$$
$$= 340 + 9.30 = 349.3 \text{ m/s.}$$

The distance of the tower from the observer is the distance travelled by sound in 5 seconds. *i.e.*

$$= 349.3 \times 5$$

$$= \mathbf{1747.5 \text{ meters.}} \quad \textbf{Ans.}$$

Example 1.4. *A body vibrating with a certain frequency sends 1.5 m long waves through a medium A and 2.0 m long waves through a medium B. The velocity of waves in A is 120 m/s. Find the velocity in medium B.*

Sol. Suppose frequency of vibrating body be n. We know that frequency remains same irrespective of the medium.

Medium A. Wave length $\lambda_1 = 1.5$ m

Velocity $V_1 = 120$ m/s

$$n = \frac{V_1}{\lambda_1} = \frac{120}{1.5} = 80 \text{ Hz.}$$

Medium B. Frequency, $n = 80$ Hz

Wave length $\lambda_2 = 2.0$ m

Velocity $V_2 = n \cdot \lambda_2 = 80 \times 2 = 160$ m/s. **Ans.**

1.12. TYPES OF OSCILLATIONS

There are following types of oscillations :

(1) *Free Oscillations*. When a body is set into vibrations and allowed to vibrate *freely* under the influence of its own elastic or gravitational forces then it vibrates with a certain time period called the *natural period*. Such vibrations are called *free vibrations* or oscillations and the frequency with which the body vibrates without any external influence *i.e.* when left to itself is called *natural frequency*. For example when a pendulum is displaced from its mean position, and it vibrates with a natural time period given by

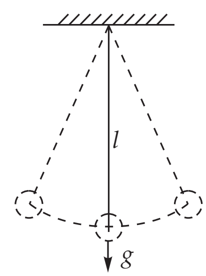


Fig. 1.9

$$T = 2\pi \sqrt{\frac{l}{g}}$$

where l is length of the pendulum and g is the gravity (see Fig. 1.9).

Other examples of free vibrations are vibrations of tuning fork, strings of musical instruments, air columns in organ pipes etc.

(2) *Damped oscillations or vibrations*. The oscillations in which amplitude decreases with time are called as damped oscillations or

vibrations. The wave form of these oscillations are shown in Fig. 1.10. The decrease in the amplitude is due to the losses taking place in the oscillator, which may be due to friction of the surface or air friction, and no means are provided to compensate these losses. It is to be noted that the amplitude of oscillations decreases but the frequency remains the same, since it depends upon the parameters of the system.

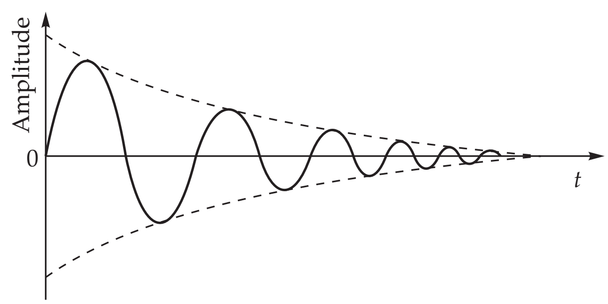


Fig. 1.10

The oscillations of a freely oscillating pendulum are damped as they go on decreasing with time.

It is to be mentioned here that friction forces are of many types but they always result in the damping (decreasing) amplitude of the oscillations.

The most important frictional force is the *resistance* to motion of the oscillator and this resistance may arise from radiation of sound waves and from the viscous fluid present in the surrounding. This frictional force is proportional to the velocity of the system and can be expressed as

$$F_r = -R_m \frac{dx}{dt} \qquad \left[\text{velocity} = \frac{dx}{dt}\right]$$

where F_r is the frictional force, R_m is mechanical resistance of the system. If R_m is also included, the equation of motion of a simple oscillator becomes :

$$m \frac{d^2x}{dt^2} + Rm \frac{dx}{dt} + kx = 0$$

where m is the mass of the oscillatory system, R_m is the mechanical resistance of the system and k is the stiffness constant.

(3) *Undamped (Forced) oscillations or vibrations*. The oscillations in which amplitude does not decrease with time (and remains

constant) are called undamped or forced oscillations or vibrations. The wave form is shown in Fig. 1.11. These oscillations are produced by a system in which some means are provided to compensate for the losses, *i.e.,* the loss of energy after each oscillation is provided by external means. We are more interested in these oscillations, as these are used in audio video systems.

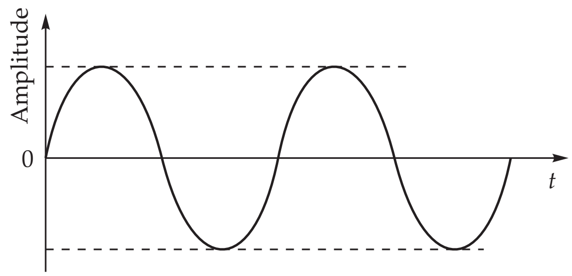


Fig. 1.11

In this, the oscillatory system is set into vibrations with the help of a strong external *periodic force* having a frequency different from its natural frequency. If a pendulum instead of allowing free oscillations is moved by hands it can be made to oscillate with the frequency, we like. In other words, it will then not vibrate with its own natural frequency but with the frequency of the motion of our hands. It is obvious that now the vibrations of the pendulum will be *forced* and not *free*. Similarly melody of stringed instruments is the result of forced vibrations of their sounding boxes.

1.13. DAMPED ELECTRICAL OSCILLATOR

The Fig. 1.12 shows a damped oscillator. The dissipation of energy occurs in resistance of the circuit. Due to loss of energy, the amplitude of oscillations goes on decreasing and ultimately they disappear.

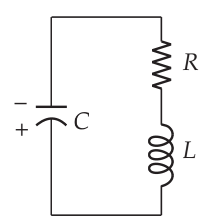


Fig. 1.12

Electrical damping is of three types (Fig. 1.13)

(1) *Heavy damping*. In this case, the charge on the capacitor decays to zero in minimum possible time.

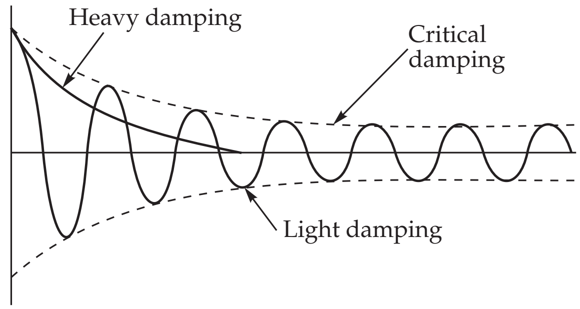


Fig. 1.13

(2) *Critical damping*. In this case, the discharge of the capacitor is non oscillatory. The rate of discharge depends upon the damping constant or the resistance R. The behaviour of the oscillator is said to be *dead beat*.

(3) *Light damping*. The discharge is oscillatory and takes place slowly. The amplitude of oscillations decreases with time, *i.e.* reduces exponentially to zero. The oscillations cease almost during the same time, in which the critically damped oscillator returns to the initial state.

Note. The damped oscillations are not simple harmonic in the sense that the maximum displacement on either side of the mean position goes on decreasing with time. The motion cannot be termed as periodic, as it does not repeat itself exactly as each swing being smaller than that of the preceeding one. However when the damping is small, the motion tends to be an SHM as well as periodic.

1.14. UNDAMPED (OR FORCED) ELECTRICAL OSCILLATOR

An oscillator, to which a continuous excitation is provided by some external agency is called an undamped or forced oscillator. It consists of

(1) *The driven system i.e.,* damped oscillator.

(2) *The external driving system*. In case of a mechanical oscillator, it is a source of mechanical force (such as a spring in old mechanical

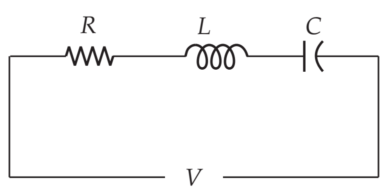


Fig. 1.14

wall clocks) and in case of electrical system, it may be an external emf. Such an electrical oscillator consists of a RLC series circuit with an external source of EMF. See Fig. 1.14.

Table 1.5. Analogies between Mechanical and Electrical Oscillators

	Mechanical Oscillators	*Electrical Oscillator*
1.	It has mechanical reactance represented by X_m and mechanical resistance by R_m.	It has equivalent inductive reactance X_L and electrical resistance as R.
2.	It has a total mechanical impedence $Z_m = R_m + jX_m$.	Its total electrical impedence $Z = R + jX_L$.
3.	Its power factor (mechanical) is given by the ratio of R_m/Z_m.	Electrical power factor $\cos\phi = R/Z$.
4.	Displacement in mechanical oscillators is represented by x and velocity is given by = $\frac{dx}{dt}$.	The equivalent displacement is the change q and the velocity is represented by the current,, given by $I = \frac{dq}{dt}$.
5.	Time period of a mechanical oscillator $T = 1/f = 2\pi$ ($\sqrt{m/k}$).	In this case, time period of electrical oscillator is $T = 1/f = 2\pi\sqrt{LC}$.
6.	The driving agent in case of mechanical oscillator is the force given by = $m\frac{d^2x}{dt^2}$.	The driving agent in this case is induced emf = $L\frac{dI}{dt} = L\frac{d^2q}{dt^2}$.
7.	The restoring agent in this oscillator is the elastic force $E = k \cdot x$.	The restoring force in this case is the voltage across the capacitor given by $v = q/C$.
8.	The mass of mechanical oscillator is given by m.	The equivalence of m is inductance L.
9.	It has springiness.	The equivalence of springiness is capacitance.
10.	It has mechanical compliance $C_m = 1/k$.	It is equivalent to capacitance C.

1.15. ACOUSTICAL, ELECTRICAL AND MECHANICAL ANALOGIES—NAME AND SYMBOLS

All acoustical systems can be converted into equivalent mechanical and electrical systems. (Table 1.6)

Table 1.6 Acoustic, Electrical, Mechanical Analogies

S. No.	*Acoustical*	*Electrical*	*Mechanical*
1.	Inertness m	Inductance	Mass m
2.	Compliance C	Capacitance C	Compliance $C_m = 1/k$
3.	Resistance	Resistance	Resistance R_m

Note that:

(*i*) Motion of the fluid (*e.g.,* air) in an acoustical system can be taken as equivalent to the flow of current in an electrical circuit.

(*ii*) The difference of acoustic pressures in an acoustic system is just analogous to the difference in electrical pressures (potential difference) across an electrical circuit.

(*iii*) The acoustical inertness is analogous to inductance, acoustical resistance is analogous to the resistance and acoustical *compliance is* analogous to the capacitance. As a whole, *acoustic impedence* is analogous to impedence of the electrical circuit.

1.16. SUPERPOSITION (OR INTERFERENCE) OF WAVES

When two or more waves reach at a point at the same time, the redistribution of energy takes place. This phenomenon is known as *Superposition* of waves.

When two or more travelling waves are moving through a medium, the resultant displacement at any point is the *algebraic sum* of the displacements due to individual waves.

(*i*) When the two waves arriving at a point are in the same phase, the resultant is a sum of two individual disturbances (waves) and a wave of larger amplitude results. This is known as *Constructive Interference*. [See Fig. 1.15 (*a*)]

(*ii*) When two waves arriving at a point are in the opposite phase, the resultant disturbance (wave) is the difference of the two individual disturbances and a wave of smaller amplitude results. This is known as *destructive interference*. [See Fig. 1.15 (*b*)]

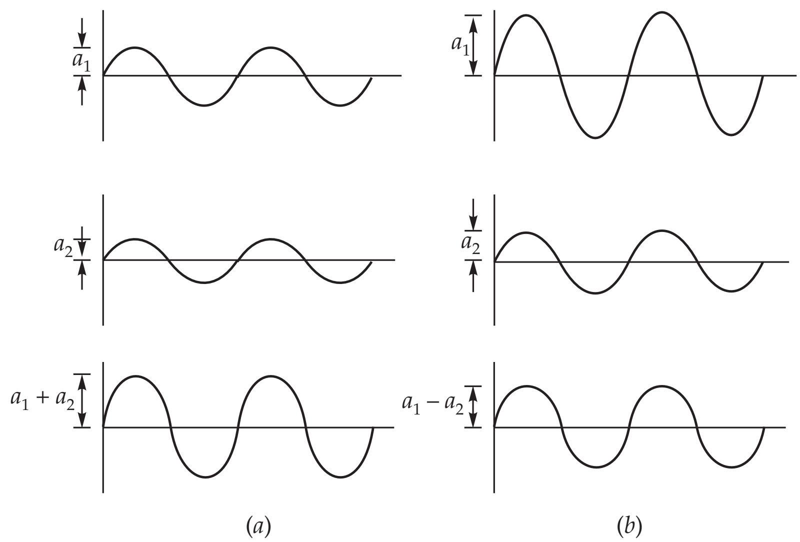


Fig. 1.15

Important

(*a*) Two waves of same frequency moving in the same direction result in *interference*.

(*b*) Two waves of slightly different frequencies moving in the same direction give rise to *beats*.

Application. The phenomenon of super position (interference and production of beats) is very much used in broadcasting, transmission and reception of A.V. signals.

The phenomenon of interference can be shown with two loud speakers.

(*i*) In Fig. 1.16 (*a*), two identical loud speakers have been connected separately with a same amplifier. Place the two speakers close to each other.

Now disconnect one speaker. Hear the sound coming from second speaker. A sound of certain intensity will be heard.

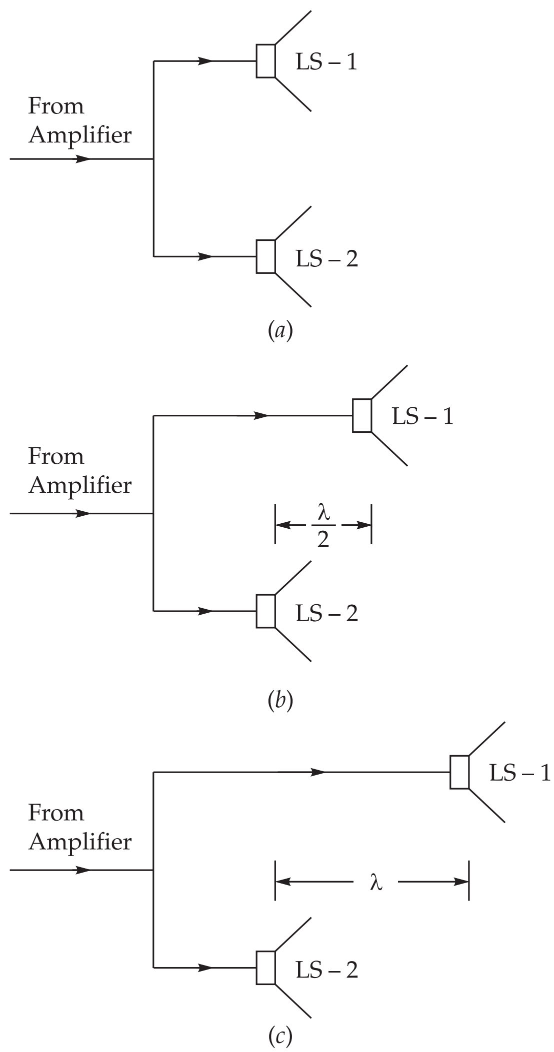


Fig. 1.16

Now connect the second loud speaker and hear the sound coming from both. A loud sound will be heard, it will be nearly double of the previous sound.

It is a case of *constructive interference*, as the sound waves from two speakers are in phase.

(*ii*) Now move one loud speaker backward by a distance $\lambda/2$, a very low sound will be heard. It is because the waves coming from two speakers are out of phase. This is a case of *destructive interference.* (Fig. 1.16 *b*)

(*iii*) Now move the speaker further backward so that its distance from first speaker becomes λ. Again a loud sound will be heard. This is a case of constructive interference. We come to the following conclusion: (Fig. 1.16 *c*)

(*a*) When distance between two speakers is an *integral multiple* of λ. Constructive interference takes place.

(*b*) When the distance between the two speakers is an *odd multiple* of $\lambda/2$, destructive interference takes place.

1.17 BEATS

The periodic rise and fall in the loudness of sound, caused by the superposition of two sound waves of slightly different frequencies is known as *beats*.

When two sound waves of slightly different frequencies travelling along the same straight line in same direction are superimposed on each other, there is alternately a *rise and fall* in the loudness of the resultant sound. This gives rise to *beats*.

One beat is formed by one *loud* and one *faint* sound. The frequency of beats is equal to the difference in frequencies of two sound waves.

The beats are heard only when difference in frequencies of two sound waves is not more than 12 as persistence of human hearing is 1/12 of a second.

If A is the amplitude of the waves having different frequencies (difference is not more than 12), the maximum amplitude will be $A + A = 2A$ and minimum amplitude will be $A - A = 0$. It means, at one time maximum sound will be heard and soon it will be silent. So one set of "maximum sound" and "no sound" is known as *beat*.

If the frequencies of the sound waves are f_1 and f_2, the frequency of beats will be $= f_1 - f_2$.

Example. If two tuning forks of frequencies 262 and 278 are sounded together, the number of beats formed = 278 – 262 = 16.

1.18 AUDIO FREQUENCY (AF)

This is the range of frequency, that a human ear can respond. This is in between 20 Hz to 20 kHz.

The range of frequency a human being produce is in between 100 Hz to 10 kHz.

1.19 FREQUENCY SPECTRUM (OR RANGE)

The table 1.7 gives the frequency range of different regions.

Table 1.7

S. No.	*Region*	*Wavelength (m)*	*Frequency (kHz)*
1.	X-rays	7×10^{-11}	3×10^{16}
2.	Ultra violet rays	1×10^{-8}	3×10^{12}
3.	Light	4×10^{-7}	7×10^{11}
4.	Infrared rays	7×10^{-7}	4×10^{11}
5.	Radio Waves		
	(*i*) Ultra High Frequency (UHF)	1×10^{-1}	3×10^{6}
	(*ii*) Very High Frequency (VHF)	1	3×10^{5}
	(*iii*) High Frequency (HF)	10	3×10^{4}
	(*iv*) Medium Frequency (MF)	10^{2}	3×10^{3}
	(*v*) Low Frequency (LF)	10^{3}	3×10^{2}
	(*vi*) Very Low Frequency (VIF)	10^{4}	3×10^{1}

1.20 REFLECTION OF SOUND

When the sound waves after striking a surface are sent back, the phenomenon of coming back of the sound in the original medium is called *reflection* of sound. The reflection of sound is similar to the reflection of light.

The practical examples of reflection of sound are: echo, thunder, reverberation, etc.

Laws of Reflection of sound

The laws of reflection of sound are the same as that of reflection of light:

(*i*) The angle of incidence is equal to the angle of reflection

$$\angle i = \angle r.$$

(*ii*) The incident ray, the normal and the reflected ray—all three lie in the same plane.

The reflection of sound is shown in Fig. 1.17 and it can be easily proved that the "sound image" A' lies as far behind the surface as A is in front of the surface (as in case of image in a plane mirror).

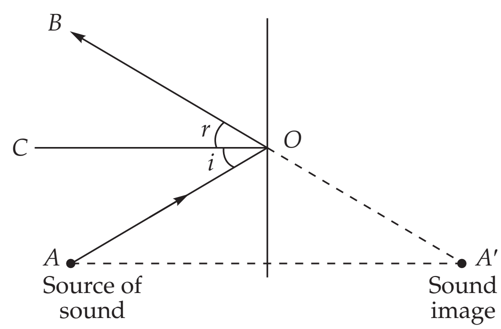


Fig. 1.17

The practical example of the sound image is that when an engine passes by a long wall of a building, its whisling can be heard behind the wall.

The phenomenon of reflection of sound is used to find depth of water by hearing echo of the sound. The laws of reflection of sound and that of light are same. The wave length of sound is 10^5 times that of light hence, the lenses and mirrors used for getting sound image are enormously large as compared to that of optical lenses and mirrors.

Applications of sound reflection

(*i*) Concave surfaces concentrate the sound waves after reflection hence these surfaces are used as *sound reflectors*. See Fig. 1.18 (*a*).

(*ii*) Convex surfaces "diffuse" sound waves after reflections. These surfaces, hence used in broadcasting studios to diffuse sound in the whole studio. See Fig. 1.18 (*b*).

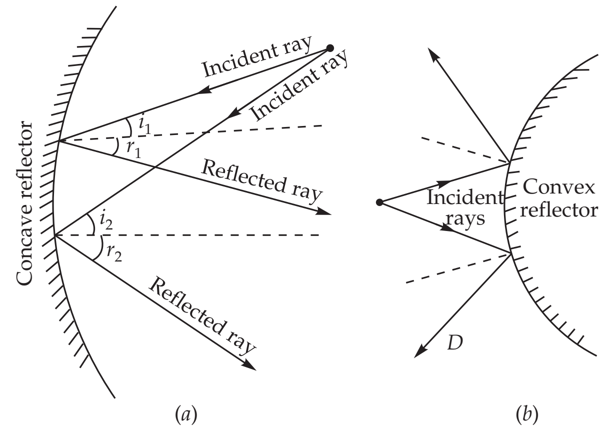


Fig. 1.18

1.21 REFRACTION OF SOUND

When a sound wave enters from one medium into another, it bends (like light rays), the phenomenon is known as *refraction* of sound. This is similar to the refraction of light.

When sound waves pass from one medium to another there is a change in the velocity of waves, as a result they are bent according to the laws of refraction of light. In other words, velocity of sound in air is 340 m/s and in carbondioxide (CO_2) is 265 m/s, hence when a sound wave enters from air into CO_2, it will bend towards normal. Similarly when sound wave travels from air into iron (velocity of sound in iron is 5000 m/s), the wave will go away from the normal.

The Fig. 1.19 shows the phenomenon of refraction of sound where, *AO* is the incident sound wave and *OB* is the refracted ray.

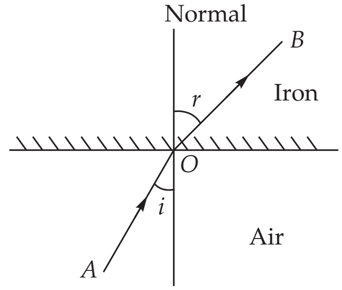


Fig. 1.19

The sound travels from air to iron, hence it is bent away the normal. That is angle of incidence $\angle i$ is less than angle of refraction $\angle r$.

If the velocity of sound in air is v_1 and in iron v_2, then according to the laws of refraction

$$\frac{v_1}{v_2} = \frac{\sin \angle i}{\sin \angle r} \quad (\angle r > \angle i)$$

The phenomenon also depends upon the density of medium and its temperature.

1.22 DIFFRACTION (SCATTERING) OF SOUND (Fig. 1.20)

Whenever there is an obstacle in the way of sound waves, a change in the direction of wave takes place. This phenomenon is known as diffraction (or scattering) of sound.

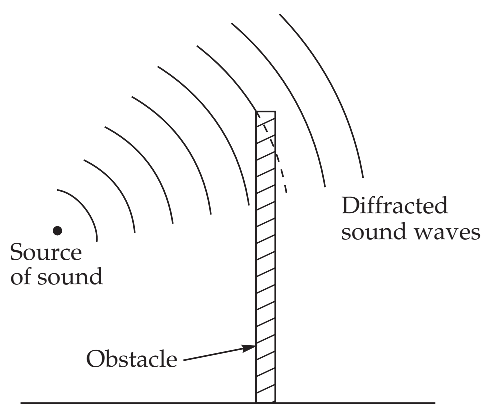


Fig. 1.20

Due to this effect of sound, we can hear a person speaking at the other side of a wall.

When a person speaks from one side of a wall, some of his sound reflects back from the wall but some crosses the wall and reaches other side of the wall. Which is called diffracted or scattered sound.

These scattered (or escaped) waves are said to be forming an *image* (or shadow) of the sound on the other side. The phenomenon is also affected by the wave length of the sound and size of the obstacle.

(*i*) A small obstacle does not produce any hinderance in the path of sound waves. It does not cause any "shadow".

(*ii*) An obstacle, which is much larger than the wave length of the sound waves forms a shadow *i.e.* diffraction of sound occurs.

This knowledge is helpful in designing acoustics of buildings.

1.23 CHARACTERISTICS OF SOUND

Important characteristics of sound are under:

1. Pitch. The sensation of *shrillness* produced by a sound is known as its *pitch*. The "Cry" of a child has a *high* pitch, whereas soft voice of a lady has *low* pitch. Greater the frequency, higher is the pitch. The sound produced by a mosquito has a high pitch, the roaring of a lion has a low pitch.

The unit of pitch is *Mel*. A pure tone of 1 kHz frequency has a pitch of 1000 mels. But frequency and pitch does not vary linearly. See Fig. 1.21. By pitch, we may compare two or more sounds. Pitch is affected by sound pressure. At increased sound pressure, pitch of a low frequency tone decreases, but at high frequency tone increases.

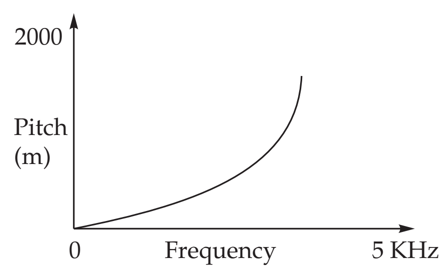


Fig. 1.21

2. Timbre. By this characteristic of sound, we can recognise two equal loud tones having same pitch but coming from different directions. Timbre, like pitch is also a complex characteristic and a function of intensity and frequency of sound.

3. Harmonic structure. Every sound wave contains a fundamental frequency and its harmonics. The fundamental frequency decides the *pitch*, whereas harmonics decide *timbre* of the sound.

4. Loudness or Intensity. The intensity of a sound is known as its loudness. Greater the loudness (*i.e.* intensity), greater is the effect on our ears.

The unit of intensity of sound is *Bel*. The bel is a logarithmic unit. More practical unit of intensity is decibel (db) which is one tenth of a bel.

The human ears donot distinguish different frequencies *linearly*. But they do it *logarithmically*.

1 Bel = 10 db

and $$1 \text{ dB} = 10 \log_{10} \frac{P_2}{P_1}$$

where P_2 and P_1 are output and input powers respectively.

The db has also a practical significance. As our ears has a logarithmic response, the unit db completely matches with this quality of our ears. For example, if loudness of sound is made 100 times, the effect on our ears will not be 100 times but only double (log of 100 = 2). If it would not happen, our ears may get damaged.

The minimum intensity of sound, which a human ear can respond is equal to the energy of 10^{-12} watts per sq. meter and the maximum intensity of sound that our ear can hear without pain is 1 watt per sq. meter.

The table 1.8 gives intensity (loudness) of some sources of sound in *phons* and table 1.9 gives in *dB*.

Table 1.8

S. No.	*Type of Sound*	*Loudness of Intensity of Sound (in Phons)*
1.	Threshold of hearing	0 (zero)
2.	Normal conversation	40
3.	High pitch sound	75
4.	Shouting noise	100
5.	Threshold of pain	150

Table 1.9

S. No.	*Source of Sound*	*Loudness in dB (Intensity)*
1.	Whispering	18
2.	Buzzing	42
3.	Normal talking	52
4.	Traffic (Heavy)	78
5.	Roaring of lion	100
6.	Thunder	110
7.	Siren	125

Intensity or loudness is

(*i*) directly proportional to the square of amplitude.

(*ii*) inversely proportional to the square of distance from the source.

(*iii*) directly proportional to the area of vibrating body (source).

Note. Intensity is also known by the amount of sound energy passing per unit area per second. It can also be measured in watts per second per cm^2 (or in *Phons*).

1.24 ULTRASONICS

The frequency range to which our ear can respond is known as "Audio Frequency". The AF range is from 20 Hz to 20 kHz.

The 20 Hz is known as our "lower limit of audibility" and the 20 kHz as our "upper limit of audibility".

Our ears neither respond a frequency below 20 Hz nor a frequency above 20 kHz.

Sound waves of frequency more than 20 kHz are known as "*Ultrasonics*".

Dogs can hear ultrasonics upto 50 kHz *i.e.* they can respond to the frequencies above A.F. The bats can produce ultrasonic frequencies and find their path by detecting the echoes after reflection from obstacles.

The frequencies below 20 Hz are known as "infrasonics". Both ultrasonics as well as infrasonics come under Inaudible Frequency range for a human being.

Production of ultrasonics

Ultrasonics can be produced by

1. Magnetostriction Oscillators
2. Crystal Oscillator

(1) **Magnetostriction Oscillators (Fig. 1.22).** It is based on the principle that "when a rod of a magnetic material is magnetised, it undergoes a change in its length".

This change in length (+ ve or – ve) has a frequency twice that of the frequency of applied magnetic field.

A rod of magnetic material (of Nickel) has two coils Q_1 and Q_2 wound on it (See Fig. 1.12). A current is passed in the coils through a triode. As a result, the rod is magnetised and demagnetised repeatedy depending upon the frequency of the supply. The length of the rod changes accordingly and its free end begins to vibrate producing ultrasonics.

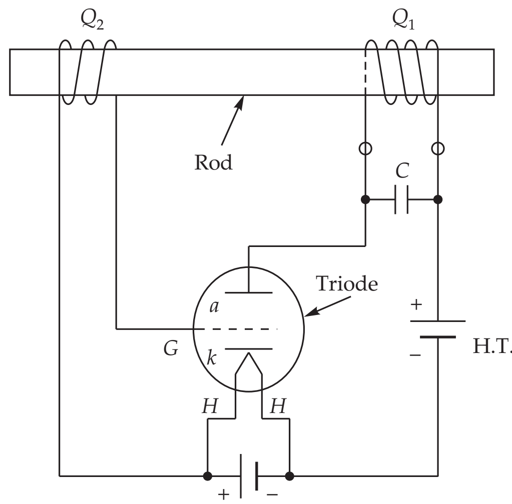


Fig. 1.22

(2) **Crystal Oscillator (Fig. 1.13).** This oscillator works on the principle of "Piezo effect". Which says that "when HF voltages are applied across the two opposite faces of a crystal (of Quartz, Rochelle salt etc.) the crystal starts producing ultrasonics.

A thin quartz crystal is placed between two metallic plates—this forms a capacitor with crystal as dielectric. The arrangement is connected across an AC supply. (See Fig. 1.23)

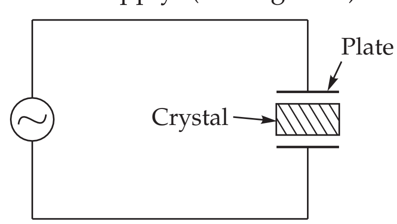


Fig. 1.23

Due to piezo effect, the crystal starts vibrating at very high frequency, thus producing ultrasonics. If the size of the crystal is selected such that the natural frequency of the crystal is equal to the frequency of the supply, "Resonance" occurs and amplitude becomes very large with this, oscillations of about 560 kHz can be produced.

The frequency that can be produced is given by

$$f = \frac{1660}{t} \text{ kHz}$$

where t is the thickness of the crystal in millimeters.

Ultrasonic frequencies are required in radio/TV broadcasting, transmission and reception.

1.25 TRANSDUCERS

A transducer is a device/equipment that converts an energy from its one form to the other. For example, microphone is a device that converts sound energy into electrical energy. Loudspeaker is another

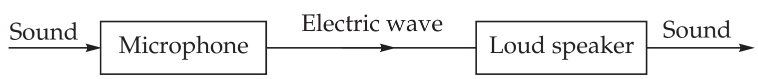


Fig. 1.24

device which converts electrical energy into sound energy. Hence microphones and loudspeakers are transducers (Fig. 1.24). Transducers play an important role in audio and video systems.

1.26 HUMAN EAR (See Fig. 1.25)

Our ear is also a transducer. When sound waves strike its *membrane*, it converts them into sensational pulses. These pulses reach our brain.

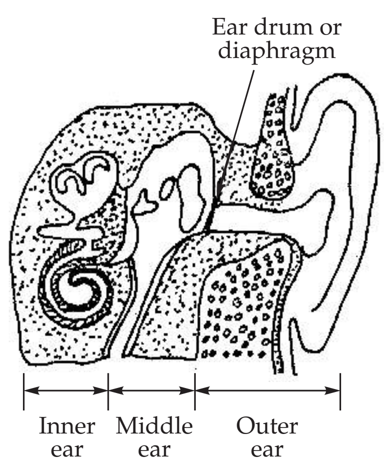


Fig. 1.25

The brain also in turn acts as a transducer and converts these pulses into the original sound, which we hear. Just as a similarity our ear acts as microphone, whereas the brain acts as a "loudspeaker".

Our ear consists of three main parts:

1. Outer ear
2. Middle ear
3. Inner ear.

1. **Outer ear.** The outer ear has a special shape, which increases its surface area and it can "catch" signals of large surroundings effectively.

2. **Middle ear.** The most important part of middle ear is the "ear drum" or *diaphragm* (membrane). The sound waves reach the membrane and produce oscillations in it. The membrane starts oscillating with a frequency and amplitude according to the pressure of sound striking it.

3. **Inner ear.** The oscillations reach the inner ear which is connected to our brain. These oscillations are converted by the brain into original sounds.

1.27 SALIENT FEATURES OF HUMAN EAR

(1) The human ear is very sensitive to sound. It can even respond to sounds having intensity as low as 0.1 Pico watt per m^2.

(2) The movement of the diaphragm may be as minimum as one millionth of a micron *i.e.* = 10^{-6} m.

(3) The response of the ear is not linear but *logarithmic*. In other words, loudness of sound hearld by our ears is not according to the intensity of sound but according to the "log of intensity of sound".

(4) Our ear can not detect a difference of 1 dB between the two sounds.

(5) Our ear can do *sum* and *difference* of two sounds. It can also find the direction of a sound even, if it is weaker.

(6) Our ear is more sensitive to higher frequencies and less sensitive to lower frequencies.

(7) Our ear is most sensitive for frequencies of 3 kHz to 5 kHz at all ages. It decreases with age. Children's ears are sensitive upto 20 kHz.

(8) Ears can detect a difference of 3 kHz in frequency and 1 db in intensity.

(9) If we listen a weaker and a louder sound simultaneously the former is supressed by the later. Movever the ear finds the direction of a sound which is received first, however weaker it may be.

(10) Ear can respond sound pressures between 10^{-4} microbars to 10^{4} microbars.

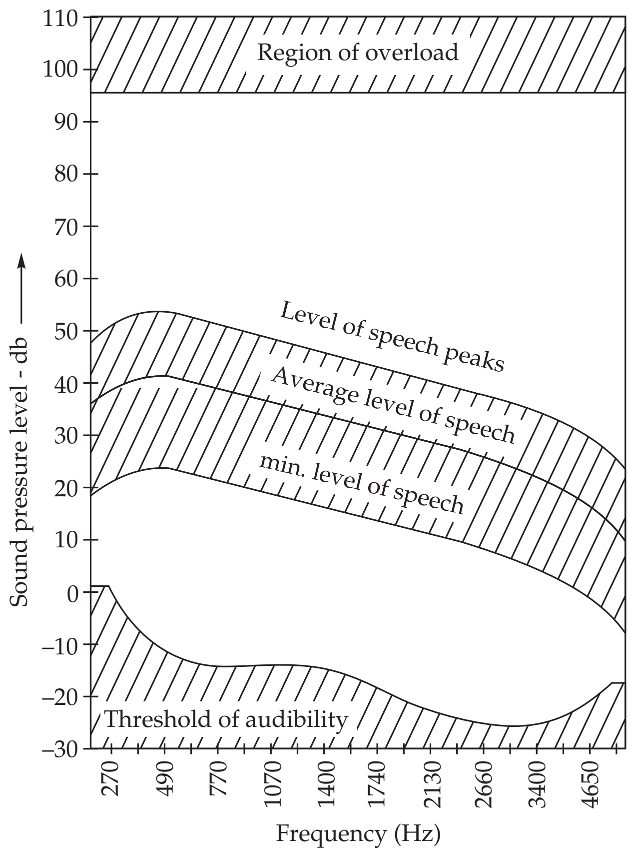


Fig. 1.26

(11) Ear can respond Audio frequency range (20 Hz to 20 kHz) and it can recognise a particular frequency sound out of many of different frequencies.

(12) The minimum sound signal a person can respond is called his *Threshold of hearing*. This value varies from man to man and age to age.

(13) Our ear judges "intervals" (ratio of frequencies f_1/f_2) and not the actual difference between 2 frequencies *e.g.* a change from 200

Hz to 1000 Hz will be recognised equally by our ears as a change from 50 Hz to 250 Hz as the ratio is same (the difference may be different) between the two frequencies.

The Fig. 1.26 shows important characteristics of human ear.

Frequency response curve for human ears (Fig. 1.27)

As mentioned earlier, the human ears respond to the whole audio frequency range (20 Hz to 20 kHz). The Fig. 1.27 shows a curve between sound intensity (in Phons) and frequency (in Hz), to which human ear can respond. At about 140 phons, we start feeling pains in the ear and this is known as *Threshold of pains*, though above 110 phons our ears feel discomfort. This intensity is called "*Threshold of discomfort*". But habitual (constant) hearing a sound above 160 phons can cause severe damage to our ear drums leading to total or partial deafness.

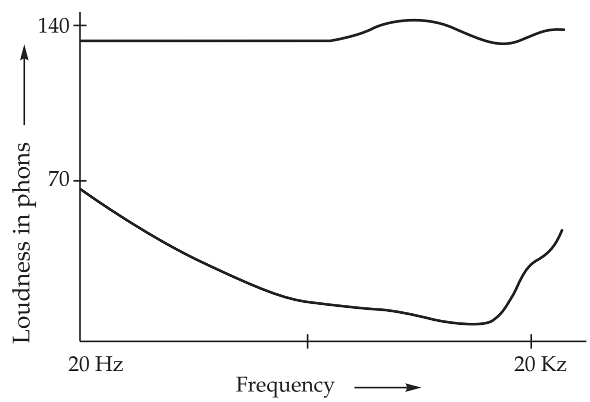


Fig. 1.27

1.28 TUNING CIRCUITS

A tuning circuit is an important component of all audio-video systems. It is basically a L-C circuit which can produce a wanted frequency. They are called tuning circuit as they are used to *tune* a particular station or channel in AV systems.

They may be

(*a*) **Series LC Circuit** [Fig. 1.28 (*a*)]. In which an inductance (L) and a capacitance (C) are connected in series.

(*b*) **Parallel LC Circuit** [Fig. 1.28 (*b*)]. In which, the inductance (L) and capacitance (C) are connected in parallel.

An a.c. supply is given to the combination.

At resonance, the value of "Inductive Reactance" becomes equal to the "Capacitive Reactance" of the circuit.

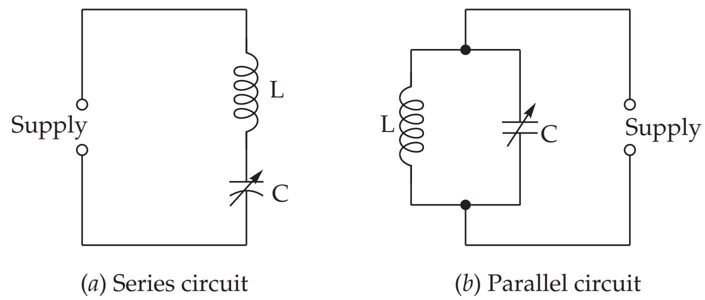


Fig. 1.28

i.e., Inductive Reactance = Capacitance Reactance

or, $$2\pi f_r L = \frac{1}{2\pi f_r C}$$ [f_r = Resonant frequency]

or, $$f_r = \frac{1}{2\pi\sqrt{LC}}$$

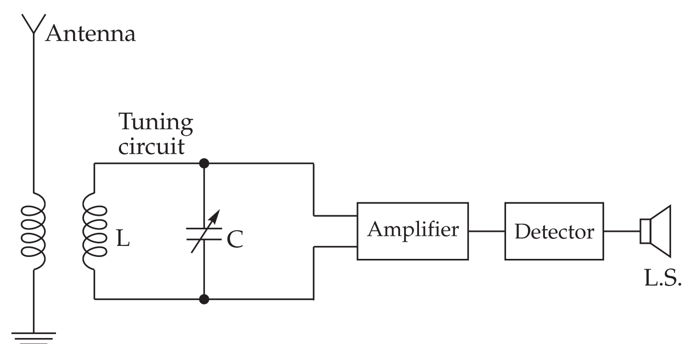


Fig. 1.29

In tuning circuits, generally L is kept fixed and C as varied. By varying capacitance resonance is established, and a particular frequency f_r is produced. Which is known as *Resonant Frequency*. The circuit, therefore is also known as resonant circuit.

For example, the frequency of Vividh Bharti radio station is 1350 kHz. By operating the radio knob, we change the value of C so that resonance is established with this frequency. Now this frequency is entertained by the radio and all others are earthed. The Fig. 1.29 shows tuning circuit of a radio receiver.

1.29 FREQUENCY RESPONSE

It shows the "response" of an equipment (amplifier or micro-phone etc.) for a particular range of frequencies. The "Gain" of a source depends upon the frequency. The curve between "Voltage gain" and "signal frequency" is known as "*Frequency response curve*".

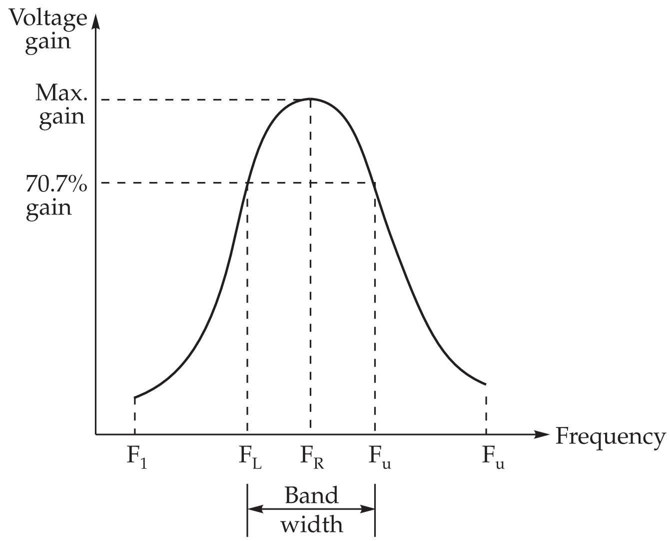


Fig. 1.30

The Fig. 1.30 shows frequency response curve of an amplifier. We can see that gain of the amplifier increases as the frequency increases till it becomes maximum at certain frequency F_r called "*Resonant Frequency*". Now if the frequency of signal is increased beyond f_r, the gain of the amplifier will decrease. In the Fig. 1.30, F_L is lower cut off frequency and F_U is upper cut off frequency,

The performance of all audio-video systems depends upon their frequency response.

1.30 BAND WIDTH

The range of frequency over which the gain of a system does not fall below 70.7% (of max. gain) is called as *Band width*.

The value 70.7% has a typical significance. It has been seen that if the gain of a system (say a loudspeaker) falls to 70.7%, our ear cannot *detect* this fall. But if the gain falls below this value, our ears will feel it. This is the reason that it is taken to be 70.7%.

In the Fig. 1.30, $F_L - F_U$ is the "Band width", during this range of frequency, the gain of the system does not falls below 70.7%. That is, remains constant.

For distortionless output, the range of frequency must be within B.W. of the system. In other words, the system will work "effectively" during this range.

If we calculate in db

$$\text{Fall in gain} = 20 \log_{10} \frac{100}{70.7} \text{ db} = 3 \text{ db}$$

Hence band width may also be defined as the range of frequency at the limits of which its gain falls by 3 db from maximum gain.

Band Width For	(*i*)	Telephone speech	: 300 to 3500 Hz
	(*ii*)	Normal programs	: 75 to 8000 Hz
	(*iii*)	HiFi program	: 50 to 1500 Hz.

1.31 DECIBEL

The gain of a system is generally expressed in *number*. For example, if we say that gain of a system is 5, it simply means that its output will be "5 times of the input".

For audio and video systems generally 'bel' (or decibel) is used as a *unit* for gain. It is a convenient unit and calculations become easier.

Bel is a logarithmic unit. It has its own practical significance. The response of ears in also logarithmic. In simple words, loudness of sound heard by our ears is not according to the intensity of sound *but* according to the *log of intensity of sound*.

For example, if the loudness of a loudspeaker is increased 1000 times, our ears will have only 3 times effect and not one thousand times, as the log of 1000 = 3.

Hence this unit tallies with the "natural response of our ears".

A bel can be defined as under:

"The log of the "Gain" to the base of 10 is called as Bel gain".

$$\text{Power gain (P.G.)} = \log_{10} \frac{P_{out}}{P_{in}} \text{ bel.}$$

$$1 \text{ bel} = 10 \text{ decibel}$$

$$\therefore \quad \text{P.G.} = 10 \log_{10} \frac{P_{out}}{P_{in}} \text{ db} \qquad ...(1)$$

On the same lines

$$\text{Voltage gain} = 20 \log_{10} \frac{V_{out}}{V_{in}} \text{ db} \qquad ...(2)$$

and $$\text{Current gain} = 20 \log_{10} \frac{I_{out}}{I_{in}} \text{ db} \qquad ...(3)$$

1.32 IMPEDENCE MATCHING

"If impedence of the source and impedence of the load are equal, maximum power is obtained at the load, and the limit of the maximum power which can be obtained is only 50%".

This is known as "maximum power transfer theorem".

At this stage, the load is said to be "matched" with the source and this is known as "*impedence matching*".

In audio and video systems, this is very important. Generally, we have amplifier as "*source*" and "loudspeaker" connected to the amplifier acts as *load*. To obtain maximum power transfer from "amplifier to speaker", it is necessary that impedence of amplifier is same as that of the "loudspeaker". In other words, "Impedence matching" between the two is essential. In such cases efficiency is *sacrificed* for more output. The arrangement is known as *Public Address (PA) System* which is a most practical example of impedence matching.

1.33 SONE

This term is used to determine increase in loudness produced by 1 kHz, Remember 40 Phone sound is equal to 1 Sone.

1.34 INTELLIGIBILITY

This factor determines the frequency range of sound required for satisfactory transmission of speech. It is also called *Articulation.*

1.35 SOUND SYNTHESIS

This is the process of producing artificial sounds. We can artificially produce sound of any musical instrument or even of an orchestra. For this, audio wave forms are generated electrically and mixed by electronic tuner. So synthesised sound is a complex form produced by electronic circuits.

1.36 TYPES OF SOUNDS

Almost all sounds are of three types: (Fig. 1.31)

(1) Speech
(2) Music
(3) Noise.

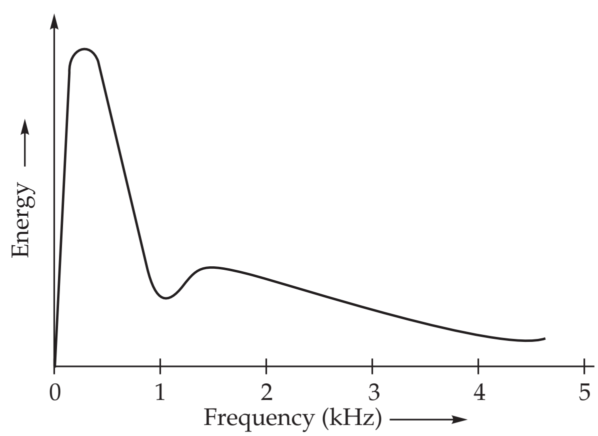


Fig. 1.31

(1) **Speech.** The speech is the common sound we speak from our mouth. This sound is generated in our *vocal tract* and radiated from our *lips*. The speech characteristic of different persons is different however, their sound pressure lies in the range of 65 dB to 80 dB, when spoken in a normal tone. The speech becomes directional at high frequencies, in other words it can then be hearld in a particular direction only.

We can generate a frequency between 100 Hz to 10,000 Hz by mouth. The Fig. 1.31 shows energy distribution versus frequency. It is clear that at high frequency, there is power loss. The production of speech is described in terms of *Intelligibility* which means how a speech can be recognised and understood by a listener. This will depend upon the energy of sound spoken, hearing power of the listener and the noise in the surrounding.

(2) **Music.** The sound, which gives a pleasant sensation to the listener is called as musical sound. The musical tones have simple harmonic structure with a regular wave form. It consists of fundamental frequency and its harmonics. The music tones are produced by vibrating bodies such as *strings*, *membranes*, and *air columns*. Different musical instruments play over different range of fundamental frequency (See Table 1.10).

Table 1.10

Musical instruments	*Frequency range (Hz)*
1. Xylophone	150—3000
2. Pipe organs	20—8000
3. Harmonica	200—1000
4. Saxophone	50—300
5. Flute	300—2000
6. Trumpet	200—900
7. Piano	25—4000
8. Banjo	130—800
9. Cello	150—1000
10. Violin	200—3000

Most of the musical instruments are "directional" *i.e.* they have their maximum effect in a particular direction. This knowledge helps us to arrange an orchestra.

Musical sound is different to that of a speech in the following ways:

(*a*) Musical sound contains different fundamental frequencies, whereas the speech contains one fundamental at a time.

(*b*) Musical sound has a wider frequency band as compared to the speech which has a very narrow band.

(*c*) Music is more sensitive to distortions as compared to the speech.

(*d*) Musical tones have Simple Harmonic Motion (SHM) with regular amplitude waves. A musical tone with one frequency can be represented by Fig. 1.32 (*a*).

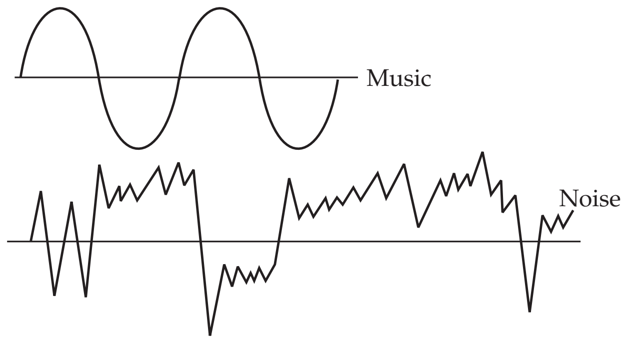


Fig. 1.32

(3) **Noise.** Noise is that, which gives unpleasant (irritating) sensations to ears. They are irregular amplitude waves [Fig. 1.32 *b*)]. Noise causes fatigue and pains in ears and hence the listener avoids it.

(A) Noises can be classified as:

(*a*) *Thermal noise.* This noise is generated in resistors due to thermal agitation of electrons. The amount of noise depends upon the value of resistor, and its operating temperatures.

(*b*) *Electrical noise or humming.* This noise is generated due to power line frequencies (50 Hz) in the audio device. The power frequencies contain its harmonics (*i.e.* 100 Hz, 200 Hz etc.) which causes noise. A good filter circuitary can reduce this noise.

(*c*) *Cross talk.* This noise enters in an audio equipment due to presence of other circuits in the vicinity, which causes inductive or capacitive coupling with the audio circuits. This noise can also be reduced by proper filtering.

(B) The noise can also be classified as:

(1) *Generated noise*. Thermal and electrical noises come under this class or category. This noise is generated in amplifiers.

When signal enters an amplifier, this noise is produced or generated due to various components (resistors, capacitors, transistors etc. of the amplifier). This noise is added to the signal (See Fig. 1.33). Generated noise further may be of the following types:

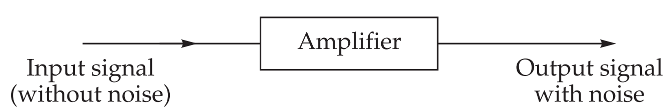


Fig. 1.33

(*a*) *Johnson nose.* This noise is produced when a signal passes through a resistor. The atoms of the resistor remain in continuous vibration and produce noise, which is added up in the signal.

(*b*) *White noise.* The vibrations produced by thermal effects within a resistor cover a wide frequency range and therefore the noise generated consists of a wide spectrum of frequencies. This wide band noise is sometimes called white noise.

(*c*) *Pink noise.* Pink noise is a signal, whose *noise power* per unit frequency *interval* is inversely proportional to frequency itself over a specified range. It is white noise passed through filter.

(*d*) *Shot noise.* The internally generated noise in semi conductors and vacuum tubes on account of random movement of charges is called *shot noise*.

(2) *Conducted noise.* The power supply to the amplifiers may itself be a source of noise as it contains ripples, spikes etc. This noise is called *conducted noise.*

(3) *Radiated noise.* There may be electrical or magnetic fields in the surrounding. Unwanted signal may radiate in to the audio device and produce noise. Such noise may be called *Radiated noise.*

The Fig. 1.34 shows sources of radiated noise.

(4) *Environmental noise*. The noise present in the surrounding is called environmental noise. It may be

(*i*) *Transient noise*. Which occur suddenly like thunder or explosion near by.

(*ii*) *General noise or Ambient noise*. The noise present all the time *e.g.* class room, auditorium.

Note. All types of noises may be broadly classified as:

(1) Electrical noise

(2) Environmental noise.

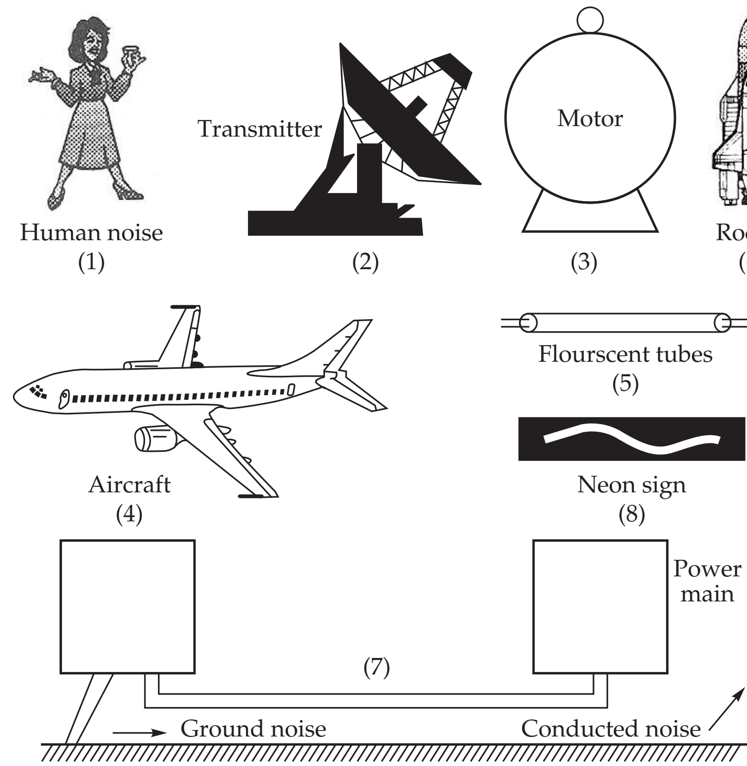


Fig. 1.34

1.37 HARMFUL EFFECTS OF NOISE

Working continuously in a noisy environment has the following ill effects on human beings:

(*i*) It produces hypertension/high B.P.

(*ii*) It causes mental fatigue and lowers efficiency

(*iii*) It may retard mental growth.

(*iv*) It may cause harm to our hearing system

(*v*) It may cause nervous breakdown.

1.38 METHODS OF REDUCING NOISE

(*i*) The *electrical noise* may be reduced by using regulated power supply, use of synchronous and servo motors in drives (like recorders), regular cleaning of recorder heads, and by using techniques (*e.g.* Dolby method).

(*ii*) The environmental noise may be reduced by using sound absorbers/insulators in the room. In general, room should be provided with thick curtains.

1.39 OTHER IMPORTANT TERMS

(1) **Signal to Noise (S/N) ratio (SNR).** The ratio of desired signal to the unwanted noise is called S/N ratio. It may be expressed as

$$\text{S.N.R.} = \frac{\text{S}}{\text{N}} = \frac{\text{Signal Power}}{\text{Noise Power}} = \frac{\text{Signal expressed in (volt)}^2}{\text{Noise expressed in (Volt)}^2}$$

A large SN ratio is always desired in audio video systems. This can be done by decreasing noise level by some means. If SN ratio is small (that is more noise), it may be a matter of concern.

(2) **Noise figure.** The noise figure (NF) is the degradation of the signal to noise ratio which may be attributed to the amplifier. It may be defined as ratio of total available output noise power (P_1) to the part of output power (P_2) resulting from thermal noise or source resistance.

Thus $\quad \text{NF} = P_1/P_2.$

(3) **Octave or Interval.** It is often convenient for testing and adjustment purposes to divide the audio frequency spectrum into *bands*. The Octave is a widely used division of frequency spectrum. An octave has a 1 : 2 ratio between lower and upper frequencies. An octave band or "Interval" includes the end points and every frequency in between. The *center frequency* of an octave band is not the mathematical average of the upper and lower cut off frequencies, but is approximately $\sqrt{2}$ times the lower frequency.

As explained above, the interval of 1 : 2 is called an "octave". For example one octave of 200 Hz is 400 Hz (200 : 400 = 1 : 2) and two octaves of 200 Hz is 800 Hz and so on.

(whereas interval between 600 Hz and 200 Hz is 3 and between 200 Hz and 600 Hz is 1/3].

Mathematically number of octaves (n) between two frequencies f_1 and f_2 is given by

$$n = \log_2 (f_2/f_1).$$

(4) **Harmonics.** Harmonics are the integer multiples of fundamental frequency *e.g.* with respect to 50 Hz, its second harmonic will be = 100 Hz (2 × 50), third harmonic 150 Hz (3 × 50) and 7th harmonic = 350 (7 × 50) and so on.

Remember 105 Hz is not an harmonic of 50 Hz as it is not integer multiple.

(5) **Overtones.** All frequencies higher than fundamental are known as *overtones*. In this way, harmonics also come under overtones.

The frequencies lower than fundamental are known as *undertones.*

1.40 ARCHITECTURAL ACOUSTICS

Architectural acoustics is the branch of engineering which deals with the production of sound effects in buildings, cinema halls, auditorium etc. The behavior of sound in enclosed spaces have been of interest to designers since long.

After Mr. sabine found a simple formula for calculating reverberation time for decay of sound in a room, the work in this line started on scientific grounds.

Later on, it was found that decay time of sound can be controlled by using absorbant materials on the walls of the room. After this, search started for materials with high coefficient of absorption *i.e.*, which effectively can absorb the sound incident upon.

1.41 REVERBERATION

The persistence of sound even after the original sound is ceased to operate is known as *reverberation.* Remember that reverberation is little different to that of an *echo.*

Reverberation time. The time taken by the sound to die completely is called reverberation time.

Actually sound is reflected back by the things placed in the room and bounces inside, till its energy is converted into heat.

Reverberation time may be defined as *the time taken by the sound to decay by 60 dB* from its original level (see Fig. 1.35). At point *O*, the sound source is switched ON and sound intensity rises upto point *A*, and then it becomes steady. At point *B*, the source of sound is

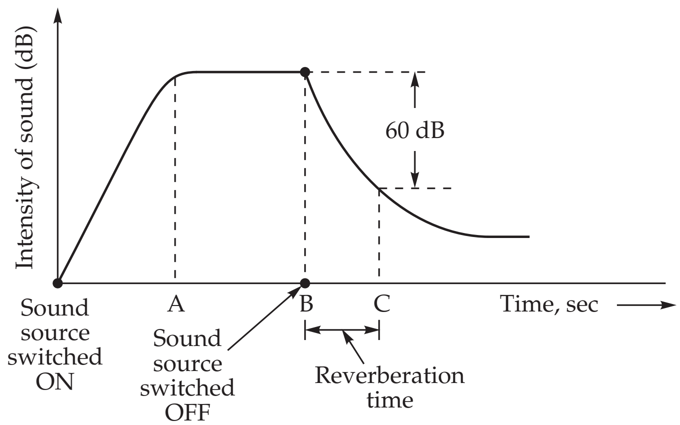


Fig. 1.35

switched off and the sound starts decaying. At point *C*, it falls down by 60 dB. Reverberation time is represented by *BC*.

1.42 SABINE FORMULA FOR CALCULATING REVERBERATION TIME

The sabine's formula for calculations of reverberation time is given by

$$t = \frac{0.161 \times V}{a} \text{ seconds}$$

where, a is the total absorption area (of sound) of the room in m^2 and V is the volume of the room in m^3.

By absoption area, we mean the total area of the surfaces in the room absorbing sound *i.e.* walls, ceiling, floor, curtains, furniture, audience etc. etc.

The table 1.11 gives approximate reverberation time (t) for few places:

Table 1.11

S. No.	*Place*	*t in seconds*
1.	Drawing room	0.5
2.	T.V. Studio	0.8
3.	Concert hall	2.2
4.	Class room	0.5
5.	Temple	2.4

Note. There is little difference between reverberation and echo. In case of echo, there is a time gap (about 50 ms) between the original sound and its echo. If the time gap is less than 50 ms, it is reverberation. The distinct echo produces irritation and hence should be totally eliminated from the room.

Reverberation has continuity of sound. A small reverberation is pleasing, therefore desirable in the design of acoustics. Total elimination of reverberation makes the sound dull and lifeless, whereas small reverberation keeps the naturality of sound.

1.43 GROWTH OF SOUND IN A ROOM

If a source of sound is operating in a room, the absorption of sound by the walls, surfaces and by the medium of the room prevents the sound intensity from becoming large. In small rooms the absorption in the medium is negligible. If absorption power of surfaces is small and growth of sound is low, when a sound source is started, reflections at the wall produce a sound energy distribution which becomes uniform with time. If the source of sound is pure tone of only one frequency, *standing waves* may be set up in the room. If the source produces a band of frequencies, the interference effect in a small room may be neglected.

1.44 DECAY OF SOUND IN A ROOM

The intensity or sound pressure level in a room decreases with time at a constant *decay rate D* in decibel per second given by

$$D = \frac{1.087\, a.c}{V}$$

where a is the total absorbent surface area, V the volume of the room and c the velocity of sound in air.

As the reverberation time t is the time required for the level of the sound in the room to decay by 60 db (see Art. 1.34)

i.e.
$$t = \frac{60}{D} = \frac{60}{1.087 \times ac/V} = \frac{55.2\, V}{ac} \qquad \left[\because \; D = \frac{1.087\, ac}{V}\right]$$

Now putting velocity of sound in air c = 343 m/s, we get

$$t = \frac{55.2\, V}{a\,.\,343} = \frac{0.161\, V}{a}.$$

The volume to be taken in cubic meters and a in square meters.

Sabine formulae brought revolution in acoustics, since reverberation time can be calculated at once if volume and absorbant surface area of the room are known. By surface area we mean, the total absorption of sound and that can be easily improved by using absorptive materials on the walls. In other words, the reverberation time t can be easily controlled.

1.45 ABSORPTION OF SOUND

When sound waves fall on a surface, the surface is set into vibrations and the damping force dissipates the sound energy into heat. Therefore the main cause of absorption of sound by materials are their *porosity* and *vibrations*.

When sound waves stike a surface, the sound is *scattered* in all directions. This phenomenon is a source of trouble in cinema halls, auditoriums and damages the quality of *acoustics* of the buildings.

The amount of absorption of sound differs from material to material and medium to medium.

The coefficient of absorption or absorbing power may be defined as follows:

"The ratio of sound energy absorbed by a surface to the total sound energy falling on it is called its *coefficient of absorption or absorbing power*".

The coefficient's value is between 0 to 1.

The table 1.12 gives value of the coefficient for some materials.

Table 1.12

S. No.	*Materials*	*Coefficient of absorption*
1.	Marble	0.01
2.	Concrete, Plaster	0.02
3.	Glass	0.03
4.	Wood	0.05
5.	Carpet, cork	0.25
6.	Furniture, Asbestos	0.30
7.	Human body	0.50
8.	Open window	0.98
9.	Brick wall	0.30
10.	Wooden door	0.05
11.	Curtains	0.18
12.	Cushions	0.21
13.	Ceiling, Floor	0.05

Note. The coefficient of absorption usually increases with increase of frequency.

1.46 ACOUSTIC FACTORS IN ARCHITECTURAL DESIGN

(A) (*i*) The sound should be sufficiently loud and uniform throughout the room or hall.

(*ii*) There should be no echo.

(*iii*) The sound should die soon, after it is ceased to operate *i.e.* Reverberation time should be less.

(*iv*) Sound absorbers should be used.

(*v*) Human body is a good absorber. This is the reason that a hall full of audience has better sound effect than when it is empty. (See Fig. 1.26)

(*vi*) The ceiling should be covered with some good sound absorbing material.

(*vii*) The hall should be free from resonance.

Echoes are heard in a room, whose ceiling is not sufficiently high. Due to repeated reflection of sound, echo remains in the room for sometime, even after the original sound has ceased.

As we have mentioned earlier, persistence of our ear is 1/12th of a second. If another sound comes to our ear within this time, we can not distinguish it. It can be easily calculated that minimum distance for an echo to be heard should be 15 meters from the listener.

(B) The following considerations are also to be kept in mind, while design of acoustics, in addition to given above:

(1) Reverberation time should be controlled. This may be done by controlling repeated reflection of the sound. For this purpose, good absorbing materials should be used. Open windows serve as perfect absorbers. The walls, ceiling should be covered with card boards, asbestos, heavy curtains etc. Audience is also a good absorber as told already a hall full of audience has better sound affects, then an empty one.

(2) Balconies, domes and such like structures should be avoided as they produce echoes and focussing effect of sound, with the same reasons, curved surfaces should be also avoided.

(3) *Acoustic treatment materials* should be used to control absorption in the hall.

The absorbing and acoustic treatment materials may be of following types:

(*i*) *Porous materials* such as curtains in which sound penetrates and the material has no tendency to produce vibrations. "Perforated panels" is another example for such materials. Ceiling is the most appropriate surface for application of these materials. False ceilings made of good absorbing materials, have become popular due to this reason.

(*ii*) *Non-porous materials.* Under this class of absorbant materials wood panels, plastic boards and acoustic tiles are very much used.

See Fig. 1.36, which gives plan of an auditorium. The arrangement will give better sound distribution and appreciating sound effects.

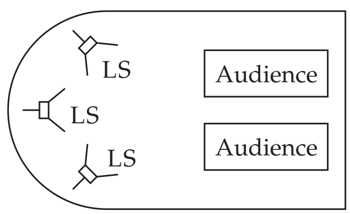


Fig. 1.36

1.47 RESONANCE EFFECT OF SOUND IN A ROOM

When a sound wave passes through an air column. At a certain relation ship between wavelength of the wave and length of the air column "resonance" occurs. The same resonance is produced in the air column in a closed pipe. A long rectangular room (length is much more than the breadth) will also resonate due to same reason given above. The resonance effect will depend upon the relation between length of the room and wavelength of sound.

On the same lines it can be explained that resonance will occur in sound devices like microphones, loudspeakers, recorders as they have mechanical parts capable of vibrating. They have a natural resonant frequency depending upon the mass of their vibrating parts. Similarly absorbing and reflecting surfaces will produce resonance effect when sound will penetrate through them.

SUMMARY

1. The consumer electronics deals with the electronic consumer devices. The audio video systems comprise majority of these devices.
2. The Audio Engineering is the branch of Engineering which deals with the study of audio (sound) energy.
3. Sound is a form of energy.
4. Audio frequency range is between 20 Hz to 20 KHz.
5. Like light, the sound also has property of reflection, refraction and diffraction.
6. Transducer converts one form of energy into another form. The human body has many transducers like eye, ear, brain etc.
7. The decibel is a logarithmic unit to measure gain, noise, pressure etc.
8. The persistence of sound, even after the original sound ceases is called reverberation.
9. Human body is a good sound absorber.
10. The tuning circuit is an important component of all audio video systems.
11. The range of frequency over which the gain of system does not fall below 70.7 per cent of the maximum gain is called bandwidth.
12. A resonator is a device that can produce resonance. An organ pipe and other musical instruments acts as resonators.

SELF TEST QUESTIONS

1. What do you understand by audio-video systems?
2. Define Audio Engineering.
3. On what factors velocity of sound depends? Write down the relation between frequency and wavelength.
4. Write short notes:

   Superposition of sound, Beats, Absorption of sound, Reverberation, Ultrasonics, Transducers, Tuning circuit, Selectivity, Frequency response, Band-width, Decibel, Impedence maching and PA system.
5. Explain reflection, refraction and diffraction of sound.
6. Explain reverberation of sound.
7. Enlist the important considerations while designing acoustics for auditoriums.
8. What are the various acoustic treatment materials?
9. Define speech, music and noise.
10. Sketch a human ear. Explain its mechanism.

11. What is the role of absorbing materials in acoustics?
12. Why an auditorium full of audience has a better sound effect?
13. Why woman's speech is difficult to be interpret?
14. Define signal/noise ratio. Give frequency response of human ears.
15. Explain the following terms: Interval, octave, Harmonics, Overtones, Timbre, Threshold of audibility and of pain, Resonance in a room.
16. What is the unit of sound pressure?
17. Give various type of noises and sources of noise.
18. How the laws of optical reflection, refraction and diffraction are applied to sound.
19. Write a brief essay on noise, reverberation and acoustics of building.
20. Define architectural acoustics.
21. Explain sabine's formula for rise and decay of sound in a room.
22. Differentiate between reverberation and echo.

OBJECTIVE TYPE QUESTIONS

Fill in the blanks:

1. Sound is a ......
2. Sound is a ...... wave motion.
3. Velocity of sound in air is ....
4. Intensity of sound at threshold of pain is ......
5. Sone is a unit of ......
6. Unit of pitch is ......
7. Sound travels ...... than light.
8. With increase in temperature, velocity of sound ......
9. Absorption of sound at low frequencies is ......
10. High A.F. contains ...... energy.
11. The A.F. range is ......
12. Energy of a sound wave is ...... at high frequency.
13. Unit of loudness is ......
14. Frequency for telephone speech is .....
15. With respect to 100 Hz, the third harmonic is ......
16. Examples of transducers are ......
17. Velocity of radiowaves is ......
18. Music-noise ratio should be ...... than one.
19. If frequency increases, wave length of a wave ......
20. A little reverberation is ......

21. Acoustics is the science of designing auditorium for creating best ......

22. Without reverberation, sound will become ......

23. Curved surfaces in buildings are ......

24. The absorption coefficient of women is ...... than men.

25. Sound effects are ...... in empty room as compared to full of audience.

26. Reverberation time in empty hall is ...... than full of audience.

27. Absorption of sound is more at ...... frequencies.

ANSWERS

1. Energy	**2.** longitudinal	**3.** 340 m/s
4. 12 W	**5.** loudness	**6.** mel
7. slowly	**8.** increases	**9.** small
10. less	**11.** 20 Hz—20 kHz	**12.** high
13. phon	**14.** 500 Hz—3500 Hz	**15.** 300 Hz
16. microphones and loudspeakers		**17.** 3×10^8 m/s
18. more	**19.** decreases	**20.** desirable
21. sound effects	**22.** dull	**23.** undesirable
24. more	**25.** poor	**26.** more
27. high.		