# |A TUTORIAL ON BASICS OF PSYCHOACOUSTICS

Reg. No. M0115

Independent project as a part fulfillment of
First year M.Sc (Speech and Hearing)
Submitted to the University of Mysore,
Mysore

ALL INDIA INSTITUTE OF SPEECH AND HEARING
MYSORE - 570 006

May, 2002

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# CERTIFICATE

This is to certify that this independent project entitled "A TUTORIAL ON BASICS OF PSYCHOACOUSTICS" is the bonafide work in the part fulfillment for the degree of Master of Science (Speech and Hearing) of the student with register No. M0115.

Mysore
May, 2002

Dr. M. JAYARAM
Director
All India Institute of Speech &Hearing
Mysore - 570 006

# CERTIFICATE

This is to certify that this independent project entitled "A TUTORIAL ON BASICS OF PSYCHOACOUSTICS" has been prepared under my supervision and guidance.

Mysore
May, 2002

Mr.ANIMESH BARMAN
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# DECLARATION

This is to certify that this independent project entitled "A TUTORIAL ON BASICS OF PSYCHOACOUSTICS" is the result of my own study under the guidance of Mr. Animesh Barman, Lecturer in Audiology, All India Institute of Speech and Hearing, Mysore and has not been submitted earlier at any university for any other Diploma or Degree.

Mysore
May, 2002

Reg.No.MO115.

## ACKNOWLEDGEMENT

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*My Muthu.............you are the best sister one could ever wish for. 7 LOVE YOU'.*

*Ammachi, your prayers have always been my guiding star.*

*Nishi, Raji, Sushu.............though we are miles apart, my love for you will never depart.*

*My B. Sc classmates, I miss the fun we had together.............still waiting for our 'SABRNJALIPFISH'*

*I thank all the IB. Sc's for their kind co-operation.*

*I thank my classmates for being their for me during those last days when every thing went hay wire.*

***I thank Lord Almighty***

***for blessing me with a loving family***

# TABLE OF CONTENTS

# INTRODUCTION

The word "tutorial" as defined by scientific and English dictionaries refers to an "instruction book" or intensive instruction in some area. It aims at providing supplementary instruction in order to present better opportunities to students and concerned professionals to actively participate in the learning process and receive feedback. The information is carefully selected and organised in a structured manner. It also evaluates the user's knowledge through different types of questions which gives him/her immediate feedback of the performance. It thus acts as an efficient guide for students and experts linked with the particular field.

An attempt has been made to construct a tutorial on basics of Psychoacousties.

Audiology is the science of the evaluation of hearing and psychoacou sties makes the scientific basis of audiology. Psychoacoustics mainly deals with the end product of the processing of auditory information, i.e the sensation and perception of sound. In its simplest defininition, psychoacoustics is the study of the psychological response to an acoustical stimulus.

Most tests in clinical audiology are based on the findings of psychoacou stical research and on changes in psychoacoustical phenomenon. Psychoacoustics link the physical parameter of sound with the sensation and perception that they evoke. Psychoacoustic findings have provided the foundation for behavioural testing in audiology and likewise promise for the future improvements and expansion in diagnostic audiology.

Without clear and complete understanding of normal hearing processes and how those processes change through out live, the evaluation of hearing disorders and their impact on communicative behaviour will be limited. The goal of this tutorial is to provide an overview of auditory behaviour and the limits of auditory ability.

From the above discussion it is clear that how important it is for students and professionals in this field to know about psychoacoustics. The present tutorial has been developed keeping this in mind. The tutorial mainly deals with the basics of psychoacoustics. It also discuss certain basic physical and acoustical concepts. The main topic has been divided into many more topics and subtopics in a simple and precise form. The main source of information has been collected from books, journals and other sources. The text in different chapters are gathered from different journals, books and other source. They were compiled and the questions prepared for each chapter were then presented to fifteen 1st B.Sc students inorder to check simplicity of language and to rule out ambiguity of the text and the questions. The suggestions also were sought to improve the text if required.

Each chapter will be followed by a set of questions of the following type:

- Fill in the blanks
- Multiple choice
- True or false
- Word grids
- Match the following
- Number games

- Acronyms
- Identification of figures and graphs.
- Completion of figures and graphs

The questions are neither too complex nor too simple. They provide the user with an opportunity to test the knowledge that he or she has gained through the tutorial. In order to cross check the results there are answers provided to all questions at the end of each section to give an immediate feedback of one's performance.

Thus, this particular independent project has been developed to serve the following purposes.

1. Give intensive information about psychoacoustics and its applications.
2. To test one's knowledge of the topic.
3. To serve as a guide for students and other concerned professionals.
4. To train and evaluate trainees during a training program.

# PHYSICAL QUANTITIES

Physical quantities may be thought of as being "basic" or "derived" and as either 'scalars' or Vectors'.

The basic physical quantities of concern here are time, length (distance) and mass. The derived quantities are the result of various combinations of the base quantities and other derived quantities and include phenomenon such as force, velocity, work etc. If a quantity can be described completely in terms of just its magnitude (size) then it is a scalar, eg., length. On the other hand, a quantity is a vector if it needs to be described by both magnitude and its direction, eg., displacement.

Following are the few physical measures.

**Force**

Force is an external effort in the form of push or pull, which (i) produces or tries to produce motion in a body at rest or (ii) stops or tries to stop a moving body or (iii) changes or tries to change the direction of motion of the body.

The following example illustrate the above definition:

i) When we push a ball lying on the ground, it starts rolling. The force exerted has thus produced motion in the ball. However when we push a heavy stone, it does not move. The effort made in this case has only tried to produce motion, but has not succeeded. In both the condition it can be said that the force is applied.

The nature of force and the basis of its measurement are embodied in three laws of Newton.

Common experience and Newton's first law of motion tell us that if an object is not moving (at rest) then it will tend to remain at rest and that if an object is moving in some direction at a given speed that it will tend to continue doing so. This phenomenon is inertia, which is the property of mass to continue in its state of rest or motion. A force (F) is needed to overcome a body's inertia. We may say that force is that which causes a mass to be accelerated (as the change in speed is acceleration), i.e to change its speed or direction. The amount of force is equal to the product of mass times acceleration (Newton's second law of motion).

$$\text{i.e } F = ma$$

Where, f is force, m is mass and a is acceleration. Because the force is the product of mass and acceleration the amount of force is measured in kg $m/sec^2$.

The unit of force is Newton (N), which is the force needed to cause a 1 kg mass to be accelerated by $1 m/sec^2$ i.e $1N = 1kg\ m/sec^2$.

**Work, Power & Energy**

In Physics, work is said to be done, only when a body moves actually through a certain distance in the direction of the force applied. In all other cases, where no external force is applied or the body fails to move actually in the direction of application of force, work done is said to be zero. Work done by a constant force is defined as the product of force (F) and the actual distance (S) moved by the body in the direction of force.

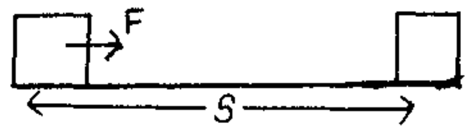



**Fig 1: Shows the force 'F applied through distance 'S'**

Work done = Force x Distance

W=FS

It should be clearly understood that F and S have to be along the same direction.

The work done by a force of 1N acting through a distance of 1 m would be 1 Joule i.e 1 Joule = 1 N/m. The Joule (J) is the MKS unit of work.

**Power**

Power of a body is defined as the rate at which the body can do the work, i.e.

Power = Rate of doing work.

= Work/time

Thus power of an agent measures how fast it can do the work. When a body takes lesser time to do a particular amount of work, its power is said to be greater and vice-versa.

The absolute unit of power is Watt, which is denoted by 'W'

$$P=w/t$$

$$1 \text{ watt} = 1\text{joule} / 1 \text{ sec}$$

Hence power of an agent is said to be one watt, if it can do one joule of work in one second.

**Energy**

Energy of a body is defined as the capacity or ability of the body to do the work. The unit of measurement of energy are also the same as the unit of work i.e joule.

Energy can assume one of two inter chargeable forms: potential or kinetic.

The kinetic energy of a body is the energy possessed by the body by virtue of its motion for eg:

i) A bullet fired from a gun can pierce through a target on account of kinetic energy of the bullet.

ii) Wind mills work on kinetic energy of air.

The potential energy of a body is defined as the energy possessed by the body by virtue of the position or configuration. For example, when we wind the spring of our watch, potential energy is stored in the spring on account of configuration of the turns of the spring. As the spring unwinds it works to move the hands of the watch. Thus the wound spring has the potential to do work.

It is due to the potential energy of the compressed spring in a loaded pistol that the bullet is released with a large velocity on firing the pistol.

## Simple Harmonic Motion

A sinusoid, also called a sine wave, describes a particular relationship between displacement and time, that is, a particular vibration.

Given figure is a diagram of a sinusoid.

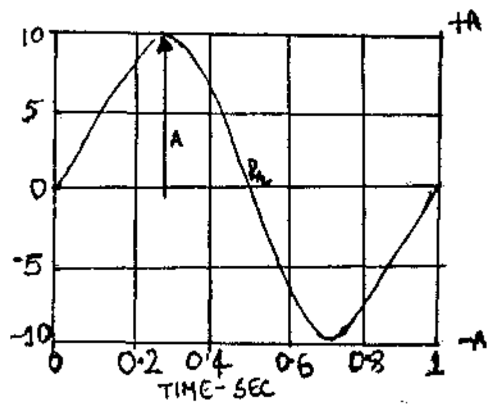



**Fig. 2: Showing a sine wave**

Displacement simply means the distance an object moves, and a sine wave describes the continuous, regular back-and-forth displacement of a vibrating object. Notice that from the starting position (displacement D=O) the motion in this example goes upward to a maximum positive distance (D= +A,+A = +10), then back to the starting position, then downward to a maximum negative distance (D=-A,-A=-10), and finally back to the starting position over the time period of 1 second ( shown in figure 2).

Four properties (parameters) characterise a sinusoid: frequency, starting phase, amplitude and period. Sinusoidal motion is often referred to as simple harmonic motion.

Simple harmonic motion is defined as a special type of periodic motion, in which a particle move to & fro repeatedly about a mean (i.e equilibrium) position under a restoring force, which is always directed towards the mean (i.e equilibrium) position and whose magnitude at any instant is directly proportional to the displacement of the particle from the mean (i.e equilibrium) position at that instant.

It can be described in terms of swinging pendulum as in figure 3.

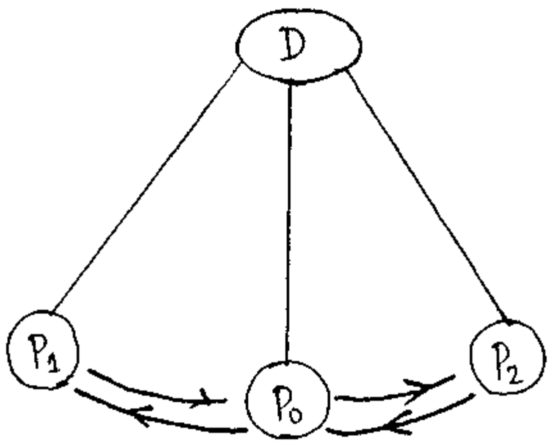



**Fig. 3 : Showing the simple harmonic motion**

Here 'D' is the amplitude of the sinusoidal motion, the rate of swing is the frequency, and the starting point of the pendulum relative to point $P_o$ is the starting phase of the simple harmonic motion.

**Frequency:**

The frequency of a sinusoid is the number of cycles it completes per second. The symbol Hz, standing for hertz (hertz is in cycle per second), is used to denote this number. If a sine wave completes 100 complete cycles in 1 second, then the sinusoid is said to have a frequency of 100 Hz.

For eg:

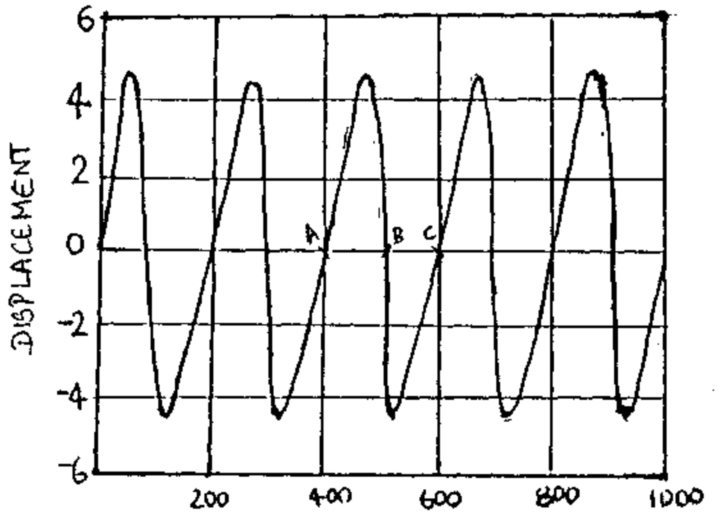



**Fig. 4: Showing a sine wave of 5 Hz tone**

A Sinusoid with a frequency of $5H_Z$ (Pr = 100/200 ms); the starting phase is 'O', the peal, amplitude is +5. One period is the time between points A and C and not between A and B or B and C. The sinusoid has completed 5 cycles in one seconds and so its frequency is 5 Hz as can be seen in figure 4.

The amount of time a sinusoid takes to complete one cycle is called its period. Thus the period of sinusoid in the above figure is 1/5 seconds.

Period is the reciprocal of frequency, or mathematically speaking, I/frequency. For example a period of a 1000 Hz tone is 1/1000, or one one-thousandth of a second.

**Starting Phase:**

The starting phase of a sinusoid corresponds to the point in the displacement cycle at which the object begins to vibrate. Starting is usually defined in terms of degree of angle. A Sinusoid is said to start at zero phase, or the start "inphase", if when the time is equal to zero (t = o) the displacement D is equal to zero i.e the phase at point A,B, C D and E is 0°, 90°, 180°, 270° and 360° respectively.

Example:

Sine wave in fig starts at zero phase.

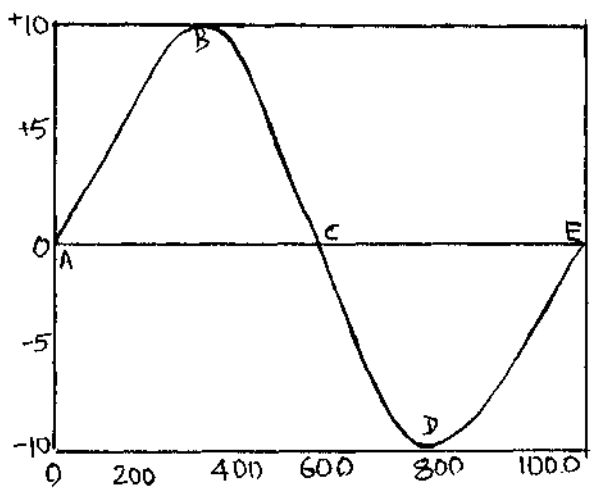



**Fig. 5. Phase of a sine wave at O°**

Sine wave in figure 6 start one-half of a period later than the zero-starting phase condition i.e 180° phase.

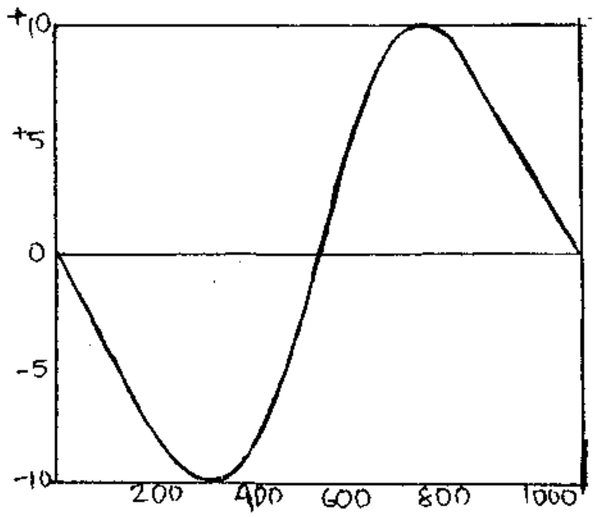



**Fig. 6 : Sine wave starting at 180° phase.**

In figure 7 starting phase is one-quarter of a period later i.e 90° phase.

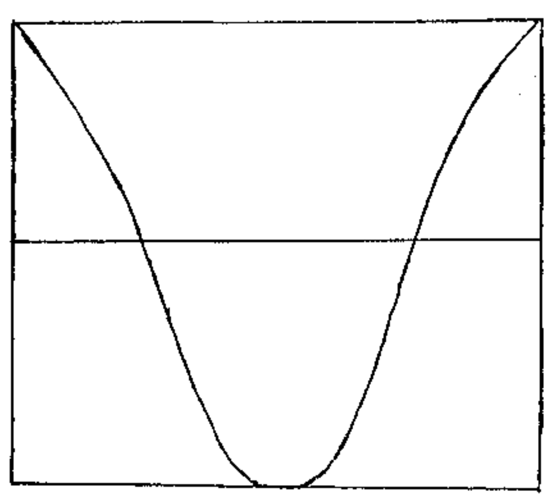


**Fig. 7 : Sine wave starting at 90° phase.**

The phase of an object that vibrates in SHM represents that portion of a cycle that has elapsed at any instant in time, relative to some arbitrary starting point. Phase is expressed in degrees, and the number of degrees contained within one completed cycle is 360. Thus, the phase angle at any point during a cycle may vary between 0 degree and 360 degrees. The concept of phase is derived from the close relationship between simple harmonic motion and projected circular motion.

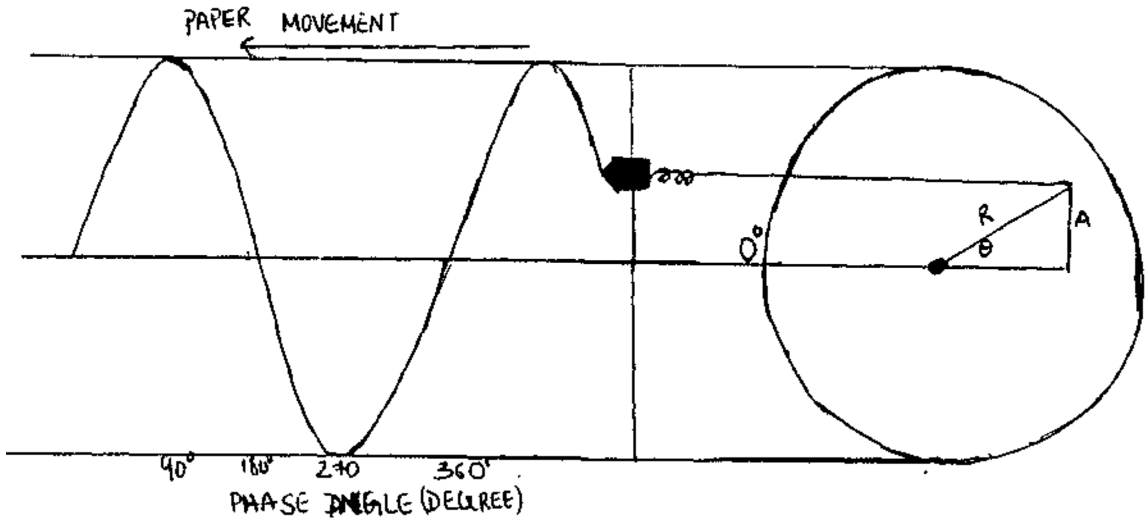



Fig 7a: Generation of a sine wave

The figure 7a shows a wheel that can be turned at a constant rate of speed. This is attached to a pen assembly that can move vertically up and down a shaft. The position of the pen at any instant in time is exactly equal to the height above or below the centre of the circle (distance 'A' in the fig). If the wheel rotates one complete cycle (360 degrees), the pen assembly will move through one cycle of vibration. Thus, 300 degrees Of rotation can be projected as one cycle of simple harmonic motion and the obtained pattern will be sine wave. Each point on the sine wave could then be specified as the number of degrees that elapsed in the cycle.

## Amplitude

Amplitude refers to the distance that a vibrator moves during vibration. The greater the distance from the rest position, the greater the amplitude.

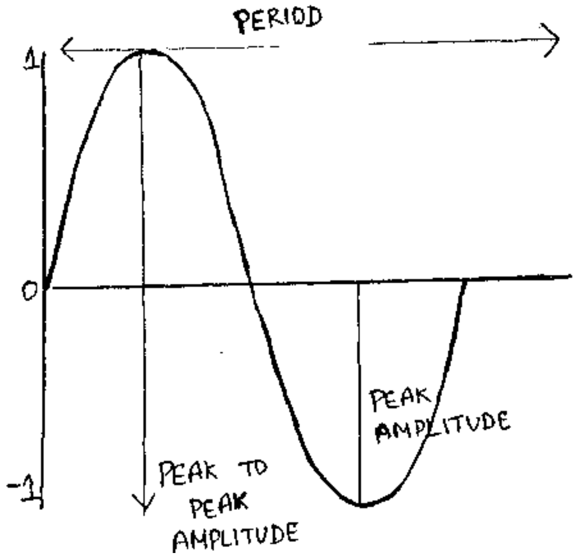



**Fig. 8 : Amplitude of a sine wave**

The amplitude of a sine wave is in a state of constant variation. One moment the amplitude is at the zero crossing line (rest position) and the next moment it is somewhere between the rest position and the maximum displacement points. For this reason, there are several ways to specify the amplitude of a sine wave like peak amplitude or peak to peak amplitude etc.

Peak amplitude refers to the maximum momentary displacement obtained by the vibrating source. It occurs once when the vibrator reaches its maximum displacement in one direction i.e either in positive or negative direction (Fig. 8).

Peak-to-peak amplitude, as the name implies, equals the distance between the maximum displacement points in both directions around the rest position. In the case of sine wave, the peak-to-peak amplitude is simply twice the peak amplitude (fig. 8). Instantaneous amplitude is the amplitude that the wave assumes at any moment of time.

**Vibrations**

The pendulum example describes one type of vibration referred to as simple harmonic motion.

The mass and spring example shown in figures may be used to describe, in a general way, all vibrations including the swinging pendulum.

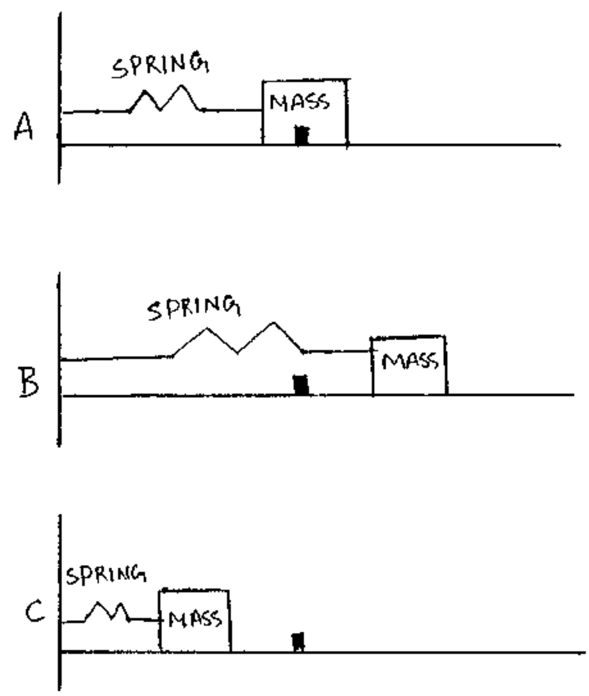



**Fig. 9: A, B and C showing how does vibration takes place in different conditions**

In fig. 9 the mass would require a force to move it, and so the mass represents the property of inertia. The spring has a restoring force and so represents the property of elasticity. If some force moves the mass out from its resting state (fig 9 A) to the position shown in fig 9 B, and then the force is removed, the spring would pull the mass back through the point of equilibrium to a position such as that shown in fig 9 C. The mass would continue to oscillate back and forth, that is, the mass would vibrate with some amplitude and frequency. If there was no resistance, such as friction, to the motion of the mass, it would continue to oscillate back and forth between the two extremes forever. Because no force is applied to the system after it is set in motion, the vibration is called free vibration.

The real world contain sources of resistance to the motion of the mass. Friction is one source of resistance that prevents the mass from continuing to vibrate forever. The mass would slowly but surely lose its amplitude of vibration.

Fig:

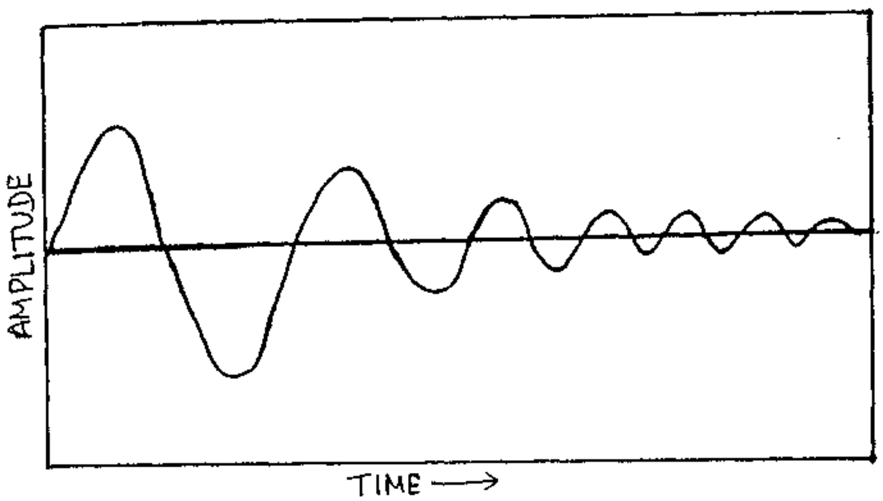



**Fig. 10: Reduction in amplitude over time**

Fig 10 describes how the amplitude of vibration would die out due to resistive forces. In this case the sinusoidal vibration is said to be damped. In real system the rate of damping is such that the ratio of each successive peak remains constant.

The second peak in the sinusoidal function might be one-half the first peak, and the third peak might be one-half the second, and so on. The ratio describes how quickly the vibration is damped.

## QUESTIONS

**Chose the correct one**

1. A physical quantity which described completely in terms of just its magnitude.

   (Scalar, Vector, Derived)

2. Force is measured in

   (kg, cm / $sec^2$, kg, m/ $sec^2$, kg/ $cm^2$)

3. Absolute unit of power

   (Joule, Watt, Newton)

4. Number of cycles per second

   (Frequency, amplitude, vibration)

5. Maximum momentary displacement obtained by the vibrating source.

   (Instantaneous amplitude, Peak-to-peak amplitude, Peak amplitude)

**Fill in the blanks**

6. Physical quantities can be -..........— and -...........-......
7. The unit of force is —.......—
8. Rate at which body can do the work is————————.
9. Two interchangeable forms of energy are——————and —......--.
10. Four parameters of sinusoid are————————, -————————, — and ————————————————.

11. We may say that——————————causes mass to be accelerated.

**True or False**

12. Work is the product of force and velocity.
13. Energy of a body is defined as the capacity or ability of the body to do work
14. Sinusoidal motion is often referred to as simple harmonic motion.
15. In simple harmonic motion particles move up and down the mean.
16. The greater the distance from the rest position, the greater the amplitude
17. Both scalar and vector quantities can be used synonymously.
18. No other conditions other than a motion due to external effort can be considered as force.
19. Inertia phenomenon is said to occur when a property of a mass will continue to stay in its state of rest or uniform motion.
20. The amount of force, F= mass x acceleration.

**Let us have fun with number (Match the correct one)**

21.

| | | | |
|---|---|---|---|
| a) | 1 Newton | : | 50 Hz (a) |
| b) | 1 Watt | : | 1 kg m/sec$^2$ (b) |
| c) | 50 cycles / sec | : | 1 N/m (c) |
| d) | 1 joule | : | joule / sec (d) |

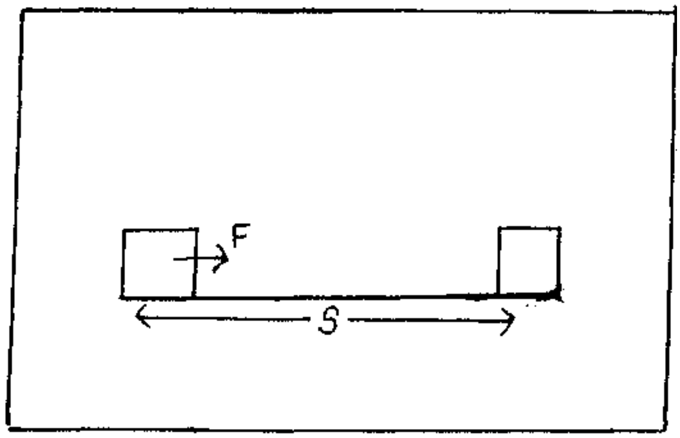
**Match the Following**

22.

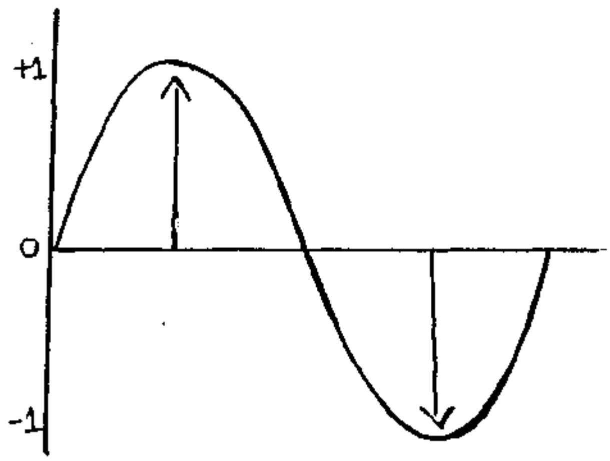
| | | |
|---|---|---|
| a) | Force | Hz (a) |
| b) | Power | Joule (b) |
| c) | Frequency | Newton (c) |
| d) | Work | Watt (d) |

**23. Identify the figures**

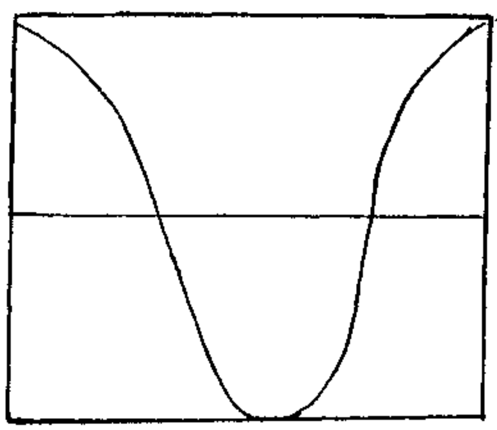
(a)



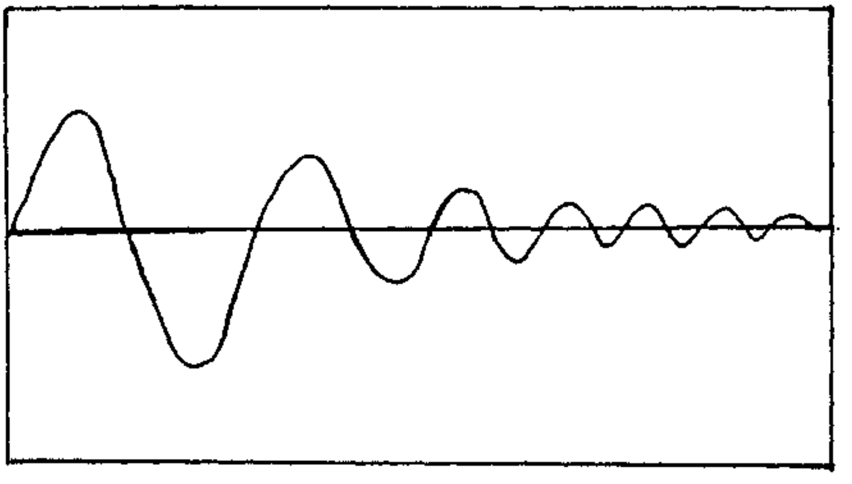

(b)



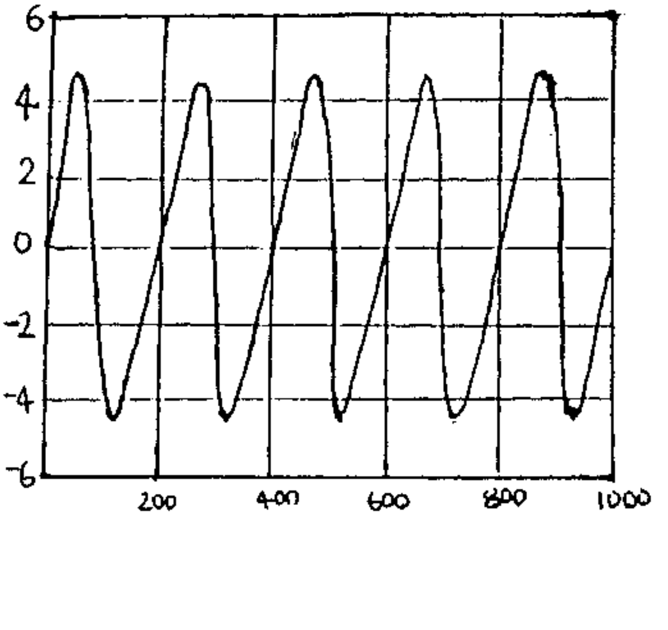

(c)



(d)



(e)



## ANSWERS

1. Scalar
2. Kg m/sec$^2$
3. watt
4. Frequency
5. Peak amplitude
6. Basic or derived / scalars vectors
7. Newton
8. Power
9. Kinetic and potential energy
10. Frequency amplitude, staring phase and period
11. Force
12. False
13. True
14. True
15. False
16. True
17. False
18. False
19. True
20. True
21. a, b

    b, d

    c, a

    d, c

22. a, c

b, d

c, a

d, b

23. A) Work

B) Amplitude

C) Phase

D) Damping of vibration

E) Sinusoid with a frequency 5 Hz.

# ACOUSTICS

Sound is one of the most familiar forms of energy. It is a physical phenomenon to which one is exposed very often. It is a some what illusive phenomenon to understand in common sense terms. Because it commands such importance in Physics, the sence of sound classically has stood apart as a major discipline known as 'acoustics'. Sound is said to be the physical stimulus which evokes the sensation of hearing. The inseparability of sound and hearing thus dictates the need for a basic understanding of acoustics.

## Sound Propagation

Because air is the medium for sound transmission in most everyday situations, we will consider the transmission of sound through air. Air consist of molecules in constant random motion. When an object vibrates in air, the molecules tend to move in the direction the object moves rather than with their noval random motion. The air molecules next to the object moves first and then pass this movement on the adjacent molecules. The molecules themselves do not move all the way from the object to the receiver they only pass along a wave motion.

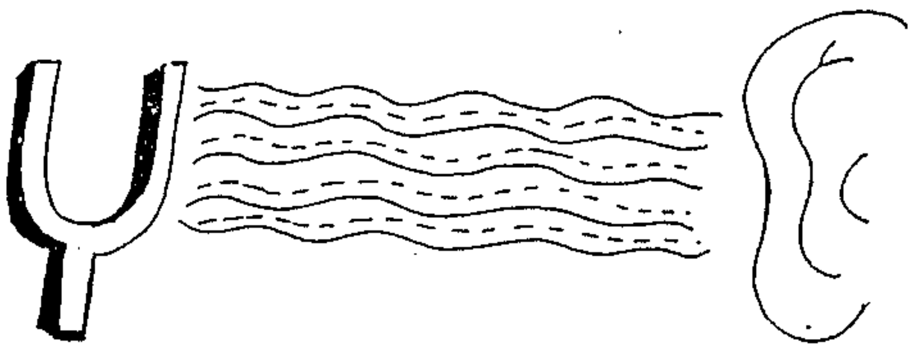


**Fig. 11: Sound waves produced by a tuning fork**

In this train like progression, the motion of the air molecules is propagated (transferred) through the air toward the ear. When the air molecules next to the ear are moved by this motion, the eardrum is vibrated, and this eventually results in our experiencing audible sound.

A propagated disturbance in air which is heard as sound is not unlike the undulatory motion on the surface of water. The waves on the surface of water are the result of a combination of longitudinal and transverse propagation.

In transverse propagation, the motion of the particles of the medium is basically perpendicular to the direction of travel of the disturbance. The compressional waves of sound, or simply sound waves in air are strictly longitudinal - the particle motion along the axis of the movement of the disturbance. Thus, sound is a disturbance in the medium involving an undulatory motion of the particles such that they vibrate back and forth along the axis of propagation.

The molecules next to the vibratory object are compressed together as the object moves outward away from its resting state, creating an area with a grate density (mass per unit of volume) of air molecules. Certain laws of chemistry/physics, the gas laws, state that, as the density of air molecules increases, the pressure increases thus, as the object moves outward, the density of air molecules next to the vibratory object increases and creates high pressure in this area. Thus an area of condensation had been generated. As the vibrating object moves in the opposite direction (back towards its resting position) the air molecules obey another property i.e, gases, like air, will evenly fill the space they occupy. Thus the air molecules fill the space vacated by the vibrating object moving in the opposite direction. As the vibrating object moves back past its resting state, an even larger vacated area is generated for the air molecules to fill. Now the density of air molecules has decreased, thus pressure is lower. An area of rarefaction has been generated.

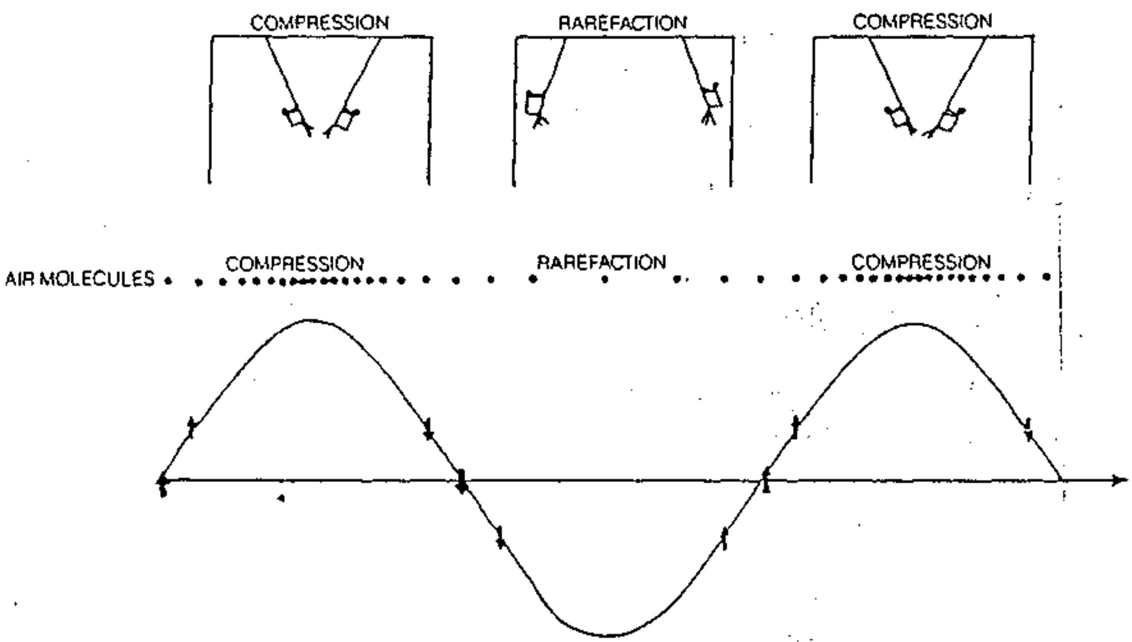



**Fig. 12: An area of condensation and rarefaction of a vibrating body**

The mere presence of molecules in air creates pressure, or static air pressure, that is proportional to the density of the molecules. As the object vibrates, the static pressure increases at any one location, then decreases, and then increases again, and so on, generating a changing pressure. This changing air pressure moves away from the vibratory object. In other words, areas of condensation are alternating overspace with areas of rarefaction. That is, molecules next to these initially moved are moved back and forth through stages of condensation and rarefaction. This pattern continues across space as the wavelike motion is propagated away from the vibratory source.

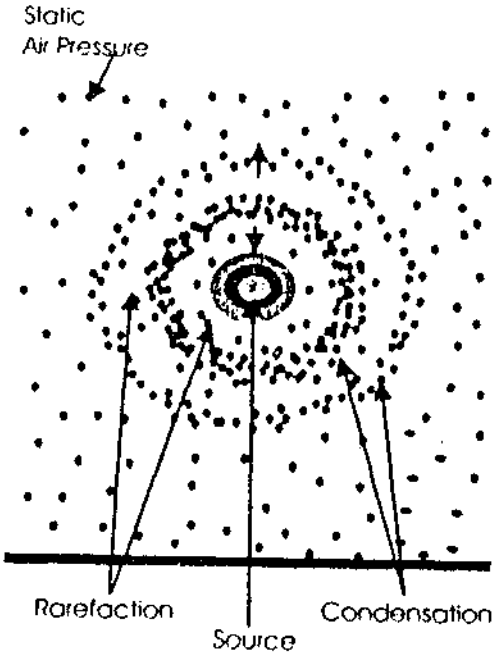



**Fig. 13: Showing the state of condensation and rarefaction**

Viewing waves on water provides an approximation to what happens in air when an object vibrates. If we push water (for instance, by dropping a stone in a pool), waves are formed. The water waves are similar to the air waves associated with sound propagation. The crests of the wave are the condensations, and the valleys are the rarefactions.

The distance between each successive condensation (or rarefaction) is the wavelength ($\lambda$) of sound. Actually, the distance between any successive identical points in the propagated wave is the wavelength. Wavelength is usually expressed in meters. The more frequently an object oscillates (higher frequency), the closer together the rarefaction and condensation become, and shorter the wave length. The speed with which the wave motion is propagated through the medium (i.e., the speed of sound) also affects the wave length. The equation relating wavelength ($\lambda$) in meters to speed of sound (c)in meter per second) and to frequency (f in Hz) is

$$\lambda = c/f$$

Wavelength is directly proportional to the speed of sound and is inversely proportional to the frequency of vibration.

The speed of sound in air is approximately 345 meter per second; although it can vary as a function of the temperature, density, and humidity of air. The speed of sound is higher in a hot, humid area than in a cold and dry place.

**Interference**

One of the more ominous properties of sound propagation is that, the farther away we are from a sound source (the vibratory object), the softer sound becomes. The sound source is considered a point, and sound radiates in a spherical fashion from the source. As they move away from the centre of the sphere, the surface area of the sphere becomes larger. Because the sound source is assumed to be producing a constant power and the area is increasing, sound intensity (I) must be decreasing as one moves away from the sound source. The area of the surface area of a sphere is $4\Pi T^2$, where 'r' is the radius of the sphere or, the distance from the sound source to the point of measurement.

From this relationship and the definition of sound intensity (I), we obtain following relationship.

$$I \alpha p / (4\pi r^2)$$

Where a means proportional to, p is to power of the sound and its source, and r is the distance from the source to the point of measurement. This in turn means that sound intensity (I) is inversely proportional to distance from the sound source squared ($r^2$).

These relationships hold good only in situations in which the sound encounters no obstacles. Sound moves most often encounters obstacles. In general, objects impede the propagation of sound wave. This impedance to sound transmission occurs whenever there is a change in the medium through which sound must travel.

The reflected sound wave may encounter the original sound wave as it propagates away from the barrier, as in fig. 14.

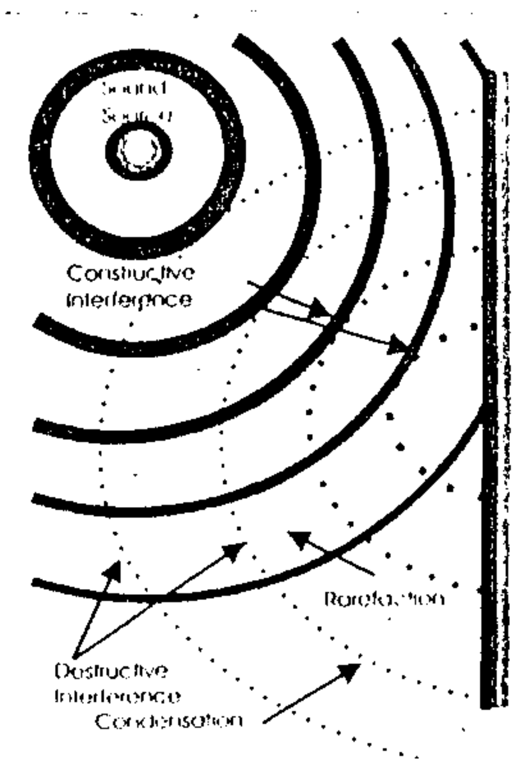
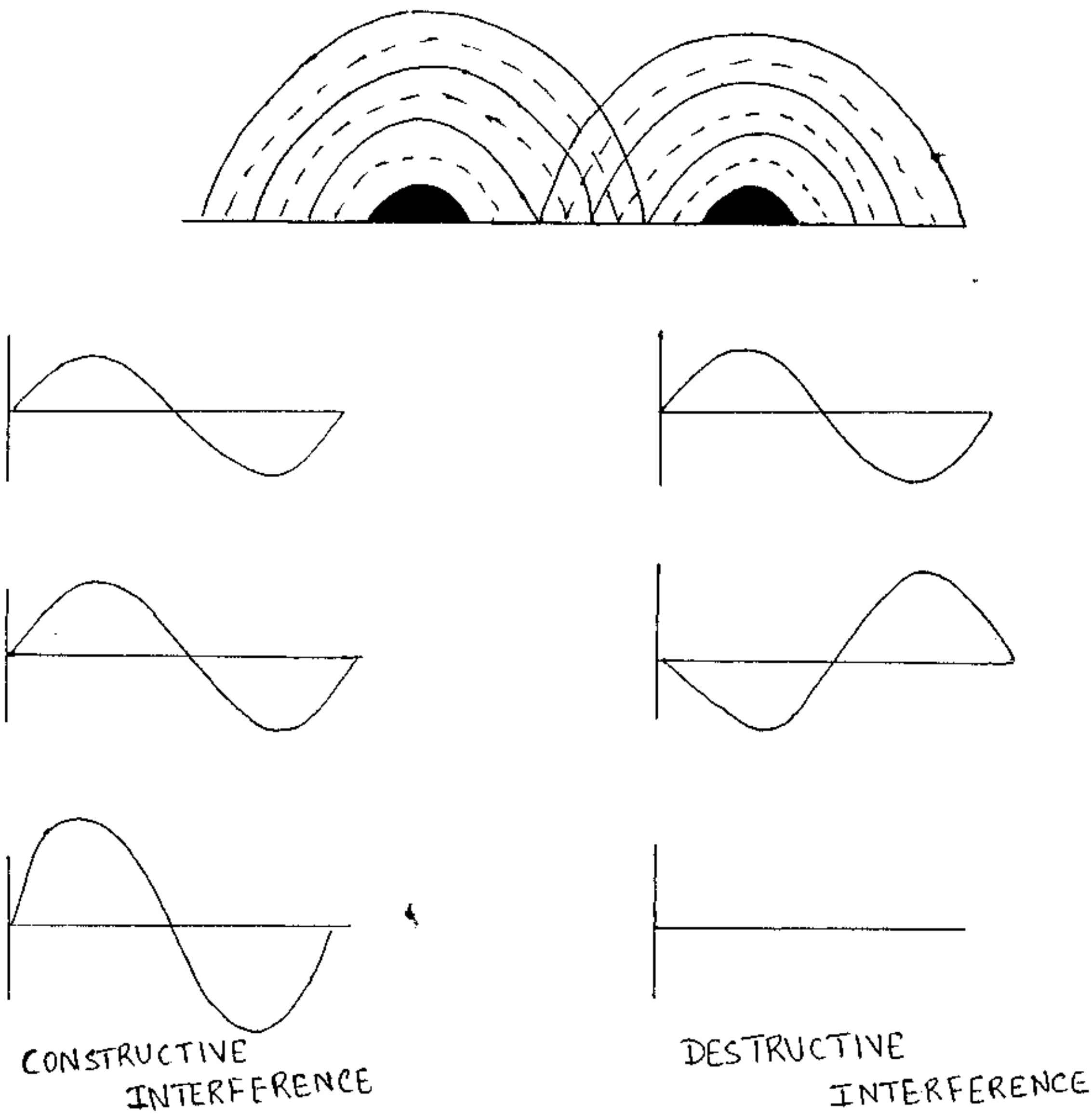



**Fig. 14: The interaction between original & reflected sound waves**

In this case, two types of interactions can occur between the original and reflected sound waves. Two points of condensation can occur together, resulting in area of greater condensation called constructive interference. Points of condensation and rarefaction can overlap, resulting in a reduction in pressure, called destructive interference. The net result will be the summing of two waveforms (reinforcement, as in constructive interference) or the subtraction of two waveforms (cancellation as in destructive interference) as can be seen in fig 15




**Pig. 15: The constructive and destructive interference of original and reflected sound waves**

## Sound Field

Any environment that contains sound is called a sound field. A sound field without any reflections is called a free field.

A truly free-field environment is almost impossible to obtain. An anechoic room is an environment in which every attempt is made to reduce reflections. This is generally accomplished by using materials and shapes that absorb rather than reflect sound. Some times a room with reflection is desired. Such environment are called reverberation (or echoic) rooms. Here the reflecting surfaces are arranged so that constructure and destructure interference leads the constant sound intensity throughout the sound field. More of this will be dealt in the forthcoming chapter - 'Reverberation'.

## QUESTIONS

### Choose the correct one and fill in the blanks

1) Air consist of molecules in——————————motion.

   (Circular, random, longitudinal)

2) In transverse propagation, the motion of the particle of the medium is ——————————to the direction of the travel of the disturbance.

   (Perpendicular, horizontal, vertical)

3) As density of air molecules increases, the pressure. . . . . -. . . . . . . . . . . .

   (Decreases, increases, stays constant)

4) As the object moves outward, the density of air molecules next to the vibratory object increases and creates high pressure in this area called ——————————

   (Rarefaction, condensation, refraction)

5) The distance between any successive identical points in the propagated wave is——————.

   (Amplitude, frequency, wavelength)

### Fill in the Blanks

6) Sound evokes sensation of———————

7) A propagated disturbance in air is similar to the——————————motion on the surface of water.

8) The waves on the surface of water are the results of a combination of——————————and——————————propagation.

9) When density of air molecules decreases, the pressure is lowered and an area of——————————is generated.

10) The crests of the wave are——————————and the valleys

**True or False**

11) Gas like air will evenly fill the space they occupy.

12) Pressure is not proportional to the density of the molecules.

13) Wave length is directly proportional to the speed of the sound and is inversely proportional to the frequency of vibration.

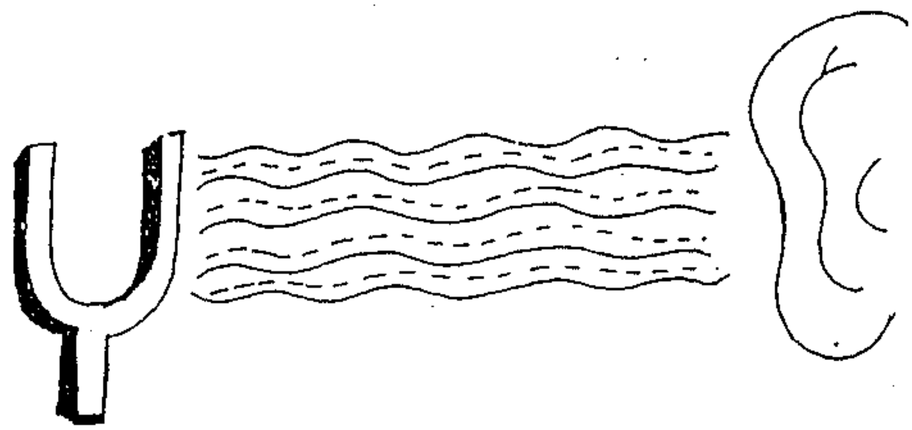
14) Sound intensity is directly proportional to distance from **the** sound source squared.

15) Word grids

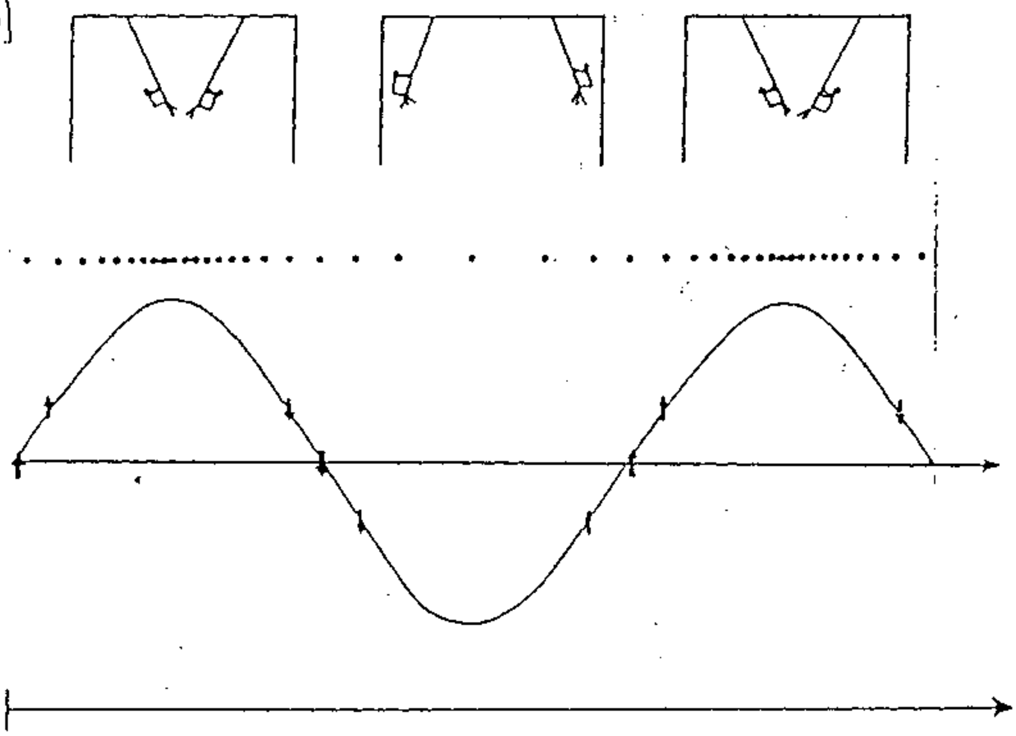
| | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| R | E | F | R | A | C | T | I | 0 | N | u | I | J | R |
| E | A | R | Y | C | 0 | B | K | G | F | Z | T | H | A |
| F | H | E | E | 0 | N | W | A | L | I | E | D | X | K |
| L | L | E | N | U | D | G | N | N | A | V | L | U | H |
| E | 0 | F | P | S | E | V | R | C | G | P | H | M | I |
| C | C | I | Z | T | N | B | X | u | N | I | E | Q | A |
| T | S | E | A | I | S | 0 | U | N | D | L | A | X | M |
| I | H | L | Q | C | A | M | P | L | I | T | U | D | E |
| 0 | I | D | M | S | T | Y | R | A | N | D | 0 | M | A |
| N | L | S | D | 0 | I | P | 0 | T | A | N | u | U | T |
| R | Y | E | T | I | 0 | W | P | P | F | P | D | I | H |
| H | I | V | B | Y | N | V | A | 0 | J | N | Y | W | U |
| D | S | I | P | M | R | I | G | Q | X | G | L | E | I |
| I | N | J | I | Y | K | N | A | L | T | S | Z | R | G |
| T | R | W | S | 0 | B | U | T | 0 | N | E | J | H | K |
| S | U | M | D | A | R | A | E | N | O | M | A | N | F |

16. Identify the figures

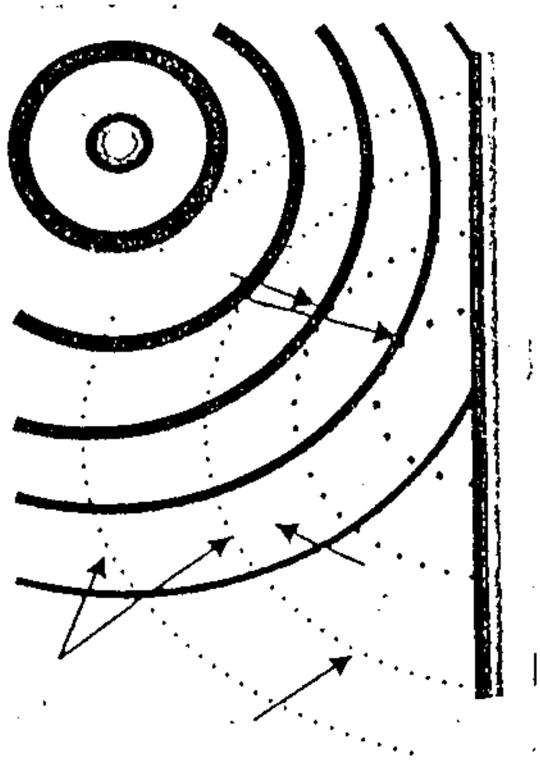
[A]



[B]



[C]



## ANSWERS

1. Random
2. Perpendicular
3. Increases
4. Condensation
5. Wavelength
6. Hearing
7. Undulatory motion
8. Longitudinal and transverse
9. Rarefaction
10. Condensation and rarefaction
11. True
12. False
13. True
14. False

15.

| | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| R | E | F | R | A | C | T | I | O | N | U | I | J | R |
| E | A | R | Y | C | 0 | B | K | G | F | Z | T | H | A |
| F | H | E | E | O | N | W | A | L | I | E | D | X | K |
| L | L | E | N | U | D | G | N | N | A | V | L | U | H |
| E | 0 | F | P | S | E | V | R | C | G | P | H | M | I |
| C | C | I | Z | T | N | B | X | u | N | I | E | Q | A |
| T | S | E | A | I | S | O | U | N | D | L | A | X | M |
| I | H | L | Q | C | A | M | P | L | I | T | U | D | E |
| O | I | D | M | S | T | Y | R | A | N | D | 0 | M | A |
| N | L | S | D | 0 | I | P | O | T | A | N | u | U | T |
| R | Y | E | T | I | 0 | W | P | P | F | P | D | I | H |
| H | I | V | B | Y | N | V | A | 0 | J | N | Y | W | U |
| D | S | I | P | M | R | I | G | Q | X | G | L | E | I |
| I | N | J | I | Y | K | N | A | L | T | S | Z | R | G |
| T | R | W | S | O | B | U | T | O | N | E | J | H | K |
| S | U | M | D | A | R | A | E | N | 0 | M | A | N | F |

16. A) Sound wave created by a tuning fork

B) Compression and rarefaction

C) Constructive and destructive interference.

# THE MEASUREMENT OF SOUND

Dependent upon subjective sensations, careful quantification of the stimulus eliciting the sensation is of paramount importance. The subjective response is often specified in terms of the physical stimulus which is sufficient to elicit.

It is the purpose of this chapter to describe the relevant measures of sound for hearing science and the terms in which these measures are specified. We consider these measures in two sections.

In section *'A'* we deal with

- Root mean square
- Spectrum
- Distortion

In section 'B' we deal with

- Decibel concept
- Octave notation

**Root** mean **square - magnitude**

In previous chapter, the amplitudes of vibratory motion or the pressure vibrations constituting sound were defined in terms of the instantaneous peak values. In other words, amplitude was merely defined as the difference between the extreme values (above or below zero) or the zero axis.

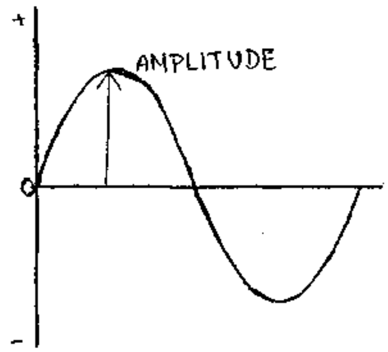


Fig. 16: Amplitude of a sine wave

It is often preferable to obtain a measure which better reflects the overall power in the sound rather than just peak magnitude, the sound are the results of fluctuating pressure changes and the rate at which work is done by the sound varies in time.

We should note however, that the effective level of a sine wave is not simply a straight average of all instantaneous amplitudes. If we do this, the obtained level would, then be zero.

Inorder to find rms, values of all positive and negative displacements are squared, so that all resulting values are positive numbers. The mean of all these values are obtained, and the rms value is obtained by taking the square root of this mean. The rms amplitude of a sinusoidal signal is numerically equal to 0.707 times the peak amplitude, or 0.354 times the peak-to-peak amplitude. The figure IT-illustrates the relationships among peak, peak to peak and rms amplitude.

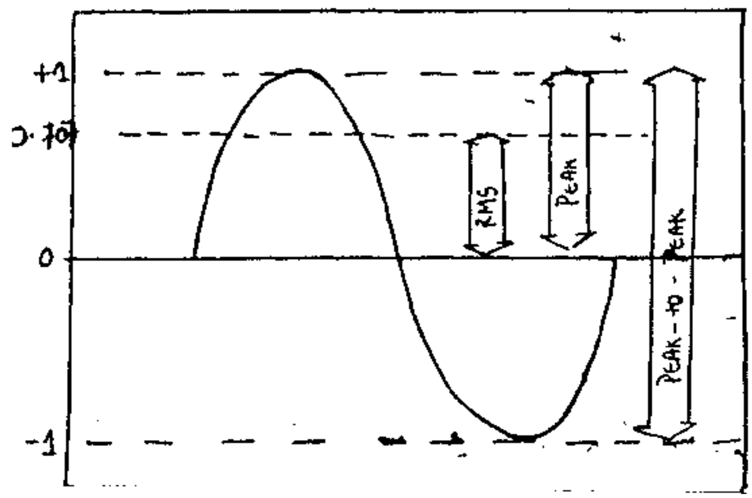



**Fig. 17: Peak, peak to peak and RMS amplitude Spectrum**

The spectrum of any sound is a plot which shows the amplitude of each frequency component in the sound. Spectrum should not be confused with waveform. The waveform of a sound is a plot that shows the amplitude of the sound at each instant in time. Consider for a moment the spectrum and wave form of a simple sound - a pure tone. Figure 18 'A' shows the waveform of a 1000 Hz tone, figure 18 13' shows the spectrum of the same sound.

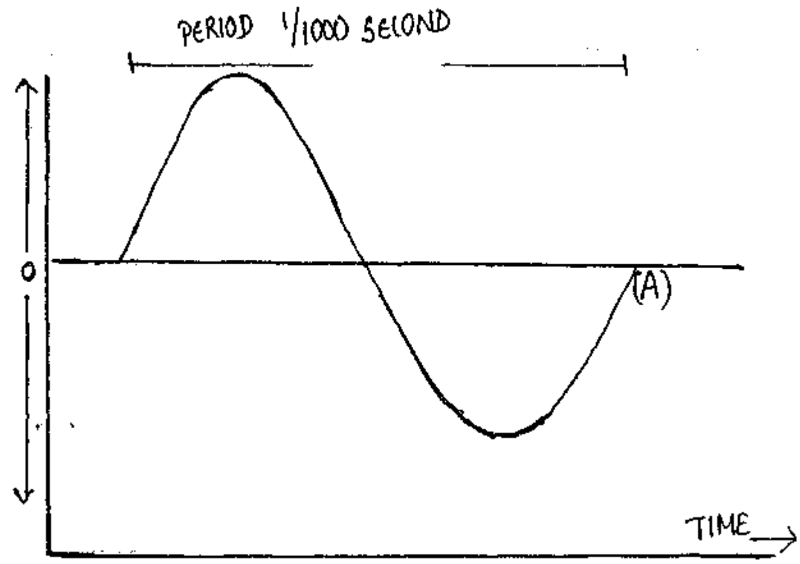



**Fig. 18 A** : **Wave form of a 1000 Hz tone**


**Fig. 18 B: Spectrum of a 1000 Hz tone**

The wave form seen in fig. 18 'A' is already familiar. The vibrating source that produces the sound yields a wave over time with a period of 1/1000 second. The spectrum of the tone is simply a straight line, or line spectrum, at 1000Hz. In other words, the entire energy of the sound is concentrated at one frequency only, 1000 Hz. The height of the line corresponds to the amplitude of the tone so that the higher the line, the grater the amplitude.

Complex tone are produced when two or more pure tones are sounded together. The resulting wave form is not a sine wave, but it is a pattern that repeats itself over and over again in time. Fig. 19(A&B) shows the wave from of the two pure tones, where one is twice the frequency of the other. Also shown in the figure is the combined wave form fig. 19C which results when both tones are sounded together. The combined wave form is different from both of the component tones, but it is periodic in that it repeats itself regularly.

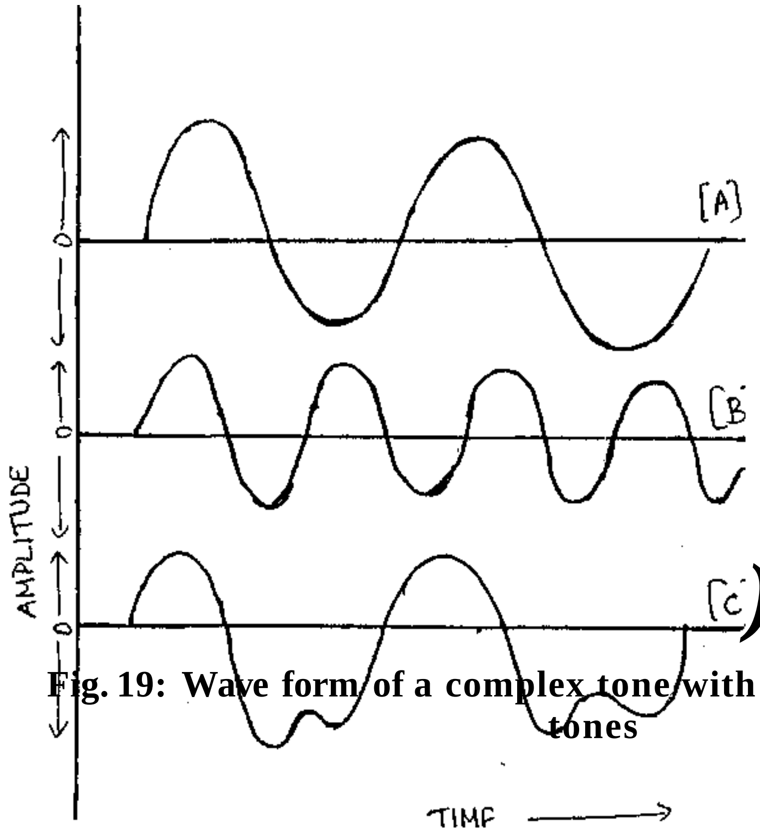



**Fig. 19: Wave form of a complex tone with their two fundamental tones**

Next een above.

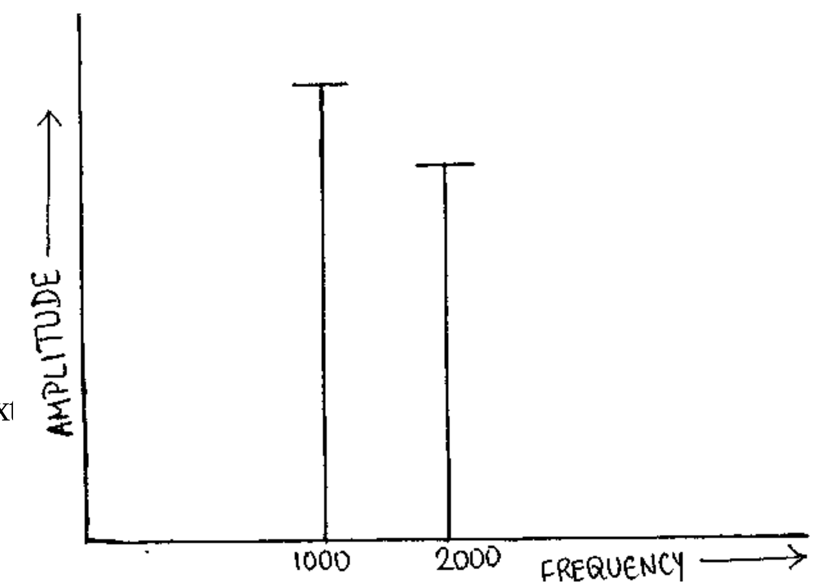



**Fig. 20: Spectrum of the above given complex tone**

Suppose that the lower frequency tone was 1000 Hz and the higher frequency tone was 2000 Hz. The spectrum is simply two lines, one at each end of the component frequencies.

Aperiodic vibration is usually referred to as noise. Many of the consonant sounds of speech that we produce and hear are, infact noise. They are produced by the individual constricting his vocal tract in one way or another, which then creates a turbulent flow of air. If we were to plot the wave form of such a sound, it would be irregular with no repeatable patterns.

**Distortion**

When a sound, vibration, or electrical signal is passed through any physical system the spectrum of the signal will undergo some degree of change. That is, the signal at the output of the system will not be an entirely faithful representation of the signal at the input; rather it will be distorted in some manner. This will intern influence the spectrum. There are three basic forms of distortion which may be introduced by the system. One type is frequency distortion, where in out put is characterized by some lose of frequency response.

A common example of a frequency distorting system is the telephone. The telephone passes only frequencies from about 300-3000 Hz (and thus act as a band-pass filters). Although this range is sufficient for speech communication, the callers voice is definitely altered in quality, as heard by the person on other end. Phase distortion may also occur. The output signal may be shifted in phase relative to the input signal. Frequency distortion is generally accompanied by phase distortion and vice versa. Finally, there is the possibility of amplitude distortion, which is perhaps the form of distortion which is most commonly associated with the term distortion. In a simple spring mass system, this would occur when the limits of elasticity of the spring axe exceeded and the restoring force is not longer proportional to the displacement of spring.

There is actually more than one form of amplitude distortion. It is demonstrated by feeding a sinusoidal signal into the input of the system in question (i.e a hearing aid) and observing the resulting output (i.e the sound produced by the earphone of the hearing aid). When there is harmonic distortion, the wave form of the signal recorded at the output will be altered, as in the figure21and is no longer purely sinusoidal.

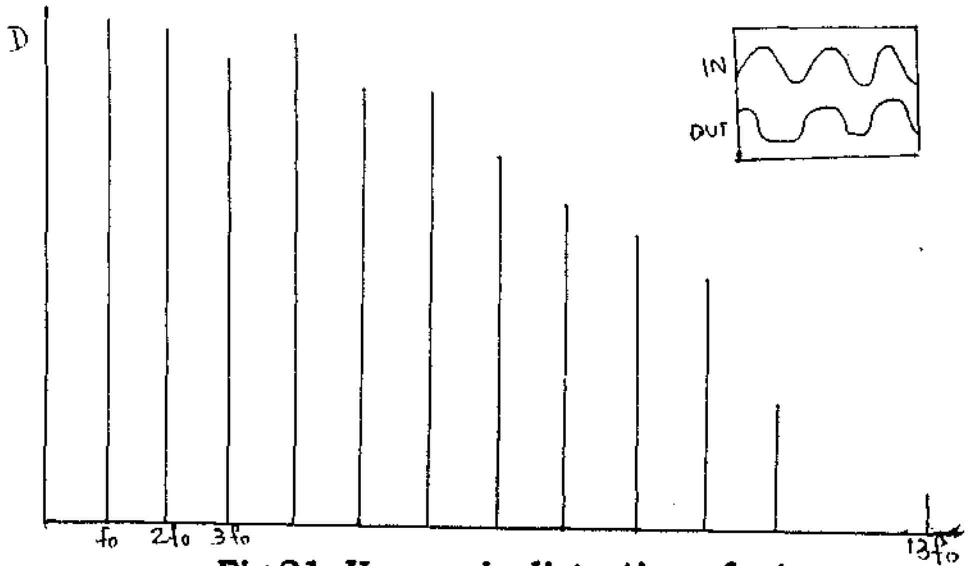



**Fig.21: Harmonic distortion of a tone**

There will be other frequency components in the spectrum - distortion products - occurring at frequencies which are integral multiplies of the fundamental frequency. It is for this reasons that these distortion products are often referred to as 'harmonics'. There will thus be second, third, fourth, and higher harmonics in the spectrum of the output signal, the first harmonies being the fundamental itself.

## QUESTIONS

**Section A**

**Choose the correct answer**

1. The plot which shows the amplitude of each frequency component in the sound.

   (Waveform, spectrum, harmonics)

2. Aperiodic vibration is usually referred to as

   (Puretone, noise, amplitude modulated tone)

3. When a signal is passed through any physical system the spectrum of the signal undergo some changes called.

   (Distortion, spectrum, modulation)

**Fill in the Blanks**

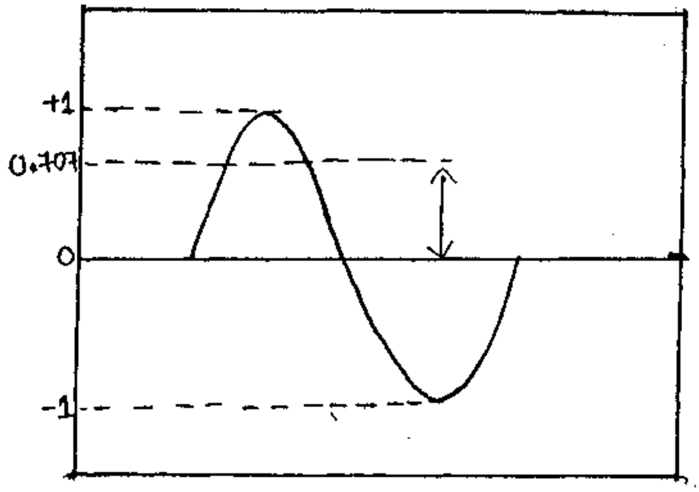
4. Amplitude is the difference between__________and__________.
5. In a sine wave, RMS value is equal to__________
6. Three types of distortions are__________-,__________-and__________.
7. The first harmonic is known as__________.
8. The distortion product of a system is__________

**True or False**

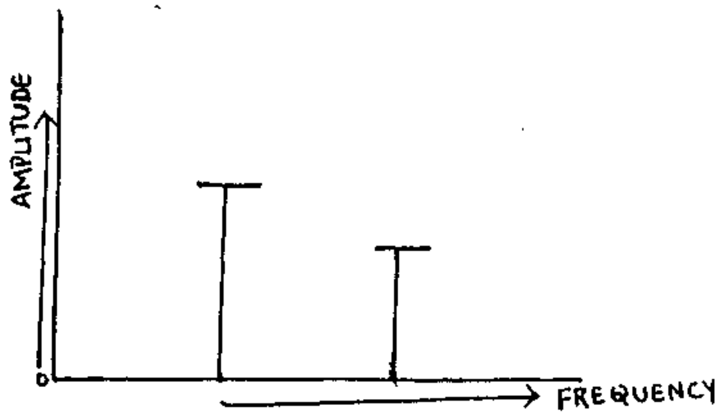
9. If the instantaneous amplitude of a sine wave are averaged the effective level obtained will be 0.
10. The square root of the mean of all the squared positive and negative displacement gives RMS.
11. Spectrum and wave form can be used synonimously.

12. State what does the figure indicate

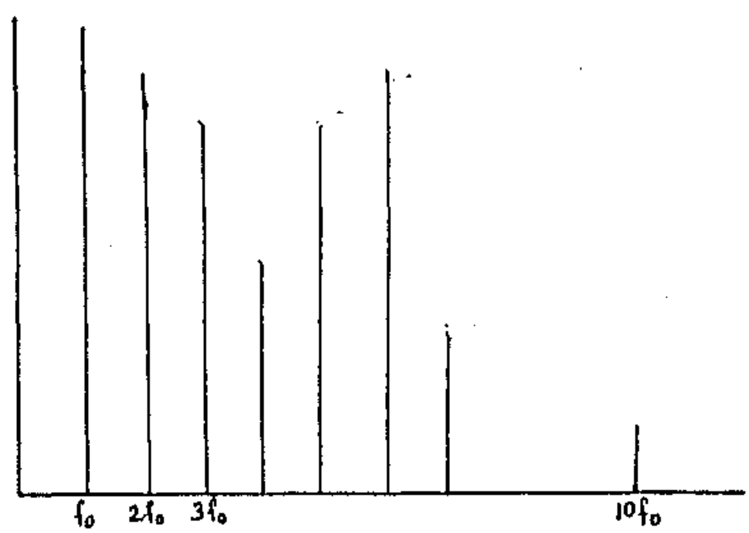
(a)




(b)




(c)



# ANSWERS

**Section A**

1. Spectrum
2. Noise
3. Distortion
4. Extreme values and zero axis
5. Peak value x 0.707
6. Frequency, phase and amplitude
7. Fundamental frequency
8. Integral multiple of fundamental frequency
9. True
10. True
11. False
12. a) RMS
    b) Spectrum
    c) Distortion

## Decibel (dB)

Intensity is the physical measure of what we perceive as the loudness of a sound. The greater the intensity of the sound, the louder it normally appears to be.

When the particles are in a state of compression, there is an increase in air pressure, when they are in a state of rarefaction, there is a decrease in pressure. Greater the pressure change, the grater the intensity of a sound. The intensity of a pure tone is measured by the amplitude of the sine wave. Greater amplitude represents greater particle displacement and therefore, greater sound intensity.

Intensity is measured in terms of two parameters

a) Power

b) Pressure.

Sound intensity depends on the force applied to the source. The sound intensity can be expressed in terms of power / unit area and pressure /unit area.

Minimum audible power to create hearing sensation for normals is $1O^{16}$ $w/cm^2$, where as maximum audible power is $10^2$ $w/cm^2$. When expressed in terms of pressure, this ranges from 0.0002 dynes /$cm^2$ to 2000 dynes/$cm^2$. The least intensity corresponds the minimum audible threshold of ear and highest represents the point at which sound produces sensation of pain. If expressed in ratio the dynamic range would be $10^{14}$ :1 (1 being the intensity of faintest sound detectable) and in terms of pressure it is $10^7$:l.

| | Pressure | Power |
|---|---|---|
| 1. Dynamic range | 0.0002 dyne//cm$^2$ | $10^{-16}$ W/cm$^2$ to |
| | 2000 dyne /cm$^2$ | $10^{-2}$ W/cm$^{-2}$ |
| 2. Ratio | $10^7$:l | $10^{14}$:l |

When the intensity is expressed in absolute units (in dynes/cm$^2$ or W/cm$^2$), it is extremely awkward to represent such a wide range of numbers on a graph scale having equal distant lines representing equal increments in sound magnitude. Because of the enormous range between two absolute values of intensity a relative unit has been devised. The difficulty in dealing with the absolute units of power or pressure, is overcome by expressing the intensity in terms of ratio between a given sound power or pressure and a standard reference power or pressure. The use of logarithmic scale to the base 10 for expressing this ratio resulted in further simplification, i.e using ratios and logarithm, wide range of numbers can be converted in a smaller scale. Hence the 'Bel' was introduced.

"Bel' has become a standard shorthand notation for expressing intensity. The 'Bel' named after Alexander Graham Bell, is defined as the log to the base 10 of the ratio of 2 intensities.

$$\text{Bel} = \text{Log10}\frac{Io}{IR} \quad \text{i.e} \quad \frac{\text{absolute intensity}}{\text{reference intensity}}$$

Thus the minimum audible intensity would be

$$\text{Log10}\frac{10^{-16}}{10^{-16}} = 0 \text{ Bel}$$

And the threshold of pain would be

$$\text{Log10} = \frac{10^{-2}}{10^{-16}}$$

$$= \log 10^{-14}$$

$$= 14 \text{ Bels}$$

Thus, in Bel the total dynamic range is equals to 14 Bels.

However, Bel is a fairly large unit since it compresses the larger range of intensity; therefore, decibel which is 1/10 of a bel is used.

Decibel (dB) = 10 log Io / IR

$$\text{Decibel (dB)} = 10 \log \frac{10^{-2}}{10^{-16}}$$

$$= 10 \log 10^{14} = 10 \times 14$$

$$= 140\text{dBIL}$$

From this we can infer that dynamic range of human ear ranges between 0-140dBIL.

In terms of pressure

dB = 10 log $\frac{I1}{I2 \text{ squared}}$ = 10 log $(PO/PR)^2$ , as Power is pressure

$$10\log (PO/PR)^2 = 20 \log (PO/PR)$$

$$dB\ SPL = 20 \log 2000/0.0002 = 20 \log 10^7$$

$$= 20 \times 7 = 140\ dB\ SPL$$

So in terms of pressure dynamic range of human ear range between 0-140 dB SPL.

Five important aspects of decibel are:

i) It involves a ratio

ii) It involves a logarithm

iii) It is therefore nonlinear

iv) It is expressed in terms of various reference levels; which must be specified.

v) It is a relative unit of measure.

1. Ratio

Ratio is shown when any number is divided by another number. If a number is divided by itself (eg, 25/25), the ratio is always 1:1.

When number with identical basis are used for division, the log of the denominator is subtracted from the log of numerator (e.g$10^3/10^2=10^1$.)

When a ratio is expressed as a fraction, the denominator becomes the reference to which the number is compared.

2. Logarithms

A logarithm (log) is simply a number expressed as an exponent, which tells how often another number (the base) will be multiplied by itself.

In the expression $10^2$ (ten squared) the log ($^2$) tells that the base (10) will multiply by itself one time (10x10=100). The exponent is the power, which tells how many times the base will be used in multiplication.

Eg: $10^3 = 10 \times 10 \times 10 = 1000$

Since, 1000, written in exponential form, is $10^3$ and log is the same as the exponent, then the log of 1000 must be 3.

A shortcut is that the log (or exponent ) equals the number of zeros following the 1:

i.e the log of 10 is 1

the log of 100 is 2

the log of 1000 is 3

the log of 10000 is 4

It is important to note that log of 1 is zero.

$10^0 = 1$ therefore, log of 1=0

This is easily confirmed since there are no zeros following the 1, and the log can be determined by counting the number of zeros follow the 1.

3. Decibel is a nonlinear logarithmic scale, as opposed to a linear scale. In linear measurement, all units on the scale are the same size, therefore 1 inches to 2 inches in on a ruler is exactly the same distance as 11 inches to 12 inches.

In a logarithmic scale, each unit in the scale is larger than the preceding unit. The distance between units is not equal as we go up the scale, but gets progressively larger. The use of a log scale, the decibel (dB), for hearing avoids very large numbers and allows us to describe the entire range of human hearing as OdB to 140 dB.

**A comparison of pressure and decibel scales as intensity increases**

| Linear scale: Pressure | Log scale/ nonlinear scales Decibels |
|---|---|
| 1 | 0dB |
| 1 x 10 | 20 dB |
| 1 x 100 | 40 dB |
| 1 x 1000 | 60 dB |
| 1x10000 | 80 dB |
| 1x100000 | 100 dB |
| 1x 1000000 | 120 dB |
| 1x10000000 | 140 dB |

**Table 1: Comparison of linear and Log scale**

4. Decibel is expressed in terms of various reference levels. An example is the temperature scale. In Celsius 0° is arbitrarily chosen to equal the freezing point of water. Here -10° means that the temperature measured is 10° colder than the freezing point of water. Here 0° is said to be reference point.

If the intensity reference of a sound system (IR) is known, the equation IO/IR may used to determine the number of decibels of the output above (or below ) the reference. As mentioned earlier, it is essential that the reference always be stated. When the reference is $10^{-16}$ watt /$cm^2$ the term intensity level (IL) may be used as a shorthand to imply reference.

5. It is a relative unit of measure and not an absolute scale.

An absolute scale is one in which the zero point of the scale represents the absence of what is being measured.

For example; 0 inches represents the absence of distance; something cannot be - 2 inches long. Like-wise 0 pound represent the absence of weight.

In a relative scale the zero point doesn't represent the absence of what is being measured. The scale value 0 is set, to some point that has particular relevance.

The decibel is a relative scale and therefore, must have an arbitrary reference or zero point. We normally measure intensity in sound pressure, and the approximate lower limit of human hearing is 0.0002 dynes/ $cm^{-2}$. This value is traditionally chosen for OdB. Therefore, if the intensity of a 1000 Hz tone is 50 dBSPL, we mean that it is 50 dB more intense than a 1000 Hz tone of OdB SPL. If a pure tone has an intensity of -10 SPL it is lower in intensity than the reference point.

## Intensity level

Under some circumstances it is useful to express the decibel with an intensity reference. A practical unit in such cases is the watt per centimeter squared (watt/$cm^2$). The intensity reference in a given system may be expressed as IR (the number of watts of the reference intensity). The measured power may be expressed as $I_o$ so that a ratio may be set up between the intensity reference and the measured intensity.

The formula used is

$$dB = 10 \times \log (IO/IR)$$

The usual intensity reference (IR) is $10\text{-}^{16}$ watt/cm$^2$. In an expression like $10^{-16}$ the exponent tells the number of places the decimal points must be moved to the right or left of the number following 1. Therefore, $10^{-16}$ watt /cm$^2$ is an extremely small quantity (0.0000000000000001 watt/cm$^2$. When the reference is watt/cm$^2$, the term intensity level (IL), may be used to imply this reference.

The decibel is a logarithmic expression. When the intensity is doubled the number of decibel is not doubled but increased by three. This is because the intensity outputs of the time signals are added algebraically.

Eg: Loud speaker 'A' creates 80 dB IL and 'B' 80 dBIL to the same point in speaker, the result is 83 dBIL

A= 80 dB B= 80dB

A+ B # 160dB

80 dB = $10^8$ watt/cm$^2$ 80x80= 2 xlO$^8$

$$\text{Total IL} = 10 \log \frac{2 \times 10^{-8}}{10\text{-}^{16}}$$

$$= 10 \log 2 + \log 10^8 = 10 (.3+8) = 83 \text{ dB IL}$$

## Sound pressure level

Audiologists are more accustomed to make measurements of sound pressure than in intensity. Such measurements are usually expressed as sound pressure level (SPL).

Because pressure ratio are known to be proportional to the square root of intensity ratio (Intensity ~ pressure$^2$)

Intensity reference dB (IL) = 10 x log (IO/IR)

Pressure reference dB (SPL) = 10 x log $(PO/PR)^2$

Because intensity is proportional to pressure squared, when determining the number of decibels from a pressure reference, IR may be written as $PR^2$ (pressure reference) and Io may be written as $Po^2$ (Pressure output).

In logarithm when a number is squared, its log is multiplied by 2; therefore, the formula for dBSPL may be written as

dB (SPL) = 10 x log $(PO/PR)^2$

dB (SPL) = 10 x 2 x log (PO/PR)

dB (SPL) = 20 x log (PO/PR)

Because the decibel express a ratio between two sound intensities or two sound pressures, decibel values cannot be simply added or subtracted. Therefore,

60 dB + 60 dB # 120

When sound pressure values are doubled, the number of decibels is increased by six

$Po = 0.0004$ dyne/cm$^2$

$P_R = 0.0002$ dyne/cm$^2$

$dBSPL = 20 \log P_0/P_R = 20 \log 0.0004/0.0002$

$= 20 \log (2 \times 0.0002 /0.0002)$

$= 20 (\log 2 + \log 0.0002/0.0002)$

$= 20 x(p.3 + 0) = 6$ dB SPL

**Application of dB**

**Hearing Level (HL)**

Originally each audiometric manufacturer determined the SPL required to barely stimulate the hearing of an average normal hearing individual. There were some differences from make to make.

The lowest sound intensity that stimulates normal hearing has been called zero hearing loss and zero hearing level (HL). Because the ear shows different amount of sensitivity to different frequencies (being most sensitive in the 1000 to 4000 Hz range), different amounts of pressure are required for zero HL at different frequencies. So audiometers were calibrated in such a way that is could test a wide range of intensities.

Audiometers manufactured in different countries had slightly different SPL values. This was resolved by international standard for organisation who adopted a new standard. They showed normal hearing to be more sensitive, resulting in the lowering of SPL's approximately 10 dB across the frequencies than the prior standards.

## Sensation Levels

Another reference for the decibel may be to the auditory threshold of a given individual. The threshold of a pure tone is usually defined as the levels at which the tone is perceived 50% of the time it is introduced. The number of decibels above the threshold of a given individual is called sensation level.

If a person can barely here a tone at OdB HL at a given frequency, a tone presented at 50 dB HL will be 50 dB above his or her threshold, or 50 dBSL. The same 50 dB presented to a person with a 20 dB threshold will have an SL of 30 dB.

**RETSPL (reference equivalent threshold)**

The RETSPL at a specified frequency for specified type of earphone and for a specified pattern of coupler or artificial ear is the modal value, at that frequency, of the equivalent threshold SPL's of an adequately large no of ears of autologically normal subjects with age limits of 18 to 30 years.

**Table 2 : Reference Threshold levels for Audiometric Earphones[a]**

| Frequency (Hz) | Reference threshold levels for all earphones[b] µPa |
|---|---|
| 125 | 45.0 |
| 250 | 27.0 |
| 500 | 13.5 |
| 750 | - |
| 1000 | 7.5 |
| 1500 | 7.5 |
| 2000 | 9.0 |
| 3000 | 11.5 |
| 4000<br>G000 | 12.0<br>1G.0 |
| 8000 | 15.5 |
| alSO R389 (1964, R 1975)<br>[b]IEC R318 coupler | |

## Octave notation

There are special ways in which frequency may be scaled or transformed. It is rare to see frequency graphed in linear co-ordinates. The usual practice is the use a logarithmic frequency scale. In hearing science, this practice is perhaps given extra importance because on a logarithmic scale 1 kHz falls approximately in the middle of the frequency range of human hearing (roughly 20-20kHz). This frequency also falls in the middle of the frequency range considered more critical for understanding speech, that 500-2000Hz. Just as the magnitude of a sound may be expressed in decibel notation, there is a notational system for frequency. Frequencies may be specified in terms of octave intervals or simply octaves. For any frequency, fo the frequency which is one octave above it is equal to 2fo. The frequency which is two octaves above it is equal to 2 x 2 fo = 4fo. The 3rd octaves is 2 x 2 x 2fo. This scheme represents a progression in powers 2. In general

$$Fn = 2^{n} fo$$

Where fn is the $n^{th}$ octave frequency, n is the number of the octave, and fo is the reference frequency.

For example, the fourth octave frequency of 500 Hz (i.e the frequency which is 4 octaves above 500 Hz) is

$$f4 = 2^{4} \times 500 = 2 \times 2 \times 2 \times 2 \times 500$$

$$= 16 \times 500$$

$$= 8000 \text{ Hz}$$

The above told equation is equally valid for finding octave frequencies below the reference frequency. In this case octave number n is negative.

For example the frequency which is one octave below 500 Hz is

$f_{-1} = 2^{-1} \times 500 = ½ \times 500 = 250$ Hz

The equation can also accommodate fractional octaves, i.e finding the frequency which is one-half octave above 1000 Hz.

$F_{½} = 2^{½} \times 1000 = 2 \times 1000 = 1.414 \times 1000 = 1414$ Hz.

In the above example, it might, at first glance, be reasoned that 1500 Hz must be the frequency one -half octave above 1000 Hz since, the frequency that is one whole octave above 1000 Hz is 2000 Hz. It is tempting to assume that the half - octave frequency will fall in the middle (1500 Hz). The subsequent errors in computation can be averted if it is clearly borne in mind that the number of the octave is an exponent of 2 and not merely a coefficient of the reference frequency.

Octave notation like decibel notation, involves a transformation in which the numerical scale is no longer linear.

In the case of logarithms equal intervals on an octave scale reflects constant ratio on a linear scale. For instance the difference between 50 and 100 Hz is 50 Hz; the difference between 500 and 1000 Hz is 500 Hz. However, on an octave scale 50 to 100 Hz and 500 to 1000 Hz represent the same interval, namely 1 octave.

**Section B**

**Pick the correct one**

1. Dynamic range of human ear ranges from

   (0-20dBIL, 0-140 dBIL, 0-90dBIL)

2. Decibel is a represented in________________scale

   (Relative, linear, absolute)

3. Frequency Range of human hearing

   (20 Hz-20KHz, 200 Hz-20KHz, 20 Hz-2000Hz)

**Fill in the Blanks**

4. Intensity is measured in terms of____________and______________

5. Decibel is a unit of______________

6. Lowest sound intensity that stimulates normal hearing has been called______________

**True or False**

7. Greater the pressure change greater the intensity of sound.

8. When a number is divided by itself, the ratio is always 0.

9. The logarithm of '1' is 1.

10. In logarithm scale each unit is larger than the proceeding unit.

11. Power is equal to pressure squared

12. 80 dB IL + 80 dB IL = 86 dB IL.

13. Identify the units.

a) $Log_{10}$ Io/lR =

b) 10Log $I_0/I_R$ =

c) 20 Log($p_0$/PR)=

d) Number of dB above the threshold =

e) Zero hearing loss / zero hearing level =

**Match the following**

| | |
|---|---|
| 14. a) 0.0002 dynes / $cm^2$ | maximum audible power (a) |
| b) $10^{16}$ watt / $cm^2$ | minimal audible pressure (b) |
| c) 2000 dynes/$cm^2$ | 40 dB SPL (c) |
| d) $10^2$ watt / $cm^2$ | 66 dB (d) |
| e) 0.02 dynes/$cm^2$ | maximum audible pressure (e) |
| 1] 60 dB SPL + 60 dB SPL | minimal audible power (f) |

**Pick the correct one**

| | |
|---|---|
| 15. a) Notational system for frequency | Decibel notation (a) |
| b) $F_n$ | 2 $f_0$ (b) |
| c) 500 to 2000 Hz | $2^n f_0$ (c) |
| d) Notational system for intensity | Speech spectrum (d) |
| e) One octave above $f_0$ | octave notation (e) |

# ANSWERS

**Section B**

1. 0-140 dBIL
2. Relative
3. 20 Hz to 20 k Hz
4. Power and pressure
5. Intensity
6. Zero hearing loss or zero hearing level.
7. True
8. False
9. False
10. True
11. True
12. False
13. a) Bel

    b) Decibel

    c) Sound pressure level

    d) Sensation level

    e) Hearing Level
14. a, b

    b,f

    c, a

    d, e

    e, c

    f,d

**15.** **a, e**

**b, c**

**c, d**

**d, a**

**e, b**

# CONCEPT OF THRESHOLD

The threshold has been viewed as a discrete point along the physical continum of a particular stimulus, i.e a certain intensity, along which there is an appropriate response, and below which there is none. This discrete point is difficult to demonstrate by virtue of the influenze of various internal and external events not under the control of the experiment.

- External events might include small but finite fluctuations in the instrumentation or procedural limitation.

- Internal events would include variations in the experimental subjects state of attention.

Consequently threshold has came to be considered more dependent on the magnitude of the stimulus.
The criteria for judging the threshold value is the elicitation of the desired response a certain percentage of the time the stimulus is presented. For example, to find the magnitude of sound which is just detectable i.e the absolute threshold, the sound is presented at different levels and the subject responses are tabulated.

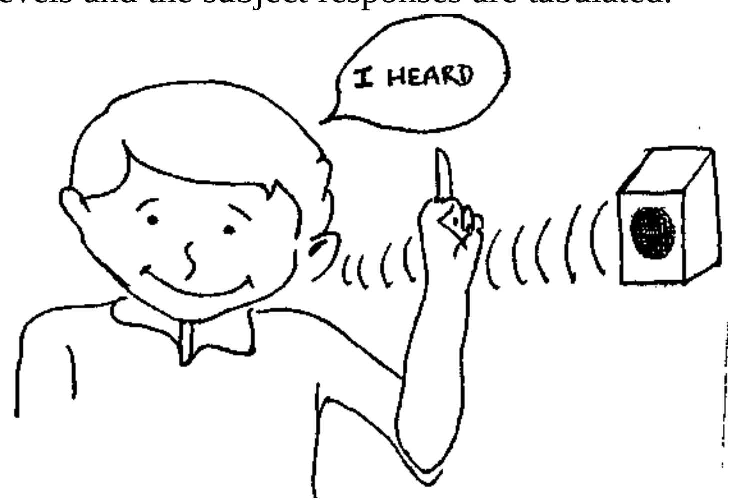



**Fig. 22: Showing an individual hearing the sound through loud speaker**

The level of which the sound is heard 50% of the time it was presented is traditionally defined as absolute threshold. Above this magnitude the subject will respond more frequently to the stimulus, where as below this value responses may or may be present. If present it will be less frequent.

In defining the threshold for a given tone, the procedure used in obtaining the measurement is an important variable. Two methods of presenting stimuli are commonly used in determining auditory threshold measurement: minimum audible field (MAF) and minimum audible pressure (MAP).


**Fig. 23: Measurement of threshold under MAF condition.**

Minimal audible field (MAF) thresholds are sound pressure levels for different pure tones, measured in a free field.

The stimuli are presented in a special room with a controlled sound environment called a sound field.

**Procedure**

- The subject is seated in a sound field.
- Distance of the subject and the loud speaker should not exceed 1 m. This is because as distance increases, intensity decreases.
- Angle between the subject and the loud speaker should be 45°. The sound produced reaching the ear is maximum at 45°.
- Select the frequency. The stimulus presented should be in dBSPL.
- Threshold values are then collected. These measurements are referred to as minimal audible field data.

In the MAF data, a number of variable that are inherent into the method must be considered.

The position of the head in the sound field affects sound reflections in the area of the ears, thus resulting in the reduction of the amplitude. This is known as "head shadow effect". This effect is determined by the angle of the head to the sound source and the frequency of the stimulus used.

The placing of subjects body in the sound field also affects the measurements, this is referred to as "body baffle effect".

The ear canal has a resonance effect, which modifies the signal from the source before it reaches the tympanic membrane. The external auditory meatus has a natural resonance in the frequency of 2500 Hz which enhances the stimulus by 10 dB to 15 dB.

The literature shows the sum of all these effects (body baffle, head shadow, and canal resonance) can cause amplification of a stimulus up to 15-17 dB.

Minimal audible pressure (MAP) thresholds describe threshold interms of the sound pressure level under earphones.

**Fig. 24: Measurement of threshold under MAP condition**

The listener listen to sound presented through earphones, and various procedures are used to determine the sound pressure that occur at tympanic membrane.

To estimate the actual pressure at the tympanic membrane in a standardized manner, the sound - pressure level is estimated from the sound level meter in a test coupler attached to the earphone during calibration. Such couplers are designed to approximate the average acoustic properties of the ear of a listener with normal hearing and contain a microphone for estimating the sound pressure level that would exists at the tympanic membrane. Sound level meter attached to the earphone through the coupler; this arrangement is often referred as an artificial ear.

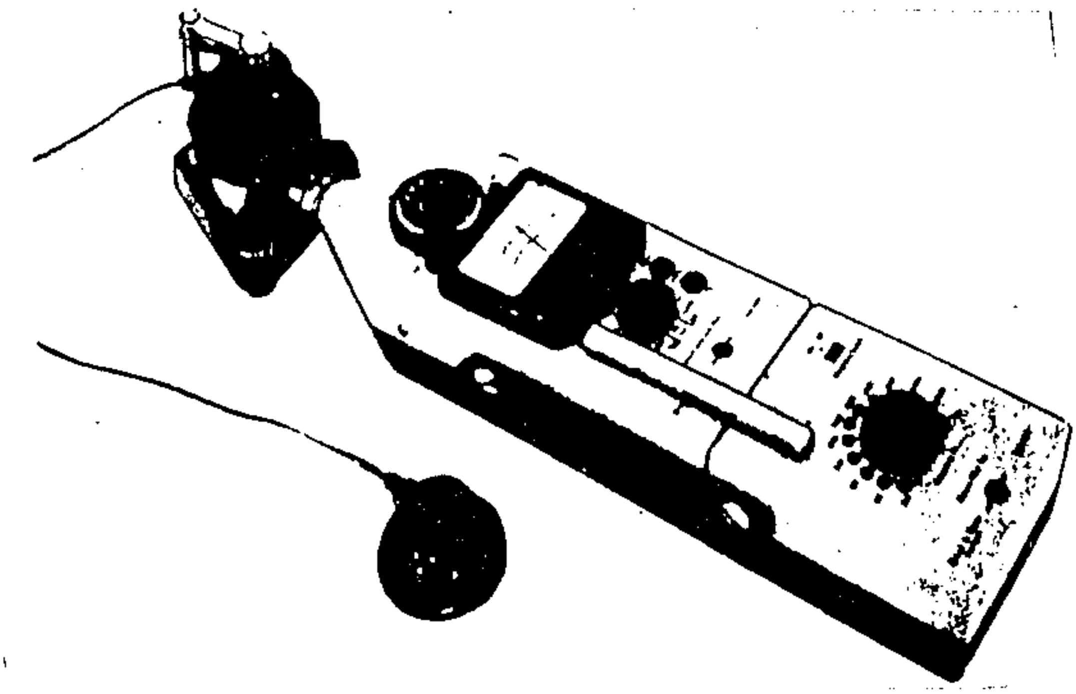


**Fig. 25: Calibration procedure using artificial ear and SLM**

## Procedure

- A probe tube microphone is inserted in the earcanal of the subject
- Place the head phone on the subjects head.
- The output of microphone is connected to an SLM or sound level meter.
- Select the frequencies and the thresholds are found at each frequencies.

Threshold obtained under MAF and MAP conditions are then plotted in a frequency vs intensity graph to obtained MAF and MAP curve as shown in fig. 26

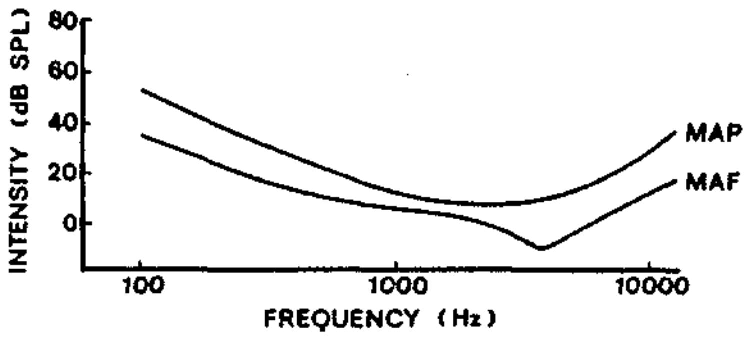



**Fig: 26: MAP and MAF curve**

An intriguing phenomenon demonstrated in the fig.26 is that the MAF curve falls consistently below the MAP curve. In other words, a lower intensity is needed to reach threshold in a sound field (MAF) than under earphones (MAP). This fact was demonstrated by Sivian and White (1993) and the discrepancy of 6 dB is called 'missing 6 dB' phenomenon (Munson and Wiener, 1952).

The MAP /MAF discrepancy might be due to

- Physiological noise picked up by the earphone. These physiological noses would partially mask the signal presented by the earphone, so that more sound pressure would be needed to achieve MAP than for the unmasked MAF.

- Binaural Summation can be stated as another reason. Under free field condition, stimulus goes to both ears simultaneously which results in threshold enhancement of 3 dB.

- Resonance frequency of external auditory canal is altered when you place head phone.
- When you place head phone, air is trapped between tympanic membrane and head phone. This results in change in pressure which inturn results in reduction in hearing sensitivity.

Although MAP may not have the apparent correlation to everyday listening situations that MAF does, the earphone technique has many advantages for research and clinical purposes. It is far easier to build an adequate sound treated room to allow earphone (MAP) measurements than it is to build a properly controlled anechoic (echo free) sound field for MAF measurements.

## QUESTIONS

### Choose the Correct Answers

1. The level at which the sound is heard 50% of the time it was presented is

   (relative threshold, absolute threshold, differential threshold).

2. In MAP stimulus are presented through

   (earphone, sound field, loud speaker).

3. In MAF angle between the listener and the individual should be

   (90°, 180°, 45°)

4. The position of the head in the sound field affects sound reflections in the area of the two ears, thus resulting in the reduction of amplitude. This is called.

   (Head shadow effect, body baffle effect, canal resonance effect).

5. The sum of body baffle, head shadow and canal resonance results in the amplification of

   (7-8dB, 15-17dBm 30-=35 dB)

### Fill in the Blanks

6. Threshold of an individual is influenced by........................and ........................which is not under the control of the experimenter.

7. Two methods commonly used in determining auditory thresholds are.........................and.....................

8. The threshold values collected throughout MAP are referred as

9. Sound level meter attached to the earphone through the coupler , this arrangement is often referred as——————————.

10. Binaural summation results in————————————threshold enhancement.

11. In MAF the stimulus are presented in a special room with a controlled sound environment called——————————.

12. Presentation of stimuli during MAF should be in——————————.

13. The placing of subjects body in the sound field affects the threshold measurement. This refers to——————————.

14. Resonance frequency of EAC is altered under——————————.

15. Physiological noise is more under——————-——————.

**True or false**

16. The threshold has been viewed as a discrete point along the physical continuum of a particular stimulus.

17. Minimal audible field threshold are sound pressure levels for pure tones measured under earphones.

18. As distance increases, intensity decreases.

19. MAF curve always fall below MAP curve.

20. Ear canal has a natural resonance in the area of 8K Hz.

21. Word grid

| | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| A | D | E | G | X | I | B | Y | F | Z | V | E | Z |
| T | H | R | E | S | H | 0 | L | D | I | M | A | F |
| C | A | N | W | L | J | D | X | N | D | N | R | T |
| E | V | F | X | M | A | Y | S | M | H | A | R | M |
| B | R | G | V | Y | K | B | M | U | A | T | E | I |
| H | E | A | D | S | M | A | D | 0 | W | X | S | A |
| V | T | N | S | T | A | F | Y | u | Y | F | 0 | K |
| S | S | H | N | I | P | F | 0 | p | K | C | N | N |
| R | P | L | 0 | M | I | L | A | H | D | M | A | P |
| X | L | N | P | U | T | E | 1 | Y | Z | T | N | E |
| Q | K | A | T | L | L | 0 | T | P | R | H | C | U |
| Y | M | K | L | I | K | R | Q | I | M | O | E | K |

## 22. Match the Following

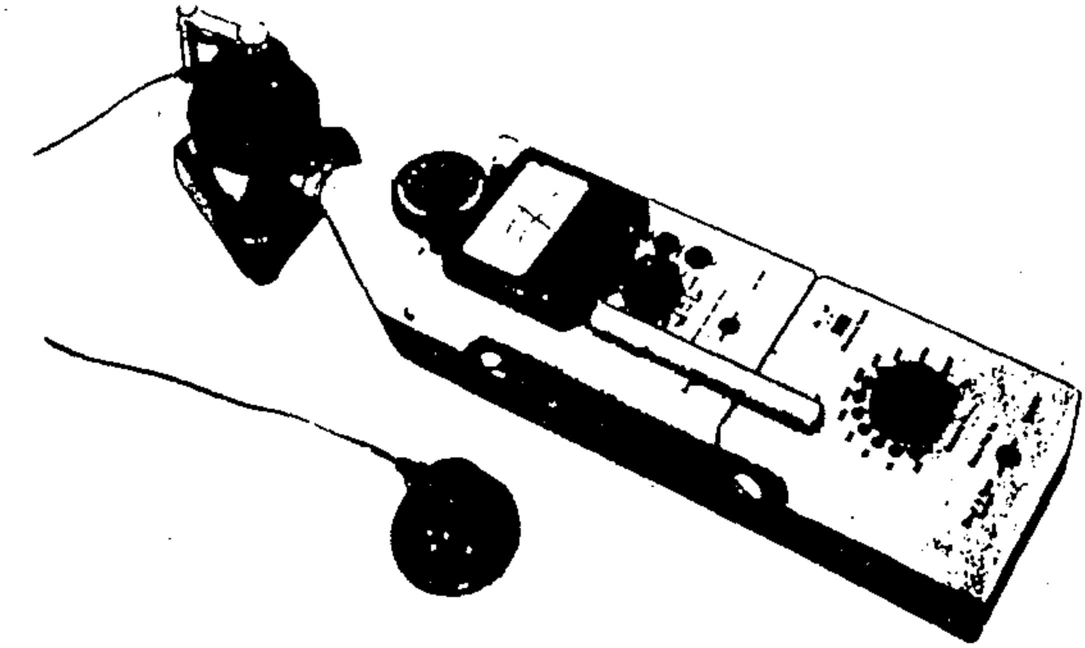
| | |
|---|---|
| a) Distance between loud speaker and the listener | 2500 Hz (a) |
| b) Angle between loud speaker and the listener | lm (b) |
| c) Resonance of ear canal | 10-15dB (c) |
| d) Enhancement of stimuli due to resonance frequency of ear canal | 45° (d) |

23. Identify the condition

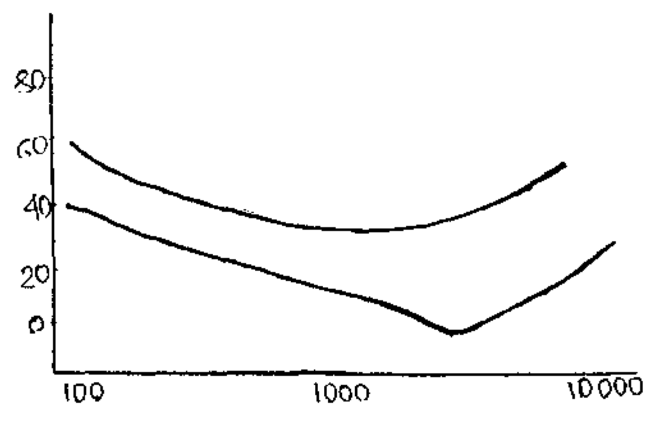
(a)

(b)

(c)



(d)



## ANSWER

1. Absolute threshold
2. Earphone
3. 45"
4. Head shadow effect
5. 15 to 17 dB
6. External and internal events
7. MAP and MAF
8. Minimal audible field data.
9. Artificial ear.
10. 3dB
11. Sound field
12. SPL
13. Body Baffle effect
14. Head phone
15. Head phone
16. True
17. False
18. True
19. True
20. False

**21.**

| | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| A | D | E | G | X | I | B | Y | F | **Z** | V | E | **Z** |
| T | H | **R** | **E** | **S** | H | **O** | **L** | D | I | M | **A** | F |
| C | **A** | N | W | L | J | D | **X** | N | D | N | **R** | T |
| E | **V** | F | X | M | A | Y | S | M | H | A | **R** | M |
| **B** | R | G | **V** | Y | **K** | B | M | **U** | A | T | **E** | I |
| **H** | **E** | **A** | **D** | **S** | M | **A** | **D** | **O** | **W** | **X** | **S** | **A** |
| **V** | **T** | **N** | **S** | **T** | **A** | **F** | **Y** | **u** | Y | **F** | **O** | K |
| **S** | **S** | **H** | **N** | I | **P** | **F** | **0** | **p** | **K** | **C** | **N** | N |
| **R** | **P** | L | 0 | M | I | L | **A** | H | D | M | **A** | P |
| **X** | **L** | **N** | P | **U** | T | E | I | Y | Z | T | N | E |
| **Q** | K | **A** | T | L | **L** | **O** | T | P | R | H | **C** | U |
| Y | M | K | L | I | K | **R** | **Q** | I | M | 0 | E | K |

22. a, b

b, d

c, a

d, c

23. a) MAP

b) MAF

c) Artificial ear

d) MAP/MAP curve

# THE AUDITORY RESPONSE AREA

The threshold of audibility define the smallest amount of pressure to which the auditory system is sensitive. Here a listener is instructed to listen carefully and to indicate when he can just detect the presence of a tonal sound. The intensity of a sound at one frequency is varied in increasing and decreasing steps until the subjects threshold at this frequency is found. Other frequencies of test stimulus are then presented to the subject until his absolute thresholds are obtained at sufficient numbers of frequencies to permit the drawing of a graph similar to the lower curve in the fig. 27. This is the minimum audibility curve for this listener.

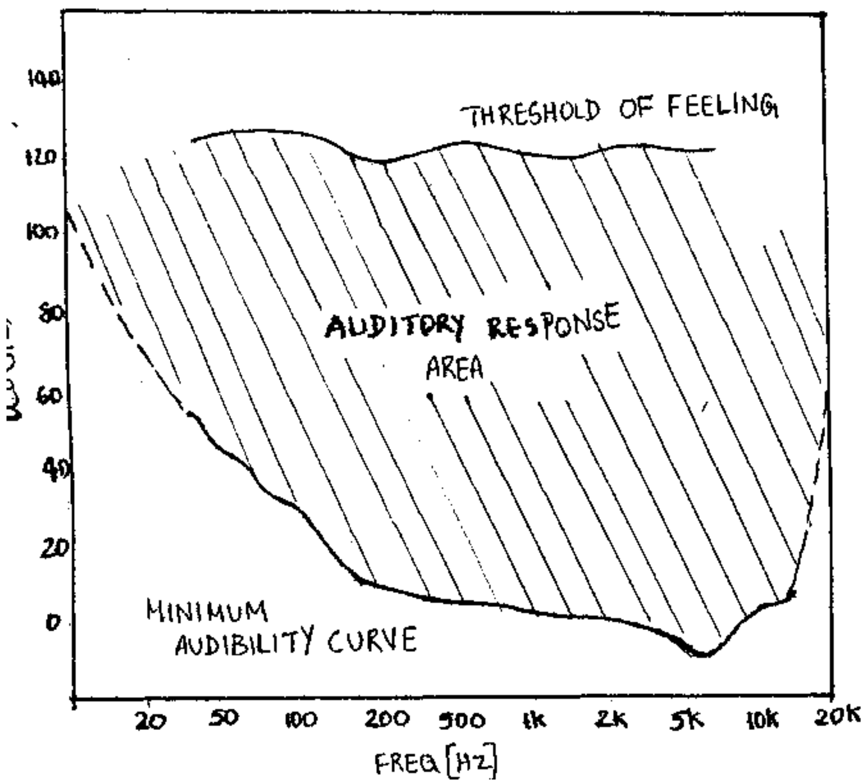



**Fig. 27: Auditory response area**

In this graph maximal sensitivity is observed around 200 Hz to 8KHz. In this range the least sound pressure is required for the listener to just detect the presence of the sound. This range of frequencies encompasses the spectral components of speech sounds which are most important for their understanding. Outside this frequency range, frequency range of hearing becomes increasingly less sensitive.

On the low frequency side of the hearing range, humans are capable of responding to sounds as low as lHz, but the test tones fail to have any tonal quality below approximately 20Hz. Thus, 20Hz traditionally has been considered the lowest frequency limit of hearing in man. Responses to tones above 20 K Hz have been reported, but this frequency has stood as the practical upper limit. Thus, the frequency limits of human hearing, typically are given as 20-20,000 Hz. These limits represent the boundaries along the frequency axis of the physical domain over which hearing exists, that is, the auditory response area.

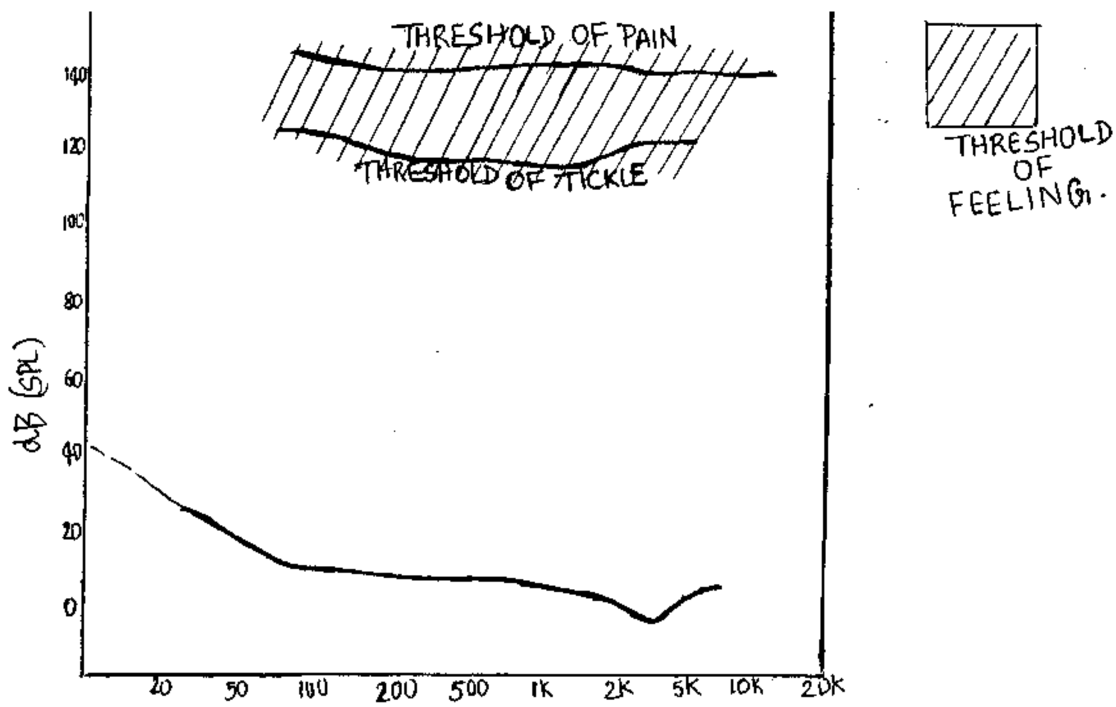



**Fig. 28: Showing threshold of pain and tickle**

If sound is made increasingly intense, it ultimately will elicit a tactual sensation or feeling depending on the individual tolerance, this may be inexperienced as a tickling sensation, but with only a little increase in the sound level of sound, pain is experienced. These sensation can be elicited at any frequency, and the level of the stimulus at which feeling is first experienced is called the threshold of feeling. The threshold of feeling is fairly constant across frequency at about 130-140 dB SPL. Level of sound at which tactual sensations are elicited are quite hazardous to the hearing organ, even for brief duration of exposure, and cannot be considered physiologically appropriate for the hearing organ. For these reasons, the threshold of feeling can be considered as the upper limit of the auditory response area. In summary, the auditory response area represents an island in the physical domain of sound which is bounded on one side by the limits of delectability of sound and on the other side by the limits of tolerability for sound.

# QUESTIONS

**Choose the correct answer**

1. Maximum sensitivity of human ear is around——————————
   (200Hz - 8KHz, lHz - 2Hz, 5KHz - 8KHz)

2. Lowest frequency limit of hearing in man-----------------.
   (20dB, 20Hz, 20KHz)

3. Frequency ranges of human hearing is-----------------.
   (200Hz - 2KHz - 2000Hz, 20Hz - 20KHz)

**True / False**

4. Upper frequency limit of hearing in man is 2000Hz

5. Depending on the individuals tolerance, the increasingly intensive sound may be experienced as a tickling sensation.

6. The threshold of feeling is fairly consistent 130 - 140dB SPL.

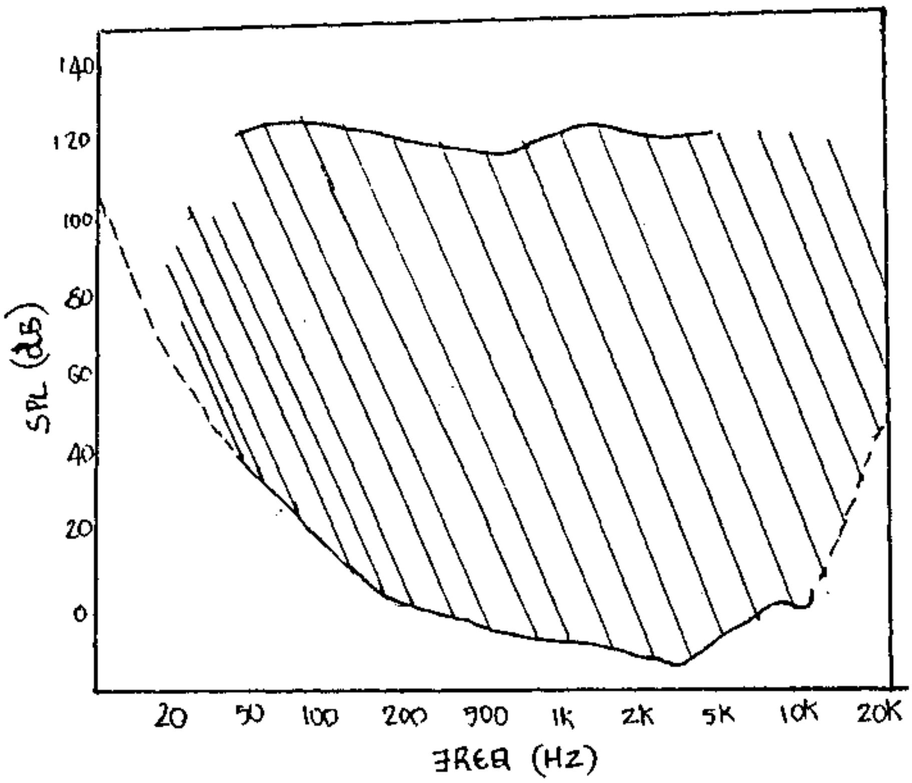
7. Threshold of feeling is not hazardous for hearing organ.

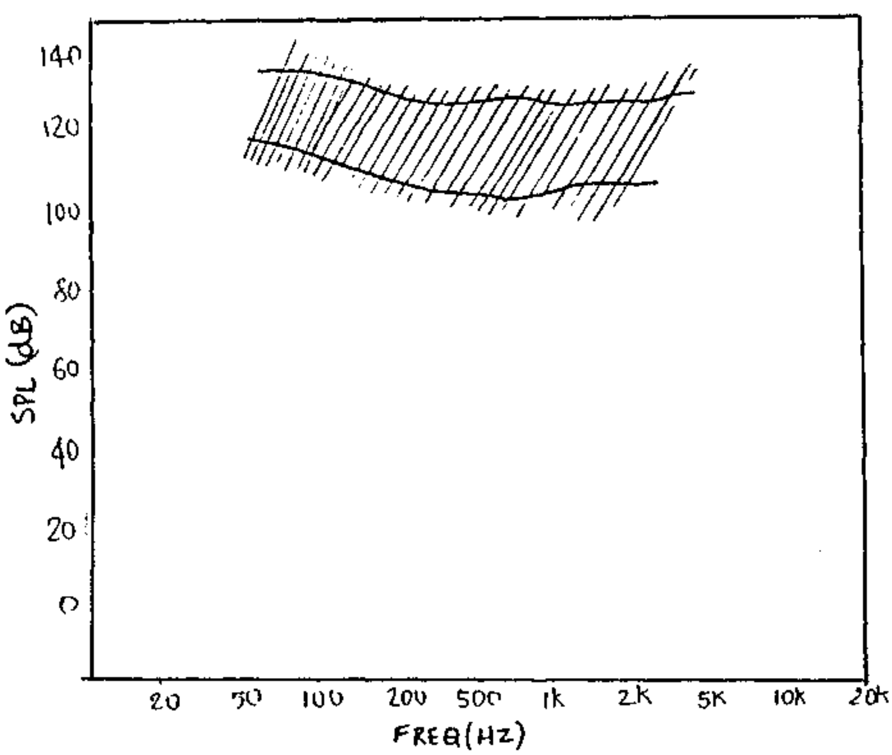
**Pill up the blanks:**

8. Threshold of feeling is the——————————limit of——————area.

9. Minimum audible curve is the——————- limit of —————— area.

10. Name the Graph



## ANSWERS

1. 200 Hz-800 KHz
2. 20 Hz
3. 20 Hz-20 KHz
4. False
5. True
6. True
7. False
8. Upper limit of auditory response area
9. Lower limit of auditory response area
10. a) Auditory response area

    b) Threshold of feeling.

# DIFFERENTIAL SENSITIVITY

In previous chapter we studied absolute threshold, or the lower limit of auditory awareness. Hearing function also depend on the systems activity to resolve fine differences between sounds which is termed as differential sensitivity (DL) or differential auditory threshold. This measures the smallest differences between two stimuli that can first be perceived by a listener. This can be called just noticeable difference or JND.

There may be a JND either in intensity, frequency or duration between two physical stimuli. The detection of the DL involves not the detection of the presence of the stimuli, but the detection of difference between two stimuli. To a listener, two stimuli may seem to equal eventhough slight difference exist between them. If the listener detects very small difference consistently then we can say that he has a high differential sensitivity (Small cited in Moore, 1982).

Thus the smallest perceivable difference between two physical stimuli or sounds is either called difference limen (DL) or JND.

Difference limen may be expressed in absolute or relative terms. This means that the minimum stimulus separation just perceived as noticeably different may be reported in the actual physical unit themselves, or the separation may be divided by the original stimulus value to form a fraction, i.e., relative difference is obtained by dividing the absolute DL by the value of the starting level. Thus if the starting level is 1000 units and the DL is 50 units, then the relative DL A I/I is

50/1000 = 0.05. This ratio $\Delta$ I/I is called Weber fraction.

The fraction obtained when the absolute increment is divided by the standard is known as the Weber fraction or Weber ratio. It derives its name from the work of Ernst Weber; who in 1834, proposed that in order for one stimulus to be judged as just noticeably different from another, the second always has to be increased by a constant proportion of the first. This relationship is known as Weber law. This can be written as

$$AI/I = K$$

Where AI represents the absolute change in the stimulus necessary to produce a judgement of just noticeably different, I represents the value to which increment is added and K represents the constant.

It has found that at lower and upper extremes of intensity ranges, the Weber fractions tend to increase. In other words, our discrimination abilities are generally better in the mid- ranges than at the extremes.

**Intensity discrimination**

The difference Limen for intensity (DLI) is generally measured by presenting two tones of the same frequency to the listeners. The first, tone, the standard, remains fixed at some intensity level. The second tone, the comparison, varies in small dB steps around the standard. The listening task is to determine when the tones are just noticeably different in their loudness sensations.

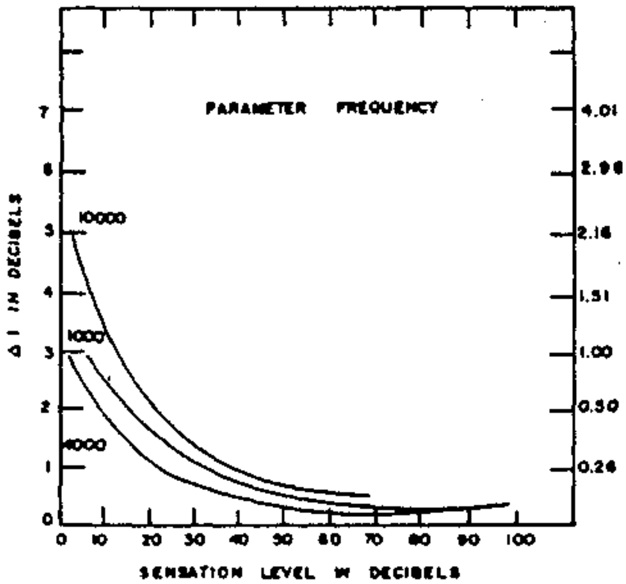



**Fig.** 29: DIFFRENCE LIMEN FOR INTENSITY IN ABSOLUTE AND RELATIVE TERMS..

Figure 29 shows the DL for intensity for 1000, 4000, and 10,000 Hz stimuli. The absolute values are listed on the left side and the relative, value are listed on the right hand side. The horizontal axis shows the sensation levels (in dB) of the standard tones. There are three separate curves for.each of the frequencies.

All the three frequencies show similar patterns, although there are difference in actual DL values. Each plots slope downward as the sensation levels of the tone are increased to about 40-50 dB, after that DL value remains relatively stable. This tells us that at low sensation levels we are less sensitive to sound intensity changes than at higher sensation levels.

## Frequency discrimination

The difference Limen for frequency (DLF) is determined by presenting two tones to the listeners. The first tone, the standard, remains fixed at one frequency at all limes the second tone the comparison, varies in small frequency steps around the standard. The testing task in this case is to determine when the tones are JND in their pitch sensation. As DLI, DLF can be expressed in absolute as well as relatives terms.

Two important aspects that should be considered in frequency discrimination are the effect of change in loudness level and frequency on DLF.

The figure 30 shows how the DL for frequency (in relative terms, AF/F) changes for various frequency tones as loudness level is increased. Although absolute of DLI vary from frequency to frequency, all of the curves show similar patterns. In general, each curve slopes rapidly downward as the loudness level is increased.

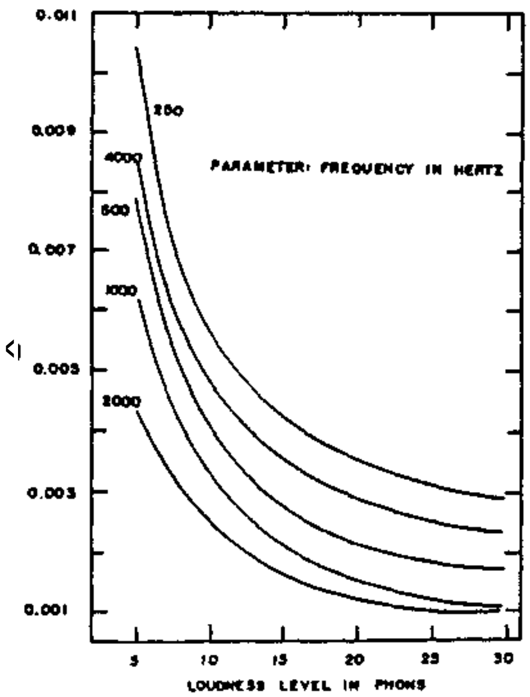



**Fig.** 30: DIFFERENCE LIMEN FOR FREQUENCY AS A FUNCTION OF LOUDNESS LEVEL

This figure tells us that, we are relatively insensitive to frequency changes at low loudness level, but as the loudness level is increased, our discrimination ability improves. The figure also shows that our overall sensitivity to some frequencies is better than others. Thus, lower the Weber fraction, the better the frequency discrimination ability.

Now we will deal with the effect of frequency. The figure 31 shows the relative difference limen (AF/F) as a function of frequency.

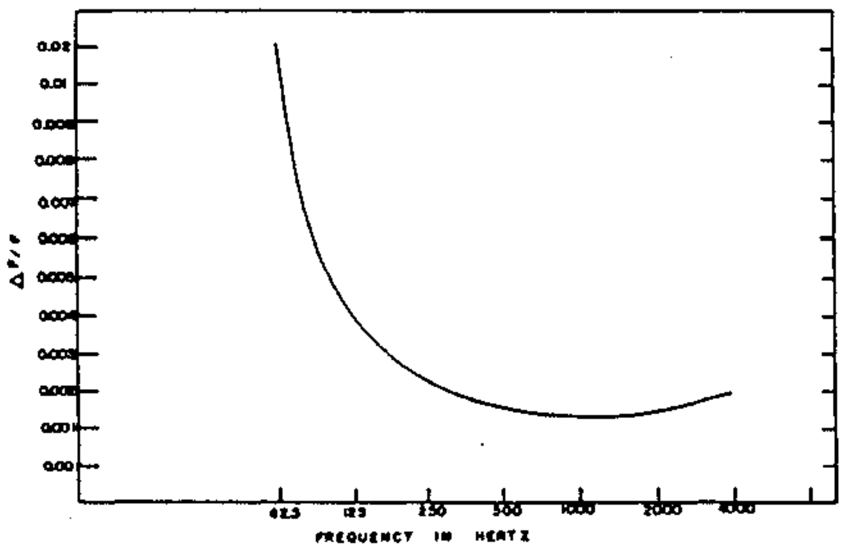



**Fig. 31: Showing relative DLF as a function of frequency**

The figure clearly shows that DLF decreases rapidly as the frequency is raised to 125Hz, but thereafter remains fairly constant.

Although the DL values remain relatively unchanged from 500 Hz to 4000 Hz, our best sensitivity appears to lie in the 1000Hz to 2000 Hz frequency region.

## Time discrimination

The difference limen for time (DLT) may be measured in a similar manner to intensity and frequency. Generally, two signals are presented in succession. The first signal has a fixed duration, and remains at a given loudness level. The second signal, the comparison, varies in small duration steps around the first signal.

The loudness level of the second signal equals that of the first signal. The subjects task in this case is to determine in their apparent lengths.

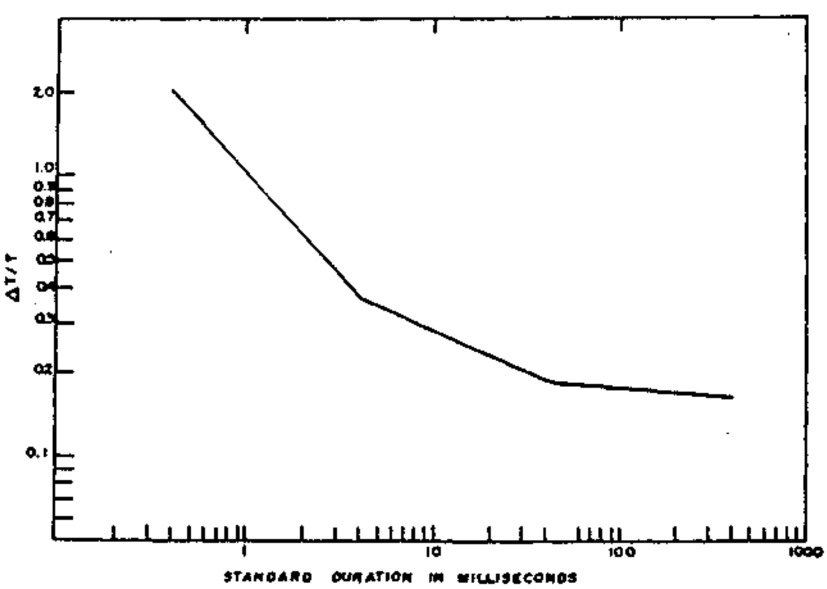



Fig. 32:. RELATIVE DLT AS A FUNCTION OF TIME

The figure shows how the relative difference limen for time. (AT/T) varies as the duration of the standard signal (T) is lengthened.

The effect is quiet clear. As the duration of the standard increases from 0.4 milliseconds (0.0004 seconds) to 4.0 milliseconds (0.004 seconds), there is a rapid decline in Weber fractions. When the standard varies from 4.0 milliseconds to 400 milliseconds (0.4 seconds), the decrease continues, but at a slower pace than before.

We may conclude that our sensitivity to time differences improves as the length of the stimulus increases.

## QUESTIONS

### Choose the correct answer

1. The smallest perceivable difference between two physical stimuli. (DL, absolute threshold, reference threshold)

2. The fraction obtained when the absolute increment is divided by the standard

   (Weber fraction, Bing fraction, Rinne fraction).

3. At lower and upper extremes of intensity range, Weber fraction.

   (Increases, decreases, remains constant).

4. Difference limen can other wise be called as

   (NPD, JND, PND)

### Fill in the blanks

5. There can be DL in————————,———————and——————between the physical stimuli.

6. DL can be expressed in———————and———————terms.

7. Discrimination abilities are better at———————range

8. As loudness level increases, discrimination ability———————.

9. Our sensitivity to detect difference improves as————————of the stimulus increases.

10. At low sensation levels we are ——————————sensitive to intensity changes than higher sensation

    (Less, more, not affected)

**True or False**

11. If the listener detects very small difference consistently we can say that he has a high differential sensitivity.

12. The Difference Limen for intensity is generally measured by presenting two tones of different frequency to the listener.

13. Two important aspects that should be considered in frequency discrimination are the effect of change in loudness and frequency.

14. Lower the Weber fraction, better the frequency discrimination ability.

15. In time discrimination the subject has to determine when the signal is same in their apparent length.

16. Acronyms

    a) DL

    b) JND

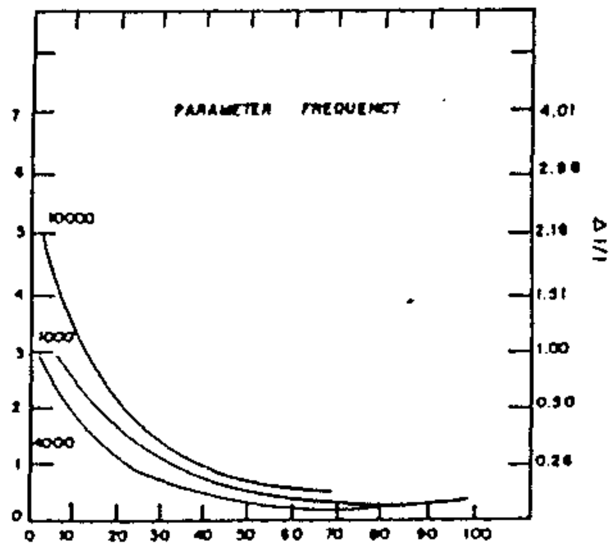
    c) DLF

    d) DLI

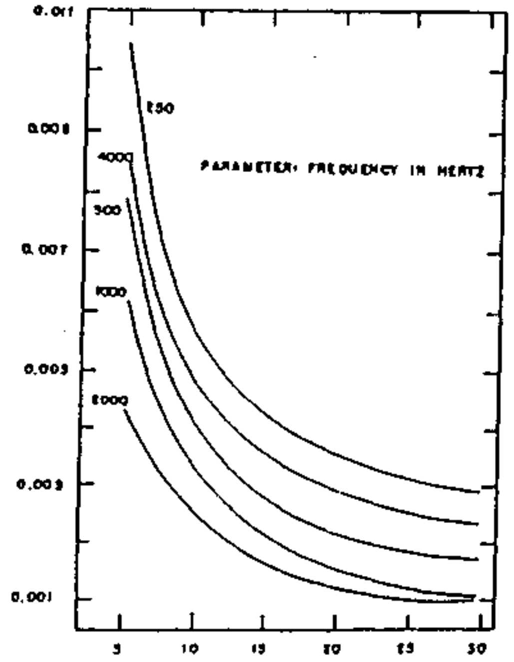
    e) DLT

17. State what represents the X axis and Y axis of the given graphs.

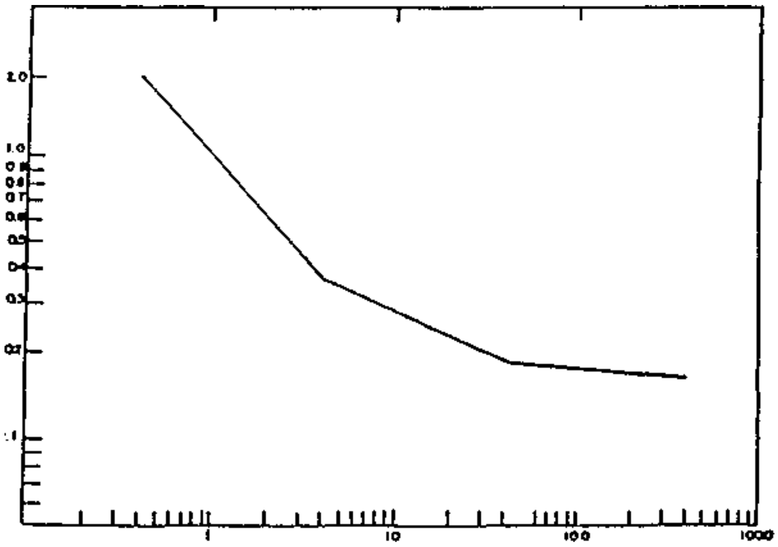
(a)




(b)



(c)



## ANSWERS

1. DL
2. Weber fraction
3. Increase
4. JND
5. Intensity, frequency, time
6. Absolute and relative
7. Mid frequency
8. Increases
9. Length
10. Less
11. True
12. False
13. True
14. True
15. False.
16. a) Difference limen
    b) Just noticeable difference
    c) Difference limen frequency
    d) Difference limen intensity
    f) Difference limen time

17. a) X axis : sensation level in dB

Y axis : A I in dB

b) X axis : loudness level in phons

Y axis : A F/F

c) X axis : Standard duration in miliseconds

Y axis : A T/T

# LOUDNESS

The intensity of a sound refers to its physical magnitude which may be expressed in such terms as its power or pressure. The psychoacoustical correlate of intensity is referred as loudness. Generally speaking, low intensities are perceived as "soft" and high intensities as "loud".

In other words, intensity is the physical parameter of the stimulus and loudness is the percept associated with that parameter.

**Loudness Level:**

Same amount of intensity for different frequencies do not produce same loudness.

e.g.: The loudness of 100 Hz at 40 dB SPL and 1000 Hz at 40dB SPL is not the same.

The intensity needed in order for tones of different frequencies to sound equally loud are called equal loudness level. The earliest equal loudness data were reported by Kings bury in 1927. The first well accepted phone curves were published in 1933 by Fletcher and Munson, and as a result, equal loudness contours have also came to known as "Fletcher Munson Curves".

A curve that relates equal loudness sensation (In dB SPL) for various frequency sounds are called equal - loudness contour. Equal loudness contours are descriptions of the frequency dependence of the loudness of puretone (Fletcher and Munson, cited in Gelfand 1997)

They can be measured easily by requiring listeners to match the intensity of a comparison tone of variable frequency to the intensity of a standard tone at 1 KHz. These tones are presented alternately rather than simultaneously. The technique often used is to present a series of levels of the comparison tone and ask the listeners to judge for each level whether the comparison is louder or softer than the standard. Thus the loudness level of any comparison tone is the level in dB SPL of the 1000 Hz tone to which it sounds equal in loudness. The unit of loudness level is 'PHON'. (1KHz is used because human hearing is most sensitive at 1KHz).

E.g., the contour labeled "40 Phons* shows the intensities needed at each frequency for a tone to sound equal in loudness to a 1000 Hz reference tone presented at 40dB SPL. Thus any sound which is equal in loudness to a 1000 Hz tone at 40dB has a loudness level of 40 phons. The loudness level of the 1KHz tone is also 40 phons. A tone which is as loud as a 1000 Hz tone at 50dB has a loudness level of 50 phons one that is loud as a 1000 Hz tone at 80 dB has a loudness level of 80 phones etc.

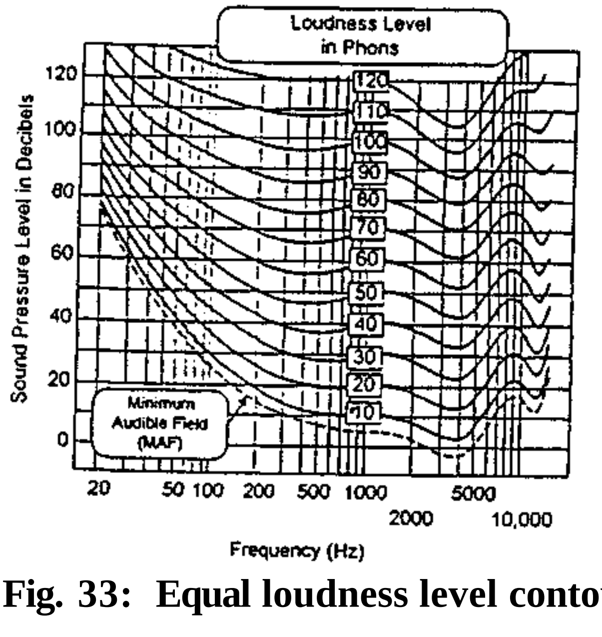



**Fig. 33: Equal loudness level contour**

The figure 33 shows equal loudness contours from 20 phones to 120 phones and also includes the absolute threshold curve (MAF).

It can be seen that the contours are of similar shape to the threshold curve at low loudness levels and at high levels the contour flattens. This means that the rate of growth of loudness differs for tones of different frequencies.

More intensity is needed to achieve equal loudness for lower frequencies than for higher ones. The phon curves tend to flatten for higher loudness levels indicating that the lower frequencies grow in loudness at a faster rate than the higher frequencies.

The sone scale is another way to measure loudness, where one sone is the loudness of a 1000 Hz tone presented at 40 dB SPL. One sone is equal to 40 phons. A stimulus that is 'n' sones loud is judged to be 'n' times as loud as 1 sone; n times as loud as the 1000 Hz, 40 dB SPL standard.

The intensity required for a subject to perceive half the loudness of 1 sone is 0.5 sones, the intensity for twice the loudness is 2 sones, and so forth. The number of sones can be obtained using the formula given below:

$$\text{Sone} = 2^{(P-40/10)}$$

Where 'P' is the loudness level of a specific frequency

E.g. P = 40.

$$S = 2^{((40-40)/10)}$$

$= 2^{0}$

S = 1.

1 sone is equal to 40 phones

a change of 10 phones will result in a two fold change in sone.

i.e. 1 sone = 40 Phones

2 sone = 50 Phones

4 sone = 60 Phones and so forth.

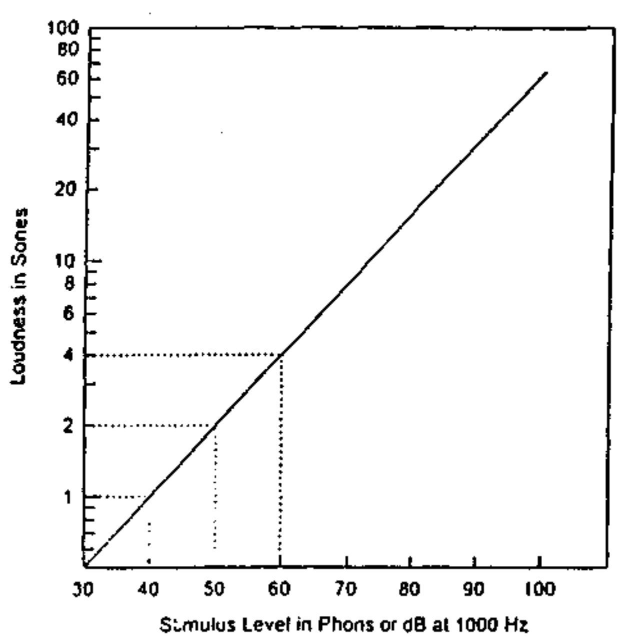



**Fig. 34: Relation between loudness and loudness level**

## QUESTIONS

### Choose the correct answer

1. Perception of intensity

   (Pitch, Loudness, frequency)

2. High intensities are perceived as

   (Soft, medium, loud)

3. Intensity needed in order for tones of different frequencies to sound equally loud

   (Rqual loudness level, equal pitch level, equal intensity level)

4. Unit of loudness level.

   (Sone, Bel, Phon)

### Fill up the blanks

5. Low intensities are perceived as__________.

6. A curve that relates equal loudness sensation for various frequency sounds are called__________.

7. Fletcher and Munson curve are also known as__________.

8. More intensity is needed to achieve equal loudness for __________frequencies than for__________ones.

9. __________frequencies grows in loudness faster than __________frequency.

10. A change of 10 phons result in__________fold change in sone.

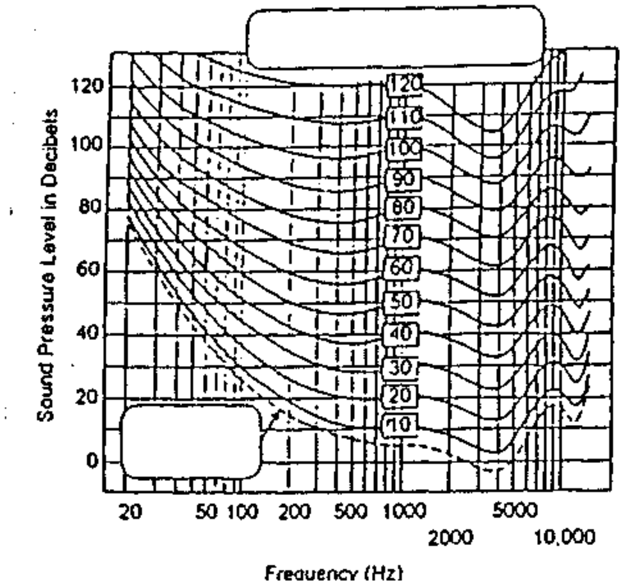
**True / False**

11. Same amount of intensity for different frequencies do not produce same loudness.
12. The loudness level of 1 KHz, at 40 dB, is equal to 60 phon.
13. Intensity required to preserve twice the loudness of 1 sone is 4 sones.
14. Loudness and loudness level are measured in the same scale
15. 50 phons are equal to 2 sones
16. Loudness is the physiological correlate of intensity.

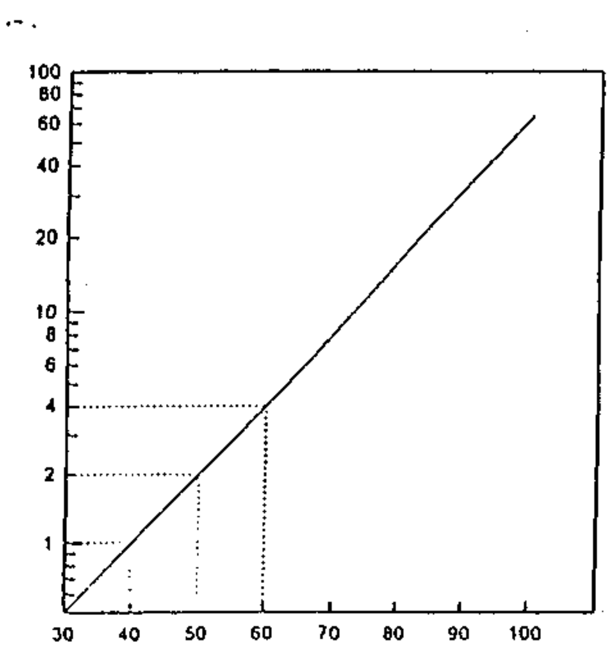
**Match the following**

| | | |
|---|---|---|
| 17. | a) Loudness | 40dB (a) |
| | b) 4 sone | phon (b) |
| | c) 1 phon | loud (c) |
| | d) Loudness level | sone (d) |
| | e) High intensity | 60 phon (e) |

18. Complete the graph




19. Level the graph



## ANSWERS

1. Loudness
2. Loud
3. Equal loudness level
4. Phon
5. Soft sounds
6. Equal loudness contour
7. Equal loudness contour
8. Low frequencies, high frequencies
9. High frequencies, low frequencies
10. 2 fold
11. True
12. False
13. True
14. False
15. False
16. False
17. a, d
    b, e
    c, a
    d, b
    e, c

18. Loudness level in phons
    Minimum audible field

19. X axis : Stimulus level in phons
    Y axis: Loudness in sones.

# PITCH

The psychoacoustical correlate of frequency is called pitch. Low pitches are associated with low frequency sounds, and sound of high frequencies are suggestive of high pitches. Pitch then is the attribute by which sounds are ordered along the frequency axis from low to high. Pitch and frequency are related, but not in a simple one-to-one manner.

**Relationship between Pitch and Frequency**

Physicists have generally used the two terms Pitch and frequency interchangeably, on the assumption that what experienced are uniquely determined by the frequency of the stimulus. Pitch is a concept determined by direct response of a human observer to a sound stimulus. Frequency, on the other hand, is determined by an observer who used the instruments of physical observation and measurements.

Pitch frequency relation can be obtained by magnitude estimation. One can also use methods in which subject adjust the frequency of a comparison tone until it subjectively sounds twice or half as high as the pitch of a test tone with a frequency set by the experimenter. This is called the method of fractionalization.

Stevens, Volkman and Newman (1937) used this method to express, pitch frequency relation. Their subjects were presented with a standard tone and were asked to adjust the frequency of a second tone until its pitch was one-half that of the standard. The result was a pitch scale in which pitch is expressed as a function of frequency (cited in Gelfand, 1997).

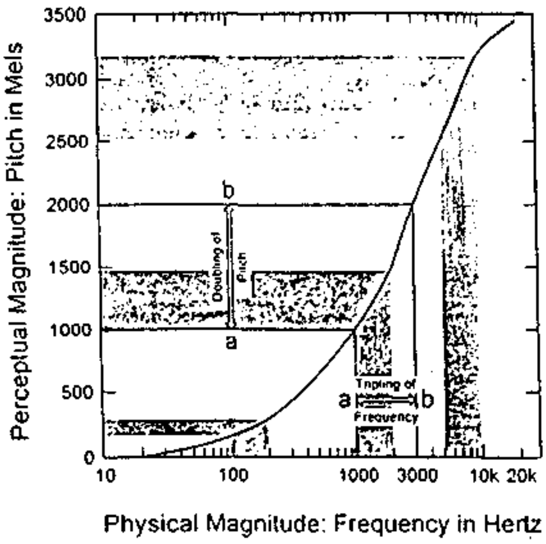



**Fig. 35: Relationship between Pitch and Frequency**

In the figure 35 frequency is shown along the abscissa and pitch on the ordinate. Pitch is expressed in units called mels. By convention, 1000 mels is the pitch of a 1000 Hz tone presented at 40 phones. This association is shown by the lines labeled 'a' in the figure. The frequency that sounds twice as high as 1000 mels has a pitch of 2000 mels, while 500 mels is half the pitch of 1000 mels, and so on. Thus in line labeled •b' in the fig. indicate that 3000 Hz is 2000 mels.

Pitch range can also be viewed in terms of intervals corresponding to critical band width or Marks'.

## Relation between Pitch and Intensity

Pitch of a Pure tone not merely depends on frequency, but also on other factors, one being intensity.

The pitch of a pure tone changes with the tone intensity. One can find that,

* For tone below 1000 Hz the pitch decreases with increasing intensity.

* For tones between 1000 - 2000 Hz the pitch rather remains constant.

* For tones above 2000 Hz the pitch tends to rise with increasing intensity.

The most widely cited observations on the effect of intensity on pitch are those of Stevens, which were based on a single listener.

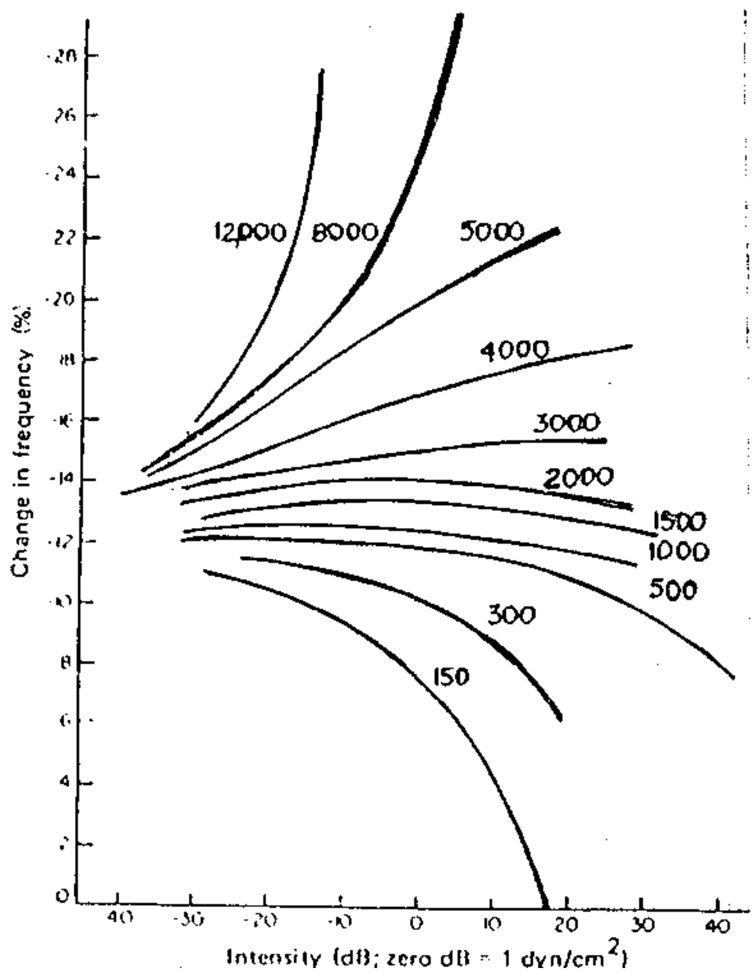



**Fig. 36: Equal pitch contours**

In general, the middle frequencies were shown to have relatively stable pitches regardless of intensity, where as the low and high frequencies had their pitches shifted progressively downward and upward respectively as a function of intensity.

## QUESTIONS

**Choose the correct answer**

1. The psychoacoustical percept of frequency

   (Loudness, intensity, pitch)

2. Low pitches are associated with

   (high frequency, mid frequency, low frequency)

3. Method in which subject adjust the frequency of a comparison tone until it sounds twice or half as high as a test tone is called (liberalization, Fractionalization, localization)

4. Pitch range can be viewed in terms of internal corresponding to critical bandwidth or

   (barks, mels, decibels)

**Fill in the blanks**

5. High frequencies are suggestive of__________pitches.
6. Pitch is expressed in units called__________.
7. 1000 mels is the pitch of a__________presented at__________-phons.
8. For tone below 1000 Hz the pitch decreases with __________ intensity.
9. Pitch rather remains constant between__________to__________ Hz.

**True / False:**

10. Pitch and frequency arc related in a one to one manner.

11. For tones above 2000 Hz pitch tends the fall with increasing intensity

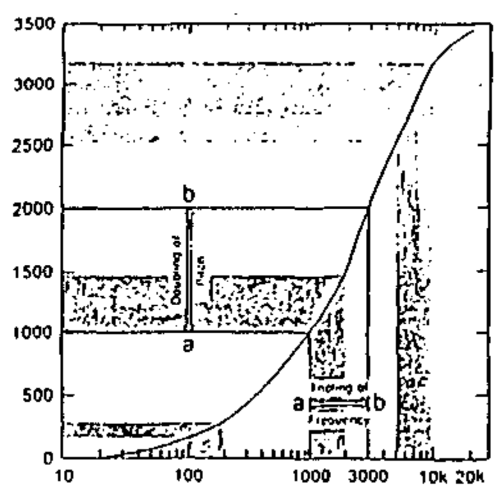
12. Pitch is an attribute by which sounds are ordered along the frequency axis from low to high.

13. Pitch frequency relation can be obtained by magnitude estimation.

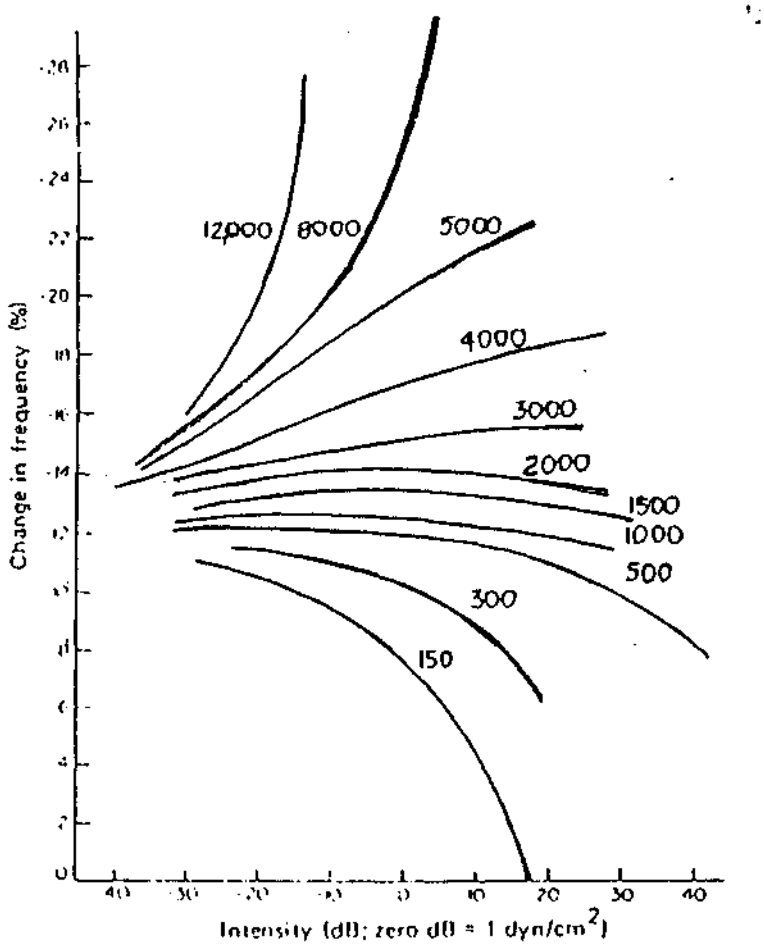
14. The frequency that sounds twice as high as 1000 mels has a pitch of 2000 mels.

**15. Identify the graph**

a) Label the X axis and Y axis of the given graph



b) Name the given graph



## ANSWERS

1. Pitch
2. Low Frequency
3. Fractionalization
4. Barks
5. High pitches
6. Mel
7. 1000 Hz tones, 40 phons
8. Increase
9. 1000 to 2000 Hz
10. False
11. False
12. True
13. True
14. True
15. a) X axis : Pitch in Mels

    Y axis : Frequency in hertz

    b) Equal pitch contour

# BINAURAL HEARING

Binaural hearing - i.e., hearing with two ears rather than one offers a number of important advantages which have obvious implications for daily living. Two most important advantages of listening binaurally rather than monaurally are localization ability, and an increased capacity to separate signal from noise (selective listening).

## Localization of sound

The source of sound can be localized in the three spatial dimensions; the horizontal plane or left-right dimension, the vertical plane or the up-down dimension, and in distance or the near far dimensions. Sound localization is therefore the result of auditory systems ability to process the physical parameters of sound (frequency, level, time/phase) that correlate with the spatial location of the sound source.

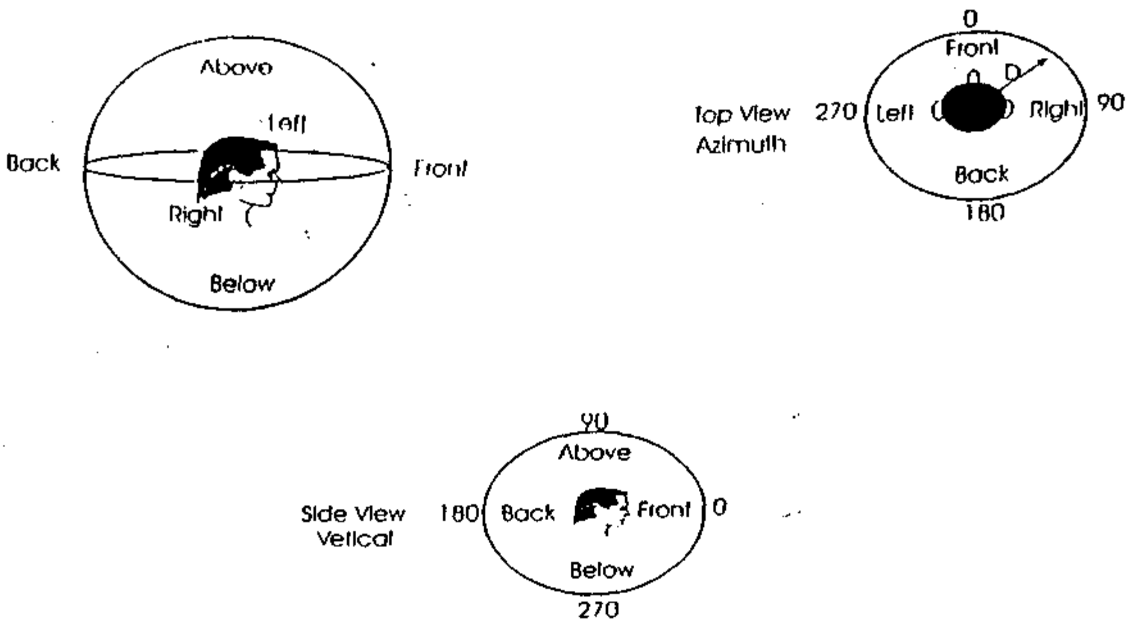



**Fig. 37: The three spatial dimension : azimuth (left-right), vertical (upand down) and distance (near and far)**

The ear makes use of two aspects of the acoustic signal in judging its origin. The comparative time of the arrival at the two ears and comparative intensity of the signal at the two ears, i.e time difference between the two ears and the intensity difference of the signal reaching to two ears.

**Intensity cues for localization**

If the origin of the sound is neither directly in front of nor behind the head, one ear must be closer to the sound source than the other. Sound dissipates as it travels through a medium because of friction between the molecules. Therefore the near ear will always get a slightly louder sound than the far ear. This small loss of intensity will occur regardless of frequency.

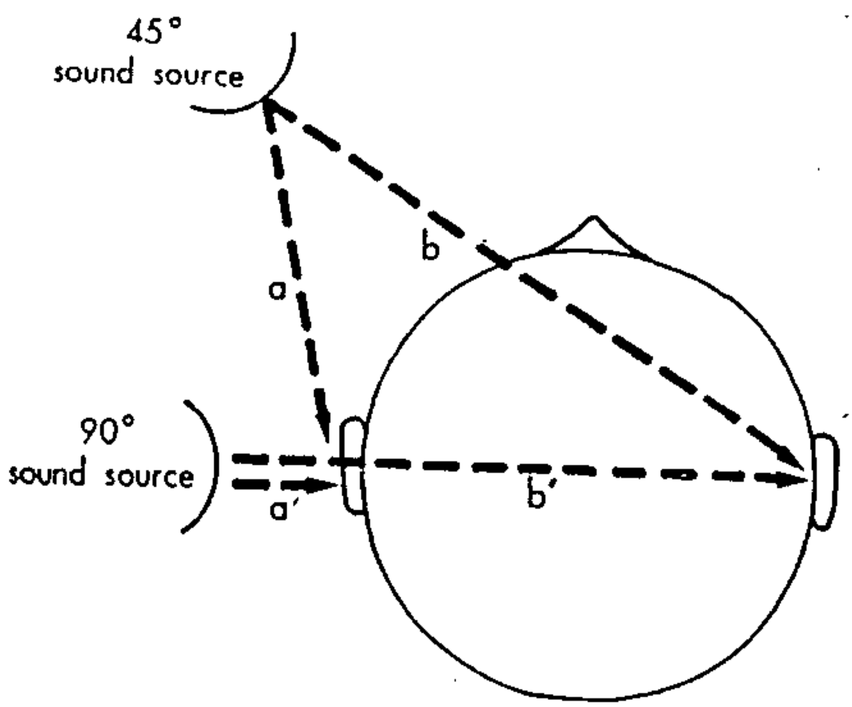



**Fig. 38: A comparison of sound reaching the near ear and far ear**

A second factor that affects the comparative intensity between the two ears is the effect of head itself on the sound, which is referred as the head shadow effect.

Low frequency tones have long wavelengths, where as high frequencies have very short wavelengths. On of the important acoustic characteristics of tone with long wavelengths (low frequencies) is that they bend around corners easily while traveling through a medium in opposition to short wavelengths (high frequencies) which are very directional and tend to reflect away from the head rather than around it.

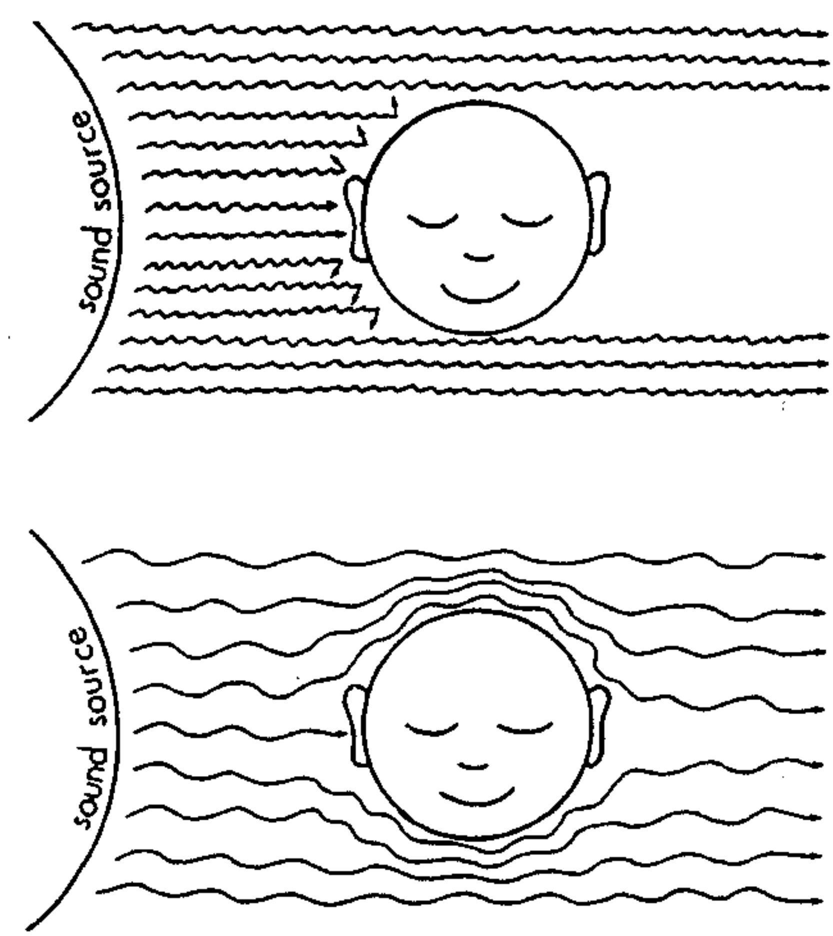



**Fig. 39: Reflection of high and low frequency tone by head**

The head shadow effect is greatest when the sound comes from the side of the head (90° azimuth) but it is also significant when the sound arrives at smaller angles to the head.

| | azimuth 30° | azimuth 60° | azimuth 90° |
|---|---|---|---|
| 500 Hz | 3 dB | 5 dB | 5 dB |
| 5000 Hz | 12 dB | 18 dB | 20 dB |

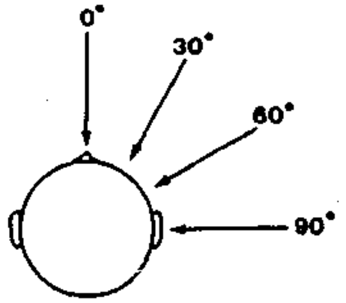



**Fig. 40: Attenuation of sound intensity when the sound wave is from different azimuth**

The head shadow effect is much more important in causing an interaural intensity differences than the relatively small loss of intensity that occurs because of the distance between the near and far ear.

The central auditory system interprets an interaural intensity difference as sound coming from the side getting the greater intensity. Because the head shadow occurs primarily at higher frequencies, the interaural intensity difference cue is larger enough to allow the localization of tone of about 2000 Hz and higher.

## Time cues for localization

Since interaural intensity differences can only account for localization of higher frequencies, there must be an alternative cue that accounts for the localization of the lower frequencies.

If we look back to the first figure in this chapter, it can be seen that the distance that sound has to travel is greater to the far ear (b) than the near ear (a). Although this is true for both the 45° and 90° situations, the difference is somewhat greater for the 90° angle. A tone that is directed at a 45° angle to the head will arrive at the near ear approximately 0.4 milliseconds sooner than the far ear; the difference will be about 0.65 milliseconds if the azimuth is 90°. The literature shows that interaural differences as small as 10 micro seconds (million of a second) have produced lateralization effects. Therefore a much larger interural time differences up the 0.65 milliseconds are more than adequate to cue the central auditory system for the localization of low - frequency sounds.

**Lateralization of sound**

A similar phenomenon that can be studied in laboratory setup is lateralization. If we put earphones on a subject and present an identical puretone to both ears, the subject will say that the tone is coming from some where in the middle of the head. We refer this as lateralization of the tone to midline. If we increase the intensity a few decibels in either ear, the sound will clearly be lateralized to that side, confirming the interpretation of interaural intensity differences by the auditory system.

The image formed from binaural presentations over headphones is sometimes referred to as a fused image because the listener reports hearing one image as if the sound source arriving at both ears were fused. A listener will not perceive a fused image if the interaural time difference and interaural frequency difference is too large. If the interaural time difference is large than approximately 2 m sec, the listener will report hearing two images, one at each ear. Also if the two ears receive independent signals differing greatly in frequency, the listener perceives two images one at each ear, and both frequencies can be identified.

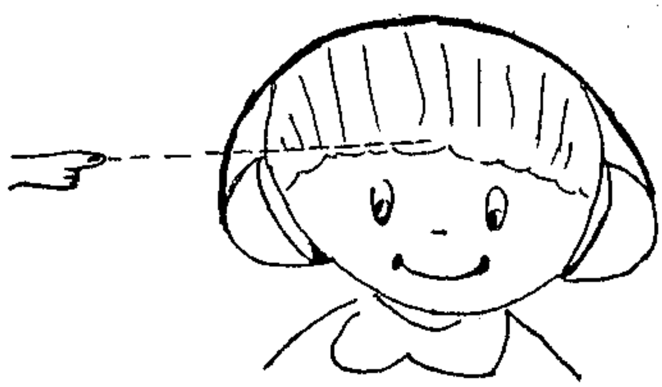


**Fig. 41 : Lateralization of sound towards midline**

As the interaural phase difference increases towards 180°, the fused image is located closer to the ear that received the tone first (leading in time). As the interaural phase difference exceed 180°, the image was located on the other side of the head (toward the ear lagging in time), and as the interaural time difference approach 360° the image is located toward the middle of the head. A tone presented with no interaural difference was perceived toward the middle of the head or at midline.

## QUESTIONS

**Choose the correct one and fill the blanks**

1. Hearing with two ears rather than one is——————hearing. (Monoural, Binaural, Circumaural)

2. The head shadow effect is greatest when the sound is at a — azimuth

   (45°, 90°, 180°)

3. Interaural intensity difference cue is larger enough to- allow the localization of tone above ------------.

   (20 Hz, 200 Hz, 2000Hz).

4. Interaural time difference is adequate for the localization of —————frequency sound.

   (low, mid, high)

5. When ITD reaches 360° the image is located toward,--------------,

   (left, right, middle)

**Fill in the blanks**

6. Two important aspects of binaural listening are ———————— and----------------.

7. The source of sound can be localized in three spatial dimensions, which are---------------------------------------and. . . . . . . . . . . . . . . . . . dimensions.

8. A similar phenomenon as localization studied in laboratory setup is

9. The image formed from binaural presentations over headphones is sometimes referred to as————————.

10. At————————degree the fused image is located closer to the ear that received the tone first.

**True /False**

11. Sound dissipates as it travels through a medium due to friction.

12. The near ear will get slightly softer sound than far ear.

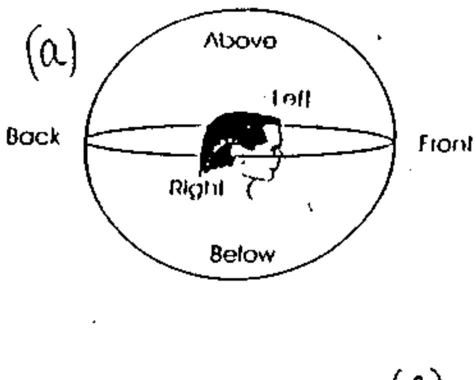
13. Low frequency tones have short wave length.

14. One of the important acoustic characteristics of tone with long wavelengths is that they bend around the corners.

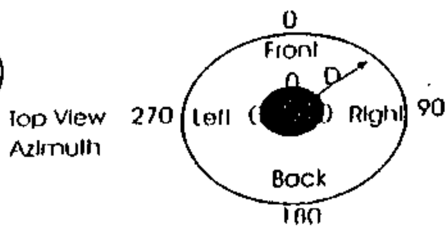
15. A listener will not perceive a fused image if the interaural time difference is too large.

**16. Identify the condition**

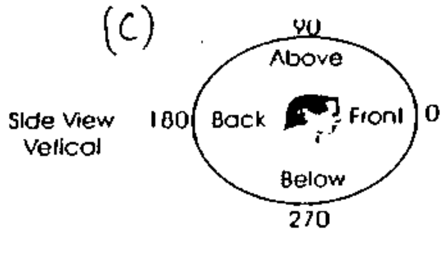
(a)



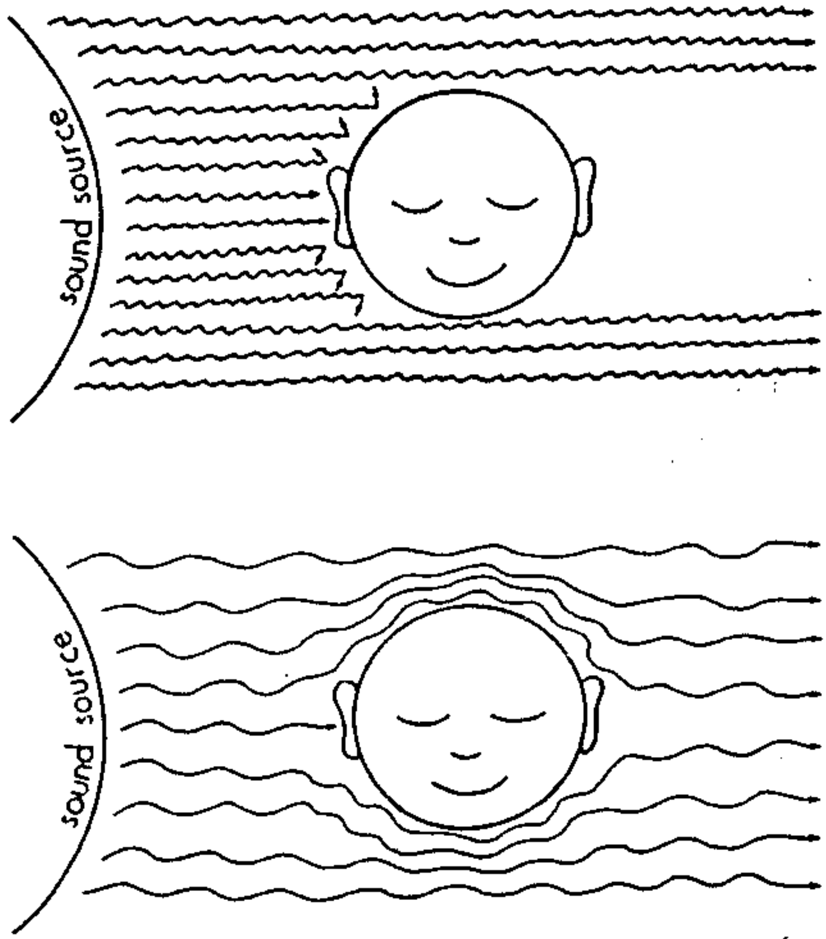

(b)




(c)



[d]



## ANSWERS

1. Binaural
2. 90°
3. 2000 Hz
4. Low
5. Middle
6. Localization ability and selective listening.
7. Horizontal phone, vertical plane, near or far dimensions.
8. Lateralization
9. Fused
10. 180°
11. True
12. False
13. False
14. True
15. True.
16. a) Left-right

    b) Up - down

    c) Near - far.

    d) Reflection of high and low frequency tone by head.

# REVERBERATION

Reverberation is a short term echo or the continuation of a sound in a closed area after the source has stopped vibrating. This results from the reflection and refraction of sound waves.

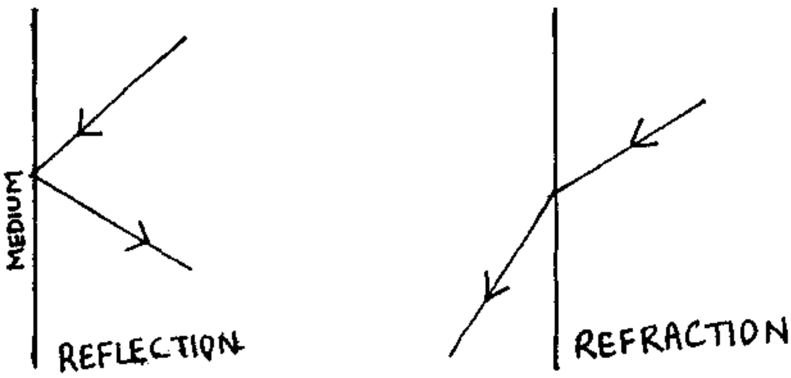



**Fig. 42: Reflection and refraction of sound stimuli**

Reflected waves are often perceived as echoes, because the incident and reflected waves arrives at the ear at different times. This, in turn, occurs because time is required for the sound wave to travel to the reflecting surface and back again. In effect, two (or more) sounds are heard. The echo is an exaggerated form of reverberation. A sound field in which multiple reflected waves are present is called reverberant field. In an ideal reverberant room, a single impulsive sound (i.e a gunshot) would bounce" around indefinitely, but in real life situations, the sound energy of the reflected waves is ultimately absorbed and/or transmitted.

In a room with hard surfaces, direct and reflected sound transmission causes the build up and decay of reverberant or semi reverberant sound fields. It consists of many, very closely packed or diffuse reflections that decay exponentially over time when a sound source ceases.

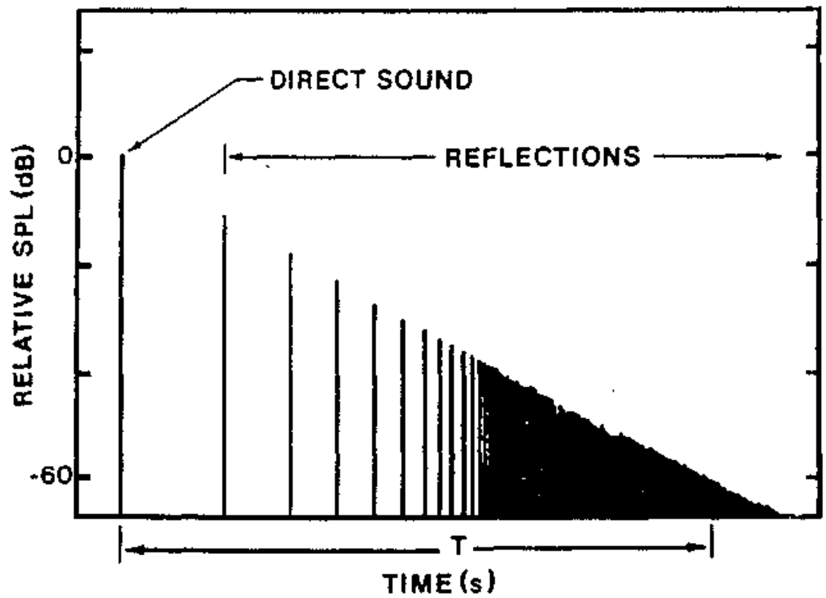



**Fig. 43: Decay in reflected sound over time**

In the above figure direct sound arrives with the greatest sound pressure, the early reflections with less sound pressure, and the reverberant reflections with steadily decreasing pressure. The early reflections are spread closer and closer together until they become a reverberant field.

The time taken for the sound to decay from a certain intensity level to a sound pressure 60 dB lower is called the reverberation time (RT). This is measured in terms of the amount of time required for the sound pressure to decay a certain level established by convention, typically one - thousand of its initial value.

Reverberation time depends upon the volume of the room and the absorbent surface. The less absorption, the longer the time of decay and greater the reverberation time. Reverberation time can be predicted by the simple relationship

$$T \; \alpha V/a$$

Where 'T' is the reverberation time, 'V' is the volume of the enclosure, and 'a' is the total absorption.

Reverberation times of rooms vary from 0 to 10 or more seconds. An anechoic room has a RT close to zero. A reverberation chamber may have a 10 second RT. The RT of a room depends primary on its absorption characteristics. An anechoic room, for example, may be completely absorptive, with no sound reflection.

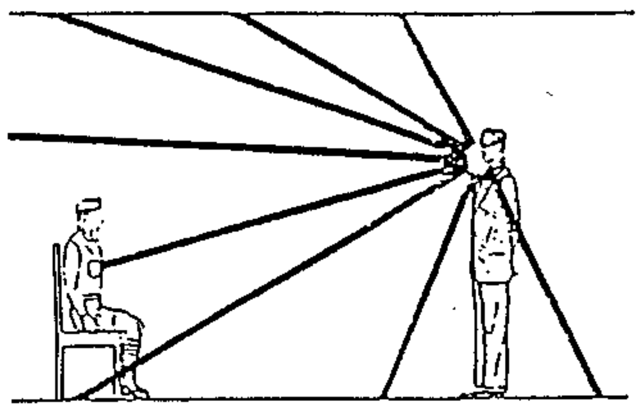


**Fig. 44: An anechoic room situation**

A reverberation chamber, in contrast, is completely lined with hard surface materials. It is so reflective that a diffuse sound field builds up, and when the sound ceases, it takes a long time for the reverberation field to decay. The reverberation chamber and the anechoic rooms represent the extremes of RT, one a completely live room' and other a completely 'dead room' (Everest 1989, cited in Berg).

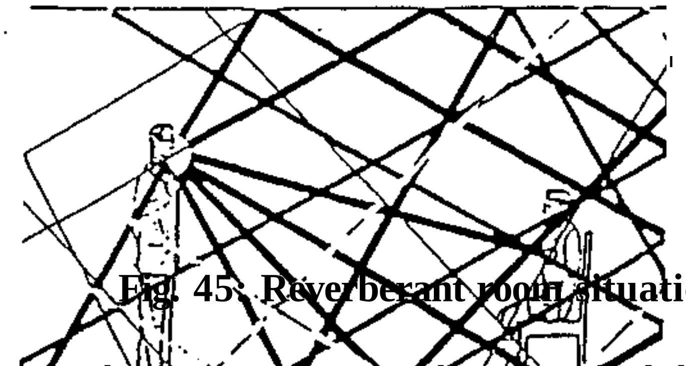


**Fig. 45: Reverberant room situation**

The reverberation time may be controlled by the appropriate choice of building material and architectural design. Below given are certain suggestions for the reduction of reverberation.

- Cover hard reflective surface with sound absorbing materials (cork, acoustic tiles, carpet etc).
- Cover the floor with carpets.
- Strategic placement of cork bulletin boards.
- Attach empty egg cartons to the walls
- Unpainted concrete walls
- Use of thick heavy curtains and bamboo screens.

## QUESTIONS

**Choose the correct answer**

1. Continuation of sound in a closed area

   (Reflection, reverberation, refraction)

2. An anechoic room have RT close to

   (zero, five, ten)

3. In a quiet room RT could be the time for 90 dB sound to decrease to

   (30 dB, 80 dB, 50 dB)

4. In the formula T a V/a; "a' is

   (Total absorption, partial absorption, minimal absorption)

**Fill in the blanks**

5. Reflected waves are often perceived as..............
6. A sound field in which multiple reflected waves are present is called
7. The time taken for the sound to decay from a certain intensity is
8. Reverberation chamber is a.................room, where as an anachoic chamber is a—————room.
9. The RT of a room primary depends on its—————.
10. In RT the sound decrease by—————of its initial values.

**State whether True /False**

11. Reverberation time depends upon the volume of the room and the absorbent surface.

12. Less the absorption greater the time of decay.

13. Greater the reverberation less is the absorption.

14. Longer the time for decay, greater the reverberation time.

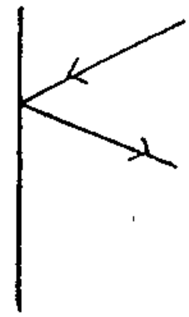
15. No diffuse field are found in reverberation time.

16. Word Grids

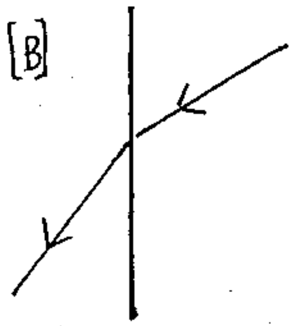
| | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| R | E | V | E | R | B | E | R | A | T | I | 0 | N |
| A | C | D | N | C | I | P | 0 | Z | E | Y | A | S |
| G | H | I | P | A | N | E | C | H | 0 | I | C | 0 |
| E | 0 | F | J | U | H | S | D | W | X | Q | I | M |
| S | B | F | T | N | A | 0 | Q | D | R | F | A | A |
| 0 | P | U | B | 0 | C | u | S | H | W | E | L | S |
| u | M | S | I | E | J | N | U | I | L | T | I | U |
| N | U | E | D | E | A | D | R | 0 | 0 | M | V | N |
| D | T | F | L | V | 0 | F | M | R | M | G | E | D |
| W | H | I | N | F | K | I | 0 | L | 0 | V | R | A |
| A | U | E | H | K | G | E | R | Y | R | S | 0 | R |
| V | M | L | A | N | I | L | V | X | K | A | 0 | A |
| E | A | D | P | T | U | D | S | O | A | H | M | N |

17. Name the condition

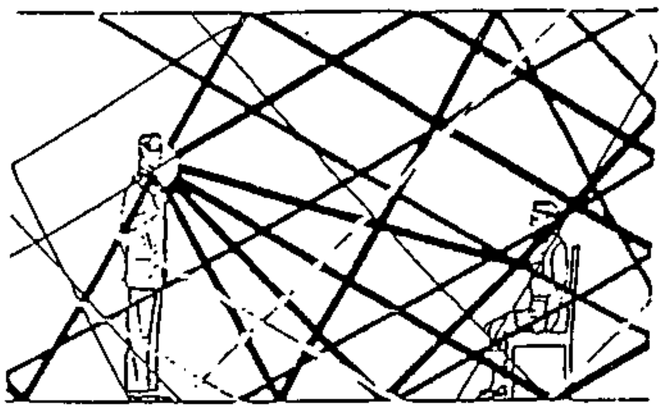
[A]



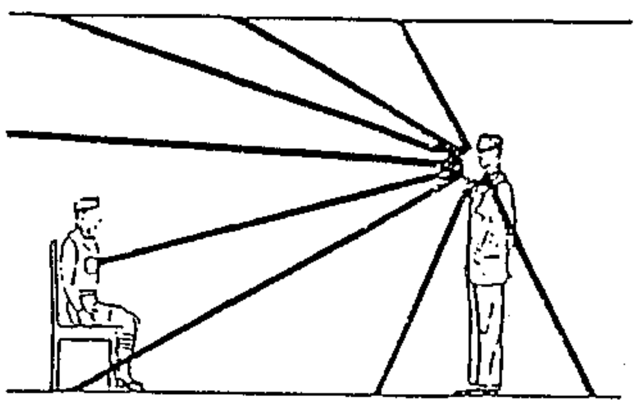
[B]



[C]



[D]



## ANSWERS

1. Reverberation
2. Zero
3. 30 dB
4. Total absorption
5. Echo
6. Reverberant field
7. Reverberation time
8. Live room and dead room
9. Absorption characteristics.
10. 1000
11. True
12. False
13. True
14. True
15. False

16.

| | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| **R** | E | **V** | E | **R** | B | E | R | A | T | I | **O** | N |
| **A** | **C** | **D** | **N** | C | **I** | **P** | 0 | **Z** | E | **Y** | **A** | S |
| **G** | **H** | **I** | **P** | **A** | N | **E** | **C** | H | **O** | **I** | **C** | 0 |
| **E** | **O** | **F** | J | **U** | H | **S** | **D** | **W** | X | **Q** | **I** | **M** |
| **S** | B | **F** | T | **N** | **A** | **O** | Q | D | R | **F** | **A** | **A** |
| **O** | **P** | **U** | **B** | **O** | **C** | **U** | **S** | **H** | **W** | **E** | **L** | **S** |
| **U** | M | **S** | I | E | J | **N** | U | **I** | L | T | I | U |
| **N** | U | **E** | D | E | A | **D** | R | **O** | **O** | **M** | V | N |
| **D** | T | **F** | L | V | 0 | **F** | M | R | M | G | E | D |
| **W** | H | **I** | N | F | K | **I** | 0 | L | 0 | **V** | R | **A** |
| **A** | U | **E** | H | K | G | **E** | R | Y | R | S | **O** | R |
| **V** | **M** | **L** | **A** | **N** | I | **L** | V | X | K | A | **O** | A |
| **E** | **A** | **D** | **P** | T | U | **D** | S | O | **A** | **H** | **M** | **N** |

17. a) Reflection

b) Refraction

c) Anechoic Room

d) Reverberant room

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