

Audio System Engineering
Prof. Shyamal Kumar Das Mandal
Department of Electronics and Communication Engineering
Indian Institute of Technology, Kharagpur

Lecture -16
Large Room Acoustics and Small Room Acoustics

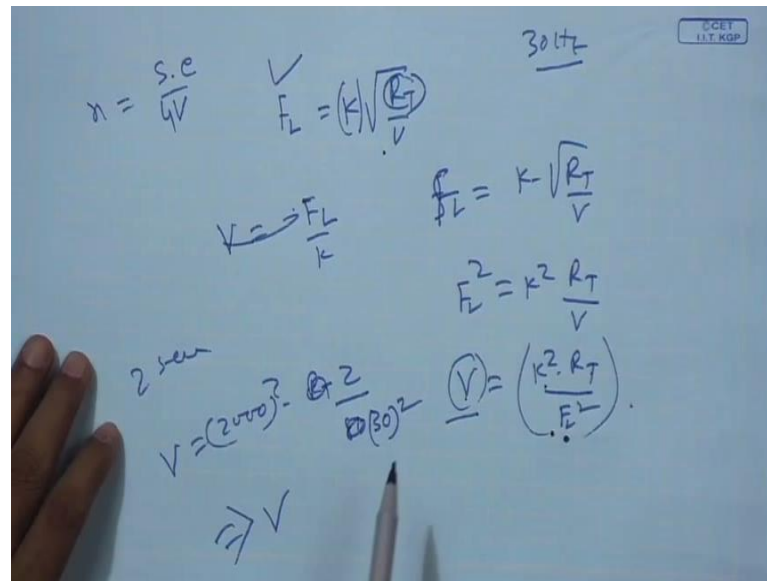
(Refer Slide Time: 00:45)

When does the statistical model apply?

- Statistical model applies to “large” rooms ones in which the reverberant field dominates the properties of the room.
 - A reverberant or diffuse field is one in which the time-averaged sound pressure is equal everywhere in the room. Sound energy flow is equally probable in all directions.
- In a “small room” the resonant standing waves—the so-called **room modes** dominate the response.

So, we have calculated the RT_{60} reverberant mean free path or reverberation time mean free path, number of reflection all those things we can calculate for a given dimension all the parameters. Now, there is a two kind of room acoustics; one is called large room acoustics; another is called small acoustics; large room acoustics and small room acoustics. A question is when does the statistical model apply, we said that about what about derive we say that RT_{60} when the statistical model is applied, when the statistical model will be applied. Do not see the slide. Let us discuss. What is when the statistical model is? In the statistical model, what I said that if the room has enough number of reflections sound, which is statistically validate that for any point, the number of reflection will be equal in the room and it create a diffuse sound field. Diffuse sound field means a reverberant diffuse sound in which the time averaged sound pressure is equal in everywhere in the room. Then we said the statistical reverberation time is applied and it is a large room acoustics.

(Refer Slide Time: 01:49)



So, if the room volume, if I said number of reflection n per second is nothing but a S into C by V . Now, s into c by V number of reflection per second is nothing but a RPS - S C by $4V$, S C by $4V$ sorry $4V$. Now, if the number of reflection is large then we apply a large room acoustics or reverberant sound field is created and statistical model of sound growth is applied. Now, in a small room, the resonant standing waves so called room modes dominate the response. In small room, number of reflection will be less and that case that it does not on the sound field is not reverberant sound field, it create by a mode room modes, sound field created by a room modes. Let us we discuss those two things.

(Refer Slide Time: 02:54)

What is the large room

$$F_L = K \sqrt{\frac{R_T}{V}}$$

F_L is the large room frequency
 $K=2000$ in SI
 R_T is the reverberation time
 V is the volume of the room
 Low frequency limit for speech is 80Hz and for music is 30 Hz.
 Let $R_T=1.6$

$$V = \frac{K^2 R_T}{F_L^2}$$

Now, what is how much volume is required to support a large room or what is the large room. So, large room equation is that if f_L is the large room frequency f_L is equal to K into RT divided by V . RT is nothing but a reverberation time RT_{60} divided by the volume of the room K is constant, f_L is the large room frequency. So, suppose I want to design an auditorium for speech using large room acoustic, large room acoustic is a reverberant field, auditorium in large room using acoustic then its volume must be support V should be f_L by K . So, I can say f_L is equal to K into root over of RT reverberation time by volume. So, I can say the f_L square is equal to K square by RT divided by V . So, V is nothing but a K square into RT divided by f_L square. So, the volume - required volume must support this equation. What is the equation value of K is 2000 in SI. So, V is nothing but a K square RT divided by f_L square.

So, suppose I want to design a speech room speech, speech auditorium for speech then the f_L is 80 hertz, if it is for music then the f_L is 30 hertz. If the volume support for 30 hertz that volume will support for high frequency also; the minimum volume is required at the lowest frequency response is 30 hertz then it support in high frequency also. So, the volume required is f square by RT divided by f_L square which is the minimum which is the volume minimum volume required to support the large room acoustics for music. For speech f_L value is 80 hertz or 60 hertz you can say the 60 also.

So, I can say I can give a problem once I civil drawing give to you, you have to decide it, whether this volume of the room what are the civil drawing is given is support the large room acoustic is not. If it is large room acoustic, treatment will be one kind, reverberation time calculate; if it is not then it has to be treated by a small room acoustic. Large room acoustic room is sound is developing using a mode, large room acoustic reverberation. So, depending on the given volume, I have to calculate whether it is support the large room acoustic or not, I would change the volume if it is required. That is why if you see suppose I want to design an auditorium RT_{60} , RT_{60} is equal 2 second then sub the music then I can calculate volume is nothing but a 2000 square into RT_{60} RT_{60} reverberation time is 2 second divided by f_L square is 30 hertz, for music it is 30 hertz 30 square. So, then I get the minimum volume which is required. So, if that civil drawing whatever the volume given is support that volumes then I go for the large room acoustics unless I go for the small room acoustics.

(Refer Slide Time: 06:50)

Room modes

- Room modes refer to the standing wave resonances that exist in an enclosed space.
- To visualize the standing wave modes recall the resonant modes on a string. When a resonant frequency excites the string a standing wave is set up with nodes and antinodes. The resonant frequencies are harmonic.
- In 2 and 3 dimensions similar standing waves exist but the resonant frequencies are not harmonically related.

I not be discussing. Now, if it is small room, large room we have already discusses large room acoustics reverberation time you can treated that acoustics small to control A, I know the reverberation time I know the volume. So, I can treat that wall with this for different observation coefficient to calculate this is desired A in Sabin. Now, for small room, room modes refer to the standing wave resonances that exist in an enclosed space if I enclose space in a small room, it create a standing wave. To visualize the standing wave modes, recall the resonant modes on a string when a resonant frequency excites the string a standing wave is setup, we know the standing wave generation.

(Refer Slide Time: 07:38)

What is an Eigen Mode

An Eigen mode is the European name for a standing wave. Standing wave are dependent upon the internal dimensions of an enclosure. The first mode will be

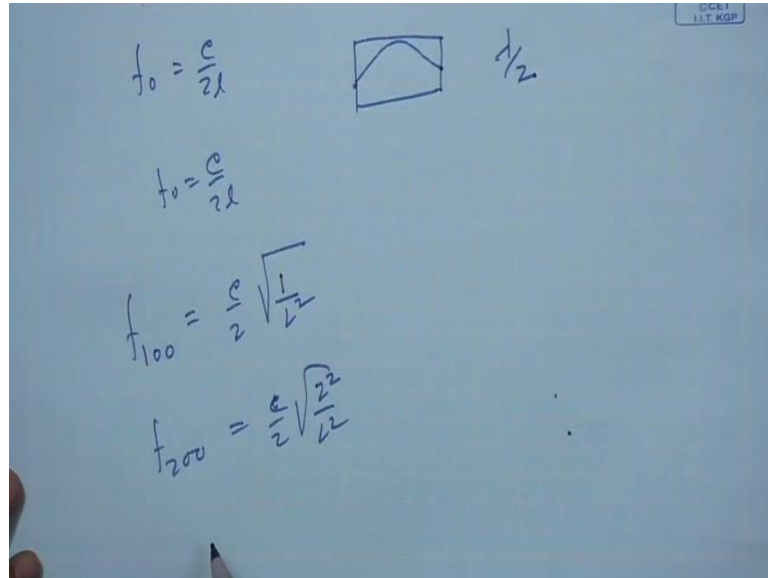
$$f_0 = \frac{c}{2l} = \frac{c}{\lambda}$$

- Modes are described by mode numbers n_1, n_2, n_3
- Room dimensions are L (length), W (width), and H (height).

$$f_{n_1 n_2 n_3} = \frac{c}{2} \sqrt{\frac{n_1^2}{L^2} + \frac{n_2^2}{W^2} + \frac{n_3^2}{H^2}}$$

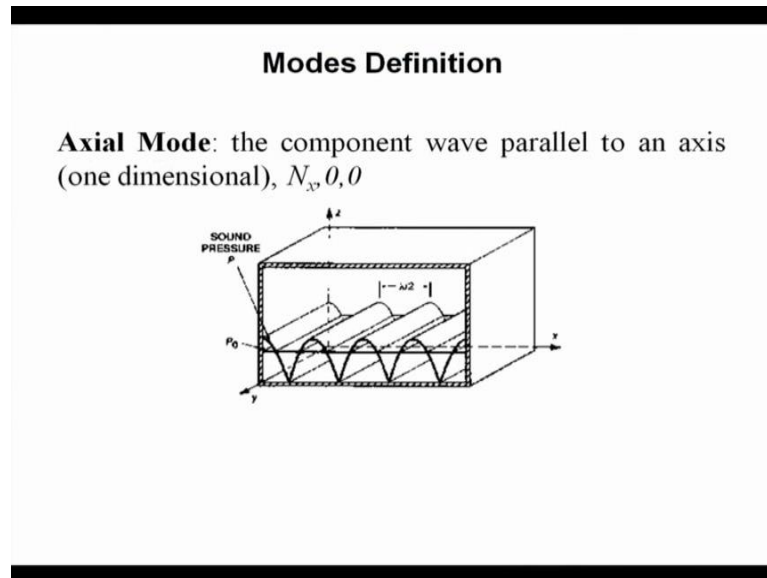
Let us go for an example. This mode is called Eigen mode, what is Eigen mode, an Eigen mode is an European name for the standing wave; standing waves are dependent upon the internal dimension of the enclosure.

(Refer Slide Time: 07:57)



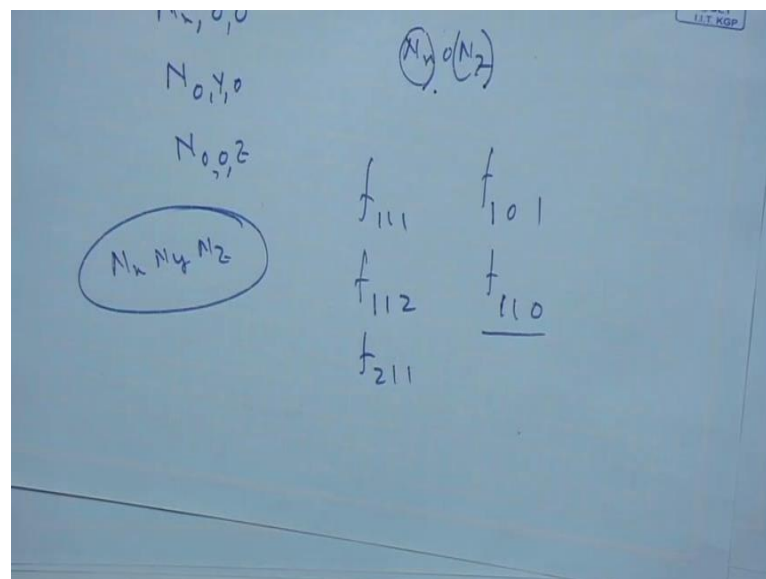
The first mode f_0 is nothing but a c by $2L$. Why, we know that. Suppose, this is an enclosure, what is the first mode, this one is first mode. So, it is λ by 2 half wavelength, so that is why f_0 is nothing but a c by $2L$. So, mode any room dimension is if the room dimension is L is length, and W is width and H is height then modes are c by 2 into root over of n_1^2 by L^2 n_2^2 by W^2 n_3^2 by H^2 . So, if it is f_{100} only with the mode is exist in with the only the length then c by 2 into root over of n_1^2 is 1^2 by L^2 . Second mode f_{200} c by 2 root over of 2^2 by L^2 I can calculate standing wave mode.

(Refer Slide Time: 09:13)



So, mode definition, now I came to the mode definition. Axial mode the component wave parallel to an axis is in one dimension. So, suppose in this room the node is created parallel to the axis only the horizontal the surface ground surface the node is created. So, only the node is created along the x, y plane only. So, all are you can say it is created alone the x-axis only standing node, so that is why it is called axial mode or it may be.

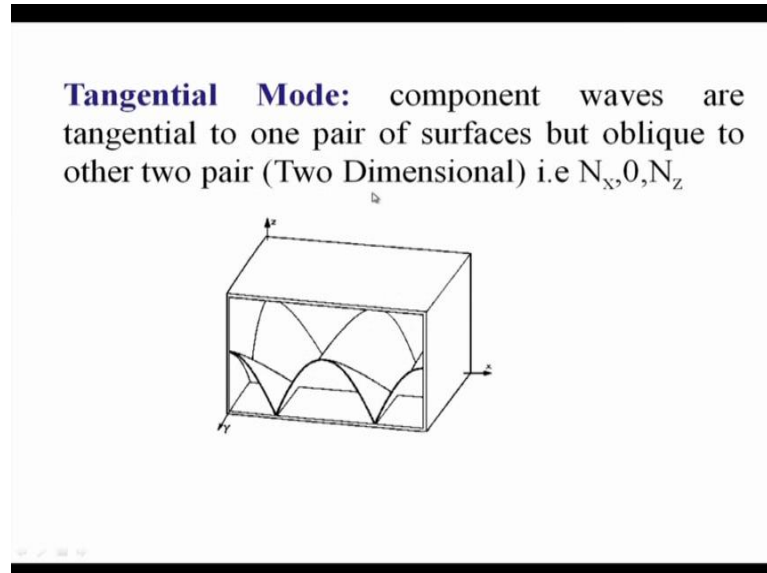
(Refer Slide Time: 09:55)



So, I can axial mode it may be create along the x-axis then it is $N \times 0, 0$; if it is created along the one y-axis $N 0 y 0$; if it is z-axis $N 0 0 z$. So, it is along the axis that is why it is

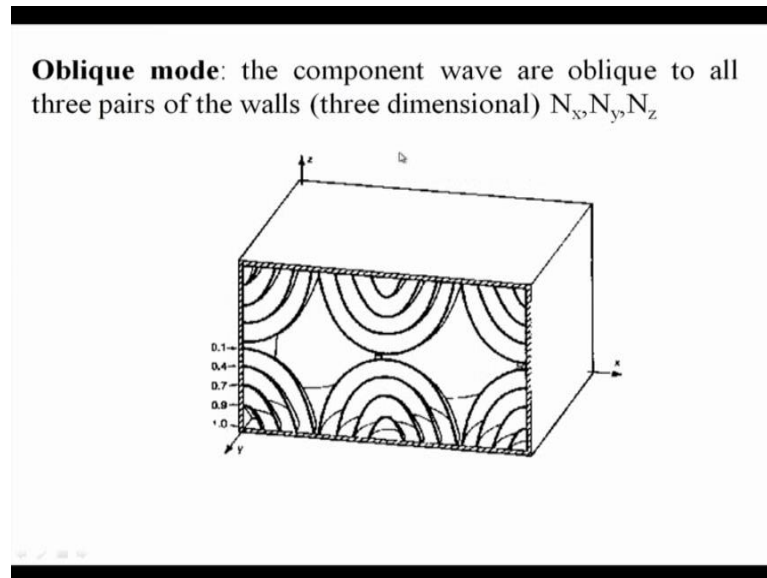
called axial mode the component wave parallel to the axis that is why in one-dimensional axial mode.

(Refer Slide Time: 10:22)



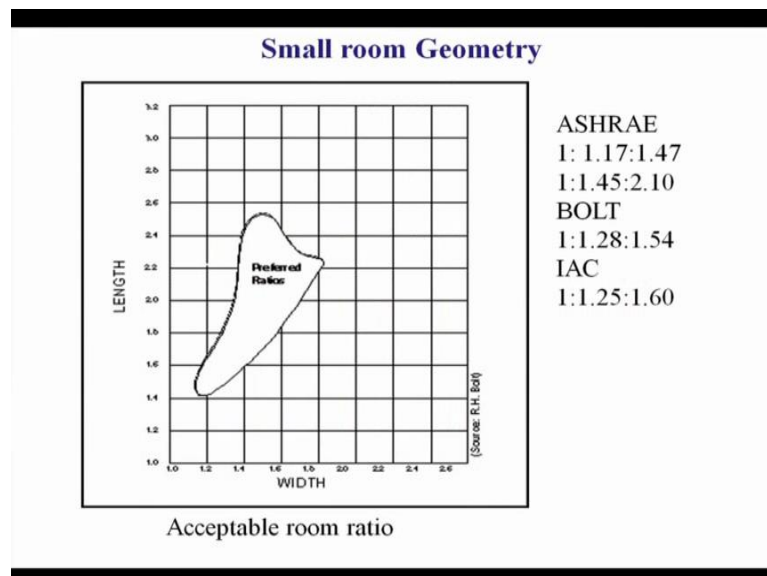
Then there may be a tangential mode. The component waves are tangential mode to one pair of surfaces but oblique to other two pair that is called two-dimensional. So, see that it is parallel to one pair of surface, but oblique to other pair so; that means, it should here two nodes. So, it is $N_x, 0, N_z$ or you can 0 0. So, the node is exist in x and z dimension other is not there similarly x and y, z will be not there, so that is why we call it is tangential mode.

(Refer Slide Time: 11:10)



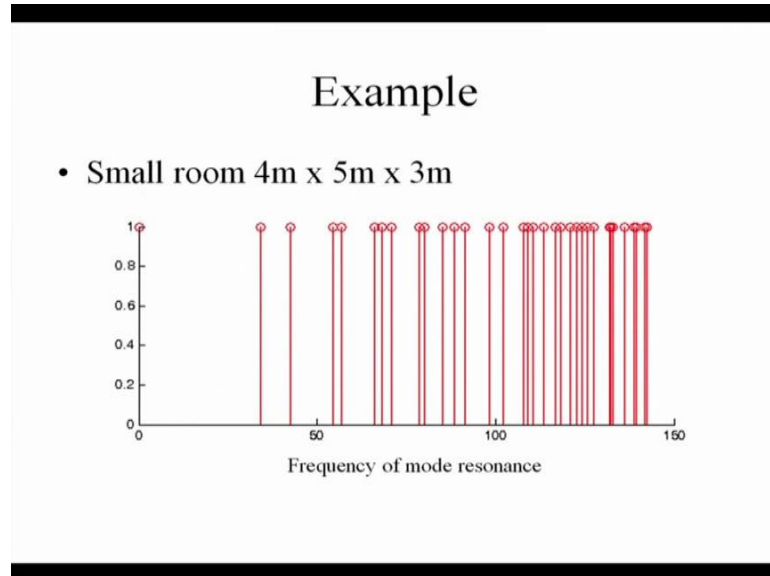
Then there is a call oblique mode. The component wave are oblique to all three pair of the surface that is x, y, z -3 I N x, N y and N z three nodes are exist, so that is f 1 1 1 first node, f 1 1 2 may be f 2 1 1. So, all kinds of combination will be possible, first harmonic second harmonic third harmonic, all combination will be possible that is called oblique mode. If it is tangential mode, only two will be there x and 1 0 1 or f 1 1 0 all are tangential mode will be there.

(Refer Slide Time: 12:00)

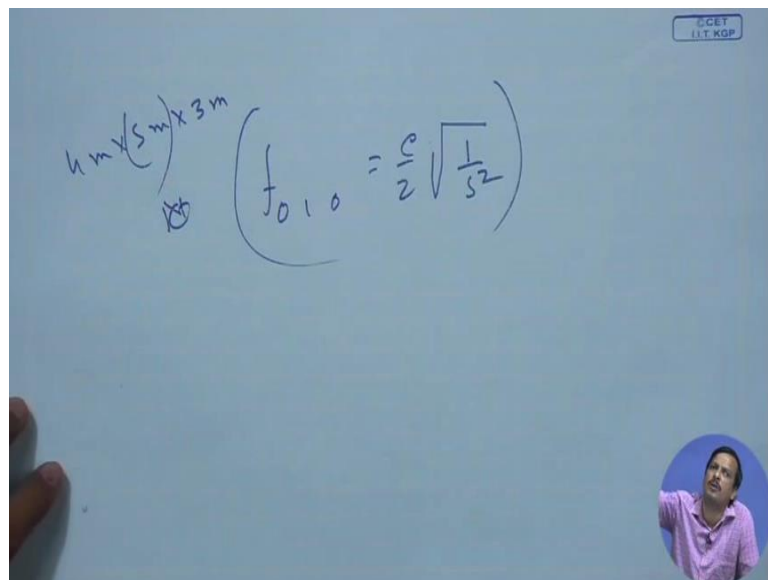


So, for example, I can say suppose I give you the room dimension, whether I can give you an example.

(Refer Slide Time: 12:14)



(Refer Slide Time: 12:16)




Lets I give an example for that node calculation. I told you that suppose I have a small room; room dimension is 4 meter, 5 meter and 3 meter. And calculate the first node lowest axial-mode lowest axial-mode. So, you can say f in any axis which is lowest when which will be lowest when the denominator is high. So, it is can be 0 1 0. So, it is it is c by 2 root over of 1 by 5 square lowest axial node frequency is this. Then I can all other

frequency will be created, so that is that way you can do it. Next one is small room, geometry acceptable room ratio for a small room acoustic that is given that is say standard length, length, width and height. So, once I one to design a small studio for speech studio small room small room acoustic then I create within this ratio.

(Refer Slide Time: 13:33)

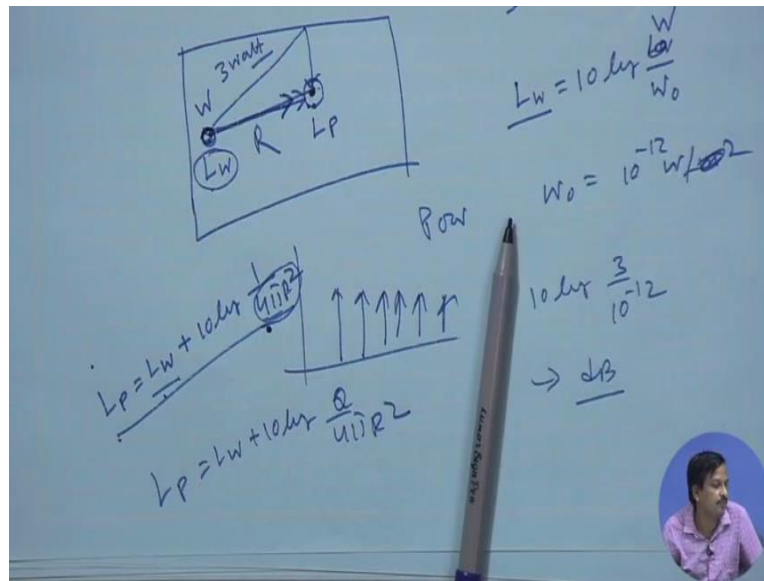
Semi-reverberant room calculations

- A room that has a mix of reverberant sound and direct sound from a source is called semi-reverberant.
- Note that most real rooms are semi-reverberant.
- The sound in many parts of the room is reverberant with energy flow equal in all directions (far from the sound source); however, near the source, the sound flow is directional



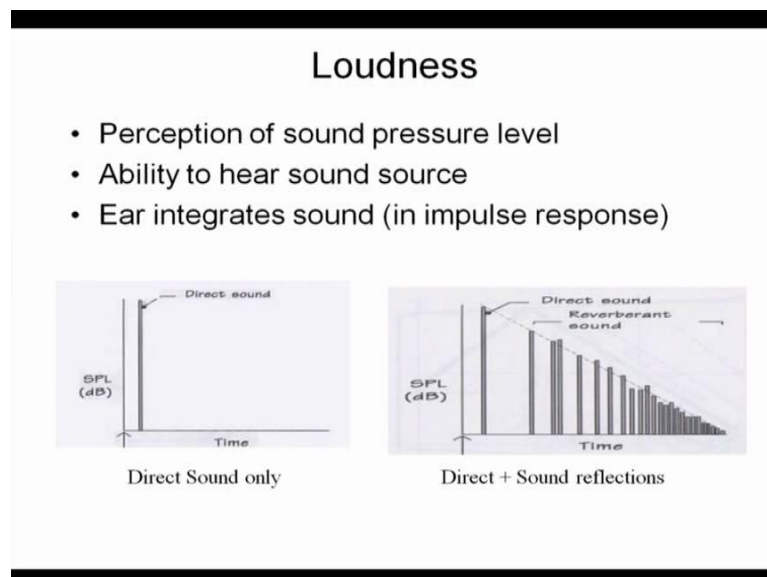
Now, semi reverberant, now I go for small room acoustic, I will discuss the acoustic treatment later on. So, I discuss the small room acoustics the acoustics field is created in node large room acoustic is a reverberant field or semi reverberant field you can say semi reverberant, there is a observation semi reverberant field. So, suppose I have a semi reverberant room in auditorium how they calculate the power and different parameter that is here. So, semi reverberant room calculation a room that has a mix reverberant sound and direct sound from a source is called semi reverberant room.

(Refer Slide Time: 14:14)



So, suppose I have an auditorium, let us draw it, and let us this is. So, I have sound source here; in here I want to calculate the power sound intensity or sound level, sound pressure level. Then since it is a semi reverberant room, there is a direct sound and there will be a reflected sound. Then the reflected sound will be there. Note most of the real auditorium in practical purpose is semi reverberant room.

(Refer Slide Time: 14:45)



So, I can say this is the direct sound; this point the sound is created or the sound pressure level will be developing based on the direct sound. So, direct sound will come here with

the pressure level, then first reflection, second reflection, third reflection, fourth reflection fifth reflection, so that way the reflection sound will be arrived here different reflection sound will be arrived here with lower amplitude. Once the path is high time will be high if the path high the intensity will be less. So, intensity will be decreasing in order number of reflection will be contribute to the sound pressure level of this point. The gap between the direct sound and first reflection is called initial delay; I will discuss that initial delay.

Now, I could calculate the source. Suppose, in this room lets I switch on this source which has a power is L W. So, what is L W in watt, in watt L the level of the sound level of this of sound power, power of this source is l w is the power of this source. So, if it is a power there it is ten log L W divided by or power of the sound source produce W watt power. So, power level is nothing but a W by reference W 0. So, what is W 0 reference is 10 to the power minus 12 watt per watt for a watt; if it is I, then it is watt per meter square 10 to the power minus 12 watt. So, if I say let us 3 watt acoustic source then it is 10 log 3 by 10 to the power minus 12 dB in dB.


(Refer Slide Time: 17:02)

Sound source calculations

- Non-directional sound source in free field. At distance R from source, direct sound is

$$L_p = L_w + 10 \log \left[\frac{1}{4\pi R^2} \right]$$
- Directional sound source (Q is directivity)

$$L_p = L_w + 10 \log \left[\frac{Q}{4\pi R^2} \right]$$
- Where W is the watts of acoustic power from source and $W_0 = 1 \times 10^{-12}$ Watts $L_w = 10 \log \left[\frac{W}{W_0} \right]$



So, non-directional sound source is a free field. So, suppose this sound source is non-directional sound source it is a non-directional sound source. So, this direct sound intensity in here will be so at point L p - sound L p at this point this is L p is equal to L W plus 10 log 1 by 4 pi R square distance from source is capital R. At R distance, it is L

p. So, it is 10 log reference power is L W distance from is R. So, it is why it is 4 pi R square, it is power it is not intensity that is why it is 4 pi R square reference power 4 pi R square. So, I said this is nothing but a 4 pi R square. Now, if put the directivity, if it is not I said it is not directional then it is; if it is directivity output then L p is nothing but a L W plus 10 log Q by 4 pi R square, Q is the directivity of the sound. What is directivity?


(Refer Slide Time: 18:21)

Directivity factor

- The directivity factor Q is a measure of the directional nature of a sound source. Q is defined as the ratio of intensity from the directional source, I_d , divided by the intensity of an omnidirectional source, I_0 .

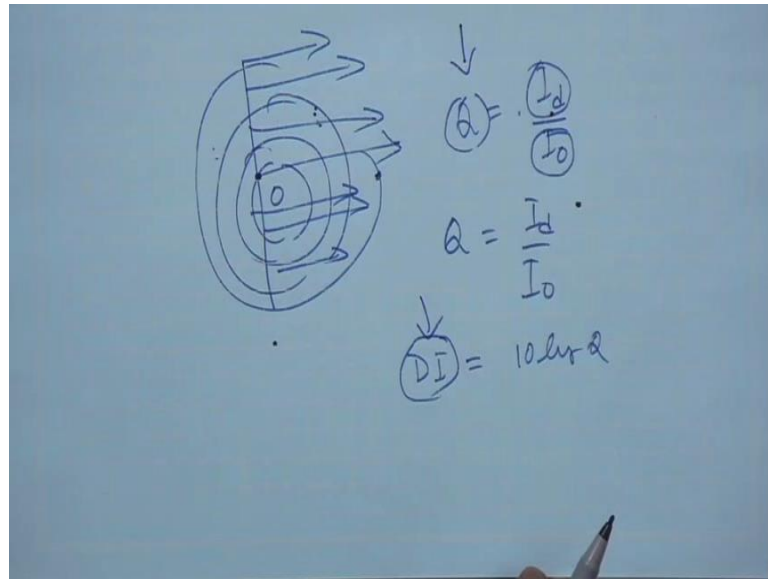
$$Q = \frac{I_d}{I_0}$$

- Directivity Index (DI) is Q expressed in dB

$$DI = 10 \log(Q)$$


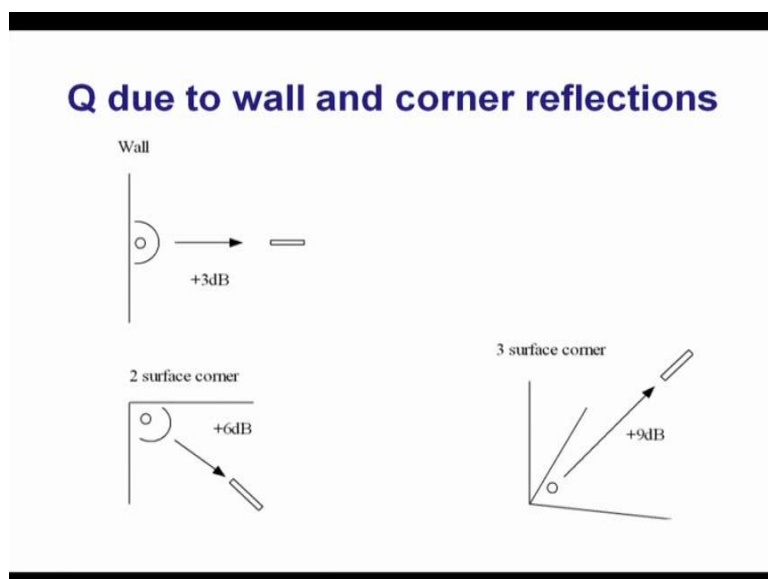
Suppose if I produce a sound in here, it is spherically propagated. Now, if I block one side, what will happen that energy will be reflected from that block surface and contributed in this way?

(Refer Slide Time: 18:28)



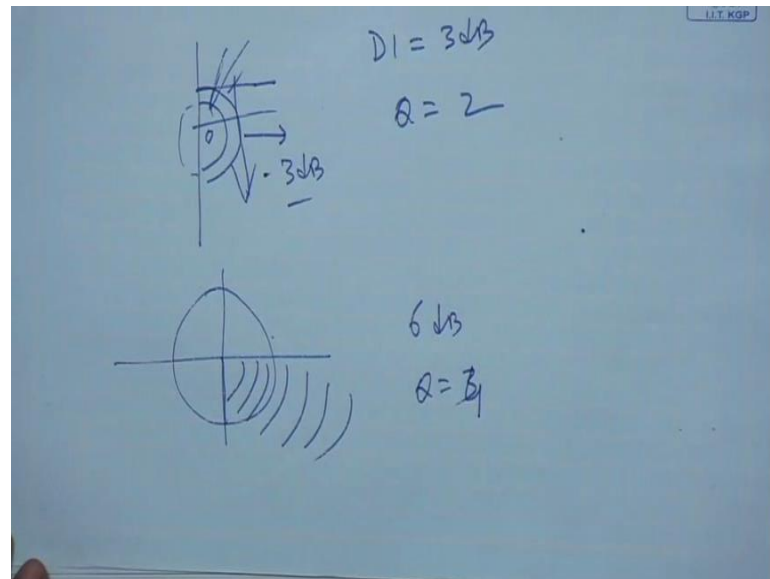
Now, I give a direction to the sound in this direction only not this direction. So, I introduce directivity of the sound. So, what is directivity, the definition of directivity Q is nothing but a I_d by I_0 it is only directional sound intensity, directional sound intensity. So, directional source I_d divided by only directional source I_0 . So, if this reflector is not there, intensity in here is I_0 . If the Lessing the reflector it is I_d , then the directivity factor is Q it is nothing but a I_d by I_0 . Q in dB, Q in dB is nothing but a $10 \log Q$ ratio ten log in dB directivity that is call directivity index, this is directivity and this is directivity index. So, this is directivity factors and this is directivity index.

(Refer Slide Time: 20:03)



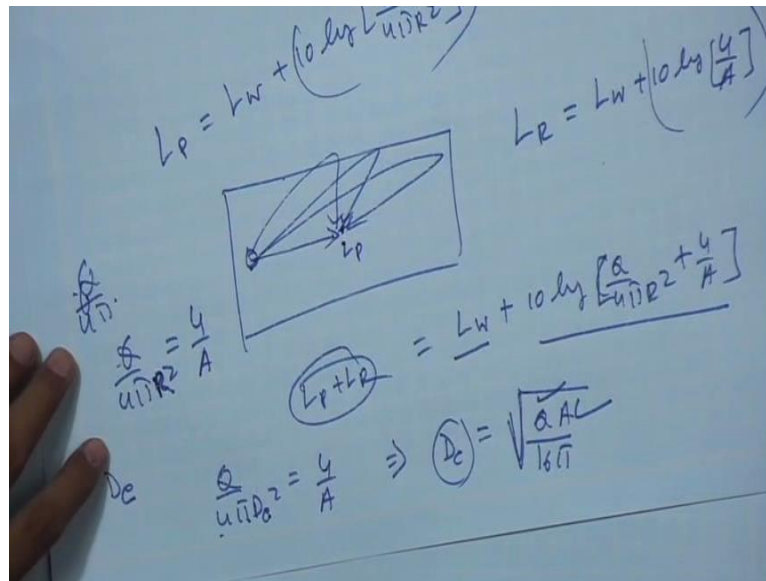
Now, let us say see this example that I have a wall.

(Refer Slide Time: 20:06)



And I have a source in here. So, what is that this side power is channelized to this wall? So, how much dB will be the gain on that how much dB the sound power will be increase double up version I channelized this way. So, it is 3 dB. So, Q is d I directive index is three dB or Q is equal to 2. Now, if I put the loudspeaker in here this corner this is the two surfaces then what will happen; once I put a surface here, then is sphere is cut down by this portion. So, it again increase by 3 dB, this portion is again channelized this, this direction. So, it is 6 dB, Q is equal to 3, sorry Q is equal to 4. If it is free surface, it will be 9 dB; Q is equal to 8, so that way I can create. So, if I put a loudspeaker on the wall - surface wall, so directivity index is 2; if I put in a corner of junction of the two walls, directivity index is increased.

(Refer Slide Time: 21:55)



So, that then direct power of the sound L_p or the point R is nothing but a L_w plus $10 \log Q$ by 4π capital R square, R is the distance from the source.

(Refer Slide Time: 22:11)

Reverberant sound

- Far from the source the decibel level of the reverberant sound is given by

$$L_R = L_w + 10 \log \left[\frac{4}{A} \right]$$

- Example—noise reduction. Change A from 45 Sabins to 120 Sabins. What is the change in reverberant sound of a 10^{-3} Watt source.

Now, what is reverberation sound? So, I said suppose this is the room and this is the source and this is point I calculating L_p . So, this is the direct power plus reflected power all reflected power will be there. So, what is the reverberant power or reflected power? So, it is L_R is nothing but a L_w plus $10 \log 4$ by A , A is the absorptivity in Sabin, 4 by A . Now, what is the total power at this point L_p plus L_R ? So, L_p plus L_R I can say it

is nothing but a L_W plus $10 \log \left[\frac{Q}{4\pi R^2} + \frac{4}{A} \right]$. So, direct power, so this is L_W will be remaining same, L_W is the L_W . So, direct power is reduces by a L_W this and this will be added up. So, L_W plus this one, Q by $4\pi R^2$ plus 4 by A .

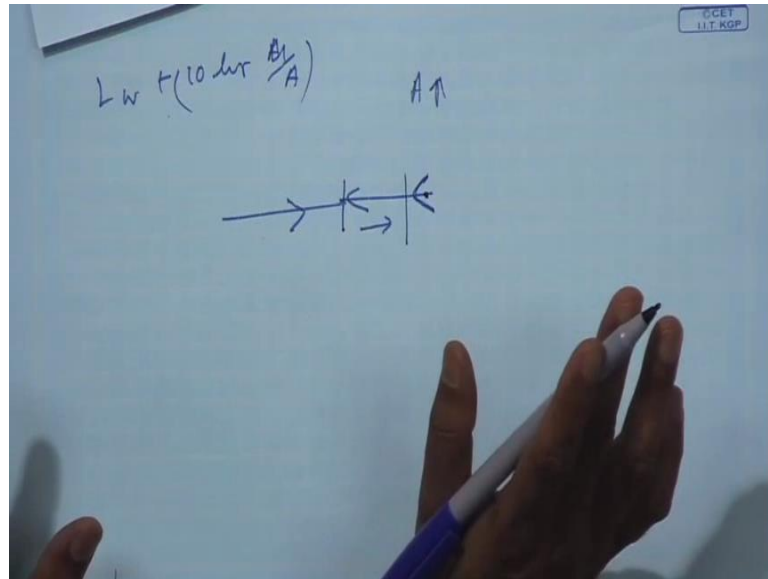
(Refer Slide Time: 23:39)

Critical distance

- The critical distance, D_c , is the distance at which the direct and reverberant sound levels are equal. $L_T = L_W + 10 \log \left[\frac{Q}{4\pi R^2} + \frac{4}{A} \right]$
- Equal when $\frac{Q}{4\pi D_c^2} = \frac{4}{A}$
- Thus, $D_c = \left[\frac{QA}{16\pi} \right]^{\frac{1}{2}}$

Now, if I say the condition is that lets I defined that find out the distance from the source or find out the distance from the source where the direct power which is Q by $4\pi R$ direct power and the reverberant power is equal. So, when it will be equal, when q by $4\pi R$ square will be equal to 4 by A . So, a distance R it should be a critical distance where that direct sound power and reverberant sound power is equal. Let us the critical distance is D_c then Q by $4\pi D_c$ square is equal to 4 by A or I can say D_c square, D_c is equal to root over of QA by $16\pi - 4$ into 4 , 16π , so it is root over of 16π . So, if I know that directivity, directivity factors and Sabin, I can find out the critical distance.

(Refer Slide Time: 25:16)




Now, there is an interesting point in reverberation time if you see reverberation sound pressure level is $L_w + 10 \log \frac{A_1}{A}$. If I increase A , the reverberation sound will be reduced. This factor will decrease $10 \log A$, if A is large then it will be minus, so it will decrease. So, this can be used for noise reduction. Change A from 40 Sabin to 100 Sabin, what is the change in reverberant sound? What is the meaning of this? A practical example is that. Suppose I want to install a machine in a room, and the machine creates noise. If I want to reduce the noise in that room, the noise created by the machine is direct sound, and the reverberation sound I can reduce. If I change A , then the reverberation sound field will increase. So, if that machine is installed in a reverberation field and I change that A , then the noise created by the machine will be reduced at this particular point, because the reverberation power level will be decreased. Only the direct sound will be there, direct sound is. So, this can be used as noise reduction.

(Refer Slide Time: 27:01)

Why is critical distance important?

- **Speech intelligibility**
 - For distances from the source much greater than the critical distance, speech becomes increasingly more difficult to understand because most of the sound energy comes from reflections. %ALCONS measures the loss of understanding of consonants. A room with a Reverberation time exceeding 1.6s there will be no listener beyond $3.16D_c$. If time raises the multiplier become lower.
- **Microphone placement**
 - General rule: microphone should be no more than $0.3D_c$ for omni-directional mic. $0.5D_c$ for directional mic.
 - The loudspeaker and the microphone should be at least $0.5D_c$ apart.



I have given a one problem on that one you can see that. So, now, I critical distance then why is critical distance is important. I calculate the critical distance, I know, but why the critical distance is important why critical distance is important is that the speech intelligibility. Suppose, I design a lecture theater then I required a speech should be intelligibility to every seat I will define that speech intelligibility. For a distance from the source much greater than the critical distance speech becomes increasingly more difficult to understand, because most of the sound energy comes from reflection. So, I say once I go beyond the critical distance, what I said the critical distance is a distance, where the direct sound reverberation sound is equal. Now, if I go that way direct sound is less reverberation sound is increases.

So, now, if I go such point where the direct sound is become less and less and only the reverberation sound is there. So, percentage measure of ALCON I will come on that loss understanding of consonants. So, then understanding of the speech is reduce or the intelligibility of the speech is reduces. A room with reverberation time exceeding 1.6 second there will be no listener beyond $3.1 D_c$; after $3.1 \text{time } D_c$, there will be no listeners he cannot receive the direct sound only reverberation sound, and he is facing the problem of speech intelligibility.

But if I put the loudspeaker on that area, then again it will be increases. So, when you design an auditorium, suppose I put only two loudspeakers in that stage and I know the


directivity of that source and total power of the source, then I can calculate that directivity directive factor. Let us too loud speaker only is installed in the front wall of the stage then I can easily calculate what should be the critical distance. Once I know the critical distance, 3.1 times the critical distance after that nobody should be there; that means I required a sound enforcement before 3.1 time critical distance. So, my next loudspeaker must be installed at 3.1 time of the critical distance within that time. So, I know where is the my speaker placement should be; I should not place a loud speaker in the front wall then within 1 meter another loud speaker within 1 meter another loud speaker which does not required wastage of acoustic power. So, I can put optimal distance loud speakers, I can use that using the critical distance.

Then it critical distance is important for the microphone plus placement also, where I should the place the microphone is the general rule microphone should be no more than $0.3 D_c$ for a omni-directional mic, and $0.5 D_c$ for a directional mic. If I want to catch then it should not be $0.3 D_c$ for a onmi-directional and $0.5 D_c$ for a directional mic, the loudspeaker and the microphone should be at least as far as D_c to get a good audio system in an auditorium. If I know that the critical distance of that auditorium then the distance between the loud speaker and microphone should be at least $1 D_c$.

(Refer Slide Time: 30:55)

Why is critical distance important?

- **Speech intelligibility**
 - For distances from the source much greater than the critical distance, speech becomes increasingly more difficult to understand because most of the sound energy comes from reflections. %ALCONS measures the loss of understanding of consonants. A room with a Reverberation time exceeding 1.6s there will be no listener beyond $3.16D_c$. If time raises the multiplier become lower.
- **Microphone placement**
 - General rule: microphone should be no more that $0.3D_c$ for omni-directional mic. $0.5D_c$ for directional mic.
 - The loudspeaker and the microphone should be at least as far as D_c



So, D_c is important that is why we have calculated the critical distance. Then effect of critical distance, it determine the maximum acoustics separation, it determine the ratio of

direct to reverberation sound. So, this is very general statement, but critical distance is very important for placing the microphone when I put loud speaker and also the distance between the microphone and loud speaker is very important. So, at least for when I design an auditorium, I should know what is the critical distance for that auditorium depending on the; if I only place the speaker in the lets place the speaker in the first in the front wall then calculate the critical distance then once you know the critical distance then you know the reinforcement required for that auditorium. So, let us stop here.

Thank you.