Acoustics

POOJA PREMNATH
S01
2110152

Acoustics : study of sound, i.e the
(i) generation
(ii) propagation
(iii) reception

- Generation : vibration of the sounding body
- Propagation : refers to the transmission of sound energy through the medium

  (a) In gas & liquids, transmission happens in the form of longitudinal waves.

  (b) In solids, propagation is in the form of transverse & longitudinal waves.

  This happens because of disturbances in an elastic medium, i.e mechanical deformations.

  ⇒ elasticity & density are imp. factors affecting the propagation of sound.
- Reception : relates to the human ear receiving the transmitted sound. (depends on the sensitivity of the human ear).

Classification of Sound (Based on Frequency)

< 20 Hz : infrasonics
20 Hz - 20kHz : audible
> 20 kHz : ultrasonics

Characteristics of Sound:

(a) **Pitch**: (physiological quantity)
high pitch: buzzing bee
low pitch : roaring lion

(b) **Frequency**: (a measurable quantity)
no. of vibrations per second
unit : Hertz

(c) Intensity: $I = 2\pi^2 a^2 \nu^2 \rho v$

a = amplitude
ν = frequency
ρ = density of medium
v = velocity of the wave.

It is also defined as the rate of flow energy per unit area.

i.e $I = \frac{\text{Power}}{\text{Area}}$

unit: W/m^2

The Human ear can detect intensities from $10^{-12}\,W/m^2$ to $1\,W/m^2$.

(d) Loudness & Perception

Loudness is a physiological quantity.
depends on the listener (perception)

For a large no. of people, change in perception is directly proportional to change in stimulus and inversely proportional to stimulus.

i.e $dP \propto dS$

and $dP \propto \frac{1}{S}$

$$dP = k\frac{dS}{S}$$

$$\int dP = k\int \frac{dS}{S}$$

$$P = k\ln S + C$$

when $P = 0$, for a particular stimulus S_0,

then $0 = k\ln S_0 + C$

$C = -k\ln S_0$

$\Rightarrow P = k\ln S - k\ln S_0$

$P = k\ln\left(\frac{S}{S_0}\right)$ or $P = K\log\left(\frac{S}{S_0}\right)$ where $K = 2.303k$

111ly $L = K \log\left(\frac{I}{I_0}\right)$

where loudness is the perception & intensity the stimulus

$$\boxed{L = K \log\left(\frac{I}{I_0}\right) \text{ is called the Weber - Fechner Law.}}$$

I_0 is the intensity when $L = 0$

↓

called standard intensity / threshold intensity

↓

has a value of $10^{-12}\ W/m^2$

In general $\boxed{L \propto \log I}$, i.e the response of the human ear towards sound is logarithmic in nature.

* Relative Intensity : $\frac{I}{I_0}$

however since intensity varies from $10^{-12}\ W/m^2$ to $1W/m^2$ a logarithmic scale is chosen.

* Intensity Level of Sound : Logarithmic ratio of the intensity of sound to the standard intensity

* Decibel and Bel

$\frac{I}{I_0} = 10^x \Rightarrow$ intensity is x bels.

$$\log\left(\frac{I}{I_0}\right) = \log 10^x$$
$$= x \log 10$$
$$= \underline{x \text{ bels}}$$

$1\ dB = 1/10$ bel

$\therefore$ Intensity : $\log\left(\frac{I}{I_0}\right)$ bels $= \boxed{10 \log\left(\frac{I}{I_0}\right) dB}$

* **Meaning of 1dB**

$$1dB = 10\log\left(\frac{I}{I_0}\right)$$

$$\log\left(\frac{I}{I_0}\right) = \frac{1}{10} = 0.1$$

$$\frac{I}{I_0} = 10^{0.1}$$

$$= 1.26$$

$$\boxed{I = 1.26 I_0}$$

⇒ if the intensity increases by 26%, the intensity is 1dB.

Sound Pressure Level

Sound measuring devices respond to the pressure exerted by sound. ⇒ Intensity is related to pressure.

$$I \propto P^2$$

$$\Rightarrow \frac{I}{I_0} = \left(\frac{P}{P_0}\right)^2$$

Relation between ~~intensity~~ & pressure & dB

$$1\,dB = 10\log\left(\frac{I}{I_0}\right)$$

$$= 10\log\left(\frac{P}{P_0}\right)^2$$

$$\boxed{1dB = 20\log\left(\frac{P}{P_0}\right)}$$

where P_0 = Zero level pressure

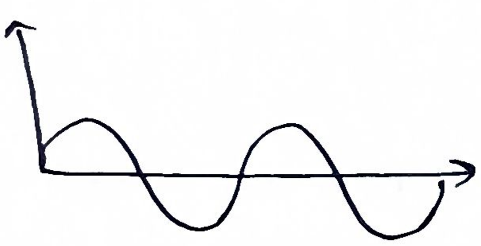
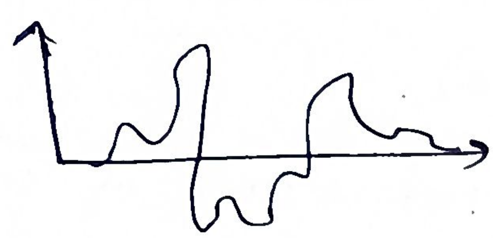
$= 2 \times 10^{-5}\ N/m^2$

Quality : a physiological quantity.

- The property which enables us to distinguish between 2 diff voices, 2 diff instruments etc
- This is because the sounds differ in harmonics & their combinations are diff. from the fundamental frequency.

Timbre : diff. waveforms as a result of diff. combinations of harmonics.

Music vs. Noise

Music	Noise
(i) produces a pleasing sensation to the ears	(i) produces an unpleasant sensation to the ears.
(ii) regularity in the shape of the curves, periodicity	(ii) irregularity in the shape of the curves, no periodicity
(iii) no sudden changes in intensity	(iii) sudden changes in intensity
	

Architectural Acoustics

Conditions for an acoustically good hall

(i) sound should be sufficiently loud in every part of the hall
(ii) quality of speech | sound should not change
(iii) no echoes
(iv) no resonance
(v) no areas where sound is concentrated
(vi) no dead spots
(vii) avoid 'echelon effect' due to flights of stairs that are equally spaced.

Analysis of Sound in a Bound Room

- Sound travels without any appreciable loss until it reaches a boundary.

Thereafter, it may undergo:
(i) Reflection
(ii) Absorption
(iii) Transmission

The direct sound varies by $1/r^2$, where r is the distance between the source and the receiver.

The indirect sound depends on:
(i) Geometry of the room
(ii) Size of the room
(iii) char. of Materials in the room.

Reverberation: • sound produced in a room lasts longer than its production rate and thus reaches the listener a number of times, once directly, & thereafter, after reflection

- The listener receives a series of sounds of diminishing intensity and they are almost indistinguishable (time between them being ~0.6 - ~1 sec).

The prolongation / persistance of audible sound even after the source has stopped emitting sound is called reverberation.

Reverberation Time: The duration for which sound persists = reverberation time. It can be defined as:

1. Time taken for the sound to fall below audible level from the instant the source has stopped sounding.

2. Time taken by sound to become 10^{-6} th the value at the time of cutoff.

3. Time taken for the sound intensity to decrease by 60dB.

initial intensity = I_i

final intensity = I_f $\qquad I_f = 10^{-6} I_i$

$$dB_f - dB_i = 10 \log\left(\frac{I_f}{I_i}\right)$$

$$= 10 \log 10^{-6}$$

$$= -60 \, dB$$

Absorption : conversion of acoustic energy into heat.

The absorption characteristics of a material are best defined by its absorption coefficient.

It is defined as:

(i) The ratio of the sound energy absorbed / unit area / unit time to the sound energy incident / unit area / unit time.

(ii) It is also defined as the reciprocal of the area of the material which absorbs the same amount of sound energy as 1m² of an open window.

units of absorption coefficient = open window units (O.W.U)
or Sabins

An open window is considered to be a perfect absorber as it allows all the sound energy incident upon it to pass through.

$$\therefore a_{ow} = \frac{E_1 / A_{ow}}{E_1 / A_{ow}} = 1$$

E_1 is the sound energy that is incident & absorbed.

$$\boxed{a_{ow} = 1}$$

$\therefore$ Total absorption of a material = $a_m \times A_m$
A_m → area.

unit = OWU; m2
or sabine m2

a_m = absorption coefficient of material

A_m is the area which absorbs the same energy as 1m² of an open window.

$a_m \cdot A_m = a_{ow} \times 1$

$a_m = \frac{a_{ow}}{A_m}$

$a_m = \frac{1}{A_m}$

Total absorption of a hall :

$A = \sum_i a_i s_i$

a_i = absorption coeff of material i

s_i = area of material i.

** Growth and Decay of Sound Energy Density in a Room.

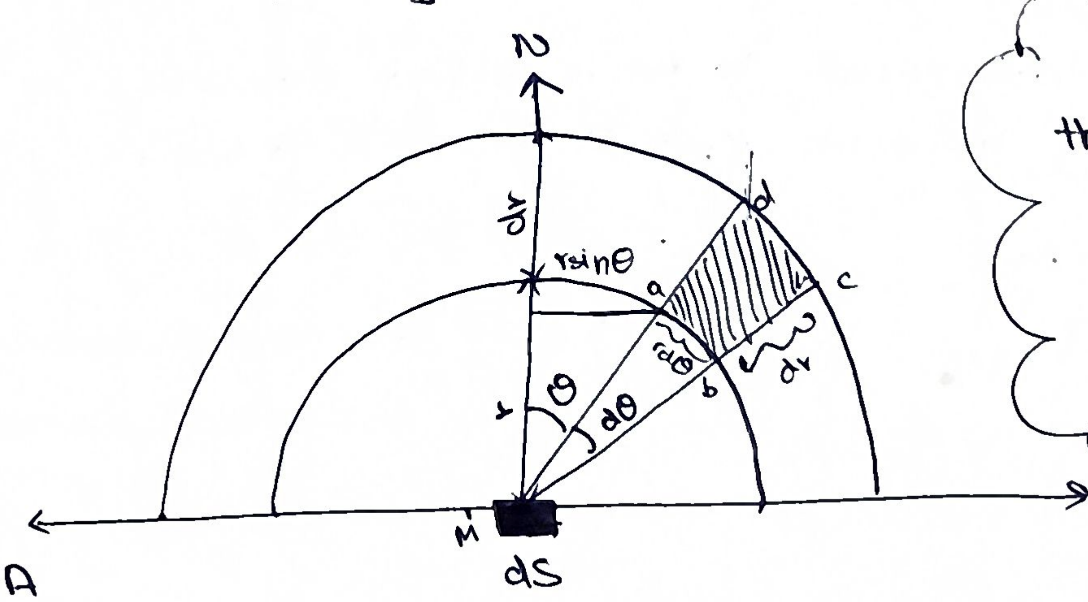
Assumptions: (i) sound energy is uniformly distributed in the room.
(ii) Frequency dependence of the absorption of sound energy is ignored.

Step 1: • To calculate the rate at which sound energy is incident upon the walls, and thereby the rate at which it is absorbed.

• Consider a plane wall AB, and a small element ds on it.

• We aim to make a volume element dV surrounding ds that contains some sound energy. This is used to calculate the amount of sound received by ds from dV in one second.

• This can be used to calculate all the sound energy from a larger volume falling on ds in a single second.



In short:
there is a small volume element dV
aim 1: To find sound energy in dV
aim 2: To find how much sound energy from dV reaches ds in unit time.

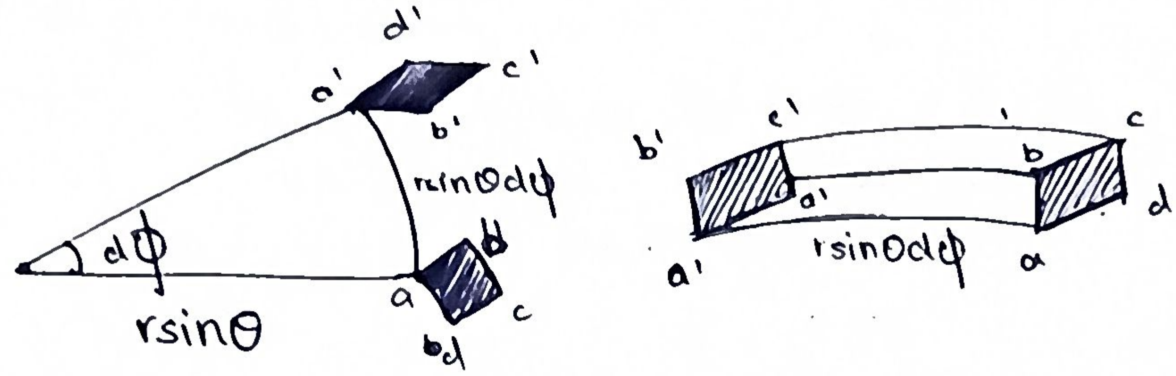
Construction: From the centre of ds, construct 2 circles of radii r and $r+dr$ such that they lie on the plane containing the normal to the surface ds.

arc length of shaded element $= r\,d\theta$

thickness $= dr$

∴ Area of the shaded region = $r d\theta \times dr$

Rotate the areal element 'abcd' through a small angle $d\phi$ about the normal MN.



circumferential distance travelled by the areal element $= r\sin\theta d\phi$

∴ Volume traced out by the areal element = area × circumferential distance

$$= r d\theta \times dr \times r\sin\theta d\phi$$

$$\boxed{dV = r^2 \sin\theta \, d\theta \, dr \, d\phi}$$

Energy density : avg. sound energy present per unit volume in the room.

energy density within the volume element $= E dV$

$$\boxed{E dV = E r^2 \sin\theta d\theta dr d\phi}$$

The sound energy present in dV travels in all directions in the room, i.e it spreads spherically.

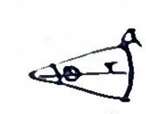
The solid angle contained by a sphere is 4π.

Energy density in dV per unit solid angle

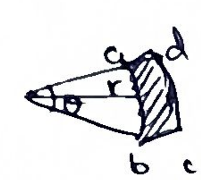
$$= \frac{E dV}{4\pi}$$

$\left(\theta = \frac{A}{r^2}\right)$ area of a sphere $= 4\pi r^2$ $\Rightarrow \theta = \frac{4\pi r^2}{r^2} = 4\pi$

plane angles & solid angles



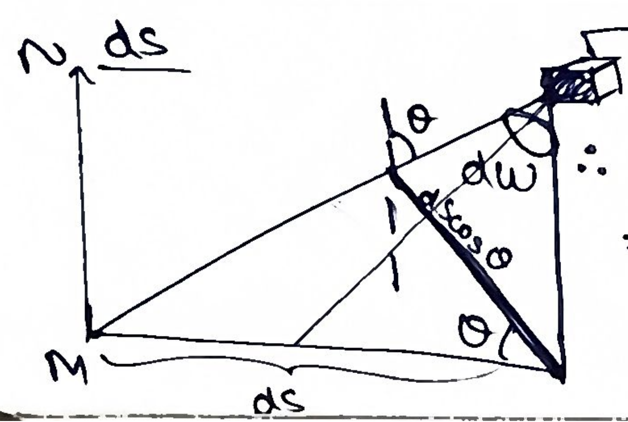
ab is an arc
radius = r
$\theta = \frac{ab}{r} = \frac{l}{r}$
θ = plane angle



abcd is a surface
angle subtended by this surface = solid angle

$$\boxed{\theta = \frac{A}{r^2}}$$

To find the amount of sound energy reaching ds



Volume element dV

∴ solid angle $= \frac{ds\cos\theta}{r^2}$

solid angle takes ⊥r length, since the distance is not ⊥r, consider a projection such that the ⊥r length can be taken.

$\therefore$ sound energy reaching $dS = \frac{EdV}{4\pi} \times$ solid angle

$$= \frac{EdV}{4\pi} \times \frac{ds\cos\theta}{r^2}$$

$$= \frac{E r^2 \sin\theta \, d\theta \, dr \, d\phi}{4\pi} \cdot \frac{ds\cos\theta}{r^2}$$

$$= \frac{E \cancel{r^2} dr \sin\theta \cos\theta \, d\theta \, d\phi}{4\pi \; \cancel{r^2}} \times ds$$

$$= \frac{E}{4\pi} dr \sin\theta \cos\theta \, d\theta \; ds \, d\phi$$

$$= \frac{E ds \, dr \sin 2\theta \, d\theta \, d\phi}{8\pi}$$

To find the sound energy reaching ds in 1sec :

integration limits : $r : 0 \to C_s$ (C_s = speed of sound, aka distance travelled by sound in 1sec)

$\theta : 0 \to \pi/2$

$\phi : 0 \to 2\pi$ (to get a hemisphere)

Thus, the sound energy from dV reaching ds in 1second is given by

$$= \frac{Eds}{8\pi} \int_0^{C_s} dr \int_0^{\pi/2} \sin 2\theta \, d\theta \int_0^{2\pi} d\phi$$

$$= \frac{Eds}{8\pi} \times C_s \times 1 \times 2\pi$$

$$= \frac{E C_s ds}{4} = \text{rate of incidence of energy on } ds$$

absorption coefficient 'a' = $\frac{\text{sound energy absorbed}}{\text{sound energy incident}}$

Sound energy absorbed $= a \times \frac{E C_s ds}{4}$

or $\frac{E C_s}{4} \times \sum_i a_i s_i$

$= \frac{E C_s A}{4}$ // where $A = \sum_i a_i s_i$

Expression for the Growth of Sound Energy in the room

Let P = power output (rate of emission of sound energy from the source)

V = volume of the room

Total energy density $= EV$

rate of growth of energy $= \frac{d(EV)}{dt} = \frac{V dE}{dt}$

rate of growth of energy = rate of supply of energy from the source − rate of absorption by all the surfaces

$\Rightarrow \frac{V dE}{dt} = P - \frac{E C_s A}{4}$ — (1)

Case where steady state is attained

$E = E_m$ & $\frac{dE}{dt} = 0$

$\therefore\ 0 = P - \frac{E_m C_s A}{4}$

$\frac{E_m C_s A}{4} = P$

or $\boxed{E_m = \frac{4P}{C_s A}}$

Considering eqn 1

$$V\frac{dE}{dt} = P - \frac{ECsA}{4}$$

$$\frac{dE}{dt} = \frac{P}{V} - \frac{CsA}{4V} . E$$

let $\frac{CsA}{4V} = \alpha$

$$\frac{dE}{dt} = \frac{P}{V} \cdot \left(\frac{CsA}{4}\right)\left(\frac{4}{CsA}\right) - \alpha E$$

$$\frac{dE}{dt} = P\alpha \cdot \left(\frac{4}{CsA}\right) - \alpha E$$

$$\frac{dE}{dt} = \frac{4P}{CsA} \cdot \alpha - \alpha E$$

$$\frac{dE}{dt} + \alpha E = \left(\frac{4P}{CsA}\right) \cdot \alpha$$

multiplying both sides by $e^{\alpha t}$

$$\frac{dE}{dt} . e^{\alpha t} + \alpha E . e^{\alpha t} = \frac{4P}{CsA} . \alpha . e^{\alpha t}$$

$$\frac{d}{dt}(E e^{\alpha t}) = \frac{4P}{CsA} . \alpha . e^{\alpha t}$$

integrating on both sides

$$\int \frac{d}{dt}(E e^{\alpha t}) = \int \frac{4P}{CsA} . \alpha . e^{\alpha t}$$

$$E e^{\alpha t} = \frac{4P}{CsA} . \frac{e^{\alpha t}}{\cancel{\alpha}} . \cancel{\alpha} + K$$

$$\boxed{E e^{\alpha t} = \left(\frac{4P}{CsA}\right) \cdot e^{\alpha t} + K}$$

constant K can be found from boundary conditions at $t = 0$

$E = 0$

$\Rightarrow 0 \cdot e^{\alpha(0)} = \left(\frac{4P}{CsA}\right) \cdot e^{\alpha(0)} + K$

$K = \frac{-4P}{CsA}$

$\therefore E e^{\alpha t} = \left(\frac{4P}{CsA}\right) \cdot e^{\alpha t} - \left(\frac{4P}{CsA}\right)$

$\frac{4P}{CsA} = E_m$

$\therefore \boxed{E e^{\alpha t} = E_m(1 - e^{-\alpha t})} \rightarrow (2)$ (at $t = 0$, $t = \infty$ $E = E_m$)

→ Expression for the growth of sound energy

graphically,

Decay of sound energy in the room

when $t = 0$, $P = 0$

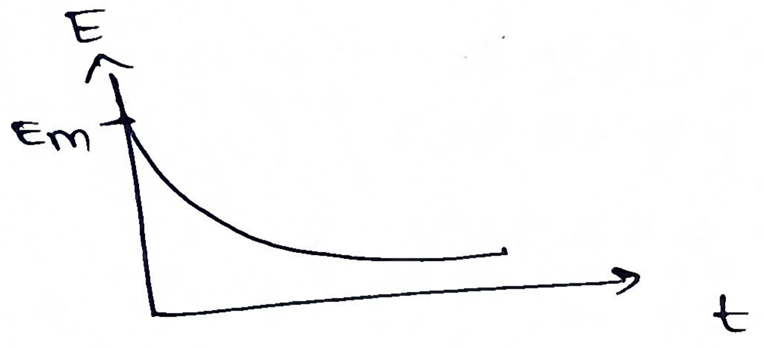
$E = E_m$

~~assed~~ sub in eqn (1)

$\cancel{E_m e^{0} = E_m(1 - e^{0})}$

$E_m e^{0} = \left(\frac{4P}{CsA}\right) \cdot e^{0} + K$

$P = 0$



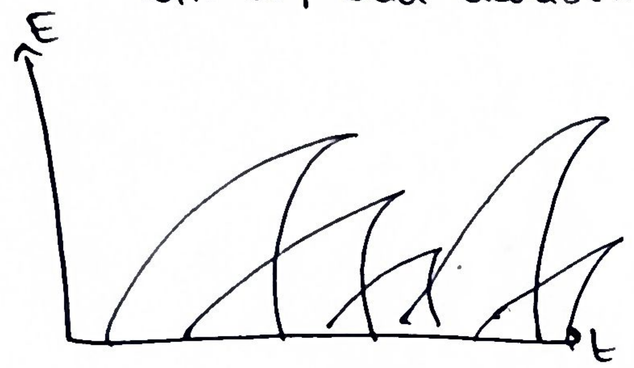
$\boxed{E_m = K}$

$\therefore E e^{\alpha t} = E_m$

or $\boxed{E = E_m e^{-\alpha t}}$ → expression for the decay of sound energy

Growth and decay in a room w/ good acoustics

in a room w/ bad acoustics



(overlapping)

Expression for reverberation time

when $t = T$

$$E = E_m \times 10^{-6}$$

$$E_m \times 10^{-6} = E_m \times e^{-\alpha t}$$

$$e^{\alpha T} = 10^{+6}$$

$$\alpha T = \ln 10^6$$

$$= 6 \ln 10$$

$$\alpha T = 6 \times 2.303$$

$$T = \frac{6 \times 2.303}{\alpha}$$

$$T = \frac{6 \times 2.303}{C_s A / 4V} \qquad C_s = 340\ m/s$$

$$\boxed{T = \frac{0.163V}{A} = \frac{0.163V}{\sum_i a_i s_i}}$$

Sabine's formula for reverberation time

can also be written as: $\frac{0.163V}{\bar{a}\,S}$

$\bar{a}$ = avg. absorption coeff   S = total surface area

Limitations of Sabine's Formula

It works only when the value of a is small.
In a dead room where $a = 1$, reverberation time should be 0.
but according to Sabine's formula:

$$T = \frac{0.163V}{s} \neq 0$$

Eyring used a statistical approach & derived the foll

$$T = \frac{0.163\,V}{-s\ln(1-a)^{-1}} \quad -(1)$$

when $a = 1$

$$T = \frac{0.163\,V}{-s\ln\left(\frac{1}{1-a}\right)^{-1}} = \frac{0.163\,V}{-s\ln\left(\frac{1}{0}\right)} = \frac{0.163\,V}{-s(\infty)} = 0$$

$$\therefore \boxed{T = 0}$$

For small values of a:

$$\ln(1-a)^{-1} = \frac{-a}{1} - \frac{a^2}{2} - \frac{a^3}{3} - + \dots$$

the higher powers can be neglected.

$$\therefore \ln(1-a)^{-1} = -a$$

$\therefore$ Sabine's formula from (1) $= \dfrac{0.163\,V}{-s(-a)} = \dfrac{0.163\,V}{sa}$ //

Measurement of Absorption Coefficient

Method 1: By calculating the reverberation time in the room with & without a sample.

T_1 = reverberation time without sample

$$T_1 = \frac{0.163V}{A_1} = \frac{0.163\,V}{\sum_i a_i s_i}$$

$$\frac{1}{T_1} = \frac{\sum a_i s_i}{0.163V} \quad -(1)$$

T_2 = reverberation time w/ a sample

$$T_2 = \frac{0.163V}{a_s S_s + \sum_i a_i s_i}$$

where a_s = absorption coeff. of the sample material

S_s = surface area of the sample

$$\frac{1}{T_2} = \frac{a_s S_s + \sum_i a_i s_i}{0.163V} \quad \text{--- (2)}$$

(2) - (1) $\Rightarrow$ $$\frac{1}{T_2} - \frac{1}{T_1} = \frac{a_s S_s}{0.163V}$$

$$\boxed{a_s = \frac{0.163V}{S_s}\left(\frac{1}{T_2} - \frac{1}{T_1}\right)}$$

Method 2: In this method, 2 power sources of power outputs P_1 and P_2 are taken. The decay times of the steady energy state to the minimum audibility levels are calculated for P_1 and P_2.

Formula for the decay of sound energy : $E = E_m e^{-\alpha t}$

if E_0 is the minimum audibility level, and t_1 & t_2 are the times that correspond to minimum audibility for P_1 and P_2:

$$E_0 = \frac{4P_1}{C_s A} e^{-\alpha t_1} \quad \text{--- (1)}$$

$$E_0 = \frac{4P_2}{C_s A} \cdot e^{-\alpha t_2} \quad \text{--- (2)}$$

(1) ÷ (2) $$1 = \frac{\frac{\cancel{4}P_1}{C_sA} \cdot e^{-\alpha t_1}}{\frac{\cancel{4}P_2}{\cancel{C_sA}} \cdot e^{-\alpha t_2}}$$

$$\frac{P_1}{P_2} = e^{\alpha(t_1 - t_2)}$$

$$\ln\left(\frac{P_1}{P_2}\right) = \alpha(t_1 - t_2)$$

$$\ln\left(\frac{P_1}{P_2}\right) = \frac{C_s A}{4V}(t_1 - t_2)$$

$$\therefore \quad A = \frac{4V \ln(P_1/P_2)}{C_s(t_1 - t_2)}$$

$$\bar{a}S = \frac{4V \ln(P_1/P_2)}{C_s(t_1 - t_2)}$$

$$\boxed{\bar{a} = \frac{4V \ln(P_1/P_2)}{S C_s (t_1 - t_2)}}$$

$\bar{a}$ is the avg. absorption coeff. of the hall,

S = total surface area

Measurement of the Reverberation Time of a Hall

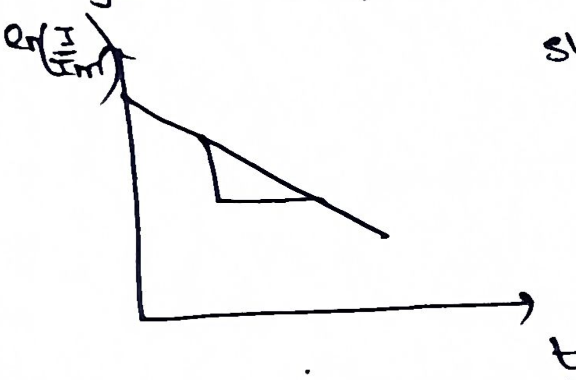
$$E = E_m \cdot e^{-\alpha t}$$

$\Rightarrow I = I_m \cdot e^{-\alpha t}$, where I and I_m are the intensities corresponding to E & E_m

$$\frac{I}{I_m} = e^{-\alpha t}$$

$$\ln\left(\frac{I}{I_m}\right) = -\alpha t$$

A graph of $\ln\left(\frac{I}{I_m}\right)$ vs. t would be a straight line with a negative slope.



$$\text{slope} = \frac{dy}{dx} = -\alpha$$

From the equation for the decay of sound energy,

$$\ln 10^{-6} = -\alpha T$$

$$T = \frac{-\ln 10^6}{-\alpha}$$

$$T = \frac{\ln 10^6}{\alpha}$$

$$\boxed{T = \frac{13.816}{\alpha}}$$

α can be calculated from the slope of the graph, or from the theoretical value : $\alpha = \frac{C_s A}{4V}$.

Factors affecting the acoustics of buildings

Factor	Cause	Effect	Remedy
1. Reverberation	Reflection from hall surfaces.	If reverberation time is large: booming sounds / confusion due to overlapping syllables. If reverberation time is low: sound is flat, loudness reduced.	optimum reverberation time to be obtained by: (i) providing doors, windows, ventilations at appropriate places (ii) providing absorbing materials at app. places (iii) using acoustic (absorbing) tiles
2. Loudness	Decrease of reverberation time.	Inadequate loudness	(i) low ceilings for reflection (ii) use P.A system (iii) use sounding boards behind speakers

POOJA PREMNATH 1
501

ULTRASONICS

Classification

- Sound waves can be classified on the basis of frequency as follows

sound

< 20Hz	20Hz - 20kHz	> 20kHz
infrasonics	audible	ultrasonics

production of ultrasonics

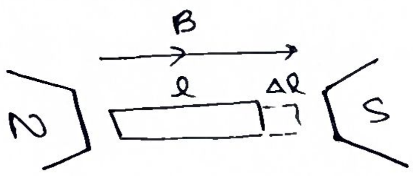
(i) magnetostriction

(ii) piezoelectric (oscillator)

Magnetostriction Oscillator

Principle : The oscillator works on the principle of magnetostriction.

When a magnetic field is applied parallel to a ferromagnetic rod (made of Ni, Fe), there is a change in the length of the rod (either contraction / expansion) This change in length is very small. (1 part per million). This is called magnetostriction



The change in length of the rod depends upon:

(i) The intensity of the magnetic field. $B \propto \Delta l$

(ii) The nature of the material (Different materials have diff. valencies, and the magnetic domains orient themselves different).

- It is independent of the direction of the field. If the nature of the material is such that it will ~~caus~~ contract on interaction w/ a magnetic field, then it will do so regardless of the direction of the applied magnetic field.

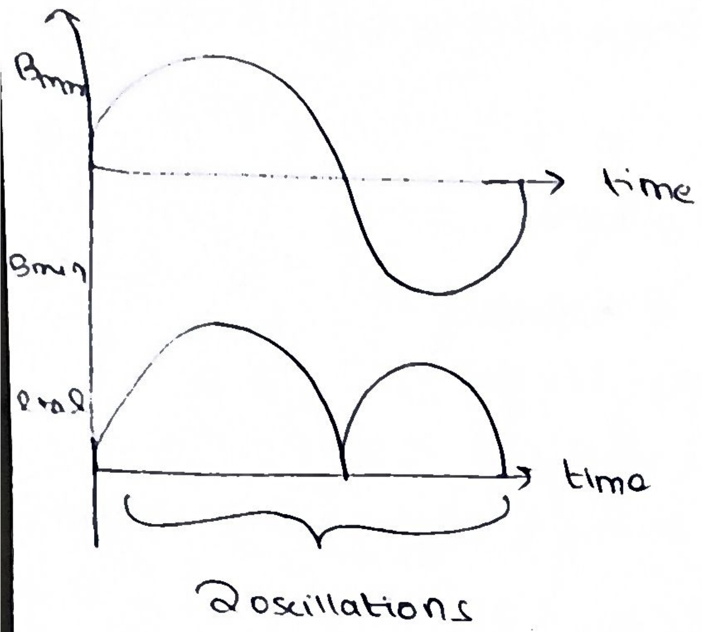
~~Magnetizing the rod~~

~~The rod can be mag~~

Generation of oscillations of the rod.

- A magnetic field can be produced by using a current carrying coil. If the ferromagnetic rod is placed inside a coil containing AC, it will suffer the same change in length for every half cycle of the AC, provided that it is already not magnetized

• Thus, the frequency of the change in length of the rod is twice that of the AC.



• Thus, it is necessary to magnetize the rod with DC initially, so that $f_{rod} = f_{AC}$

• The frequency of the AC can be adjusted to match the frequency of the rod, when they both match, resonant frequency is obtained, which produces large oscillations.

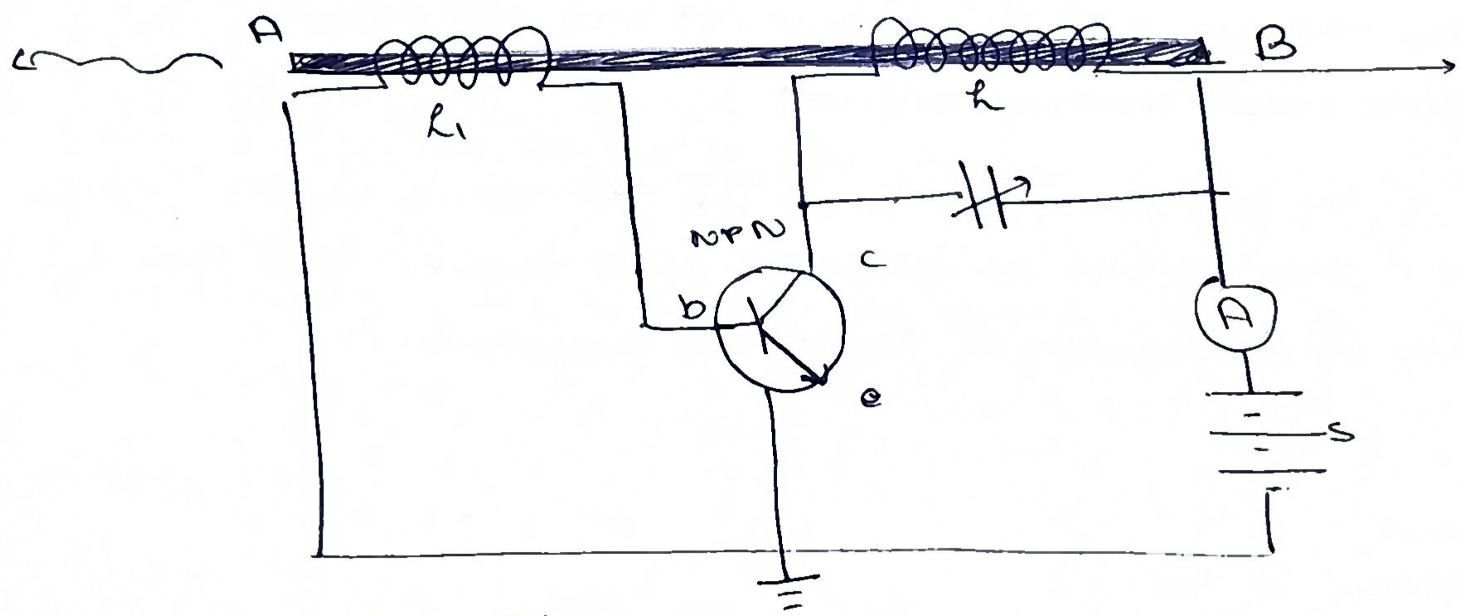
$$f_{rod} = \frac{1}{2l}\sqrt{\frac{E}{\rho}}$$

l = length of the rod

E = Young's modulus

ρ = density of the rod

Construction



• The ends of the ferromagnetic coil are bound by 2 coils L & L_1

• The coil L is connected to a tank circuit and then to the collector of the transistor and the coil L_1 is connected to the base of the transistor.

• The frequency of the oscillatory circuit can be adjusted by adjusting the

: value of the capacitor.

The DC circuit that is connected can be used to control the biasing. The emitter is forward biased and the collector is reverse biased in the transistor.

Working

- The rod is first magnetised ~~and in~~ by using direct current.
- The switch is closed and current flows through the circuit. Current first flows through the coil L. This generates an emf, and the rod starts vibrating due to the magnetostriction effect.
- The emf is carried over to the coil L_1 as well. The induced emf is fed back to the base of the transistor.
- Thus, the transistor acts as a feedback, which enables the constant vibration of the rod.
- The frequency is adjusted using the capacitor, and it is made to match the frequency of the rod to attain resonance.
- At resonance, the vibrations are of large amplitude, and thus ultrasonic waves are produced on either side of the rod.

$$\frac{1}{2\pi\sqrt{LC}} = \frac{1}{2l}\sqrt{\frac{E}{\rho}}$$

Merits

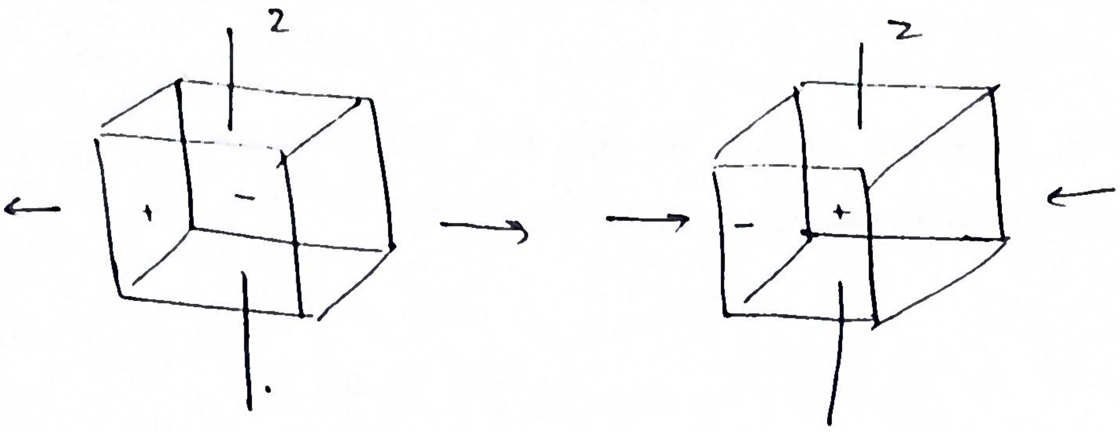
- simple circuit
- large output power can be produced
- alloys of Ni, Ni, Fe are easily available

Demerits

- hysteresis losses and eddy current losses
- dependent on temp.
- length of the rod cannot be reduced more than ~~beyo~~ a certain extent, frequencies beyond 300kHz not possible

Piezoelectric Oscillator

Piezoelectric oscillation : When one pair of opp. faces in certain crystals like quartz, tourmaline are subjected to pressure, charges develop on the other pair of opp. faces (which is not along the optic axis). Similarly, if tension is applied instead of pressure, charges again develop, but opp in magnitude.



tension

pressure

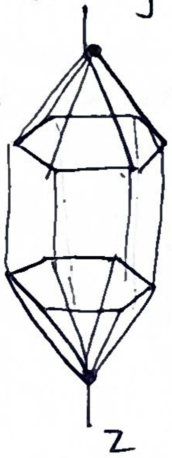
Inverse Piezoelectric Effect

When an AC current is applied on a pair of faces, the other pair of faces elongate & contract alternatively producing oscillations.

Transducer : A material which converts 1 form of a signal to another is called a transducer. eg. quartz, rochelle salt.

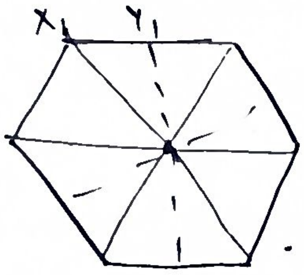
X-cut & Y-cut Piezoelectric Crystal

- Naturally occurring quartz is hexagonal bipyramidal in shape.



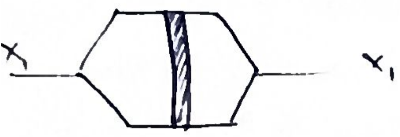
- The longest edge is the z-axis or the optic axis.
- The crystal is first cut $\perp^r$ to the optic axis.

- The cross-section is hexagonal in shape.



- When opp. corners of the hexagon are joined, we get X-axes or electrical axis
- when the mid-pts of opp. sides are joined, we get Y-axes or the mechanical axis.

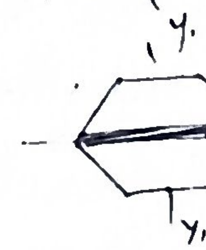
X-cut crystal : An X-cut crystal is obtained by cutting the hexagonal piece along the Y axis. ~~A longit~~. This leads to longitudinal mode of vibration.



$$f = \frac{1}{2l}\sqrt{\frac{E_f}{\rho}}$$

where l is the length of the quartz crystal for the fundamental mode of vibration.

Y-cut crystal : when the hexagonal piece is cut along the X-axis leads to transverse mode of vibration of the crystal. (5)

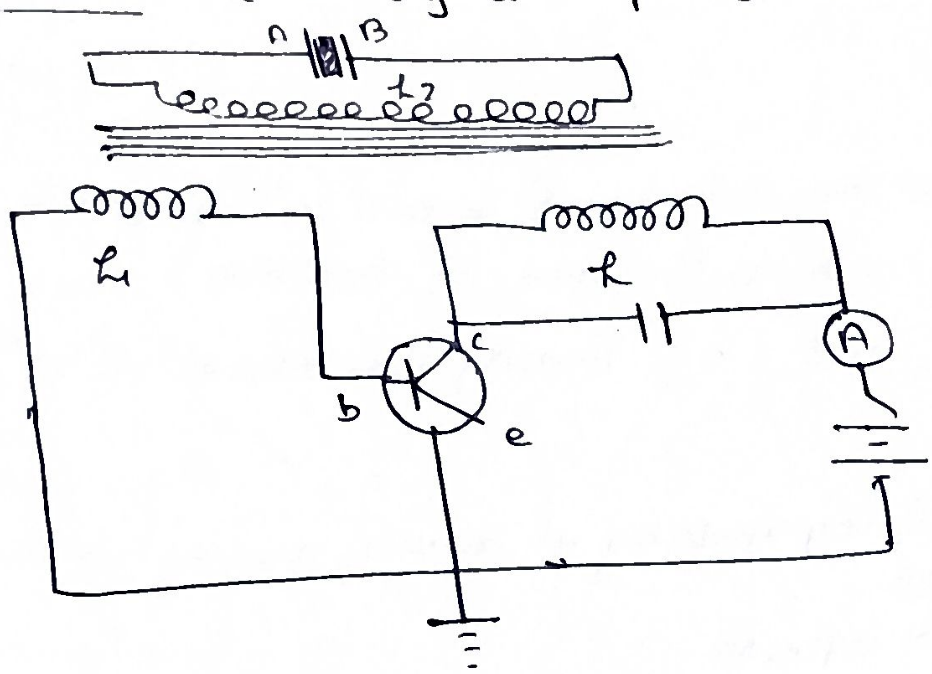


$$f = \frac{1}{2t}\sqrt{\frac{E_y}{\rho}}$$

If there are p turns, $f = \frac{p}{2t}\sqrt{\frac{E}{\rho}}$ or $\frac{p}{2l}\sqrt{\frac{E_y}{\rho}}$

t = thickness of the crystal for the fundamental mode

Construction : The cut crystal is placed between 2 electrodes, which are polished



thickness 't' is much lesser than the length 'l'
→ higher frequencies can be produced.
⇒ y cut crystal is preferable

- Coil L is connected to a tank circuit, coil L_1 is connected to the base of the transistor
- The power supply enables the correct biasing of the transistor
- The frequency of the setup can be varied using the capacitor.

Working

When the power is switched on, current flows through the LC circuit. There is an induced emf.

- By transformer action, A & B are also induced w/ an emf.
- Thus, the crystal experiences alternating oscillations, the frequency of which ~~depends on~~ lies in the ultrasonic range, emf from L_1 fed to base of transistor.
- The frequency of the AC is made to match with the natural frequency of the quartz crystal to attain resonance with vibrations of max. amplitude.

Resonant frequency is given by: $\frac{1}{2\pi\sqrt{LC}} = \frac{p}{2l}\sqrt{\frac{E_y}{\rho}}$

$$\text{or } \frac{1}{2\pi\sqrt{LC}} = \frac{p}{2t}\sqrt{\frac{E_y}{\rho}}$$

merits: • does not depend on temperature
• high frequencies can be generated
• a range of frequencies can be obtained using diff. transducers

~~Acoustic Grating~~

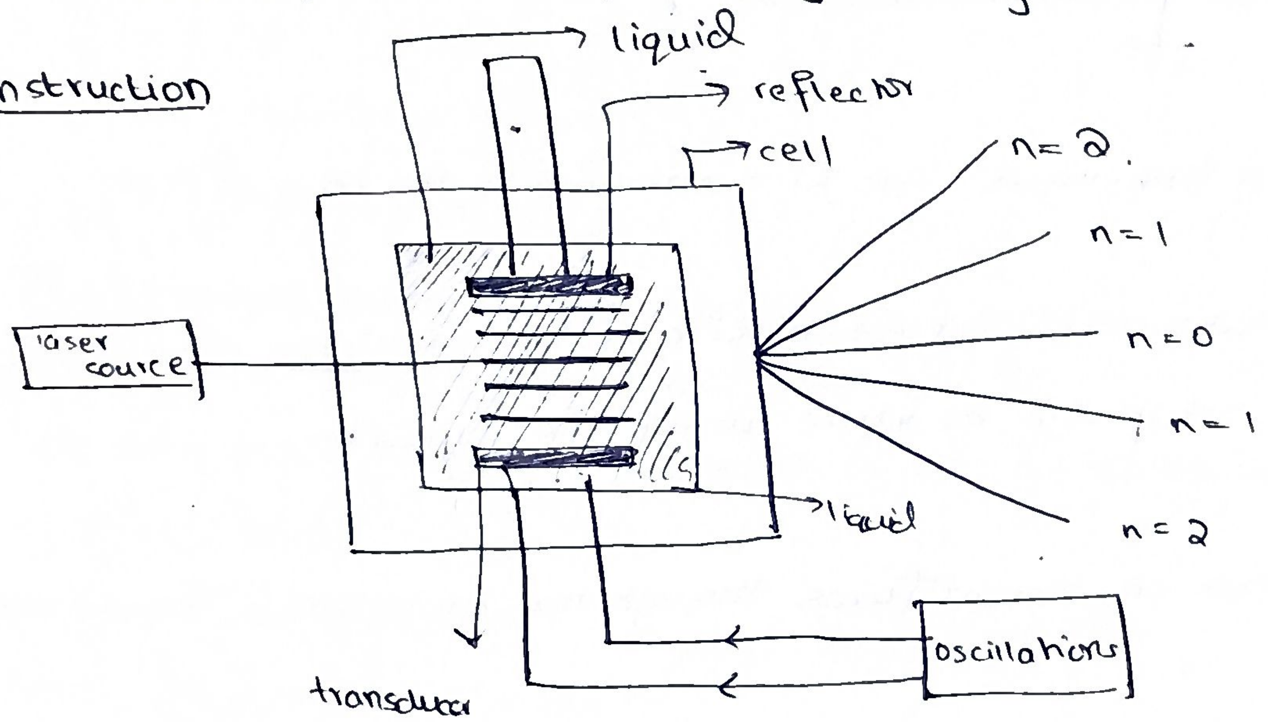
demerits: • high cost
• cutting & shaping of crystals is complex & expensive

Acoustic Grating

• It is sometimes required to find the velocity of waves in a medium, and thereby estimate the depth. (as in the case of SONAR). Here $d = vt/2$. t can be measured, and by finding the value of v, d can be calculated

• The value of v can be found by making an acoustic grating in the medium

Construction



• The setup consists of a cell, filled with the liquid in which the value of v is to be measured.

• There is a transducer placed at one end and a reflector at the other end.

• Ultrasonic waves pass through the medium and get reflected, standing waves are set up.

• There is a periodic variation in density, and thus refractive index as well.

• This resembles a diffraction grating, and since the grating is formed by ultrasonic waves, it is called acoustic grating

• A monochromatic beam of light is passed through this set up and there is a diffraction pattern is obtained. The angle of diffraction can be measured ~~easy~~ and thereby V can be found too

λ_u = wavelength of the ultrasonic waves

d_u = grating element

θ_n = angle of diffraction

λ_ℓ = wavelength of the laser source

condition for diffraction: $d_u \sin\theta_n = n\lambda_\ell$

$d_u = \frac{n\lambda_\ell}{\sin\theta_n}$

$d_u = \frac{\lambda_u}{2}$ $\lambda_u = 2 d_u$

$$\lambda_u = \frac{2n\lambda_\ell}{\sin\theta_n}$$

$$v_u = \lambda_u \cdot f_{osc}$$

$$\boxed{v_u = \frac{2n\lambda_\ell f_{osc}}{\sin\theta_n}}$$

Ultrasonic Pulse Echo System

The ultrasonic pulse écho system is used to detect flaws, irregularities, to evaluate the dimension of the flaw etc.

An ultrasonic pulse echo system consists of the following units:

(i) a master timer

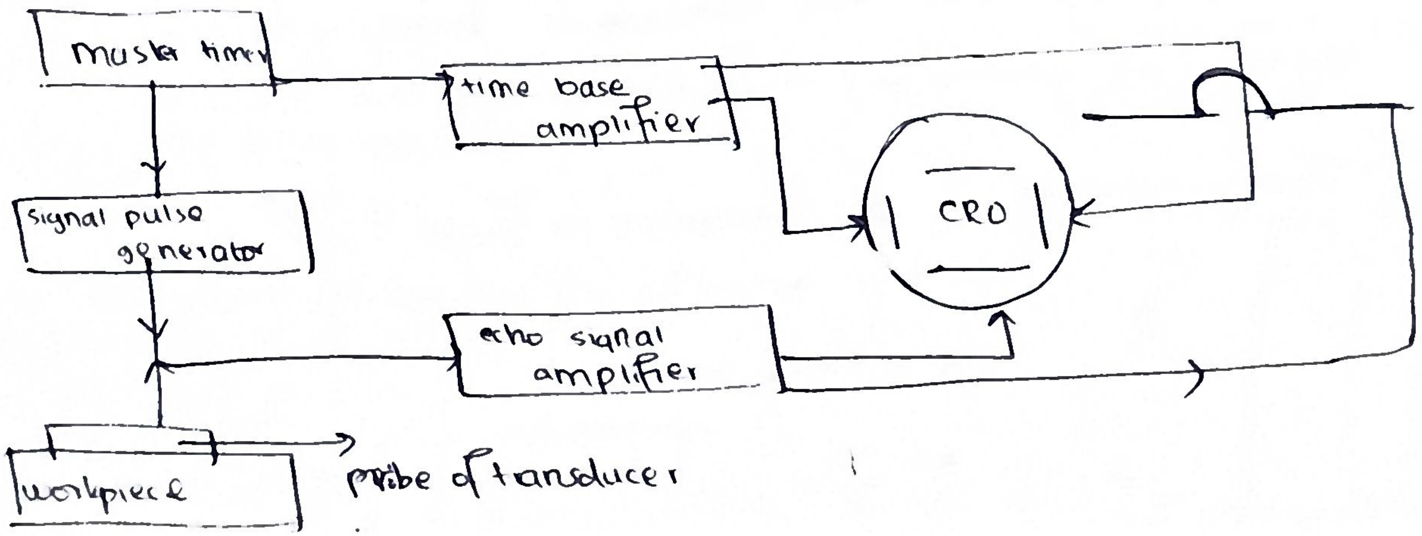
(ii) a signal pulse generator

(iii) signal transmitting transducer

(iv) signal receiving transducer

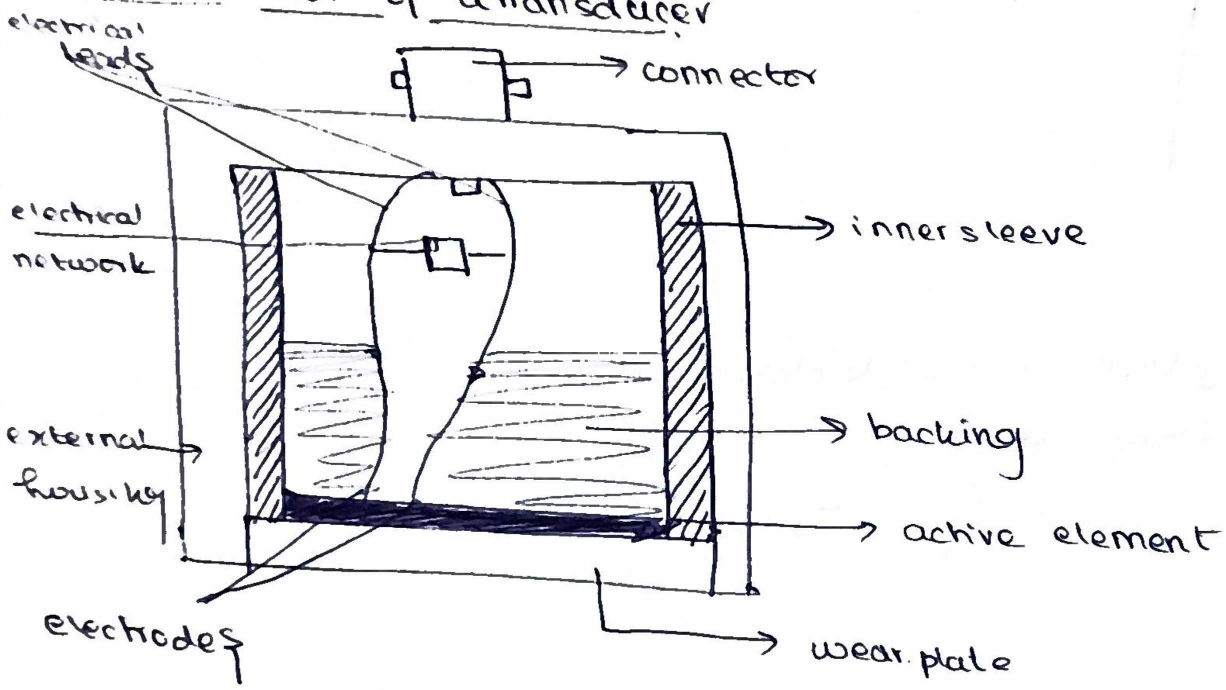
(v) echo signal amplifying setup

(vi) a display device like a CRO



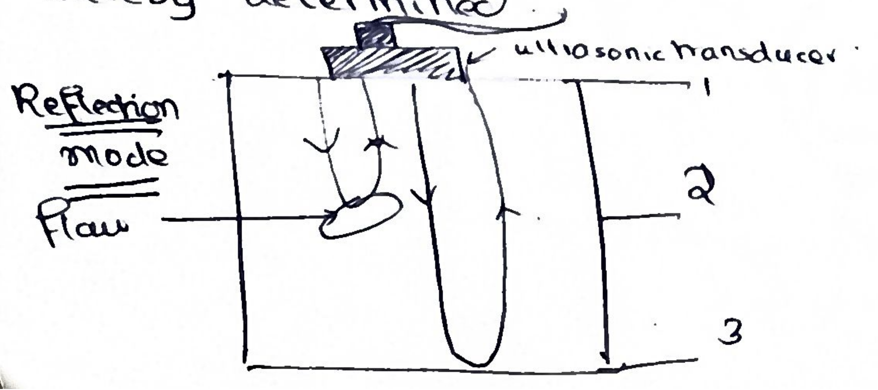
- The master timer triggers the signal pulse generator at periodic intervals.
- The signal pulse generator sends a burst of AC current to the transducer which produces ultrasonic waves
- If there is a flaw in the material, then the ultrasonic waves reflect back
- The reflected pulse causes the receiving transducer element to vibrate causing an alternating voltage across the transducer.
- This signal is amplified, and sent across to the display device, the CRO.
- When the master timer initially triggers the signal pulse generator, it actives the linear time base circuit of the display device.
- Hence, on the CRO, the reflected signal strength vs. time is displayed.

Cross-section of a transducer



Identification of Flaws

- Ultrasonic flaw detection is a comparative technique.
- A technician generally identifies echo patterns from good parts and from representative flaws.
- These patterns are then compared with that generated from a test piece. The condition / extent of damage / magnitude of the flaw is thereby determined



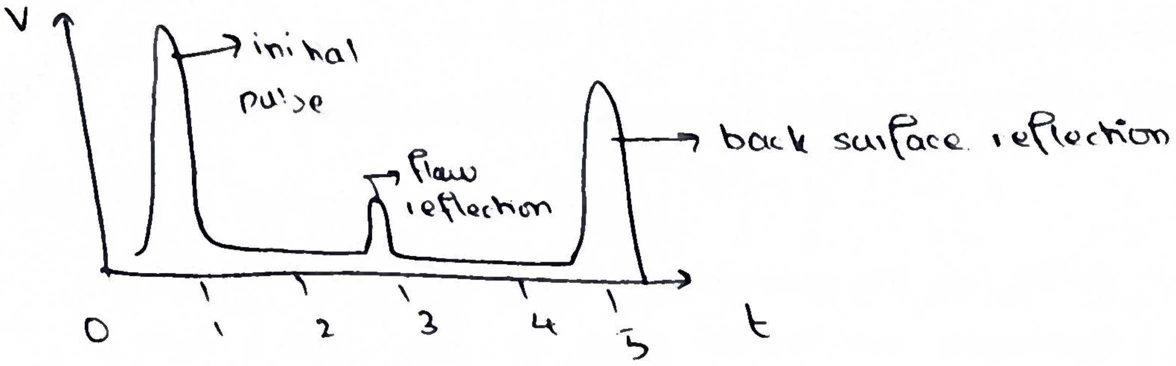
d = distance to flaw

v = velocity of sound in medium

t = time taken

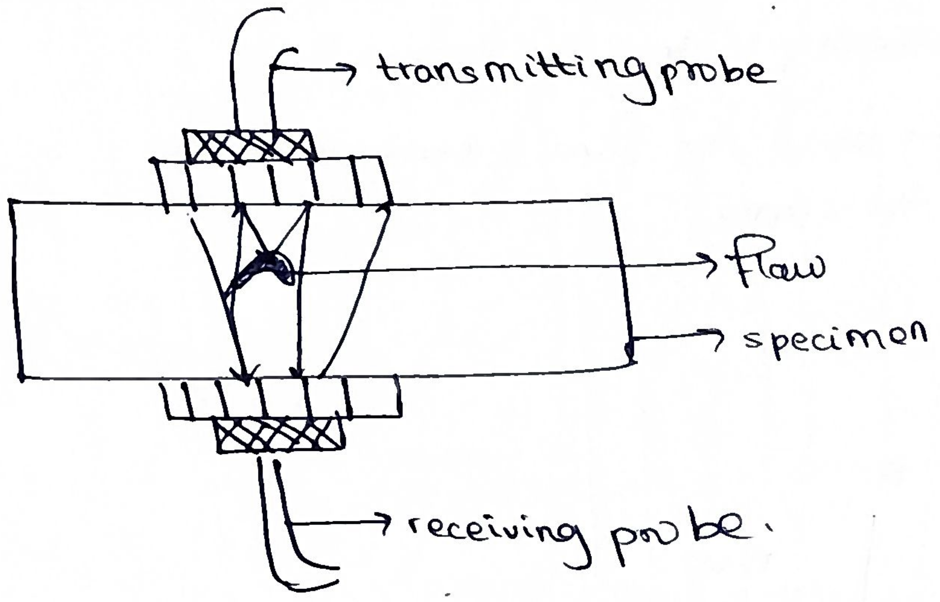
$v = \frac{2d}{t}$ $\boxed{d = \frac{vt}{2}}$

* echo pattern indicating a defect



merits :• only one side of sample is required for study
• sample can be of any shape
• depth of flaw can be measured

Ultrasonic Flaw Detection in Transmission Mode



• In this method, a transmitting probe is fitted onto one side of the specimen & and a receiving probe on the other.

• When there is no defect the receiving probe receives the entire signal.

• Otherwise, it receives only a part of the signal due to ~~back~~ partial reflection at the flaw.

Demerits

• Both sides of the specimen are required

• Both ends of the specimen should be parallel to one another.

• Only the presence of the defect is made known, its location is not known.

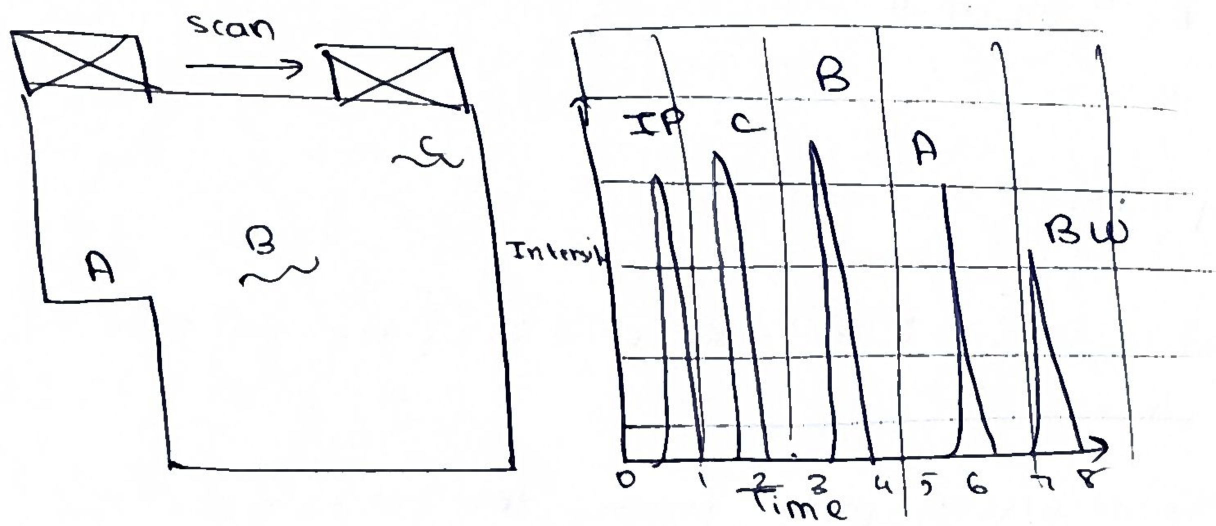
Data presentation modes. A, B and C Scans

The ultrasonic pulse echo data comprises of
(i) the time elapsed between the ~~refl~~ transmitted and the reflected signal
(ii) the intensity of the echo.

- There are 3 ways in which this data can be presented. The 3 most common NDT (Non-Destructive Testing) methods are the A-scan, B-scan and C-scan
- Each mode presents a different method of evaluating the data.

A-mode

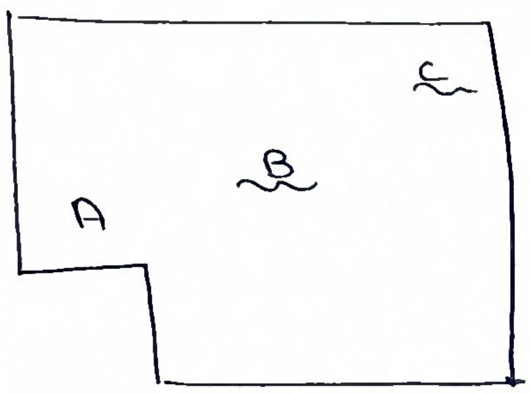
- The A-mode or amplitude mode presents the intensity of the echo received as a fn of time.
- The time is along the X-axis, and intensity is along the Y axis
- Using the A-mode, the depth ~~of the~~ at which the flaw is present can be identified based on the position of the signal.
- Consider the following example:

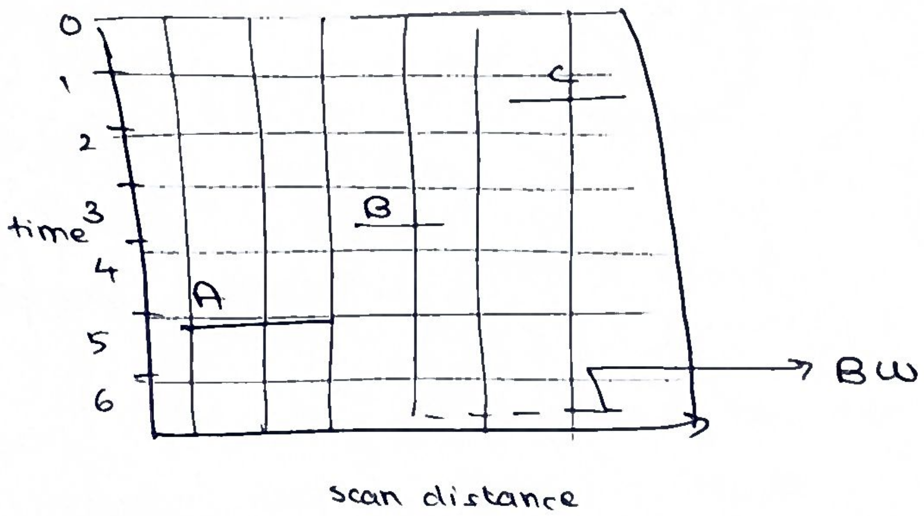


- Initially, at the beginning of the horizontal sweep, only the initial pulse is detected, which is near time 0.
- The signal after reflection from the back wall shows up at a later time, since ~~there~~ the signal has to travel a longer distance
- The defect B shows up somewhere near the middle of the X axis since it is approximately halfway between the front and back of the sample.
- The defect C shows up at a lower time since the signal does not take much time to undergo reflection

B-Mode

- The B-mode or the brightness mode has the travel time (time of flight) along the Y axis, with zero at the top left corner and the scanning distance along the X-axis
- The mode gives the linear position of the transducer and the linear dimensions of the flaw. The B-mode presents the cross-sectional view of the test specimen.
- The B-mode scans are generally generated by placing a trigger on the A-scan. Whenever the intensity is large enough, a pt. with appropriate brightness is produced on the B-scan.

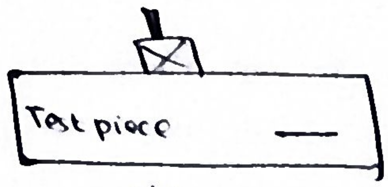




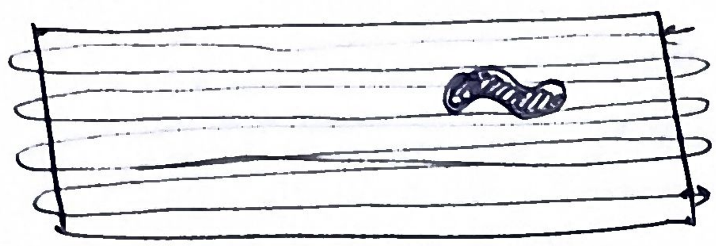
- Initially, the B-mode scan produces a line corresponding to the length of part A, then the back wall line BW is also produced.
- When the transducer moves over the flaws B and C, lines correspond to their lengths, at the respective depths are produced.

C-mode

- The C scan mode presents a plan-view of the location and size of the defect.
- The plane of the image is parallel to the plane of the transducer.
- The images are captured using automated data acquisition systems.
- A data collection gate is set up at the A-scan, which tracks the time of flight, and the x-y coordinates of the transducer are also monitored
- Accordingly the relative signal amplitude is measured and colored in shades of grey.
- If the pulse strength is high and the discontinuity small, 3D images can also be obtained



side view



scan patlein

c-scan presentation