Acoustics & Noise Control
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Module – 17
Lecture – 22
Human factors in acoustic engineering

Fine in the last class we did look at the different ways in which the units of sound got transformed. Firstly, from the Pascal value to the decibel value which is the logarithmic scale, and then above the decibel value we had different ratings the a b and c rating which was meant to capture the human factor of this acoustic engineering.

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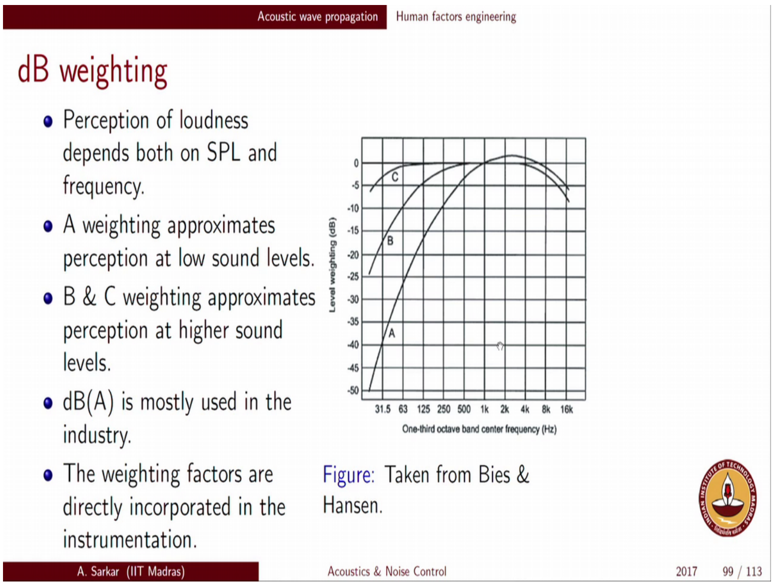

The fact that we perceive different loudness for different frequencies which is captured through this a weighting criteria; a weighting is the one which is like uniformly used except for very sort of exceptional cases, you will mostly get to see the values of sound or noise rather would be reported in terms of dBA, and in particular if you recall the CPC noise regulations, the centre pollution control board noise regulations are in terms of dBA.

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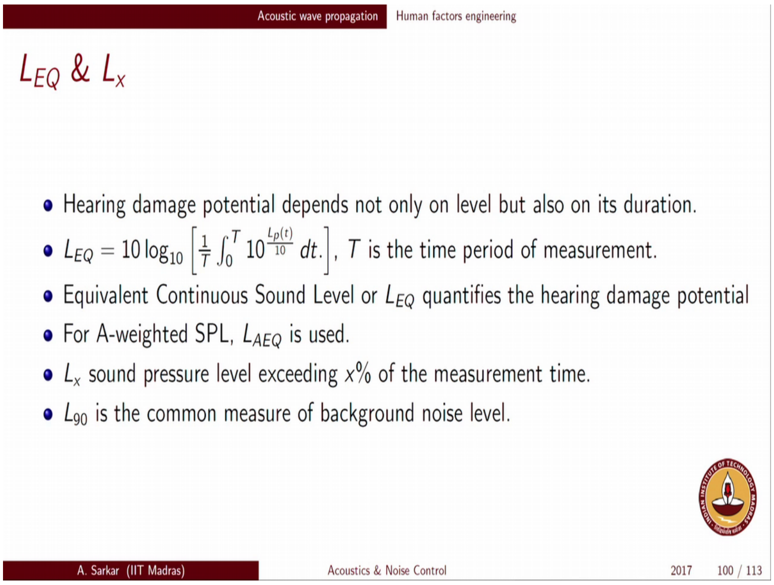

So, now you know what dBA means n comparison to dB. So, we move forward and we will define a few more terms which are mean to incorporate the different factors as human factors, associated with these ideas of noise and it is perception. So, definitely the hearing damage potential that a person encounters when he or she is expose to noise, is going to depend not only on the sound level, but also on it is duration. So, if you can have a higher level, but if it is for a very small duration then hopefully it should not be cause be that much hazardous in comparison to a lower level of sound, but acting over a longer period of time right. So, therefore, we need to now incorporate a certain more effects, which sort of the captures this effects especially if the noise is transient. As we know machines do not encounter a fixed load condition they do change for example, in our vehicle as you change the speed of the vehicle, the engine operating conditions chain slightly and the road noise; obviously, will change.

So, whenever you have an intermittent operating condition and where in it is not possible to say that the noise is steady, then you must have a record of these rather a major of how hazardous the noise is right. So, towards that end we are going to develop a few parameters by which we can quantify these issues of intermittent noise. So, the first factor which comes is in is called LEQ this is somewhat like an equivalent sound. So, what we do here is that we understand that the logarithmic addition simply tells us to take the sum of 10 to the power of Lp by 10. You would have see that that is how the sound addition was basically done, let me just turn back to show you that formula yeah something like this is this is the averaging formula.

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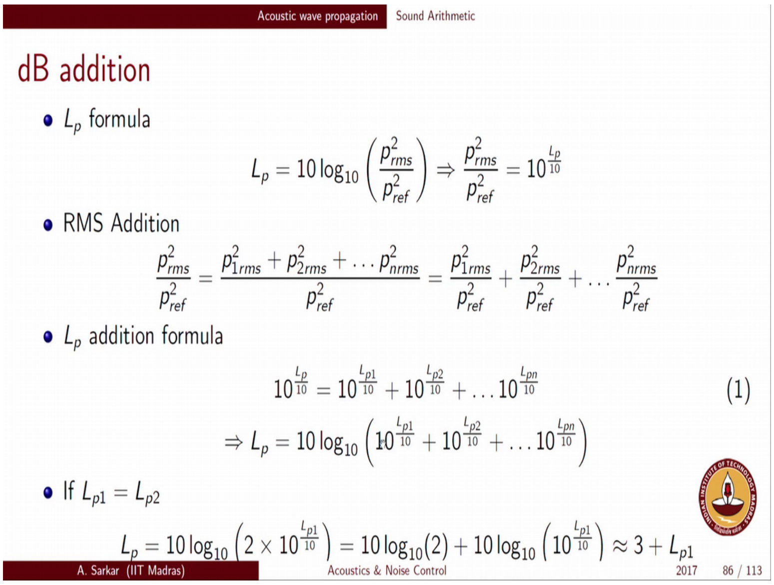

So, if you look back at the sound addition formula which is this yeah. So, when you have to add sound at different levels, you simply within the argument of the logarithm you have to be add 10 to the power Lp 1 by 10, 10 to the power Lp 2 by 10 and so on up to the n sources.

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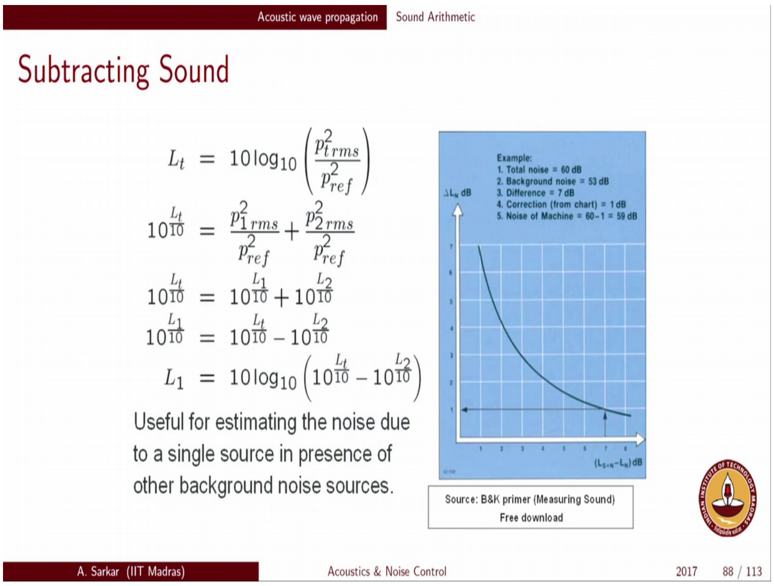

So, now, what we tend to do is that associated with each time interval, there is a different sound pressure that is the idea of intermittence sound associated with different time intervals you would be recording a different value of spl.

So, you cannot report a single value of Lp, but rather Lp itself will be a function of time which is what I have meant by this notation. Lp is now a function of time and for each of these time windows you will have to just instead of the summation you will have to take an integration and divide it through by the number of the entire time interval of your interest. So, instead of the summation as you used in the decibel addition formula you will have to integrate this out, integrate this over a period, but then normalize it again by having out 1 by T division, and then 10 log 10 of that. So, that will give you the equivalent sound pressure level when the sound pressure level itself is changing with time right. Remember we had said that we will do harmonic calculations we will take steady state.

But this is an exception to that situation this is not a steady state condition, here we are accounting for the fact that in time sound pressure is changing and we have a record of how it is changing with time. Then this is almost like a dB averaging formula, but just that instead of the summation it is actually a integration.

Other than that the formula is all nice and fine. So, if we have to report the sound for a vehicle which is let us say running from 0 to 40 and then again from 40 back to 0, then as the vehicle is running at different speed conditions it may be different road conditions also then it will generate different sound pressure levels we will take a record of that put it in this formula, and then at the end of the day we will say that the LEQ of this vehicle of the equivalent sound pressure level of this vehicle is going to be such and such. But this is not what is asked for in a pass by noise, pass by noise is much simpler let us we a had discussed, but this is just as a matter of reporting and capturing the fact that it is an intermittent noise rather than a steady state noise.

So, this is called equivalent continuous sound level or LEQ, this is what will quantify the hearing damage potential in some sense. Because it captures both the magnitude of the sound pressure level as well as time duration of it because the integral of the both sorry integral of the a sort of mean square effect is properly captured and the average value is correctly reported. And then instead of taking these Lp values directly in terms of decibel, if you take the Lp values in a weighted decibel scale and you know what is the difference between a weighted scale and the standard scale is then instead of LEQ we will have LAEQ. So, that is equivalent sound in the A weighting factor.

So, that also something pretty easy to understand Lx is a pretty useful measure it is if we have we if we report for example, in this recording studio we wish to make sure that say that the background noise is less than a certain value let us say 20 dB or so, right. What do I mean by background noise? Obviously, I cannot say that even if I keep my mouth shut and there is no noise, if we have to take a measurement the sound will not always be measure to be same right. Because the disturbance at times will more some people in the corridor will be shutting at certain times, may be at lunch time you will find people at the corridor speaking away, but at other times it may possibly be a quieter right.

So, we need to have a measure let us say 90 percent of the time what is the value of this sound that is recorded. So, that will be L 90. So, if for example, L90 would be a background noise level; that means, for the 90 percent of the time sound pressure level will exceed this level right. So, if L 90 is reported as 20 db; that means, for 90 percent of the time the sound level is more than 20 dB. Similarly the converse situation if L10 is reported as 20 dB; that means, for only 10 percent of the time the noise exceeds 20 dB so; that means, there is a particularly quite room. Only for 10 percent let us say I have my house in a village and it is perfectly quite their it is far from the highway, and I report that L10 for that house is about 40 dB.

That means only for 10 percent of the time the noise exceeds 40 dB right; where as L90 would mean for 90 percent of the time in an urban setting I would rather have L90 to be 40 dB, which means for 90 percent of the time the value which will be exceeded is the floor which will be exceeded is 40 dB right. So, the 90 percent of the time the value recorded will be more than 40 dB, where as in the first example for 10 percent of the time the value will be more than 40 dB. So, L10 is a most stringed measure. If you are reporting in terms of L10 it is a very aggressive and stringent measure, L90 is a very relaxed or a it is a relaxed measure because you are basically reporting that for 90 percent of the time the sound will exceed this value right.

Just like we had this quantity LEQ and Lx, we will again sort of change our game a little bit and that is again dictated by a lot of experimentation on human subjects and it so happens and that if the change in sound level is just 3 dB then we are just able to perceive it right.

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Acoustic wave propagation Human factors engineering

Perception of Loudness

Change in Sound Level (dB)	Change in Apparent Loudness
3	Just Perceptible
5	Distinctly Perceptible
10	Half or Twice as Loud
20	Much quieter or Louder

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What as been shown by experiment a lot of extensive experimentation is this that what is a barely perceptible change and loudness is 3 dB not 3 dBa, but 3 dB right. So, 3 dB is the bare minimum change that you need to make in your products, says that your customers are at least able to perceive a change in it is acoustic characteristics right and 5dB will be distinctly perceptible.

So, if you really want to make sure that you win hands down in terms of a improved acoustic characterization, you have to have a target of at least 5 decibels right. In other words if you just change by two decibels, probably one and half decibels that will probably go unnoticed right in terms of the perception of your customer. So, that actually makes the target pretty stiff. So, a bare minimum of one and half dB or 2 dB does not at all cause any perceptible difference, 3 dB is sort of the threshold after which you can change have a perceptible improvement in the effect of noise difference, but what is distinctly perceptible is 5 dB and again in terms of perception a 10 dB difference is apparent to us as half or twice the loudness this is pretty strange, but that is how our human hear responds to it. If we play out sound at two different decibel levels, the values of which differ by a 10 decibel and this difference is just a arithmetic difference.

The arithmetic difference is 10 dB which means 80 dB and 90 dB sound, perceptibly it appears 90 dB is twice as loud right. So, we need to incorporate these perceptions also in another new scale.

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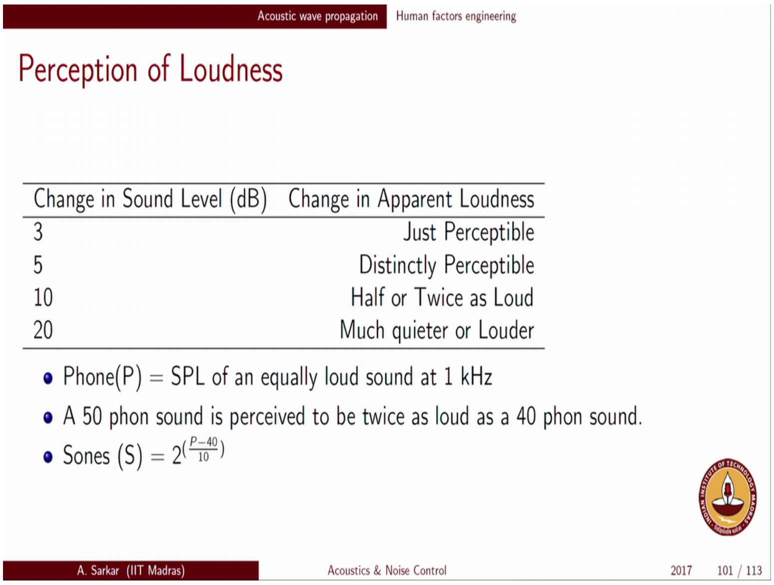

Decibel we understand is not able to incorporate these perceptions that you know a 10 dB difference is half or twice as loud. So, we define this new scale which is called phone. So, phone is the SPL of an equally loud sound at 1 kilo hertz. We have already seen the chart for equal loudness contour right. So, this was roughly the chart for equal loudness contour. So, phone is the SPL of an equally loud sound at 1 kilo hertz right. So, 1 kilo hertz as I said is just the reference frequency. So, here you will see that at this 31.5 and equally loud sound is about 40 right at 1 kilo hertz.

So, 81 dB at 30 sorry 79 dB at 31.5 hertz is equally loud as 40 dB at 1 kilo hertz right. So, using that idea this phone measure is taken in which is the SPL of an equally loud sound at 1 kilo hertz. So, if we have a 79 point 79 dB sound at 31.5 hertz, the SPL of an equally loud sound is actually 40 dB at 1 kilo hertz. So, the phone is 40, even though the dB value is 79.5, the phone will be just 40. And to get into this effect that a 50 phone sound will now be perceived as twice as loud as 40 phone sound, because basically phone is SPL measured in dB. So, we understand just as an example of 50 any 10 dB difference will now be equivalent to a perception of twice or half the loudness right.

So, 50 phone sound essential means an SPL of 50 dB at 1 kilo hertz 40 phone sound essentially means a sound which is equally loud as 40 dB at 1 kilo hertz, and these two sounds will have a perceptible difference of being half as loud or twice as loud depending upon which way you look at it. So, therefore, to capture this effect of twice or half we invent this new scale which is Sone right. So, sone will be 2 to the power phone minus 40 divided by 10. So, if you now see put P as 50, then the sone associated with the

50 phone sound is going to be 50 minus 40 by 10 is 1. So, sone is going to be 2 right for a 40 phone sound the sone is going to be 1 right. So, now, we have calibrated this perception correctly in terms of sone.

So, 40 dB sound sorry 50 phone sound will appear in sones scale as two sones right where as a 40 phone sound will appear as 1 sone. So, thereby you get the feeling that the loudness has doubled for a 50 phone sound than in a 40 phone sone. So, these are all lot of nitty gritty details have been incorporated just to capture the idiosyncrasies associated with human factors, the human perception of hearing. And still lot of active research is continuing to incorporate these effects even better.

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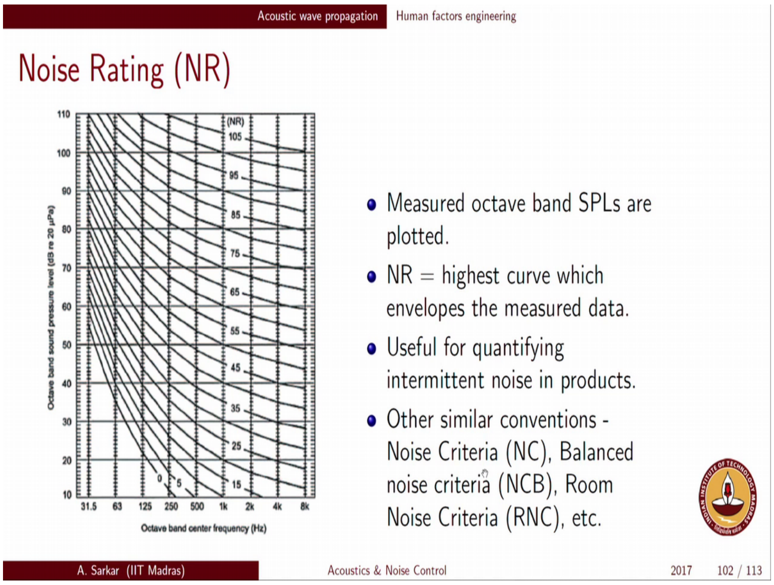

But I think this is where we will sort of let it go, but probably with one more example again in case of intermittent noise.

Let us say for an air conditioner an air conditioner works with or refrigerator for example, these days the compressors are far more silent, but at least maybe if you look at those refrigerators 10 years back, they used to have a very noisy compressor and once the compressor use to get switched on, then there will be a huge noise impulsive noise because of that compressor starting to work. So, that is an intermittent noise, you do not want to quote the noise of a refrigerator. So, both of these conditions are like an extreme condition that is the extreme noise will be heard when the compressor is just switched on and if the compressor is switched off again you will have the lower limit right.

So, a manufacturer a refrigerator manufacturer cannot possibly quote one of these two extreme noise and get away, in terms of it is favorable or unfavorable situation. What has to do in such cases is that one has to quote the noise rating associated with the machine. So, noise rating curves sort of standardized curves, and the these are this is one such graph this has been taken from the textbook by b (Refer Time: 17:03) Henson. So, this what we do here for quoting the noise rating of any particular industrial component is the following.

So, as you note the x axis is in terms of octave bands, these are all in geometric progressions, these are the center frequencies of different octave bands. So, we will measure out the octave band SPLs on using certain instruments which will talk about how to measure the noise, but we will plot these SPLs as a function of frequencies taken in octave bands right. And then we will find out which of these curves encloses the plot under all conditions right. We understand that the frequency spectrum can possibly change because it is an intermittent noise, it is not going to remain in steady state, it is an intermittent characteristics.

So, we will keep taking under all possible condition compressor on compressor off and whatever other conditions that you can think of, for all possible conditions under which this machine is suppose to operate we will have to take the readings for the SPL as a function of it is octave band frequencies, and then over laid on this noise rating curve. The curve which just encloses this entire spectrum I mean spectrum for all possible such conditions, that will be quoted as the noise rating of this machine.

So, it is a pretty aggressive criteria, that you have to when you quote the noise rating and typically I think few days earlier I got bottom juicer mixer, a food processor kind of thing even I when I read the manual it quoted the noise rating value the NR value right that was nice to see. So, this is where because you know in the juicer kind of application food processor kind of application, you have intermittent noise as oppose to a steady noise right. So, in such cases this inner values have to be quoted and as I said they are useful for quantifying intermittent noise in products, but again you know this there are various incarnations in various avatars of this kind of graphs, noise rating or NR is one of them, there will be similar depending upon the standardization issues the other conventions also are there, which are called noise criteria, balanced noise criteria, room noise criteria and so on.

Each of them essentially gives you one such chance, but the process of arriving at the NR or the NC number is exactly the same. Take the reading over lied on these curves and find out which is the highest curve, which just encloses the entire spectrum in octave bands. If you can do that that value is going to be the appropriate in NR or NC value, and the reason why we have different ratings is because that in different situations you have different perception of sound.

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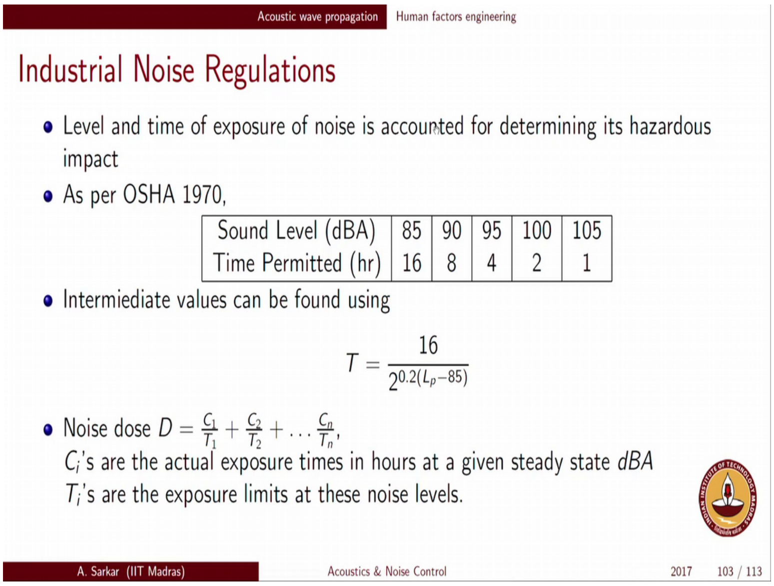

For example in room you will have a room which is pretty large, it is having or in auditorium acoustics case the perception is going to be very much different than let us say in my small living room where I have a refrigerator. So, therefore, you will keep seeing different conventions as per the different applications that are being used. Noise rating for some reason is sort of quite well known for home appliances and things like that, but there are other criterias also and each of these criterias have evolved for particular reason, which is mainly dependent on applications.

So, with that I think I will close my discussion on human factors of engineering by no means this is an exhaustive account of it, there are many more issues associated with human factors in engineering and in acoustic engineering and it is a active field or search even as we talk, is lots of work that is going on which mainly deals about experimentation and trying to find out how we perceive sound and then to develop you know ways and means by which we can incorporate these human perception into our instrumentation, and thereby have a testing procedure which reflects the perception of human perception of sound, rather than just quoting a Pascal value or the pressure value because that possibly does not relate to our perception of loudness right.

Because as I said the perception depends upon how our ear canal sort of picks of those signals and get them interpreted by the brain. So, that is there is lot of anatomical issues which complicates the perception of sound, which cannot be sort of dealt with in this course in any further details. So, I will switch gears now or may be one more we have a one more just one more slide associated with this, which is about industrial noise regulations yeah.

So, as we said the level and time of exposure of noise is accounted for determining it is various hazardous impacts because one such thing was LEQ standard, the LEQ measurement was hopefully able to capture the level and time of exposure. But then LEQ is little complicated to calculate because you will have to essentially take a record of the entire sound level for the entire period of the time, and then do an integration which typically factor the people will not like it right. So, you need a easier simpler formula yet which captures this issue that you have to account for both the sound pressure level as well as the time of exposure. So, as per this convention which is occupational safety and health act, I guess that is called OSHA 1970 and I could not find a revision of it, but that is also will stick with OSHA 1970.

So, as per that convention it was found out that healthy human can possibly be exposed to 85 dBA of sound at for about 16 hours, 90 dBA of sound for about 8 hours, 85 dBA of for about 4 hours, 100 dBA for about 2 hours 105 dBA for about 1 hours. Please note that in terms of time it is having an in terms of dB you are having a 5 dB change in difference. So, that is what we said even in the last slide when we said a 5 dB difference is what is like roughly, the I mean 5dB difference is just about perceptible, but in actually in contrast what to a to the calculation of phones and sones where we said the loudness is in terms of is like doubling in terms of perception, but here this is the time that is permitted for exposure towards these noise levels, without causing any health hazards.

So, as per this standard if you have a factory where in let say your steady state sound or LEQ is suppose to be 85 dB, then you have the any worker can be permitted to work in double shift up to 16 hours. But if it is 90 dB it is only 8 hours that a worker should be expose to. And especially in the western countries these rules are pretty stringently applied and there are some tests and clearances that as to obtain and the worker cannot be subjected to more than these sound levels for these time periods right and if the value cannot read of from this chart you can use this formula which basically is given by this. So, this gives if for example, you have an 87 dB sound generated in a particular environment, than the time of exposure for this 87 dB sound without causing any health hazard is going to be rid of using this formula right. Roughly it gives you the same I mean it gives you the exactly the same values which have quoted in this table.

So, you can see that if Lp is 90 then 90 minus 85 is 5, 5 multiplied by 0.2 is ones and then 16 divided by 2 is 8. So, that is how it goes. So, the factor here is of time is getting how right this is the time that is permitted without causing any health hazard, this is not exactly loudness this is the health issue in terms of both physiological and psychological issues associated with noise exposure. So, the point is this that there is a factor which is called noise dose. So, the noise dose is given by this the formula which if you know particular environment like let say in IIT, In Gajendra circle you may have a different value of sound pressure level, at my office room you will may have a different value of sound pressure level at the play ground you will have a different value of sound pressure level.

So, you will have to calculate the total noise dose which means that the numerator of each of these factor is the actual exposure time to each of these noise levels and the denominator capital is are the exposure limits as given by these OSHA regulations. So, the total value of this noise dose should be within unity. So, instead of having just a steady state noise at 85 dB, you may have an intermitted noise of 85 dB for 50 percent of the time and 90 dB for 50 percent of the time, in which case you have to calculate the noise dose for that particular environment and if it exceeds the value of unity then as per OSHA that environment is not a healthy and conducive environment as for as the health issues associated with this noise exposure is concern ok.

So, again this is one more case where we have talked about human factors of engineering, the factors which decide as to what are the how we can control the amount of exposure of noise and make sure that the inhabitance associated with a particular community are not subjected to any health hazards due to this exposure to noise ok.

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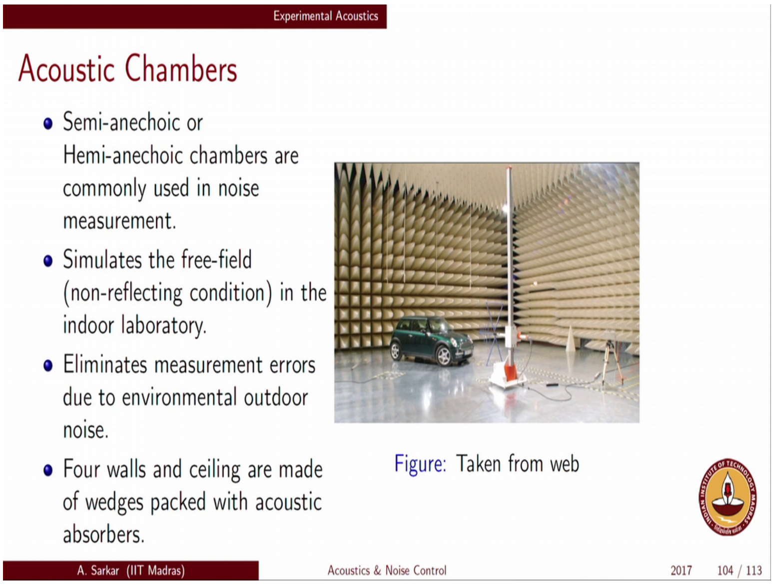

So, now we move on to the next topic which is about experimental acoustics; here we will learn different instrumentation procedures associated with acoustics. The first important thing that we have to learned while taking our measurements in acoustics is that, acoustic measurements cannot be just taken in any random situation there are specially designed chambers which are called acoustic chambers, where in we should take such acoustic measurement and that is not because of some you know strange egoistic reason, there is a good reason why we have to use acoustic chambers. As we have known very well by now is that the boundary conditions in a acoustics can really cause a lot of difference in the perception of sound.

If you have an enclosure then the sound is not allowed to escape and that it will keep reflecting back and keep reverberating back and you will possibly hear a different sound then if the same product was there in free field. In the free field you will not hear the reflected sound, you will only hear the incident sound right where as in an enclosure you will not only hear the incident sound, but also hear the super imposition of the incident sound together with the reflected field, and that reflection can be pretty complicated to analyze if the room is not a very standardized room or something like that. So, therefore, we need to, but at the same time it is not possible for a product to be always taken in to a free open atmosphere and do the measurement. In fact, if I take the vehicle some like this and in to the open road and open ground and try to measure the acoustic performance of this vehicle, now the problem is my noise measurement will be corrupted by the outdoor noise also right.

So, it is not possible to actually take this the product for which you want the acoustic measurement to be done out in to the open atmosphere and take a measurement, that will require you a big open field where in the outside noise does not contaminate your measurements. So, what is done is this that you design something called an acoustic chamber.

So, these acoustic chambers have acoustic absorbers padded in the four walls as well as the ceiling; however, it is not possible to do this acoustic absorbing lining on the floor because if you do it, then there is no chance of the vehicle can be moved in to this room right. So, such rooms are called semi anechoic or hemi anechoic, because it is the reason it is called semi anechoic is because all, but one floor is left without the acoustic treatment. And the work of this acoustic absorber is going to be such that it will be absorb all the incident sound that is incident on these walls right and by the construction of it, it is a there is a reason why they have this wedge type construction I will just sort of intuitively hand wave an explanation, you would heard about ferrys black body in your high school optics.

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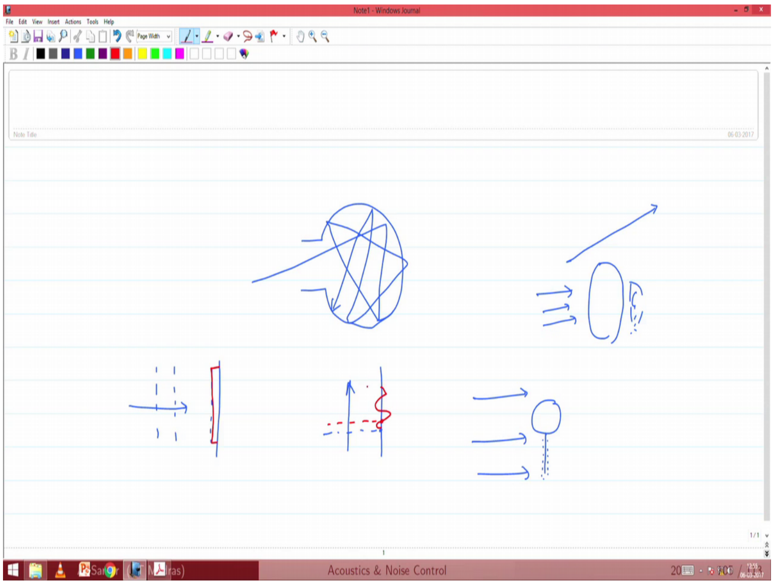

So, this is supposed to be a black body where in which will observe all the light. And the reason it is of in this shape is if you will remember is that any light that is incident on it will keep going through having multiple reflection and, but with each reflection it will decay right.

A similar effect happens with these wedges that you see here. So, the wedges create this ferrys black body like a situation here, the wave that is incident on to the wedge gets mostly absorbed, the part which gets which reflected will now get incident in to the neighboring wedge. And again the neighboring wedge will observe the remaining part and with a few reflections hopefully you can achieve a near perfect absorption. So, there are bench marks associated with these absorbers and these observing materials must follow suitable guidelines such that near complist absorption takes place. Because then if a complete absorption of the incident wave does take place, you will have the situation of a non reflecting condition. So, that is as good as saying that you are in the free field because there is no reflections, there only thing that happens is the incident wave reflections never come back.

But at the same time this room also isolates any acoustic disturbance which may be happening outside the room to penetrate in to the room, because this is a room after all and these isolations together with the concrete structure of the room will ensure that will sound which is outside the room does not percolate in to the room. Similarly the doors and the windows here are they are no windows in acoustic chambers the doors are appropriately sealed; because if the door does not have a ceiling then the sound from outside will leak through the small slits that are they are in usual doors right. So, when we take measurements we shut this door get the ceiling appropriately done, and then we can expect not only will the inner sound escape outside, but also the outside sound does not interfere in to the interiors.

So, it is actually a reciprocal thing. If the outside sound cannot escape sorry if the sound inside the room cannot escape outsid, the outside sound also cannot penetrate in to the room. So, that is how we take the measurement. So, semi anechoic chambers or hemi anechoic chambers are commonly used for products that we would likely to be interested, but there are fully anechoic chambers also where even this floor sort of is having had acoustic treatment; obviously, as I said you cannot bring a car in such a room, but if there are if you are interested in let us say speakers, loud speakers right and you really want to get the complete anechoic condition for the sake of your calibration. Then possibly in such situation it make sense, to have an arrangement where after all a speaker is not that big you can tie it up in some sort of a fully arrangement, and make it sort of hang in (Refer Time: 34:32) with the help of hopefully arrangement, but then the sound that is emitted will be enclosed with in a completely anechoic space.

Where in only the incident waves exist the reflected waves are going to corrupt. So, at least in such sort of products we will have situations for fully anechoic chamber; however, for most industrial purposes, I would think that it is semi anechoic rather than fully anechoic chamber which will be more useful as I said it simulates the free field or non reflecting condition in the indoor laboratory, it eliminates the measurement errors due to environmental or outdoor noise. The outdoor noise cannot corrupt the measurements done indoors. As I said all these 4 walls and the ceiling are packed with these acoustics absorbers and the instrument that is required to measure these acoustic or sound pressure levels is called microphone. By the way there are different kinds of micro phone it is not just a instrument for a measurement of sound but for example, recording of sound is also done using a microphone.

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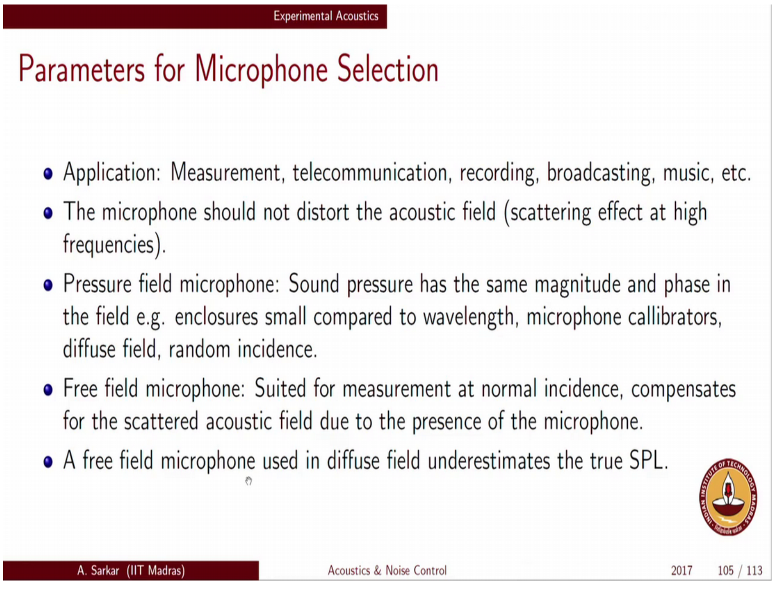

Communication your mobile phone that also as a microphone. The essence of it is this that the acoustic pressure just like as it hits our hear drums, it is converted to an electrical signal which is interpreted by the brain which is the boss of as for as the processing unit human processing unit, is concern. The same thing essentially happens in a microphone; when ever these acoustic pressure waves is incident in a microphone using some techniques it is converted to a voltage sinking, and from there all it can have different purposes. When it is for your telephone then these voltage signal is essentially transmitted through your carrier wave which is a what your service provider gives you, and then that wave is send to the receiver and then the receiver has waves and means to demodulate the carrier wave frequency and the frequency of your speech and you hear it readily right.

Each of these application whether it is communication, whether it is recording, they will have different kinds of micro phones that is understand the micro phone that I am using is a here right now is for the recording purpose, this is not meant for measurement purpose right. We are going to concentrate on the measurement part of it each micro phone will have different characteristics; the measurement micro phone is what we are going to look at it in this and the next talk. So, those microphones are called the condenser microphones, we will talk about those condenser microphones possibly in the next class.

But let us just quickly give you some introduction about the requirements and the parameters which characterized the microphone. The one important requirement of a micro phone is that by placing the microphone you should not distort the sound pressure field. Just like you know if you have a flow meter and you are interested in measuring a flow through a pipe, and the fact that you are putting a flow meter in this pipe distorts that flow field then it will be an erroneous measurement right. Similarly if you have an object which is sort of which is facing this sound waves, then fact that the there is an obstacle towards it essentially distorts the sound field this distortion is technically called scattering.

Let me just explain myself a little better. So, you would know at least again to give the analogy from optics, you would know that if there is a wave that is incident on to an object, just behind the object you will have the a region of shadow there is no light which will pass through, but may be this light ray will pass through. So, at some angles you will be able to see that the optic field is undistorted.

So, the moral of the story is this whether I place the object or whether I the object is absent, the field that I wish to measure is going to be different right. The difference is going to be very minimal if the size of the object is very minimal for example; as we are talking we are having some dust particles definitely in these rooms but these dust particles as so, small possibly they do not scattered or they do not create an obstruction right, but then similar thing can happen even for acoustics.

So, if you have let say an acoustic plane wave and you wish to measure this condition where in the acoustic plain waves are setup in a particular environment, the fact that you are bringing in a microphone right and what is the condition that this solid object will demand. Let say this micro phone is made out of rigid solid material and the microphone does not start vibrating the body of the microphone does not start vibrating the micro phone is actually placed on a tripod stand also and you can feel that the tripod stand is pretty much rigid right.

So, if it is rigid then what is demanded is all along the body of this microphone, you must have a 0 velocity condition right because the microphone should not is rigid it will not vibrate. So, therefore, it demands that the particle velocity of the microphone particles, which are in neighborhood with the air particles surrounding them should have a 0 velocity right. But then that is not possible because you have a plain wave condition every particle should vibrate so; that means, the fact that you have introduced another instrument in the field definitely means that this field is going to get distort right. It is not going to remain a plain fill plain wave or if it was spherical wave even that is going to assure you that you have 0 velocities exactly at the boundary of your solid surface right. Because your instrument is like a rigid solid it is going to demand that at all along the boundary of your instruments surface you should have a 0 velocity that is not possible.

So, therefore, you the incident field will get distorted and that distortion is called scattering and there is a whole lot of theory as to how you can find the scattered wave field because of the presence of some scatterers, we would not go in to that, but just again from analogy with optics we will make this remark that if the scatterer is really small then the effect of this distortion is going to be minimal, but then there is nothing like small or big it is always small or big with reference to something. So, if the scatterer is small in comparison to the wavelength of the sound then you can have this condition that the effect of this presence of this scatterer is going to be minimal it will scatter the sound in a minimal fashion. So, you could almost neglect this scattering effect in case it is small.

So, one of the important conditions that is demanded for good acoustic measurement is that the presence of the micro phone should not distort or scattered this acoustic field, and this will actually imply that your microphone should be small. Like if the microphone is small you will be able to satisfy this condition, I mean small in comparison to the wave length of the sound that you are measuring right. So, therefore, you will see microphones coming in different sizes, please understand the smaller size is better in terms of scattering, but smaller size will be worst in terms of sensitivities. So, these are the two competing factors which goes for your microphone selection and through this I am possibly the next talk we would like to understand these concepts that how do you make a proper selection of microphone. There are different kinds of measurement microphone available I mean there is two main kinds I should say one is called pressure field microphone, the other is called the free field microphone.

Actually pressure field microphone works under the condition that the sound pressure will have the same magnitude and phase within the field of your measurement. So, that is possibly true only when you have a small enclosure something like a Helmholtz resonator right. A resonator which is like a cavity and that cavity if it is very small then again small in comparison to the wavelength of the sound then you can assume that all points have practically the same pressure and the same amplitude of the pressure and the same phase of the pressure; in such situations pressure field microphone are supposed to do a good job where as the free field microphones are the once that we should use which will work better in an anechoic chamber which is in a free field setting

So, these a sitter suited for a free field setting a setting where in you have only an incidence wave and nothing else the reflected wave does not come back. Whereas pressure field or random incident microphone generally assumes that there will it will work in a setting where there are multiple waves, all combining in such a fashion that the phase in case of a random incidence microphone is going to have the same value at all possible points. Where as in a free field microphone it is going to be under the setting that there is a single wave right and if there is a single wave that is going on then; obviously, the phase associated with different points will be different right. And the fact that you are talking about a free field situation which means that you are talking essentially in about a condition, where there is a single incident where the wave in which things will get now complicated is that for a free field microphone the orientation of the microphone matters. Whereas for a pressure field microphone the orientation does not matter, because anyway it is supposed to work under the condition where all points will have the same pressure and the same phase.

So, it really does not matter how you hold your microphone; where as a free field microphone will work best and it is suggested that the way you hold it is such that the diaphragm on the front face of the microphone actually faces the incident normally incident wave right because that will what causes the maximum response of the diaphragm of the microphone. I will just explain myself with one picture. So, essentially a microphone will have a diaphragm, if you have a sound wave which is incident directly on it then the entire pressure the pressure surfaces as we know are going to be normal to this incident direction so on. The diaphragm you are going to have the sort of uniform pressure at any instant of time right, but between different instance it will fluctuate, but at a certain instant it will be uniform. If instead on the microphone surface, you have let say the other extreme where the incident wave just traverses tangential; then what will happen is that this point will have a certain pressure where as the neighboring point will be at a different face right there is a phase difference between this blue and red.

So, therefore, the pressure associated with this blue plain and the red plain will be different. So, different points in your diaphragm will see different phase of the pressure where is here the diaphragm was being loaded by a constant pressure across it is entire surface, here you will see the diaphragm will be loaded by some kind of a harmonic pressure right, which will sort of complicate things, which will lead to erroneous measurement because the calibration has been done the sensitivity charts has been done for a uniform pressure impinging on the diaphragm surface.

So, therefore, for free field microphone it is imperative that you use this microphone with the knowledge that it is supposed to work best when it is pointing directly along the direction of the incident wave. It should not point the other way down or it should not point in a perpendicular direction in which case you are going to have erroneous measurement. A free field microphone if we use this free field microphone in a diffused field condition if we make this mistake of using this free field microphone in actually a situation, where there are multiple waves and you know the phase is all random, then the measurement that will be obtained is going to be underestimated and rightly.

So, because it is not sort of having the capability to factor in this effect that the phase there are multiple reflections that are coming in. Free field microphone is designed on the basis that there is only one wave the incident wave which hits the structure. So, as per that the calibration and the sensitivity charts will be made available to you. So, if you use in the condition of an enclosure and in an enclosure which is spoil, and such that there are multiple reflections happening then these multiple reflections will actually not be properly accounted resulting in an underestimation of the measured SPL values.

So, I guess we leave it at here and we will take a deeper dive into the way microphone and different parameters of the microphone work, will actually derive the sensitivity of a condenser microphone for you and we will show you the attributes of a good what we think is a good microphone for at least the testing purposes. But as I said depending upon the applications the microphones differ, depending upon the economics the microphone differs. For example, in a phone; obviously, the microphone has a different purpose different economics in this caller mike it has a different economics, but essentially the microphone is anything which can convert this acoustic pressure signal into a voltage signal. So, to that extent we must understand that this is the generic use of microphones, but we are interested only in the testing acoustic testing microphones not anything else.

Thank you.