

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

**Lecture -17**  
**Large Room Acoustics and Small Room Acoustics (Contd.)**

(Refer Slide Time: 00:31)

**%Articulation Loss of Consonants**

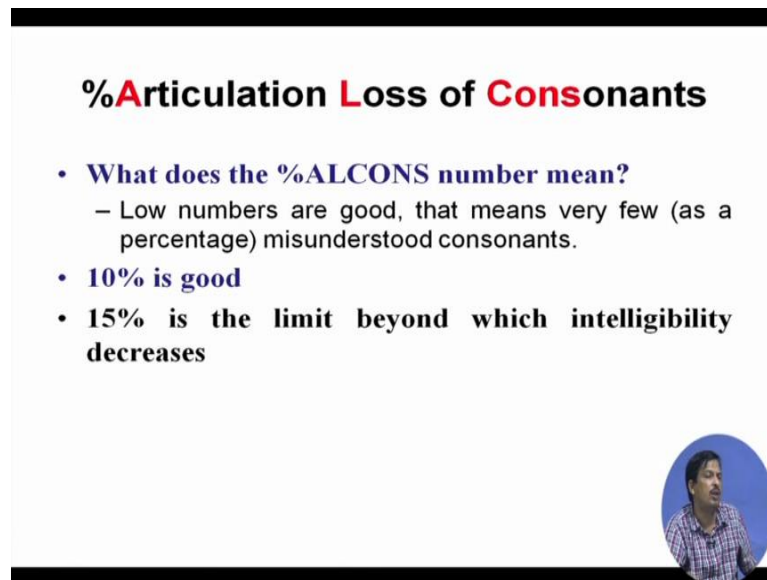
- %ALCONS formula

$$\%ALCONS = \frac{200R^2T_R(n+1)}{QV}$$

- R **Distance from speaker to listener**
- $T_r$  Reverb time
- Q **Directivity factor**
- V Room volume
- n Number of reinforcing loudspeakers


So, we have discussed about that critical distance, we calculate the critical distance and the important of the critical distance. Now, there we have mention the percentage ALCONS, what is percentage ALCONS. ALCONS is articulation loss of consonant percentage, ALCONS percentage of articulation loss of consonant that is why it is called percentage ALCONS. The formula percentage ALCONS is 200 into R square. R is the distance,  $T_R$  is the reverberation time, n is the number of the reinforcing loudspeaker plus 1 divided by Q is the directivity factor and V is the volume. So, if all the parameters are given, I can calculate the percentage ALCON.

(Refer Slide Time: 01:07)



**%Articulation Loss of Consonants**

- **What does the %ALCONS number mean?**
  - Low numbers are good, that means very few (as a percentage) misunderstood consonants.
- **10% is good**
- **15% is the limit beyond which intelligibility decreases**



Now, what will happen for percentage ALCONS, what is percentage ALCONS. What does the percentage ALCONS number mean? The low number means if the percentage is very low percentage articulation loss of consonant. So, if the percentage is very low then the percentage of loss of consonant is low. So, it is good for if the percentage ALCONS value is low means less number of consonant are lost. So, in intelligibility of the speech, if it is measured based on the percentage ALCONS then the percentage ALCONS should be low to make the speech more intelligible we have loss of articulation is less.


So, there is a value is called ten percent is good and 15 is the limit beyond which intelligibility decrease. If it is more than 15 percent the speech intelligibility of that auditorium is very less. Why, I said that percentage loss of consonant is also acceptable; now if it is human speech, if you see there is a lot of consonant and normal combination not there. All consonant are no that do not that important to understand that what is speaker is saying. Even if you see when we expose to a such talking or speech lectures even if some of the word or some of the syllables, some of the consonantly is dropped then also you can understand that speech so that is why there is a some redundancy of the consonant clarity on exist in the speech. Although there is some consonant is not clear then also you can understand that speech that is why percentage ALCONS of 10 percent means 10 percent articulation loss of consonant is acceptable for an auditorium design.

Now how do we calculate percentage ALCONS the formula is given. So, if the parameters are given room parameter means R is the distance from the speaker to the listener. So, I know that for this seat, what should be the percentage of ALCONS, if I want to calculate, and then I know the reverberation time of the auditorium, V - volume of the auditorium directivity pattern of the auditorium, depending on the placement of the loudspeaker and number of reinforcing loudspeaker on that auditorium.

(Refer Slide Time: 03:28)

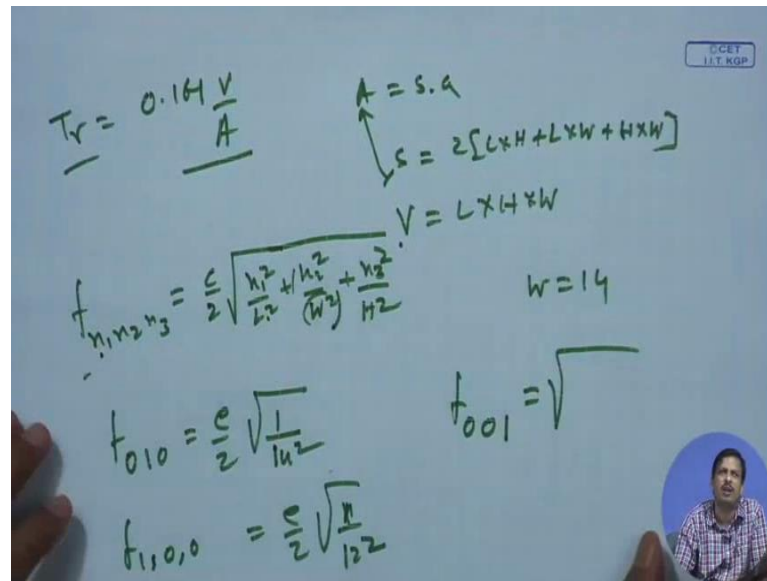
### Large Room Example

- Room dimensions 12 m x 14 m x 6 m
- $\bar{\alpha} = 0.2$ 
  - Calculate A and  $T_r$ .
  - What are the lowest 5 standing wave frequencies?
  - If a  $3 \times 10^{-6}$  W average output acoustic source is placed in the center of the front wall find
    - The reverberant level in dB
    - The total dB at a distance of 3 m from the source
    - The critical distance
    - %ALCONS at R=3 m, 9 m, and at 15 m from the source



Now, for the example, let us do mathematics that suppose I given the room dimension is given 12, 14 and 6 meters, this is a squared kind of room. And there A is given average acoustic is given, calculate capital A in Sabin and  $T_r$  means reverberation time, can you calculate  $T_r$ .

(Refer Slide Time: 03:52)



How do  $T_r$ ,  $T_r$  is nothing but a  $0.161 \text{ V by A}$ . What is  $A$ ,  $A$  is nothing but a  $s$  into  $a$ , what is  $s$ ,  $s$  is nothing but a surface area of a square room. So,  $2$  into length into height plus length into width plus height into width, I know the diameter dimension and I calculate  $s$ , then I calculate the length into height into width, I calculate the volume. Once I know  $v$  and  $s$  once I know  $s$  I calculate  $a$ , then I calculate  $v$  then calculate reverberation time. What is the lowest five standing wave frequency? What I said  $f_{n_1 n_2 n_3}$  is nothing but  $a$ , what is the formula  $c$  by  $2$   $c$  by  $2$  root to the power of what is the formula.

(Refer Slide Time: 04:46)

**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).

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

I told you the formula in the last lecture, I give you that formula, formula is nothing but if I see this formula is nothing but a c by 2 root over of n 1 square by L square plus n 2 square by W square plus n 3 square by H square, so length width and height. So, if I say lowest standing wave frequency, lowest five standing wave frequency I said. So, what is the lowest frequency? So, what will be f, f will be lowest while f all is 0. So, if lets that what is the width, width is maximum width is 14. So, if this width is large, so this parameter will be low. So, I can say f 0 1 0 will be the lowest frequency that is C by 2 root over of 1 by 14 square.

Then the next highest when the height will be there f 1 0 0 that is nothing but c by 2 root over of 1 by 12 square, then f 0 0 1 is root over of that things. So, now that way you can calculate the lowest five standing wave frequency. Similarly, if I now I said if I c into ten to the power minus 6-watt average output acoustic source is placed in the centre of the front wall find reverberant level in dB.

(Refer Slide Time: 06:35)

Handwritten notes on a whiteboard showing calculations for reverberation level. The notes include a diagram of a room with a speaker on a wall, and formulas for sound power level ( $L_w$ ), reverberation level ( $L_R$ ), and total sound level ( $L_T$ ).

Diagram: A rectangle representing a room with a speaker on one wall. The directivity factor is labeled as  $Q=2$ . The sound power level is labeled as  $3 \times 10^{-6} \text{ W}$ .

Formulas:

$$L_w = 10 \log \frac{3 \times 10^{-6}}{10^{-12}} \text{ dB}$$

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

$$= L_w + 10 \log \frac{4}{A}$$

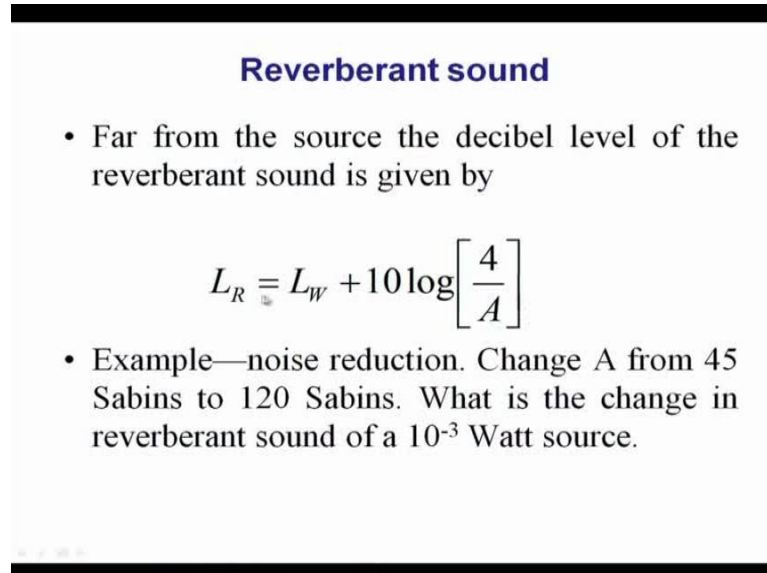
$$= \frac{\text{dB}}{A} \quad R=3$$

$$L_T = L_w + 10 \log \left[ \frac{Q}{4\pi r^2} + \frac{4}{A} \right] \text{ dB}$$

So, centre of the front wall this is the room this is the room. I placed a loudspeaker centre of the front wall, what is this 3 into 10 to the power minus 6 watt. Now, if it is placed then what is the directivity of the loudspeaker, it is placed on the single wall. So, Q is nothing but a 2. Once I know a Q, then what is L R reverberation sound level is nothing but a source plus 10 log 4 by A you know that formula. What is the formula reverberation sound level at any point on that room is nothing but a if you see I just open

that lecture number 1, I see the formula is going the formula not lecture number 1 it is in lecture number 2.

(Refer Slide Time: 07:40)



**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.

So, it is nothing but a, that formula  $L_R$  reverberation sound level is nothing but a  $L_w$  is the sound source level plus  $10 \log 4$  by  $A$ . So, what I said to find out the reverberation level in d B power level in d B. So, it is power reverberation sound power. So, power is nothing but a source. What is  $L_w$ ,  $L_w$  is nothing but a  $10 \log$  power 3 into 10 to the power of minus 6 divided by 10 to the power of minus 12 is the reference for 0 d B. So, that d B, d B, I got this  $L_w$ . Once I knew the  $L_w$  then plus  $10 \log 4$  by  $A$  I know the  $A$  from the previous slides, so  $A$  is nothing but s into a. So, I got the  $A$  4 by  $A$  and get that d B. Then I said the total d B at distance 3 meter from the source. So, if 3 meter from the source then total power  $L_T$  is nothing but a detect sound power plus reverberation power. So, what is  $L_w$  plus  $10 \log q$  by  $4 \pi R^2 r$  is the distance plus 4 by  $A$  that is the formula. So, I know the  $Q$ ,  $Q$  is equal to 2,  $R$  is equal to 3, I know that things and I know  $A$ , so can find out  $L_t$  also in that.

(Refer Slide Time: 09:29)

Handwritten notes on a whiteboard:

- Top left:  $A=2, Q=2$
- Center left:  $\% \text{ALCONS} = \frac{200 \cdot R^2 \cdot T_R \cdot (n+1)}{Q \cdot V}$
- Bottom left:  $R=3, Q=2, \frac{15\%}{T_R (n+1)}$
- Top right:  $\frac{Q}{4\pi D_c^2} = \frac{4}{A}$
- Middle right:  $D_c^2 = \frac{Q \cdot A}{16\pi}$
- Bottom right:  $D_c = \sqrt{\frac{Q \cdot A}{16\pi}}$

Critical distance, what is critical distance when I said critical distance of both that distance where the direct sound power and reverberation sound power is same. So,  $Q$  by  $4\pi D_c^2$  square is nothing but a  $4$  by  $A$ . So,  $D_c^2$  square is nothing but a  $Q$  into a  $16\pi$  and  $D_c$  is nothing but a root over of  $Q$  into a  $16\pi$ . Now, once I know that then what is  $D_c$  for this. This room I know  $A$ , I know  $Q$  equal to  $2$ , I know  $A$  and then I can calculate the  $D_c$  in meter. Then I said percentage ALCONS percentage ALCONS, calculate the percentage ALCONS. What is the percentage ALCONS formula is percentage ALCONS,  $A L C O N S$  is nothing but a  $200$  into  $R$  square  $R$  is the distance  $T_r$  is the reverberation time  $n$  is the number of reinforcement speaker and  $Q$  into volume.

So, what is the formula in the problem what is given  $R$  is equal to  $3$  for previous,  $R$  is equal to  $3$ , and  $Q$  is equal to  $3$ ,  $T_r$  I have to calculate, how do I calculate I know that  $T_r$ , I calculated from here  $16 - 16 V$  by  $A$ . So, I calculate  $t_r$ . So,  $t_r$  I know and  $n$  I said is only single source is placed. So, single source is placed means  $n$  plus  $n$  is  $1$ , so  $1$  plus  $1$ . So, I know this factor, this factor, I know this, I know  $Q$ , I know volume. So, I calculate the percentage ALCONS and if I say commands on your percentage ALCONS. So, if it is above  $15$  percent, I can say this is not a good room for speech intelligibility, I can calculate the percentage ALCONS. So, I am going to design a auditorium, if somebody said can I can you calculate that distance from the stage distance from the source

(Refer Slide Time: 11:51)

**Initial signal delay gap(ISD)**

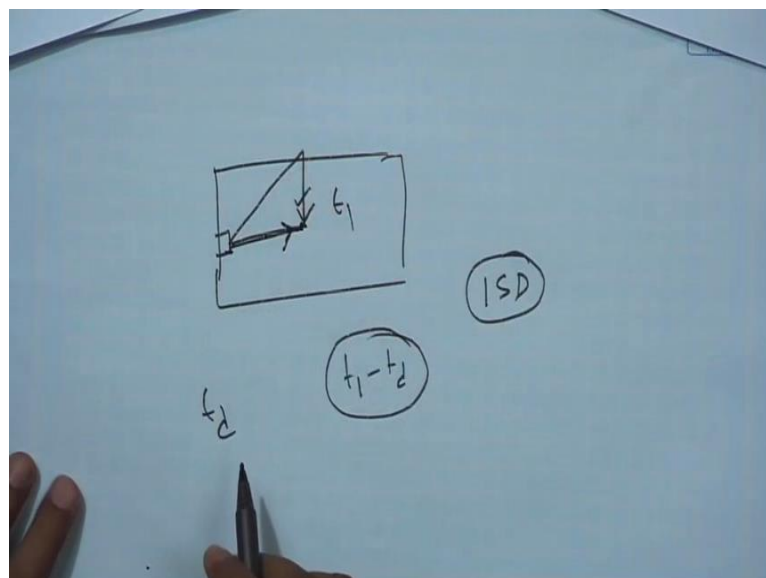
It is the time between the arrival of direct sound LD at a listener's ears and the arrival of first significant reflection.

This is the fundamental room parameter.

For small room ISD is in the order of 1 to 5 ms

Lets two speaker I place in the two side of the wall of the front of the auditorium by stage side of the auditorium then from speaker to any distance I can calculate the percentage ALCONS speech intelligibility then the another term is called initial signal delay gap. So, I said then if in any enclosure room if I hope if I play a sound source then at any point of from the source it will be direct sound and there will be a reflection sound.

(Refer Slide Time: 12:15)





So, I can say if this is an enclosure room and if there is a source in here and if this point there will be a direct sound and there will be a reflection sound from the path of the reflection. So, initial signal delay gap is the time between the arrival the direct sound. So, time lets  $t_d$  is the direct sound time arrival and host reflection lets host reflection term in  $t_r$ . So, the time delay  $t_r$  minus  $t_d$  is the initial signal delay gap. So, this is calling I S D, I S D initial signal delay gap. So, at any point, I can say the distance the time taken by the direct sound difference between the time of reaching a direct sound and fast refection is called initial signal delay gap this is the fundamental room parameter this is used how I S D is used. So, for a small room, for the recording studio, it should be 1 to 5 milliseconds the initial delay should be 1 to 5, millisecond.

(Refer Slide Time: 13:31)


**Live End Dead End(LEDE)**

For design of control rooms for monitoring recording studio this is used

**Front half of the room was made as absorptive** as possible and the other half (the half to the near of the listener) was made as **reflective and diffusive** as possible.

$$L_T = 10 \log( 10^{L_D/10} + 10^{L_R/10} + 10^{L_n/10} )$$

$L_T$  - > Is the total sound in decibels  
 $L_D$  - > Is the direct sound in decibels  
 $L_R$  - > Is the reverberant sound in decibels  
 $L_n$  - > Is the ambient noise level in decibels



Now, another parameter live end and dead end for the small room this parameter is very much useful, live end and dead end. For design a control room for monitoring recording studio this is used. So, front half of the room was made absorptive as possible, and other half was made reflective and diffusive. So, in the small room front half is absorbed it, and other half is reflective. So, the total power  $L_T$  it is continuing by the sound source reflection power and ambient noise. So,  $L_R$  is the reverberation sound, direct sound and noise is contributing to the total sound level. So, direct sound, reflective sound and ambient noise if there is the ambient noise that noise also contributing the sound power.

(Refer Slide Time: 14:30)

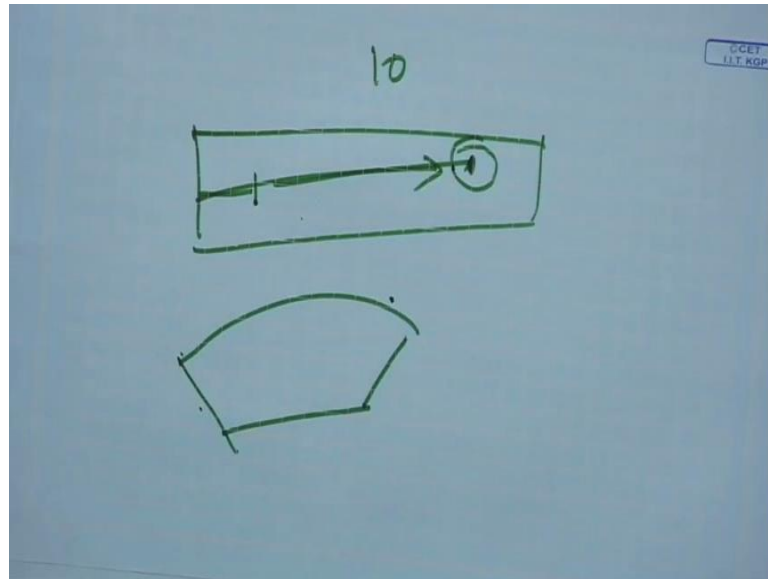
### **Factors to watch for in room**

- Curved surfaces, especially concave curved surfaces
- Absolutely parallel walls such wall cause flutter echo.
- Absorption on the ceiling. Unless the ceiling is very high (over 60 feet) the placement of absorption on its means the sound system lost some useful reflecting surfaces.
- Potential ambient noise sources
- Extra wide or round audience seating

Next factor two reach for a room factors to watch for a room that means, what is the meaning, to design an acoustic studio which are the factor you should look. First is curved surface especially concave curved surface that should not be any concave curved surface. Then absolutely parallel wall such wall that wall cause flutter echo. When you design an auditorium there should not be too absolutely parallel wall. Then absorption on the ceiling if you put absorptive material on the ceiling then you lose sound energy enlarge the ceiling is very high. Now, the placement of the absorption on it means the sound system lost some useful reflection.

Now if it is above 60 feet, what will happen that it clears an echo that in that case to reduce the echo we put absorptive material, but if the ceiling height is not above 60, we have 60 feet then you should not put absorptive material on the ceiling because it unnecessary lose that acoustic energy. Potential ambient noise source there should not be a potential ambient noise source just close to the auditorium. Then the ambient noise level will be increased extra wide or round audience seating. This is very important what I said that in critical distance that after the critical distance the direct sound intensity is less than the reverberant sound then it is creates an effect that the clarity of the speech will be clarity of the sound will be the is m part.

(Refer Slide Time: 16:35)



So, we want that audience should be as close to some factors of critical distance. So, if I make a let call like this and that critical distance is this. So, it may be a ten times the critical distance which means a some person is sitting in here the direct sound and the reverberation sound direct sound is no intensity. So, is there only the reverberation sound? So, intelligibility of loss, so what I said that if you if you make a extra wide range sitting audience extra wide 120 degree you know that extra wide sitting arrangement then this can be avoided.

(Refer Slide Time: 17:09)

### Early Reflections

- The timing of the first reflection is an important aesthetic parameter in auditorium acoustics.
- If the first reflection is delayed by greater than about 35 ms then we hear an **echo**—an undesirable effect.
- Best values obtained by evaluating “good” concert halls are less than 35 ms. 20 ms for an “intimate” hall.
- Even in the presence of reflections we can localize the sound source. If similar sounds arrive at the ear within 35 ms the direction of the source is the direction of the first arriving sound. Note that we only hear one sound—not an echo.

Next, early reflection, the time of first reflection is an important aesthetic parameter in auditorium acoustics. If the first reflection is delayed by greater than the 35 millisecond then what will happen echo will come which is not desirable for an auditorium. So, the first reflection must be within the 35 millisecond, best values obtained by evaluating good concert hall are less than 35 millisecond which is 20 millisecond for an intimate hall; I will come what is intimate means of all those things. So, 20 millisecond is good, but not above 35 millisecond in that case echo will be hard.

Even in presence of reflection, we can localize the sound source if the similar sound arrived at ear within 35 millisecond, the direction of the source forgets about the reading. So, what I said the human being perceives the direction of the sound based on the past arrival of the sound which is past arrival is the direct sound. So, although it will be perceive sound is the direction of the direct sound. So, if the reflected sound is come within 35 millisecond, it does not create another sense is on the same sound. So, we have not hard echo. So, the direction of the sound will be the direction of the direct sound. If it is greater than - 35 millisecond, then we can hear a sound is come from the direction of the source and a sound is come from the reflection of the source the same sound is reflected back reflected.

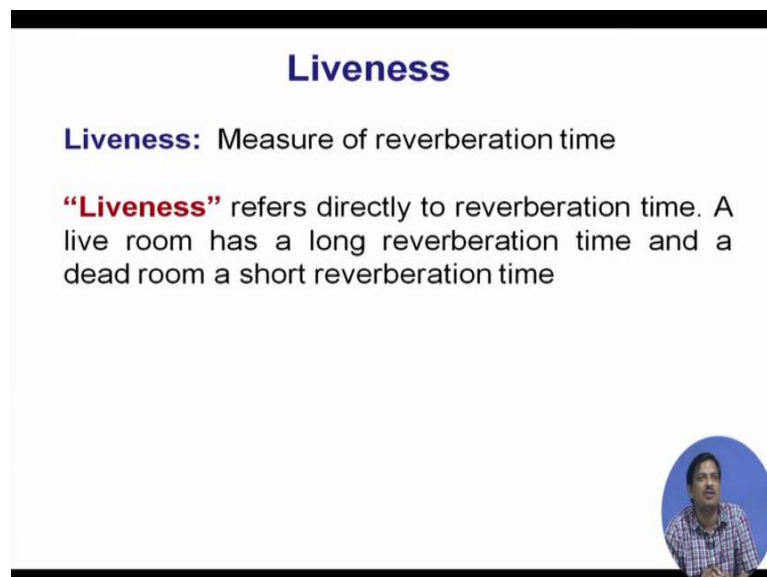
So, echo as if you create an echo what is that direction of the echo direction of the echo is the reverse of the source if you see. If I create a sound in here if it is reflected back and produce an echo after 35 millisecond it reach my ear while heard first sound which is great sound, next sound is the reflected sound assume which is come from my front side. So, the sound which is I perceive direct that is the direct sound direction of the direct sound, if the reflected sounds are within the 35 millisecond.

(Refer Slide Time: 19:33)



Then there is some acoustics parameter for any acoustics room that is the some parameters a measurement parameter one is called liveness, intimacy, fullness and clarity, warmth and brilliance, texture, blend and ensemble. So, all are the seven acoustic parameter which we have to know which are define the quality of the acoustic treatment of the room.

(Refer Slide Time: 20:02)



So, what is liveness, liveness is a measure of reverberation time. So, liveness referred directly to reverberation time, a live room has a long reverberation time and a dead room

has a small reverberation time. So, anechoic chamber the liveness is dead zero this reverberation time is zero, because there is no reflection. So, there is no reverberation time. So, live room of both rooms where the reverberation time is much larger, so that is called liveness.

(Refer Slide Time: 20:33)


### Intimacy

***Refers to how close the performing group sounds to the listener.***

**Room said to be “intimate” when the first reverberation arrives within 20 ms of the direct sound.**

This condition is met easily in a small room, but it can also be achieved in large halls by the use of **orchestral shells** that partially enclose the performers.

Another example is a **canopy** placed above a speaker in a large room such as a cathedral: this leads to both a strong and a quick first reverberation and thus to a sense of intimacy with the person speaking.



Then intimacy refer to how close the performing group sound to the listeners then what is the human tendency suppose a workers they performing on the stage you know that you have to buy a high value tricky to sit in the front of the orchestral; that means, if the audience is feel it is close to the orchestral. So, intimacy is that refer to how close the performing group groups sounds to the listeners if the listener is sitting on the front seat how he feel that he is close to the orchestral. So, room said to be intimate when the first reverberation, reverberation arrived within 20 millisecond of the direct sound. So, if the first reflection to the listeners arrives within 20 millisecond.

So, I make arrangement of the all reflector such that that direct sound and reflection sound difference between the two sounds in time is 20 millisecond then a person will feel that the I have intimate with that the orchestral which is performing in the stage. So, this condition is met easily in a small room. So, if it is small room the reflection part is less. So, it can be easily met, but it can also be achieved in a large hall by use the orchestral shells that particularly enclose the performers. So, enclose the performer means you increase the directivity. So, the reflected sound will come the reflected sound

insensitivity will be very high then you know that in chart also that this is the area how you can increase the intimacy then fullness and clarity fullness. What is fullness? The amplitude of the reverberant sound relative to the direct sound is referred to the fullness.

(Refer Slide Time: 22:23)

### Fullness and Clarity

**Fullness:** The amplitude of the reverberant sound relative to the direct sound is referred to as **Fullness**. More the reflected sound more 'full' will be the hall.

**Clarity:** Ability to distinguish and articulate individual sound. The opposite of fullness, is achieved by reducing the amplitude of the reverberant sound.

A fuller sound is generally required of Romantic music or performances by larger groups, while more clarity would be desirable in the performance Mozart or in speech

$$C80 = 10 \log \frac{\text{Energy arriving within 80 ms of direct sound}}{\text{Energy arriving later than 80 ms after direct sound}}$$

So, I said the amplitude of the reverberation sound relative to the direct sound. So, if the reverberation sound by direct sound is actually defined the fullness. So, more the reflector sounds more the fullness. So, fullness is nothing but a reflected sound divided by less direct sound intensity, reflected sound intensity by direct sound intensity. Amplitude means it can contribute the intensity. So, if that is the case then the fullness will be more if the reverberation sound intensity is high. So, reverberant sound is pre dominating in the direct what the direct sound means close relation. So, if it is equal then the fullness is 1.

But what is clarity ability to distinguish and articulate individual sound, the opposite of the fullness. So, if it is suppose in a stage that is a three performer, somebody is playing guitar, somebody is playing sitar, somebody is performing vocal list. So, clarity is the ability to distinguish that articulator individually. So, how am I able to individually identify those sound is defined by the clarity which is opposite to the fullness if the reverberation sound is a ensemble sound direct sound is coming from the sitar then there will be a there will may be guitar and there will be a vocal list. So, if I close my eyes, I



understand that yes individually, I can identify the sitar is sitting in this side, guitar is sitting in this side, and vocal list is sitting in this side.

So, I identify that thing that direct sound give me, but if the reverberant sound is also very intense then what will happen the fullness is very high, but the clarity will be confused, because all reflection will come to get together to me. So, clarity will be confused. So, clarity is the opposite of fullness of the sound. So, clarity C 80 is defined by energy arriving within 80 millisecond of the direct sound and energy arriving later than 80 millisecond after the direct sound. So, what is the reflected sound within 80 millisecond of the direct sound? Some energy will come and energy that is defined by the clarity.

So, while more clarity would be, a fuller sound is generally required of romantic music or performance by larger groups. So, I required ensemble or you can say that pool sound that I do not want to clarify who is performing because there is a huge and large group is performing. So, individual artist I do not want to identify. So, what I want I want a pool sound, sound intensity should be very high. So, in that case I go to the fuller sound, but the clarity sound suppose somebody giving a lecture then I want to listen that lecture and somebody is playing a or somebody is vocal list and there is a sitar there is a guitar I want interested on the vocal list. So, I critically identify the vocal list. So, clarity is very important to me. So, in that case that is why the fullness and clarity is defined

(Refer Slide Time: 26:03)

### **Warmth & Brilliance**

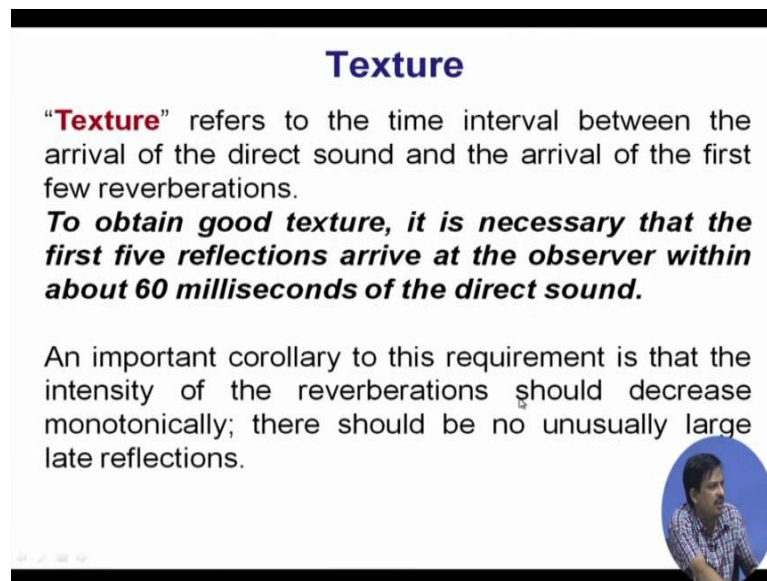
*refer to the reverberation time at low frequencies relative to that at higher frequencies.*

**Above about 500Hz**, the reverberation time should be the same for all frequencies. But at low frequencies an increase in the reverberation time creates a **warm sound**, while, if the reverberation time increased less at low frequencies, the room would be characterized as more **brilliant**.



Then which parameter refer to the reverberation time at low frequency relative to that at higher frequency; that means, above about 500 hertz the reverberation time should be the small for all frequency. But at low frequency an increase in the reverberation time create a warm sound while if the reverberation time increase is less at low frequency it create a brilliant sound. So, this is the definition of warmth and brilliance then texture, texture refer to the time interval between the arrival of the direct sound and the arrival of the first few reverberation sound this is five reverberation.

(Refer Slide Time: 26:33)




**Texture**

“**Texture**” refers to the time interval between the arrival of the direct sound and the arrival of the first few reverberations.

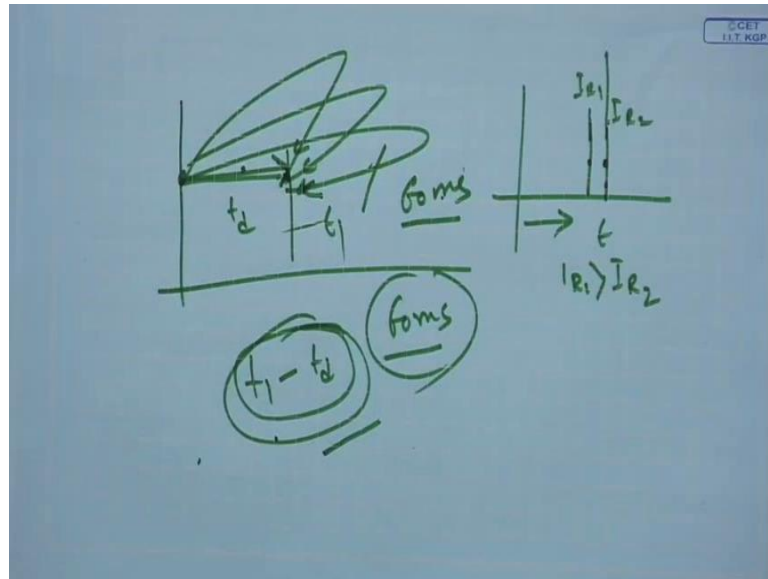
***To obtain good texture, it is necessary that the first five reflections arrive at the observer within about 60 milliseconds of the direct sound.***

An important corollary to this requirement is that the intensity of the reverberations should decrease monotonically; there should be no unusually large late reflections.



So, this texture is defined by the direct sound arrive interval between the arrival of the direct sound and first few reflection sound. So, to obtain a good texture this is necessary that the first five reflections arrive at the observer within about 60 millisecond of the direct sound that is the good texture within the 60 millisecond some reflection must at least 5 or 6 reflection must come then I give a good texture room. So, an important corollary to this requirement is that the intensity of the reverberation sound should decrease monotonously. So, there is assumption that is that is that the intensity. So, actually what is the meaning, meaning is that suppose this is the source. So, I said the texture will be good.

(Refer Slide Time: 27:03)



If the direct sound the time interval between the reaching of the direct sound and few five lets five or six reflective reflection five reflection first five reflection as minimum as possible if it is decrease the texture will be good. So, it should be within 60 millisecond at least then the texture will be said as it at good structure, so as low as possible. So, difference time. So, I said the first I have measure the direct sound arrival at this point and then measure the first reflection arrival time second reflection arrival time when the peak reflection arrival time is ten let say this is  $t_1$  and direct sound is  $t_d$ . So,  $t_1$  minus  $t_d$  is the texture, it defines the texture.

So, if it is within 60 millisecond then we said texture is good, but suppose there is such a reflector the peak reflection or fourth reflection come intense compare to the first reflection then I feel little bit of uncomfortable some time sound is bouncing and then going down bouncing and going down. So, I do not want that things. So, generally what happen since the first reflection path and second reflection path is first reflection path is less than the second reflection part as per the inverse square law the 3 d B down will be there decade. So, I can say the first reflection intensity if it is  $I_{R1}$  if it is  $I_{R2}$   $I_{R1}$  must be greater than  $I_{R2}$  because of path is different large number of part is taken in the second reflection it is the time. Now if say let second reflection is very high then it create uncomfortable. So, then texture is not good. So, in that texture will be good. So, if this third reflection come within 60 millisecond and the intensity of the each reflection gradually decrease, linearly decrease then we said the texture of the room is good.

(Refer Slide Time: 29:56)

### **Blend and Ensemble**

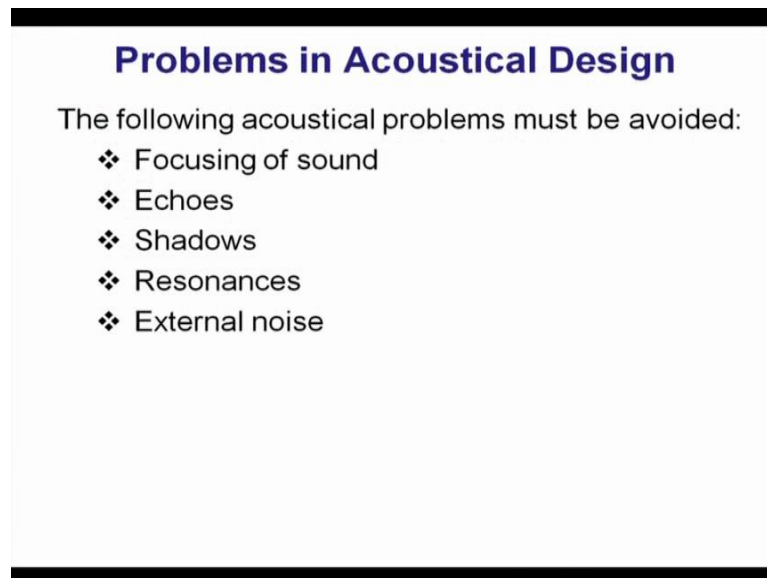
“**Blend**” refers to the mixing of sounds from all the performers and their uniform distribution to the listeners.

“**Ensemble**” refers to the ability of the members of the performing group to hear each other during performance, enhancing the ability of the players to play together effectively.

Then the blend and ensemble blend refer to the mixing of sound from all the performer and their uniform distribution to the listener. What is blend? Blend means blending sound blending, so mixing of the sound. Suppose, in this stage, there is a lot of musician. So, blending means the mixing of the all sound and uniformly distributed to the listeners. So, if a listener is sitting in front of the stage and the listener in the sitting in the just far away from the stage both listener should not say the first listener is heard that guitar is playing loud and second last listener is heard, now tambala is playing loud that kind of thing should not be there. So, blending of the sound must be proper to all listeners then ensemble refers to the ability of the member of the performing group to hear each other during the performance.

If you say in the stage, any audio stage, any musician perform if you see there is a reflector on the stage that is call monitor reflector. So, that since the performer is good sitting in the stage they cannot heard the reverberation sound if is the reverberation will come from the backend of the stage and the reverberation sound will be not audible to may intensity is very less. So, what they want, they want a extra sound reinforcement to hear that other performer sound that is why there is a reflector is placed on the stage or monitor is placed on the stage to performer performing group to hear each other during the performance that is called ensemble.

(Refer Slide Time: 31:45)



**Problems in Acoustical Design**

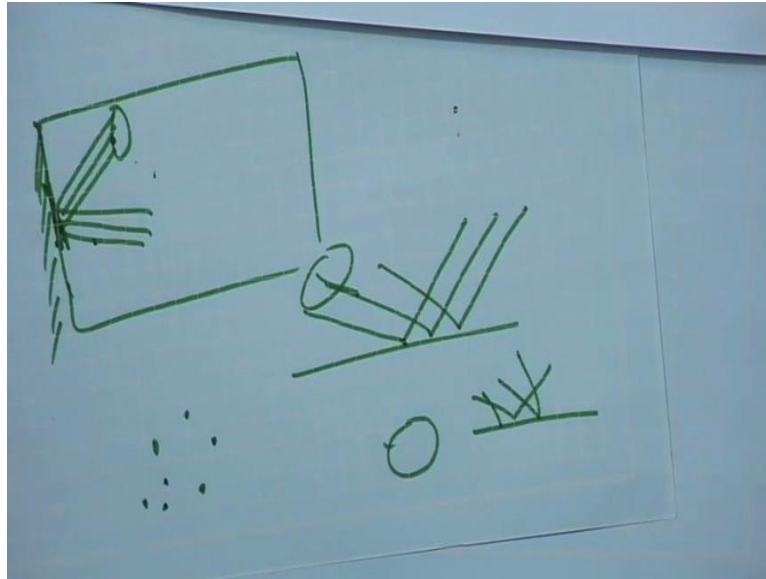
The following acoustical problems must be avoided:

- ❖ Focusing of sound
- ❖ Echoes
- ❖ Shadows
- ❖ Resonances
- ❖ External noise

Then what is the problem in acoustics design that is the few problems in acoustical design we have to avoid focusing of sound when I design auditorium there should not be any particular place where the sound is focused. We have to have avoid echoes there should not be any echoes we have to have avoid shadow what is sound shadow we discussed diffuser in diffraction if there is a obstacle of the sound it create a sound diffraction.

So, if due to the diffraction shadow will be created. So, I do not want a shadow region in the auditorium then I want do not want the resonance there will be a resonance is happen in the auditorium because I want a reverberate sound feel in a auditorium then I want external noise I should not want the external noise source should enter in the auditorium. So, those are the factors I would taken care during the design acoustic design of the auditorium one is called focusing of sound there should not be suppose I want to design an auditorium.

(Refer Slide Time: 32:56)



Let I want to design an acoustic room or auditorium like this and I put a regular reflector here. So, if a sound is come in this direction and it will be going this direction. So, this portion sit will be focused. So, I do not want the focusing of the sound. So, I do not want there will be regular reflection in the surface. So, if I want to avoid the regular reflection what I required I required a diffuser surface what is diffuser. So, what is regular reflection regular reflection is that if the surface is very, very smooth. So, there will be a regular reflection angle of reflection and angle of this ray will suppose the sound is coming from this direction and it going in a signal direction. So, I want irregular or diffused reflection; that means irregular random reflection. So, in that case all direction the reflection will happen. So, each point of the auditorium, the reverberation field will be same.

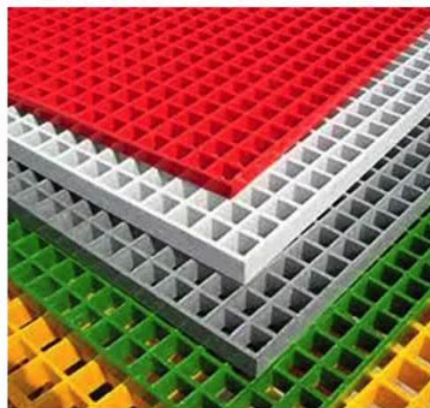
(Refer Slide Time: 34:18)

### How do diffusers work?

- Two basic methods
  - Random scattering from a roughened or textured surfaces. Easy to make but not predictable in response.
  - Diffraction by profiles that possess “all” necessary grating spacing to ensure a uniform diffraction pattern.

So, I have to treat the wall of the auditorium or the reflector of the auditorium is by a diffused reflector. So, what is the diffused reflector, there is some material is available on diffused reflector can like that random scattering from a roughened and textured surface. So, if the surface is very rough then there will be random reflection will be happened, but easy to make I can make a rough surface of the auditorium I can instead of polishing the wall of the auditorium I can make a rough surface, but not predictable in response I do not I cannot predict the response. So, I required a control diffuser.

(Refer Slide Time: 34:52)




So, what are the control diffusers if you see this kind diffuser is there an acoustic tile which is treated in the wall.

(Refer Slide Time: 34:58)

### Quadratic residue method

- A method of designing a multilevel diffuser that operates over a greater wavelength range.
- Sequence of depths  $d_n$  is generated by
$$d_n = \frac{\lambda_0}{2p} s_n$$
- Where the sequence  $s_n$  is defined by
$$s_n = n^2 \bmod(p)$$




So, quadratic residue method one of the design method for diffuser is quadratic residue method I am not going very details, but at least you should know what are the diffuser is available a method of designing multilevel diffuser that operate a over a greater wavelength range. So, frequency response and design of the diffuser there is a relationship. So, if you see there is a acoustic style there is a quadratic, so there is a if you see this is the tiles there is a depth. So, this depth is not random, this depth is designed if it is quadratic residue diffuser depth is nothing but a lambda 0 is the fundamental wavelength divided by 2 p into s n. p is an integer, s n sequence which can be generated using a n is a integer n various point 0, 1, 2, 3, 4, 5, 6. Let us p is equal to 16 then I can get a s n sequence then I can get the depth sequence.

(Refer Slide Time: 36:06)

### Well width and diffuser bandwidth

- Maximum well depth should be 1.5 times wavelength of lowest frequency of operations
- Well width should be 0.5 the wavelength of the highest frequency of operation
- Highest frequency to lowest frequency define the operating bandwidth of the diffuser


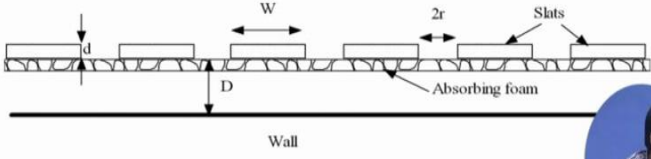


Then there will be a well width and diffuser bandwidth. So, if you see there is a slot like this way, wall then we have a slot, slot diffuser you put a slot.

(Refer Slide Time: 36:13)

### Slot Absorber

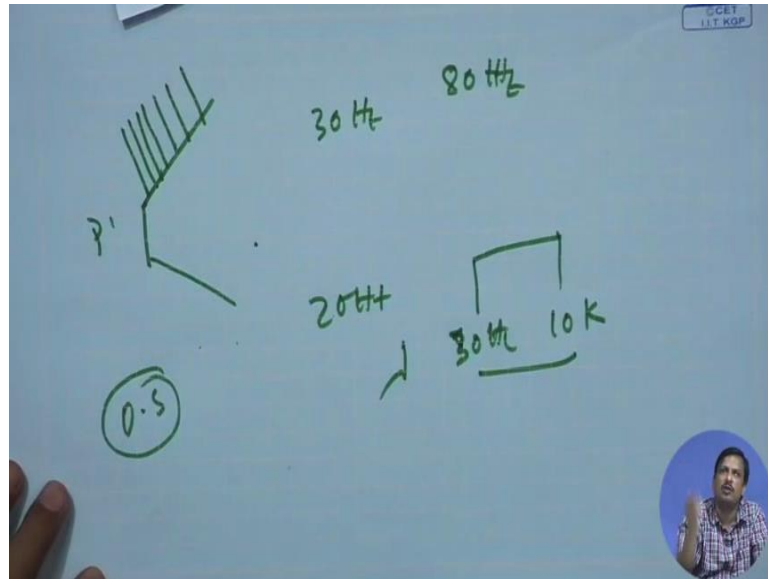
- One example of absorber based on Helmholtz resonator
- Slotted panel that is spaced away from one of the walls of the enclosure.



If you see the auditorium the front part to just enclosure to the stage this curves to the stage there is a slot wooden slot, slot is placed on the wall. So, this slot width and depth is not random it has to be a designed best on that requirement. So, you know the frequency of the operation of the auditorium then we have to design the depth on based on that equation.



(Refer Slide Time: 36:23)



So, maximum while depth should be 1.5 times the wavelength of the lowest frequency because if it is support from the lowest frequency you support for the higher highest frequency. So, the maximum while depth should be 1.5 times the lowest frequency of operation of the auditorium. So, if it is a music studio, a music auditorium then it is 30 hertz; if it is only for a speech auditorium then it is 80 hertz is the lowest frequency. So, I know the lowest frequency then I know the wavelength and then 1.5 times of the wavelength of the depth. Then while width should be 0.5, the length of the highest frequency of the operation, it will be the 0.5 times the highest frequency operation. So, of I say music 20 kilo hertz then I know the lambda then I find out the width then the highest frequency to the lowest frequency define the operating bandwidth of the diffuser. So, if a diffuser work for lets 30 hertz to 10 kilo hertz then I can calculate the depth and width of the slot then I according to do treat that thing.

(Refer Slide Time: 38:15)

### Uses of the Slot Absorber

- Reduce low frequency reverb time without affecting high frequency reverb time.
- Suppress low frequency standing wave resonances
- Absorption can be varied by placement of foam either close to opening or set back between the wall and the slats.

So, there is a number of other diffuser also there is a slot absorber is there that use slot absorber to reduce the slot absorber. So, there is a diffuser. Now there is a absorber also what I said I have to make the I have to make the auditorium such that there is a no focusing of sound that is the job of the diffuser. So, all the reflective surface must be treated as a diffusive reflection no poly surface if it is poly surface regular reflection which has to be avoided.

Then I have to avoid the echo. So, if my far end lets the auditorium is a huge large auditorium if the far end is the very lengthy then may be the reflection come in 65 milli more than 65 millisecond. So, I do not want the reflection more than 65 millisecond reflections should come. So, all the far end must be treated as the abortive material which can absorb the most intensity of the sound. So, there will be reflection insensitively greatly reduced. So, that there will no echo. So, the far end of the auditorium must be absorptive and the sometime some absorption absorptive material is required in the middle also because that also created echo for some point.

(Refer Slide Time: 39:53)

### Perforated panel absorber

- Yet another version of the damped Helmholtz resonator

p=perforation percentage;  
D=air space;  
t=effective hole length (panel thickness +0.8\*hole diameter)

Use meters for all measurements.

$$f = 5.4 \sqrt{\frac{p}{Dt}}$$

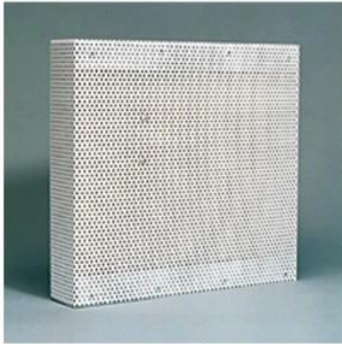
So, absorption there is some absorptive material also required. So, what kind of material is there for absorption I required a slot absorber I can use a slot absorber, you can read the slide for that use of the slot absorber or I can use the perforated panel absorber where the frequency is formula is given.

(Refer Slide Time: 40:02)

### Industrial Panel absorber

- Absorption coeff.

– 125 Hz	0.22
– 250 Hz	0.77
– 500 Hz	1.12
– 1000 Hz	1.00
– 2000 Hz	0.78
– 4000 Hz	0.57



So, perforated panel absorber like this; this is an absorber acoustics absorber. So, absorption coefficient is given for different frequency. So, while you buy a absorber from the market or acoustic styles from the market the absorption coefficient with the

frequency will be rate in that packet. So, you can use that material once you ticket the wall using that material you know the total absorption a in Sabin then you that will affect the reverberation time.

So, when you design the auditorium, you should thought or you should explain which portion of the auditorium should be absorptive, which portion of the auditorium should be reflective and which portion of the auditorium should be more absorptive. So, that kind of treatment you will make when you design that auditorium. But for design any studio or acoustic room or even if you are drawing room acoustics that it should be there is no focusing of sound that is no echo, there is no shadow, there is no resonance and external noise must be minimized. If you see that built an auditorium near the highway line and I do not take care about the highway noise to pass the auditorium then auditorium is useless, because the highway noise is easily enter that auditorium. So, how to avoid the external noise keep the external noise source most away from the auditorium there is the basic consideration for auditorium design.

(Refer Slide Time: 41:39)

- Room treatment depends greatly on the purpose of the space—classroom, musical auditorium, small vs large space...
- Main parameters that affect experience—reverb time (large spaces), early reflections, standing wave resonances (small spaces).
- Control methods—absorptivity, diffusers, low frequency traps

Now, there is another things room treatment. So, I said that room treatment is how you design the auditorium it depends on what, what is the purpose of the designed auditorium. If it is auditorium is design for a classical music or performance of the classical music or orchestra then the design tit band will be different. If it is only for a

lecture theatre like class room then the reverberation time requirement will be different and the treatment of the acoustic treatment will be different.

So, next class, we will discuss about how to design an auditorium acoustics for which part of an auditorium should be treated in which kind of material and why. And the most common phenomena are that there should not be any parallel wall while you design and acoustics auditorium or even in room acoustics. If it is parallel it creates a, so there will be at least some sliding of the wall which is good for the design a very good acoustics room.

Thank you.