

Acoustics
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Module 7 Lecture 4
Room constant, and Sabine's coefficient

So in the last couple of lectures we have talked about sound propagation in regular rooms and also in irregular rooms. In case of irregular rooms we developed a parameter reverb constant and while developing this parameter we had assumed that there is no damping in the air but the walls themselves have some dampening effect and they do absorb some sound. The context of all this discussion regular rooms and irregular rooms is that if there is some sound which is generated in a room how does it propagate and it has direct application in areas of in places in acoustics places lecture halls, auditorium, kettles places where do there recording even small rooms.

I wanted to develop this reverb time further and also what we will see today is what does air do to this whole notion of reverb time but before I start doing that I wanted to generate some thinking that if you have a let us say big room lecture hall where you sit and take lectures and there are 200, 300 students sitting in the room, what are different things as an engineer we would think which will help you design a good lecture hall can you site some parameters we do not have to be very well defined engineering but what will make as a listener a good room and it will distinguish it from a bad room.

No time delays,

Student: because base and the sound reaching us.

Professor: Why should that matter if the speaker generates a sound and it reaches you after in an upside case 5 minutes later, what does that do? Why is it bad or good?

Student: It will we lose the synchronization and the harmony.

Professor: But first later reaches you after 5 minutes second letter reached you after 5 minutes plus some Δt so.

Student: If we do not watch him then it is okay.

Professor: Okay, understood there is no synchronization between his lip movement and understood, okay. So okay that is one that know significant delay between the sound source and in the time when the sound is coming out and when it reaches your ear because there is will be a disconnect between visual perception and auditory perception so that a good, you wanted to say something.

Student: Sir there should not be any nodes and in there room the sound uniform all over the room.

Professor: Sound should be uniform all over the room, okay and why is that good or bad or why is that good.

Student: I mean, if one person is sitting like this and if he changes his position he should hear the same volume like the same sound.

Professor: Second anything else?

Student: There should not be echo.

Professor: There should not be any echoes, so why are echoes bad?

Student: Then we will hear double sound like.

Professor: So echoes is there, anything else?

Student: Sir the main importance should be on the vocal frequencies.

Professor: Main importance should be on the vocal frequencies, why?

Student: For vocal frequencies should be less.

Professor: In the lecture room?

Student: Yes.

Professor: But not necessarily in a place where you are paying visit.

Student: Yes sir.

Professor: It depends on the application that how the room behaves should be (())(4:03) to whatever is the band in which sound is being generated. What about noise from outside?

Student: Noise cancellation.

Professor: You want to minimize that noise which is coming from outside has to be minimized echoes you already mentioned. So we will also cover some of these other things which we talked about.

Student: There should be no resonance with objects.

Professor: No resonance?

Student: I have seen many a times this chairs and if it is a specially metallic objects they start to vibrate and make sound in the base room.

Professor: In the room.

Student: If there is a loud bass, objects begin to chatter.

Professor: So there should not be any buzz, chatter, rattle, tricks these kind of. Yes, so we will cover some of these we will definitely cover echoes, we will cover some criteria so that we can figure out what is a good permit not good acceptable noise level as it comes from outside. We will also cover a term called room constant we have talked about reverb time and we will also introduce a term called room constant so that is all that is the spectrum things we will talk today but we will start by including the effect of air damping on reverb time.

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The whiteboard contains the following handwritten notes and equations:

- Top left: $T = \frac{60V}{1.08\alpha c}$
- Top right: $\alpha' = -S \ln(1 - \bar{\alpha})$
- Middle left: D_0 (with a double underline)
- Middle right: d and $D_0 e^{-md}$ with $m \ll 1$ and $\frac{1}{m} = \frac{1}{4\alpha}$ written nearby.
- Center: $D(t = n\Delta t) = D_0 (1 - \bar{\alpha})^n e^{-nmd}$. The term $(1 - \bar{\alpha})^n$ is highlighted in yellow, and e^{-nmd} is highlighted in cyan. A note to the right says "After n reflections".
- Below the center equation: $= D_0 (1 - \bar{\alpha})^{\frac{t}{\Delta t}}$ and $= D_0 (1 - \bar{\alpha})^{\frac{t}{\Delta t}}$ (with $\frac{1}{\Delta t} = \frac{c}{4V}$ written to the right).
- Bottom right: $\frac{d}{\Delta t} = c$ and $\frac{1}{\Delta t} = \frac{cS}{4V}$.

We had seen that reverberation time is $60 \frac{V}{1.08 \alpha c}$, right? (5:29) and we had defined a prime c as minus S natural log of $1 - \bar{\alpha}$. So what we will do is we will modify this particular equation by incorporating the damping of damping generated due to air. Let us say D not is my initial energy going into the system and once sound travels a distance d which we had defined in earlier lectures as mean free then just because of air this d not it goes down and it becomes d not e to the power of minus where m is fairly small compared to 1 and its dimension is 1 over length so dimension could be of m is 1 over meters or 1 over feet.

So using the logic we had used in our previous lecture we can say that D which is at the energy density at a given point of time time equals $n \Delta t$ equals D not $1 - \bar{\alpha}$ and e^{-nmd} this is after n reflections, this part the one in yellow is because of the reflections of the wall this part is because of the role of the air so you have n reflections so it is n times m times d . We know that so we can develop this further so that is D not $1 - \bar{\alpha}$ and n is what t over Δt , right? And then the other exponent part is minus again I will replace n by t over Δt so it is t times md over Δt .

Now we know that d over Δt is what? D is the mean field part, c , right? Velocity of sound so I will put that in here so I get D not $1 - \bar{\alpha}$ t over Δt e^{-tmc} we also know that 1 over Δt is basically cs over $4V$ we had seen this earlier, so I modify this further D_0 $1 - \bar{\alpha}$ average α so I get cst over $4V$ exponent minus tmc .

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The image shows a whiteboard with handwritten mathematical derivations. At the top, there is an equation for D involving a complex fraction with terms like $\ln(1-\bar{\alpha})$, $\frac{Sv}{4v}$, and $4m\omega$. A red box highlights a term $-\{s \ln(1-\bar{\alpha}) + 4m\omega\}$. Below this, the expression is simplified to $D_0 e^{-\frac{d}{4v} \frac{Sv}{4v}}$. Further down, the reverberation time T_{RIR} is calculated as $\frac{60 \cdot V}{1.086 \cdot d_{air} \cdot C}$. At the bottom, there are two notes: (i) $m \rightarrow$ depends on ω , as $\omega \uparrow \rightarrow m \uparrow$ and (ii) $d'_{air} \approx d_{air}$.

So now what I do is like I did for earlier case I express this entire thing also in an exponential format so I get D equals D_0 not exponent so I get $\ln(1 - \alpha)$ as set over $4v$ minus tmc and if I rearrange my stuff what I get is exponent minus $s \ln(1 - \alpha)$ plus $4m\omega$ times ct over $4v$ I can make this whole thing as a prime here because now it includes the effect of damping also of the air.

So I get $D_0 e^{-\alpha'_{air} \times t}$ over $4v$, so this is essentially very similar to the earlier expression we had developed for a room with no damping due to air so I can use the same logic stream and I can say that reverb time if I have to include the effect of air is basically 60 times volume divided by 1.086 α'_{air} times velocity sound in air. So couple of things one is that m is m parameter which is it actually very strongly depends on ω so this depends on ω and the relationship is that as frequencies go up m goes up.

So as ω goes up we see through experimental data ω also m goes up. So once again we have to be careful that when we find the value of a prime its specific for a particular frequency or an arrow frequency band is important to understand. Second thing is that a prime air is approximately equal to a prime with for small values for small rooms if my v is a small then this term will tell in significance compare to the other term.

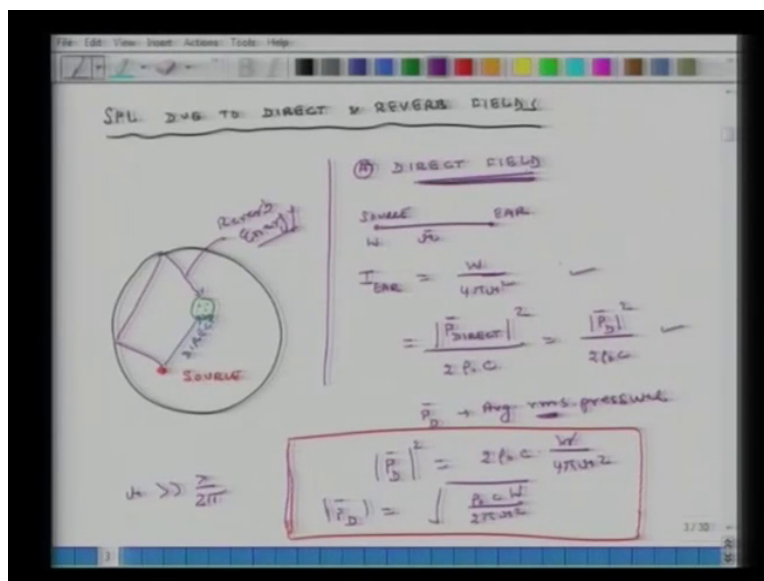
So for small rooms I may still get away without including the effect of air damping but for very large rooms, auditoriums, halls it should not use the damping effect. This completes the effect of

air damping and what kind of an influence it has on reverb time. So does the reverb time go up or go down as I include the effect of air damping it go down because this term goes up and physically what it means is that what is the original fundamental definition of reverb time.

Student: Sir the sound decays by the 60 decimal.

Professor: Yeah, which is a factor of a million so it will take lesser time to decay so that floor level if I include the effect of air damping which makes sense. So this completes our discussion on the reverb time and now what we will do is we will move to a new concept and this is also used in architectural acoustics and it is essentially trying to develop a ratio of how much energy is received at the ear of the listener which is directly coming from the source and how much energy is coming from reflected you know due to reflections.

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What we will discuss is SPL due to direct and reverb field, I have a big I have my source and then I have a person some energy is coming directly to the individual direct and some other energy goes and it as a consequence of reflections it comes so this is my reverb. The minimum amount of reflections are reverb energy is what direct field there is no reflection the minimum amount of reflections which a reverb energy will have will be 1 at least 1 will have see at least 1 reflection.

In very large auditoriums typically if you are far away from the source in general the reverberant energy dominates compared to the direct field we will see that as we develop the mathematics of it. And the SPL depends on several parameters it depends on volume, it depends on how far you are from the distance of the source, depends on power of the source and it depends on alpha and a bunch of other parameters. So first we will do is direct SPL direct field, so I have a source this is my ear or microphone.

Student: Sir reverberant energy how can it dominate? (14:45).

Professor: We will see that, and this is the distance between the source and the ear I mean which is the intensity at my ear and let us say the source is emitting w watts of energy, so I mean is what? Based on some of the earlier concepts we have developed power per unit area, right? So it is watts over $4\pi r^2$ and we know that we can also write this as $P_{\text{direct}} / (4\pi r^2)$ I have to average this divided by $2\rho c$ and I can write this in a short form over $2\rho c$.

So again note that P_{D} is average rms pressure from this relation and from this relation what I get is P_{D}^2 is essentially $2\rho c$ times whatever is the strength of the source divided by $4\pi r^2$ or P_{D} equals the square root of $\rho c w / (2\pi r^2)$. So this is my first relation and we will apply the standard disclaimer that this everything of this nature is valid to the extent my r is what?

Student: (16:36) $\lambda / 2\pi$.

Professor: Yes, this is really large than $1/6$ of the wavelength of the frequencies we are considering.

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⑤ REVERB. FIELD

(i) INPUT POWER = $W(1-\alpha)$ → Power flow into the room ①

(ii) Power going out = $\frac{\text{Energy going out}}{\text{Time}}$ ②

= $\frac{\text{Energy lost per reflection}}{\text{Time bet. two reflections}} = \frac{\text{Energy lost per ref.}}{\Delta t}$

$\Delta t = \frac{d}{c} = \frac{4V}{5c}$

$D_{Re} = \frac{|P_{Re, rms}|}{4\pi c^2}$

$E_{Re} = D_{Re} \cdot V = \frac{V |P_{Re, rms}|}{4\pi c^2}$ ③

Total energy lost per reflection = $D_{Re} \cdot V \cdot \alpha$ ④

So this is my direct so now what I will do is I will develop something for the reverb field we do it for a reverb field, so for the reverb field we have to think how the sound develops in the room, if the room has absolutely no dampening surfaces and sound does not leak out of the room then as my speaker is pumping energy into the system what will happen to the reverb energy over a period of time it will just keep on going on increasing infinitely.

If there is at least one dampening surface which absorbs some energy or if there is air which is also absorbing energy then the reverb field the way it will behave is that it will keep on growing but the rate of growth will start over a period of time it will start slowing and it will become flatten it will flatten out and at a certain after a certain point of time whatever is being pumped in to the speaker is getting absorbed by the walls and there is some sort of an equilibrium.

So what we are trying to do in this analysis is that we are trying to find that what is that equilibrium state, so it is not the (())(18:07) state what we are trying to figure out is what is that equilibrium state at which the reverb field does not grow over a period of time. So in this analysis what we will essentially do is we will compute whatever is going in input power and whatever is going out which is the output thing and once they are equal then I can that is my condition for equilibrium.

Input power equals what? What is the input energy going in the system or power going into the system, it is.

Student: Whatever is pumped by the (ρ) (18:40).

Professor: No but it has to see at least one reflection that is going in so what is that value, what is being pumped by the system w ?

Student: w times 1 minus infinity.

Professor: So this is power flow into the room, now what we will do is whatever is going out so power going out equals energy going out divided by energy going out divided by time, right? I will number them 1, 2 this is 2 power going out is essentially power loss per reflection, so that is energy lost per reflection and then what is the time? Time is Δt , right? So that is time between two reflections so this is energy lost per reflection over Δt and we know that Δt equals what d over c and it is also over v over S_c .

So now D_r we define the term D_r it is reverb and energy density and that is essentially E_R upper case R implying its reverb pressure rms the whole thing square divided by ρ not c square this is from earlier notes and E_R is reverb energy in the room, so what is reverb energy in the room in terms of D_r ? D_r is reverb energy density reverb energy in the room is?

Student: into Δt .

Professor: This is energy density D_r is energy density unit of energy density is joules per cubic meter for instance in a (ρ) (21:21). So total reverb energy in the room is what?

Student: (ρ) (21:25)

Professor: Times volume, so it is D_r times volume so that gives me V P_R dot rms the whole thing square over ρ not c square 3. And total energy lost each reflection will be how much in terms of E_R ?

Student: E_R into 1 minus α .

Professor: E_R times α energy lost each reflection is E_R , so total energy lost per reflection E_R times α which is same as D_r times V times α , 4. So whatever is the flow going in we know w 1 minus α and that equals whatever is going out, so now we equate them we have to do one more step this is the total energy lost per reflection so total power which is decaying per

reflection which gets attenuated per reflection or it gets deducted per reflection total power will be how much?

Student: Sir w times alpha.

Professor: No no, in terms of this we know now we know what is energy loss per reflection is D_R times V times alpha, so in terms of this power divided by delta t.

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Power decay/reflection = $\frac{D_R \cdot V \cdot \alpha}{\Delta t}$ ←

$$W(1-\alpha) = \frac{D_R \cdot V \cdot \alpha}{\Delta t} = D_R \cdot V \cdot \alpha \cdot \frac{0.5}{4V}$$

$$= \frac{V \cdot |P_{R,rms}|^2 \cdot \alpha \cdot 0.5}{R_0 c^2 \cdot 4V}$$

$$|P_{R,rms}|^2 = \frac{W(1-\alpha) \cdot 4R_0 c}{3 \cdot \alpha}$$

$$R_0 = \frac{3 \cdot \alpha}{1-\alpha} \text{ m}^2 \text{ or ft}^2$$

$|P_{R,rms}|^2 = \frac{4 \cdot W \cdot R_0 c}{R}$

 ← Remarks: Pressure

$$|P_R|^2 = \frac{4 \cdot W \cdot R_0 c}{R}$$

$$|P_D|^2 = \frac{W}{4\pi r^2} \cdot R_0 c$$

$|P_T|^2 = W \cdot R_0 c \left[\frac{4}{R} + \frac{1}{4\pi r^2} \right]$

 ← UNCORRELATED

SPL = $10 \log_{10} \left[\frac{|P_{T,rms}|^2}{P_{ref}^2} \right]$

So power loss I will not say lost power decay per reflection equals D_R times V times alpha over delta t and this equals whatever is going in which is w times $1 - \alpha$, right? So w times 1

minus alpha equals $D_r V \alpha$ over Δt and I substitute the value of Δt as $c s$ over $4 V$ so what I get is $D_r V \alpha$ times $c s$ over $4 V$ and now what I do is $(\alpha)(23:53)$ and my if I make use further this particular relation D_r equals $P_R \text{ rms square}$ divided by $\rho c \text{ square}$ and I plug in there what I get is V times $P_R \text{ rms square}$ divided by $\rho c \text{ square}$ alpha bar times $c s$ over $4 V$.

So I do the math and I simplify this and essentially what I get is reverb rms pressure which is averaged in time and space the square of it is basically whatever the speaker is pumping in times $1 - \alpha$ over $s \alpha \text{ bar}$ times $4 \rho c$ and now I introduce a new term called room constant R and R is $s \alpha \text{ bar}$ divided by $1 - \alpha \text{ bar}$ and the unit of R is meter square or $(\text{m}^2)(25:07)$ square.

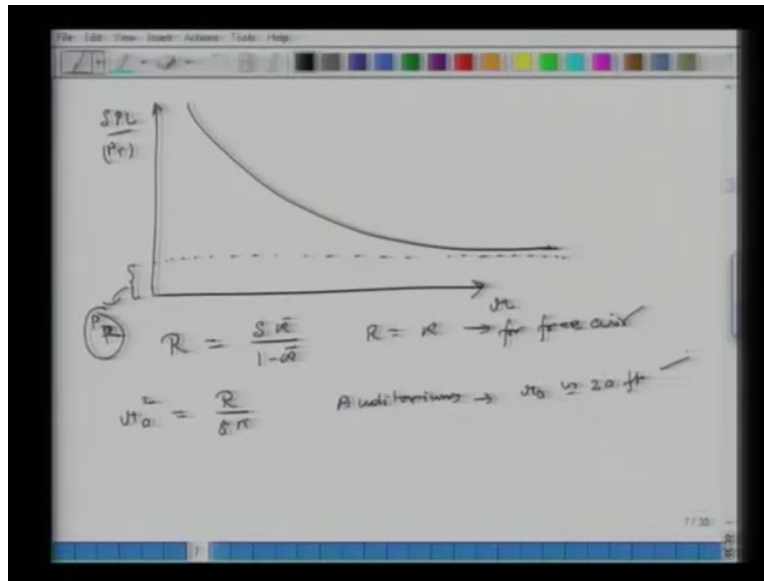
So $E_R \text{ rms}$ is basically $4 W \rho c$ over R , so that is my reverb pressure I will rewrite these two relations, my reverb pressure is $4 W \rho c$ over room constant and my direct pressure is what watts over $4 \pi \text{ square}$ times ρc . So what you see is that your reverb pressure does not change in the room and the direct pressure very rapidly it decays with radius, so to answer your earlier question if you are reasonably far away from the source within the room the reverb pressure will dominate whatever you are going to do and we will put some numbers on that and my total pressure and I can add the energies basically this is $P_R \text{ square}$ is directly proportional to reverb energy $P_d \text{ square}$ is directly proportional to the direct energy and I can add the energies I cannot add the pressures themselves but I can add the energies. So $P_t \text{ square}$ is basically $W \rho c$ times 4 over R plus 1 over $4 \pi r \text{ square}$. So this is the relation for my total pressure.

Student: Sir there is instance when we consider this straight line, sir the part time will be minimum, in that case a direct pressure will decay and when we consider the reflected then the distance travelled is more it will decay more than how?

Professor: Yes, but the defused pressure the reverb pressure it does not change from point to point it will decay more when you are in related terms, see all you are doing you have to compare your direct pressure in a square way one over $r \text{ square}$ way decays as he move out your diffuse pressure is not changing within the space a diffuse pressure is not changing within the whole room. So if you are fairly far away from the source in a direct pressure will be fairly small while the diffuse pressure will still remain in.

This relation is basically addition of two energies that is one thing I wanted to say and this works only to the extent that both these sources are, what correlated or uncorrelated we have talked about this earlier the source of the music is uncorrelated. And my sound pressure level basically $10 \log_{10} \frac{P_{t \text{ rms}}}{P_{\text{ref}}}$ basically it is a square of this and square of this.

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So now I plot my SPL with respect to r and the curve is going to look like somewhat like this, this is my r r is the distance between the listener and the source and this is my sound pressure level and what I get is something like this an asymptotic behavior where what is this value this corresponds to P_d this is my total P_t so P_t converges to P_d as I move away from the source.

So this is not P_d I said that by mistake this is reverb field, another question that I know that room constant is s times α bar over $1 - \alpha$ bar if there are no walls in the room so that it is just I am having a speaker in a field and I am just running it, then what is r what is the value of r?

Student: Infinity.

Professor: Infinity? For if I am too close to the room not that to the source then my direct field is dominating, basically that is what this duration is saying and as I move further there will be a threshold where this term equals this term and once I have crossed that particular threshold then the $\frac{4}{R}$ term start dominating compared to $\frac{1}{4 \pi r^2}$. That particular value is where I have this equivalence between $\frac{4}{r}$ and the other term $\frac{1}{4 \pi r^2}$ that we

call it r not, so r not is basically r square average size auditoriums r not is approximately equal to 20 feet about 6 meters.

Student: Sir over 8π , sir here it should be $2\pi r$ square.

Professor: 4 wavelengths and where is p direct, yes so r not is about 20 feet for reasonably sized auditoriums, if I make my auditorium bigger then this r not also goes up because r goes up and for small rooms this is around two feet less than a meter, so even in a room like this most of the sound which you are hearing is essentially reverb in a room (31:40) not direct. Consider a scenario where you have a recording equipment and there is a bunch of people who are generating creating some music there instruments and all that and if you are too far from the music then the microphone will essentially capture the reverb field, if it is too close to the musical instruments then it will capture predominantly the direct field now the limitation of capturing the direct field only is that each of these instruments they have their polar patterns based on their geometry and all that.

So if you are too close then based on each instrument's polar pattern the microphone will take selectively some frequencies more than other frequencies, so that will not necessarily be a faithful reproduction of the sound (32:44). If you are far then you do not capture that but essentially what you are doing is you are taking a peaks and nulls and you are picking up a lot of information contained in the modes and the natural frequencies of the room. So again that distorts the faithfulness of the music reproduction.

Student: Also sir if we are far the absorption of the air absorption will be more.

Professor: Air itself, but that you can amplify but yes air absorption.

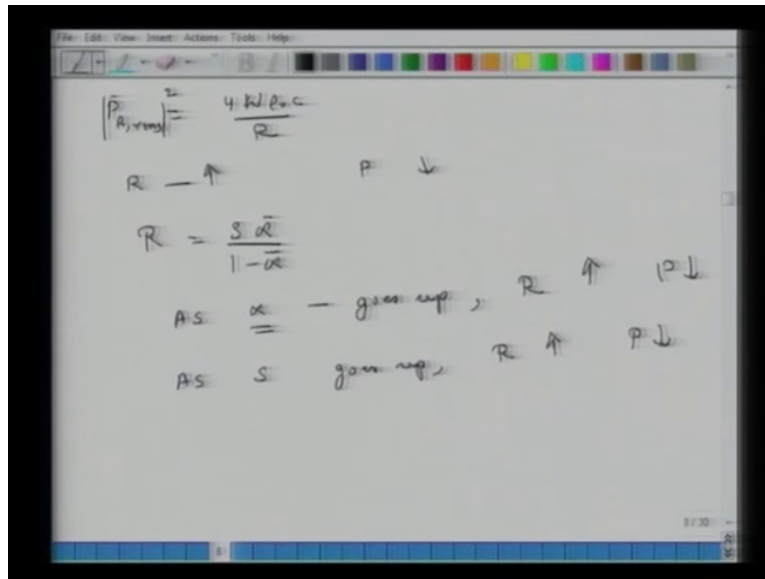
Student: And it depends on the frequency?

Professor: It depends on the frequency but a lot of times these rooms are not that big recording studios are not that large but yes I mean if you want to do this recording and sitting in the auditorium and where the performance is happening on the stage and air effects will also become dominant will become prominent, so that will also tackle some of the higher frequencies. So it is very difficult to reproduce live music whatever you hear in a room when live music is being played faithfully capturing that in a recording on a CD or a DVD or on a cassette it is very difficult and

there are technical reason behind that, it is not that the companies do not know what they are doing but it is difficult.

The lot of times they try to put there recording instrument at this value r not so it is like some sort of a compromise.

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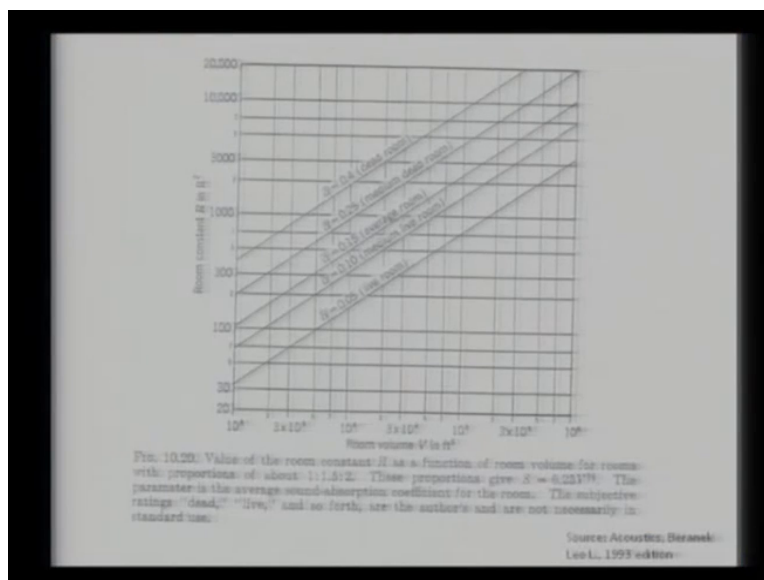
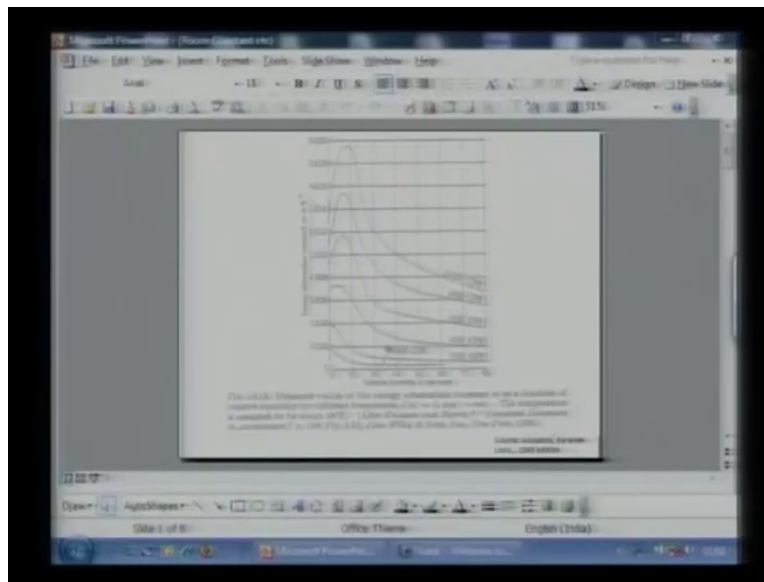


Other thing is that we know that room constant is $4w\rho c$ over R , no I am sorry P_{rms} , right? So this is my room constant so as room constant goes up as it goes up my P goes down and so if I have a larger room my reverb energy will go down in that room, now we know that R is $s\alpha$ over $1 - \alpha$ so what that means is that as α goes up what happens goes up or down?

Student: Sir depends.

Professor: Alpha is always less than 1, R goes up so as alpha goes up which means the absorption coefficient of the room is going up R will go up and P will go down more sound is getting absorbed so less reverb energy. Similarly as s which is the surface area of the room goes up R goes up and P goes down I just wanted to curve.

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So first thing I wanted to show in this overhead is that what you are seeing here is the dependence of alpha on humidity which we have not explicitly not talked about in earlier lecture also but if I increase the relative humidity in a room then my alpha it actually decays from 10 to 20 percent it goes up as I move from 10 percent to little humidity 20 percent but after that it goes down significantly. So humidity is a very important parameter as we are trying to figure out what is the value of alpha and that has an impact on the time constant of the room.

And what this is showing is that so as alpha changes how does room constant change and this is also plotting the value of for larger rooms as alpha goes up we expect room constant has to go up

and as alpha, I am sorry as b goes up for a given value of alpha room constant is going almost linearly and as I reduce my alpha those linear relationship they just move in parallel. The other thing is so we will again start again from this relation or actually we will start from this relation and we will go further.

(Refer Slide Time: 37:17)

The image shows a whiteboard with handwritten mathematical derivations. The top equation is:

$$SPL = 10 \log_{10} \left[\frac{W \rho_0 c \left(\frac{1}{4\pi r^2} + \frac{4}{R} \right)}{P_{ref}} \right]$$

Below this, it is expanded as:

$$= 10 \log_{10} W + 10 \log_{10} \rho_0 c + 10 \log_{10} \left(\frac{1}{4\pi r^2} + \frac{4}{R} \right) + 94$$

Then, the Power Level (PWL) is defined as:

$$PWL = 10 \log_{10} \left(\frac{W}{10^{-13}} \right)$$

Substituting PWL into the SPL equation, it is boxed as:

$$SPL = PWL + 10 \log_{10} \left(\frac{1}{4\pi r^2} + \frac{4}{R} \right) + 0.5 \text{ dB}$$

Finally, the relative SPL is given as:

$$\text{Rel. SPL} \Big|_W \rightarrow 10 \log_{10} \left(\frac{1}{4\pi r^2} + \frac{4}{R} \right) + 0.5 \text{ dB}$$

So SPL is log 10 and we had in this rho w rho not c times 1 over 4 pi r square plus 4 over r this whole thing divided by e ref square. So now I can break it and that is essentially 10 log 10 of w plus 10 log 10 of rho not c plus 10 log 10 log this bracketed term and if I also take the log due to this P ref I get a value of 94 P ref is 2 times 10 to the power of minus 5 I take that log and then multiply by 10 I get 9 (38:11).

Now we know that like SPL we had also shown earlier that PWL decibels in power is basically 10 log whatever the power is being measured divided by reference power and that reference power is 10 to the power of minus 13. So if I plug this back here.

Student: Minus 9.

Professor: Minus 9?

Student: Difference power we should not take minus 10 to the power minus 9.

Professor: I think this is fairly, we will check that.

Student: () (38:45) is for intensity.

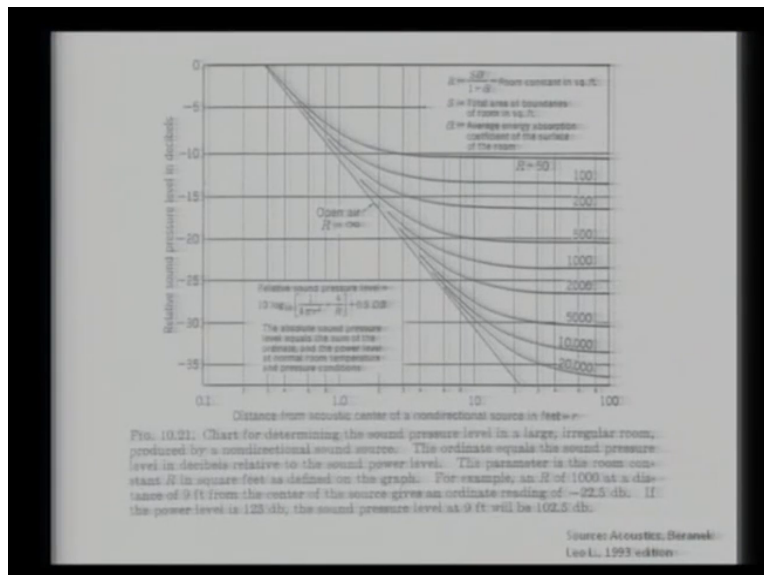
Professor: 10 power of minus?

Student: 9 is for intensity.

Professor: So the story will not so that is a number, so if I plug this back in here basically what I get is SPL equals PWL plus 10 log this number I mean this bracket term and then I get another some constant term and that constant will change whether I am going to SI units or I am going to feed counts system but essentially it is PWL plus 10 log this whole bracket term and this c is 0.5 db for British system and why I am talking about British system is some of the curse which you will see have charts which are based on the British system but for SI system this is what is there minus 36, anyway.

So hope British system this is what you have, okay. So if I plot the relative SPL then essentially what does relative SPL means that I normalize it with respect to w itself then relative SPL is essentially dependent on 10 times log of this whole bracket term plus this constant 0.5 db my reference here I am relative to this w.

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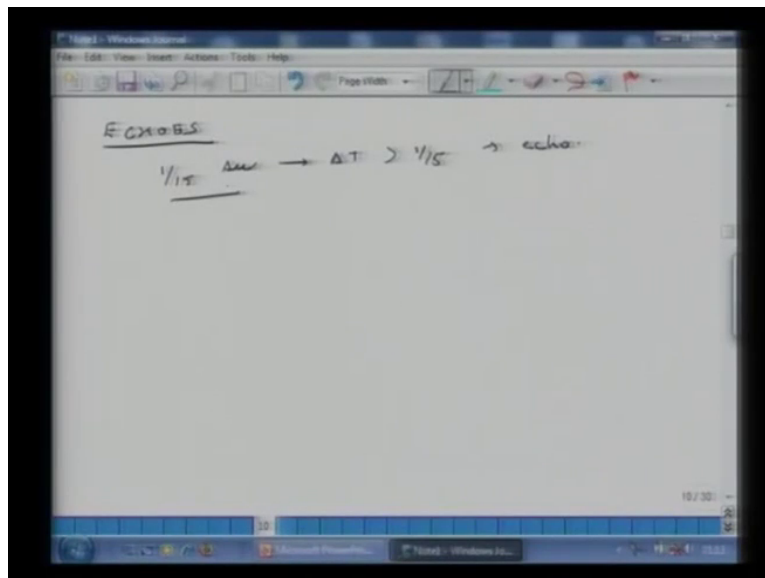


So what I wanted to show you was this particular chart, so what you are seeing here are relative sound pressure level in decibels for which we had developed the relation just now and here you are increasing r as you move away from the source and then what you have are different curves.

So for r equals infinity which is (∞) (40:53) here you have a straight line, okay. Now let us consider this and essentially what it shows is that let us say minus 20 degree line and let us I am only 2 or 3 feet away from the source if I am just 2 or 3 feet away from the source then the relative sound pressure level is fairly close to whatever I have for a room which has no walls because this point on the curve is fairly close to this r equals infinity line divides, okay.

But as I move away depending on what is the value of room constant my sound pressure level starts becoming constant and it is significantly different from r equals infinity line. So just wanted to show how r plays an important role in this calculation that is what the chart I am showing. For instance this thousand r equals thousand line when I am at 20 feet and I am here where it is the relative sound pressure level is fairly constant but if I for the same curve on r equals 1000 then I become get closer and closer and let us say I am only 2 feet away then this point is very close this point is very close to an open response room with no walls.

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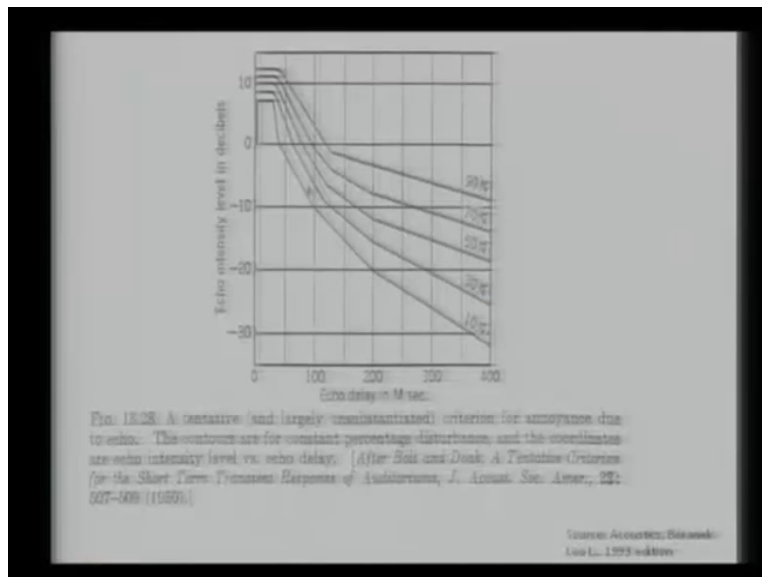
So the next thing I just wanted to talk about is echoes and there is some sort of a standard definition for an echo what is an echo and what is not an echo when intuitively we all know that whenever sound gets reflected it is an echo. This is source which emits sound it gets reflected comes back to the listeners ear, if the delta t between the original sound and the reflected sound when it hits the ear is less than 15th of a second then the perception of echo is not that great if it exceeds 150th of a second then you say, ohh there is an echo there is some sort of a fuzzy

engineering definition so it is like a 150th of a second if it is delta t is greater than 150th of a second then you say that there is an echo.

Now in fairly large auditoriums not even large auditoriums even in moderate size rooms you here this echo in auditoriums there the reverb time is in several seconds echoes will be there and they are always there but the impact these echoes they have on the our understanding and perception with sound it not only depends on whether the ear is hearing the echo but also it also depends on how strong is that echo in terms of SPL.

So if the difference of original sound and the echoes SPL is significantly large then even though that echo is there it does not have a very strong (())(44:25) on our assessment of the large hope.

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So in that context I wanted to show you another chart so what you are seeing here is echo delay in milliseconds 150th of a second is about 67 milliseconds 67, 70 milliseconds. So on the horizontal axis you have echo delay in milliseconds and on the vertical axis it is echo intensity in decibels and this is relative to the original sound level. So if echo intensity is equal to original sound level then what is the db level difference between the two.

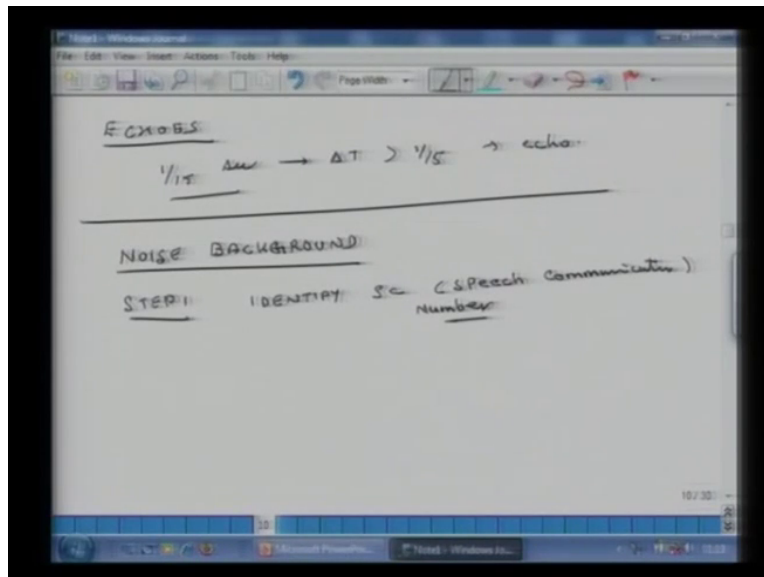
Student: 0.

Professor: 0 db? So 0 db line will be just on the top of the curve on the graph, so and then you have different curves here so we will take one point let us say delay of 100 milliseconds minus

10 db of a difference so let us say this is minus 10 db of a difference. So that falls on the 10 percent curve what that means is that out of every 100 people who listen to an echo where the delta t or delay time is 100 milliseconds and the echo intensity is 10 decibels less than the original sound 10 percent of the people will get confused 90 percent will be fine actually it is other way 10 percent will be fine, 90 percent will, no 10 percent will get confused and 90 percent will be fine.

Now if I increase the intensity level then I have a 90 percent curve and the decibel level is over 0, so I do not know how they constructed above 0 but essentially as you increase the decibel level of the echo more and more people get confused. So this also gives you some approximate idea how to ensure that whatever echoes you are hearing in the room they are below certain threshold so that most of the people do not have a poor view of the sound in room.

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And then the last one is noise background, okay. So the noise background could be for instance I am just there is some fan running in this podium there is a noise coming from this fan source it could be due to there may be power generator in the room or outside the room it could be an air conditioner, it could be AC ducts sending in air, it could be people walking outside the auditorium, it could be traffic there are so many sources of noise.

And there are again some fairly good guidelines what kind of a noise level is acceptable because you cannot eliminate the noise absolutely and what is not acceptable. So to do that if you go by

the (47:40) book what it says is that you have two step, so step 1 identify what is your need so identify this parameter S_c and I see is a speech communication you know speech communication number and there is a chart we will show that chart.

So for a broadcast studio where you have a person speaking or a musical instrument playing and that is being broadcast over a wide range that number is very low because you need very little amount of noise coming in from outside.

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TABLE 18.10 Criteria for Noise Control†

Type of room	S_c criterion curve
Broadcast studios	15-20
Concert halls	20-25
Legitimate theaters (500 seats, no amplification)	25
Music rooms	25
Schoolrooms (no amplification)	25
Homes (sleeping areas)	25
Conference room (for 50)	25
Assembly halls (amplification)	25-30
Conference room (for 50)	30
Motion picture theaters	30
Hospitals	30
Churches	30
Courtsrooms	30
Libraries	30
Small private office	40
Restaurants	45
Coliseums for sports only (amplification)	50
Secretarial offices (typing)	55
Factories	40-65

† L. L. Beranek, J. L. Heynolds, and R. E. Wilson, Apparatus and Procedures for Predicting Ventilation System Noise, *J. Acoust. Soc. Amer.*, 28: 313-321 (1956).

Source: Acoustics, Beranek, 4th ed., 1993 edition.

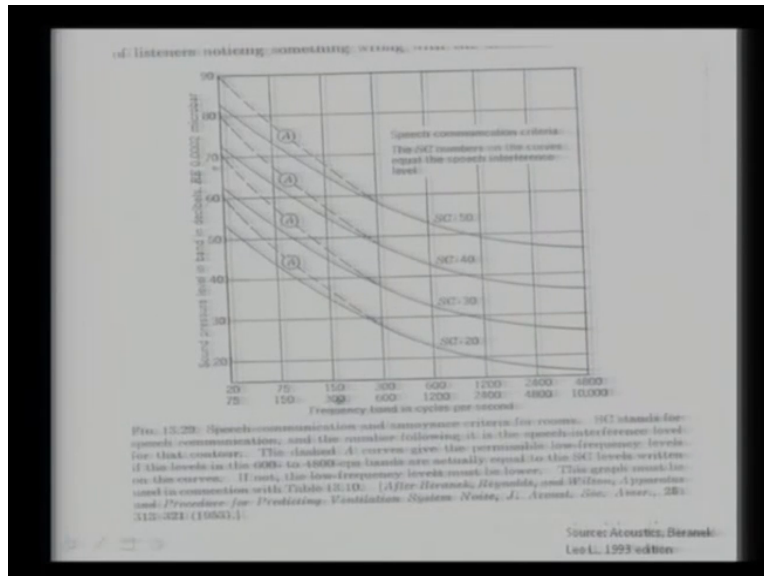
ECHOES

$\frac{1}{15}$ ΔT \rightarrow $\Delta T > \frac{1}{15}$ \rightarrow echo

NOISE BACKGROUND

STEP 1 IDENTIFY S_c (Speech Communication) Numbers

STEP 2 IDENTIFY MAX SPL LEVELS



For a factory that Sc number is fairly high, so for instance again broadcast studio that Sc criteria we have to the Sc number is 15 to 20 then you go to music rooms it goes up 25, then you go to the conference room where people also chat while they are listening and participating still remains 25.

Then code rooms, libraries, small private restaurants, restaurants is 45 noisy, factories are very noisy so it could be anywhere between 40 to 65. So that is the step, figure out what is this Sc criteria curve and then in step 2 then what you do is you refer to another table and you identify max SPL levels for each frequency or for different frequency match you figure out what are the frequency level which are permitted and this curve helps you do that, so suppose you are in that broadcast studio you pick up this curve Sc equals to 20, okay and you say ohh my range of frequencies which I am going to use in this room are 300 to 10,000 hertz so my SPL noise coming out from the outside or unwanted sources has to be below this dark line so it is fairly straight forward but these are like some sort of guidelines.

Now there are also dash lines on this curve and the only time you use those dash lines is than all or most of your noise is in this 300 to 4800 band, if your noise content is also in this particular band below 300 hertz then you stay with the dark line but you can use this dash line if most of your noise is above this 300. Basically what it means if since most of your noise is coming in that higher frequency range you can have lower noise floors for lesser peak so that.

There is some qualitative measures which you can use to design sounds, so we have talked about reverb time, we have talked about direct and reverb energy densities and how it relates to hall characteristics, we have talked about room constant, we have try to address the role of echoes, exteriors noises and in the next lecture we try to cover some application of whatever we will not definitely new topic we will try to cover some applications and you had mentioned.

Student: (0)(51:37)

Professor: Speech definition?

Student: Not exactly sound definition or like there are overtones and many things which act to a particular note from a particular instrument, so what exactly is the criteria of which distinguishes one instrument from a (0)(51:55).