Acoustics & Noise Control
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Module - 16
Lecture - 21
Sound Power Level

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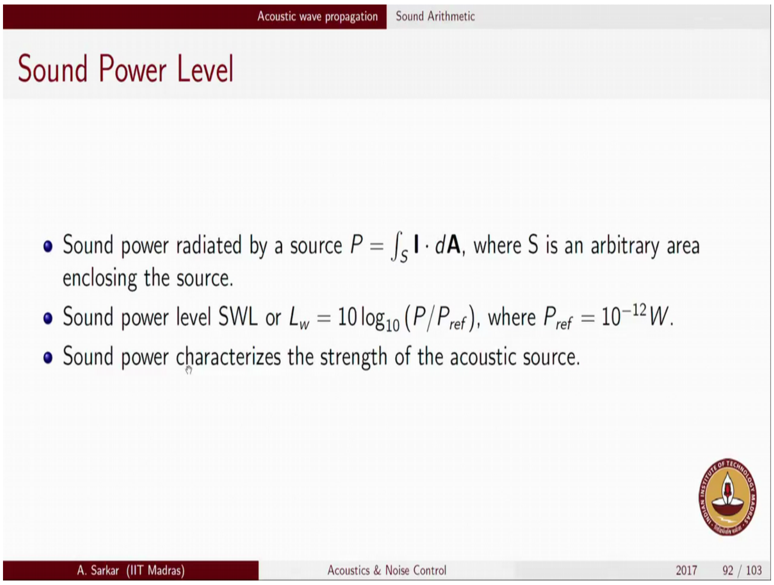

In the last class we looked at various arithmetical operations related to the sound pressure level expressed in dB. So, from sound pressure level we move on to a new concept of sound power level. So, sound power level is defined in the following fashion. For any acoustic source sound power radiated by a source is given by P equals to integral of I dot dA. Where I is the intensity, so intensity I can have to can have two connotation instantaneous and active or time average intensity.

Generally, we will take the time averaged intensity because that is more meaningful; in the sense that this will represent the average power flow per unit area. So, if you remember intensity is the power flow per unit area active, intensity the average intensity time averaged intensity per unit area. So, intensity dot producted with the area; that means intensity multiplied by the taking into account the normal of the area and the intensity itself is a vector quantity. So, I dot dA this quantity will denote what is the total amount of power flowing through an infinitesimal area of magnitude dA and having a unit normal indicated by its vector.

So, if you integrate such pieces of power flow across all these individual differential elements you are going to get the total sound power that is being radiated by the source. So, this is what we will do, that we will construct a hypothetical surface around the source of our interest and over each infinitesimal area of this hypothetical surface we will perform this surface integral operation and then determine the sound power. So, this is very important concept. And we will elaborate soon how sound power level calculations and how inferences based on sound power level is going to be very useful in determining the acoustic problems.

So, sound power level expressed in dB now. So, sound power just like sound pressure was originally defined in terms of Pascal, but then we sort of transform the units to a decibel scale. Same here the sound power as given by this formula will be transferred to a logarithmic mix scale and that is given in the following fashion. So, sound power SWL, sound power level abbreviated as L W recall that for sound pressure level the abbreviation was L P and this is time it is L W. So, that will be 10 logarithm to the base 10 P by P ref; where P is calculated as per the above formula and P ref (Refer Time: 03:14) reference number the reference number is taken as 10 to the power minus 12 watt.

So, that is sound power level expressed in dB. Again so this units is also decibels, but we understand the decibel associated with sound pressure level is different to the decibel associated with the sound power level. So, the important feature to understand is a sound power characterizes the strength of the acoustic source; it does not depend upon the perception at a particular point. And I will come to it in a moment where I elaborate what the difference between sound pressure, sound intensity, and sound power, and in what circumstances which of these quantities to be is to be used.

However, at this point we will emphasize the fact that sound power is what will determine how strong the acoustic sources irrespective of the location of the receiver irrespective of the orientation of the receiver.

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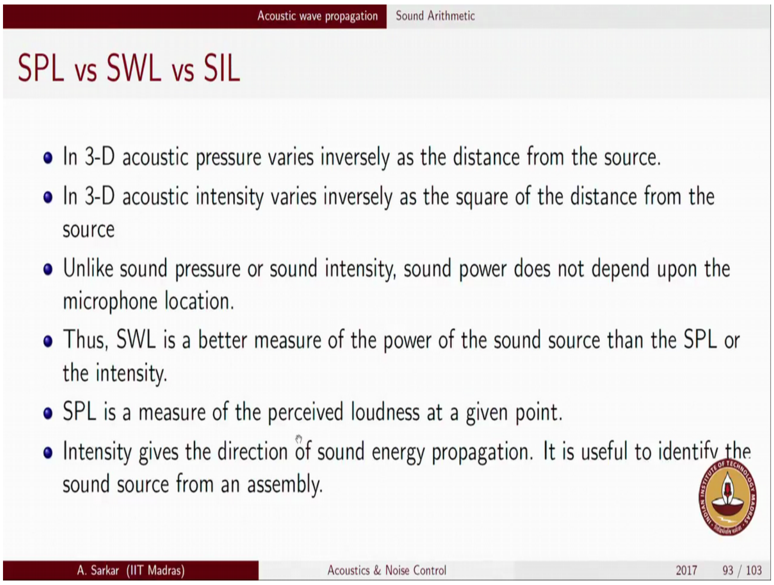

So, here we go. We have define three different quantities: sound pressure level, sound intensity level, and sound power level. SPL is sound pressure level, SIL is sound intensity level and SWL is sound power level.

So, we understand that when we did the surface spherical waves there we saw that the acoustic pressure would vary inversely from the distance of the source. This is quite true for any sort of acoustic processes that happens in three dimensions. That acoustic pressure varies inversely from the distance of the source. And a very intuitive proof of that is that as the source is taken farther and farther away from the receiver the loudness obviously decreases. Therefore, it is quite apparent that this acoustic pressure level will vary from the distance from the source.

In other words if you wish to characterize the source then it also depends upon where the receiver location is right. So, it the sound pressure level at a particular location depends upon the distance of that location from the source. So, this is one aspect which sort of is different from the sound power level idea.

So, in similarly if the orients if you are talking about directive sound a sound which is directed in a particular direction; as an example my voice is directed in front of me whereas the person who is equidistant to you, but on the other side of my face will possibly here more feeble sound. So, that proves that acoustic pressure is directive it is not going to be the same in all directions. Therefore, acoustic pressure at a given distance does not match have much meaning in terms of characterizing the source. it possibly is the most important thing in terms of perception of loudness. If you know that this is the point where you wish to hear least amount of sound then the sound pressure level at that point quantifies what is the amount of sound that will be perceptible at the given location.

But as far as the source characterization is concerned these two factors fact that it depends upon distance, it depends upon orientation sort of complicates this quantity acoustic pressure as far as characterization of the source goods. So, we need a better idea to characterize the source. Similarly, the intensity we have already figured out the formula for intensity; the formula for intensity as we have found out few classes before was that it is a quadratic it is a product of pressure and velocity; pressure times velocity conjugate real part of it multiplied by half is what the formula we derive.

So, we understand it is a product of pressure, amplitude and velocity amplitude for sure. And that intern in if you relate the pressure and velocity through the impedance calculations so in turn it will turn out that they acoustic intensity depends as a varies like the square of the acoustic pressure, because velocity and pressure are related through impedance relation. Now, if you take pressure times velocity it is effectively pressure into pressure conjugate which is magnitude of pressure square.

But then we know that the pressure falls inversely as the distance. So, magnitude of pressure squared of it will fall inversely as the square of distance from the source. So, the problem associated with sound pressure level carries over to sound intensity also, because sound intensity also is not independent of the source location it is very much dependent upon the source; sorry upon the distance between the source at the receiver.

So, the acoustic intensity definitely like pressure will vary with distance, but the manner in which it varies is slightly different than the acoustic pressure quantity. Intensity varies inversely as the square of distance in contrast to the pressure which varies inversely as just the distance. Therefore, unlike sound pressure and sound intensity what we have a sound power does not depend upon the location of measurement. So, in this is respect neither does not depend upon orientation because the total power that is emanated by this source is what we are measuring. I will just illustrate this idea in the notes here. So, the concept of sound power level goes in this manner.

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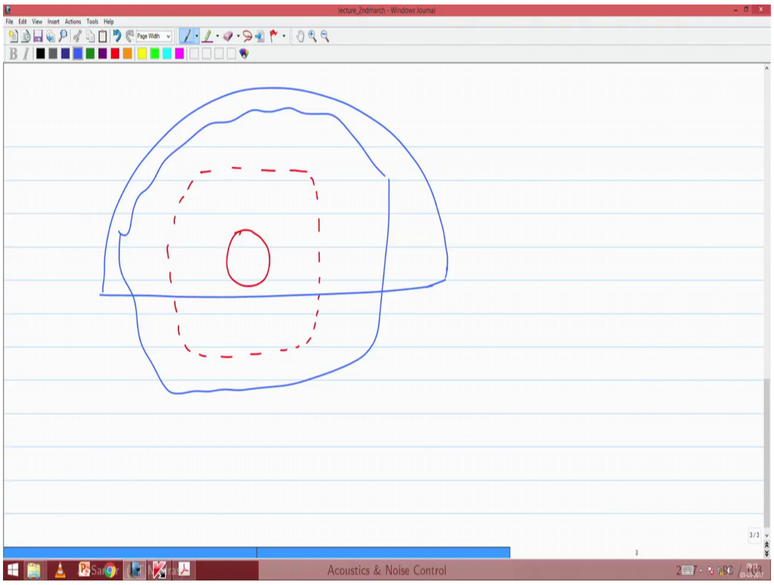

Suppose you have a source you enclose this source with whatever surface you feel like, you could enclosed it through this surface or you could enclose it through another surface or you could enclose it through a hemispherical surface also. Any manner in which you enclose this source the total power, because you are calculating the power that is radiated by the source and as per first law of thermodynamics power has to be conserved. So, in whichever fashion you enclose this source the amount of power with short of diverges from the source is going to be independent of the choice of this enclosing area. In whichever way you enclose it does not matter.

Therefore, this quantity sound power if measured or computed will actually characterize the source irrespective of the measurement locations of the measurement process; other than the fact that you have to take care of the manner in which you do the test. So, provided you do the test in a correct procedural fashion you can be sure that it actually characterized the power that is emanated by this acoustic source without any precondition on the location of the microphones on the manner in which the test has been conducted.

So, in this fashion a sound power result is more meaningful in characterizing the source, whereas a sound pressure result or a sound intensity result probably means much more in terms of the perception as perceived by the receiver at a particular location. Therefore, all you are norms in terms of you know if you have a certain benchmark to attain in for a pass by norm those will be in terms of sound pressure level. But then sound pressure level will not be too much useful or at least would be of limited usage in terms of diagnosing the problem.

If you wish to really diagnose that out of all the sources that are present in your vehicle which is the source which is responsible for creating the maximum sound power then possibly you need to take a sound power measurement or a computational simulation of such a quantity; that will throw much more light.

So, SWL is a better measure of power of the sound source than SPL or intensity, because SPL or intensity depends not only on the source, but also on the receiver. So, that is the complicating feature. Whereas in SPL is going to measure the perceived loudness at a given receiver location. So, the acid test will be obviously to reduce SPL at the desired location, if it is the driver location then at that location the SPL should be reduced. if it is the location where the microphone is held at the time of passerby test that is the location of your interest in SPL at that point will be reduced.

But let us say you are not meeting the test conditions or you are failing to meet the benchmark results in that case you have to go back and think as to which component is responsible for emanating noise which beats the benchmark. And therefore, you have to look at sound power of each of these individual components and then isolate that component possibly redesign it or at least refine it such that the sound power associated with that component comes down.

Intensity also is a very useful tool in sort of mapping the regions where, like as I keep saying there are number of different components which are possible sources of noise and the first step in going through the noise control exercise is to identify which component is a principal noisemaker. So, yesterday as we talk that our sound arithmetic formula give us one interesting manner in which we can sort of rang the sound. Intensity mapping technique is also very useful in this regard. Let me try to quickly illustrate how intensity mapping technique is done. So, what we will do is that again we will enclose.

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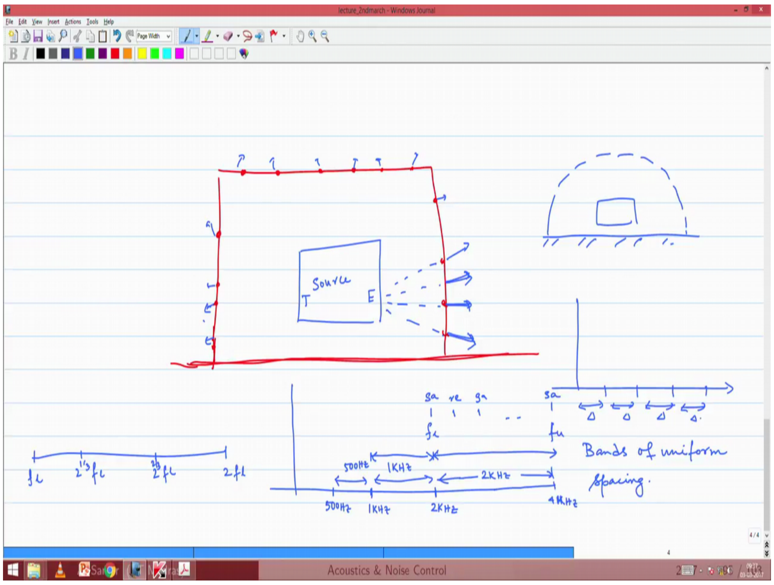

Let us say we have this is our sound source; we will construct enclosure and I am just doing it in 2 D so that it is easy for me to draw, but it applies to 3 D as well. So, will construct a enclosure of this kind and obviously a part of it is just the floor. So, what we will do is we will measure sound intensity at various points in this enclosure and let us say that there are multiple sources here within this is what we are taking as our product there are multiple sources.

So, for example, if you are talking about the power train of your vehicles on one side is the engine the other side is there transmission, you are interested to know whether the engine is the principal noise maker or whether the transmission is the principle noise maker. If you start taking the measurement or accordingly even in simulation if you can perform a simulation where you can find out at each of these points what are the intensity vectors.

Suppose the intensity vectors are looking somewhat like this; these as much smaller and in some random fashion, whereas here you see all these rays seems to be diverging from one point. So, it is seeming like these rays are actually emanating from the engines. So, this conclusively proves that the engine is principally responsible for the power flow across this enclosure, whereas all the other intensity vectors are much small. So, therefore, the chances that the transmission is responsible for the total sound power are minimal.

The looking at these vector plots and some of these vector plots you are going to plot in your next assignment; looking at these vector plots of intensity this is just the time average intensity though what you have been asked in your assignment is the instantaneous intensities. So, looking at these vector plots it is actually possible to figure out whether these vectors are emanating from a source like which is the acoustic son if you make call it. So, that is the acoustic hot spot.

So, then in an assembly of components it is actually very easy to visualize and pinpoint the location of the principal noise maker and choose accordingly the further steps in the noise control process. So, intensity mapping techniques are one very often relied and time tested means by which you can do a noise source identification exercise.

So, as I said therefore, all the three are useful in their own way SPL, SWL and SIL-sound pressure level, sound power level, and sound intensity level. So, again sound power level also will quantify that which of the components within the assembly is at the sub assembly is most trouble maker in terms of the total noise.

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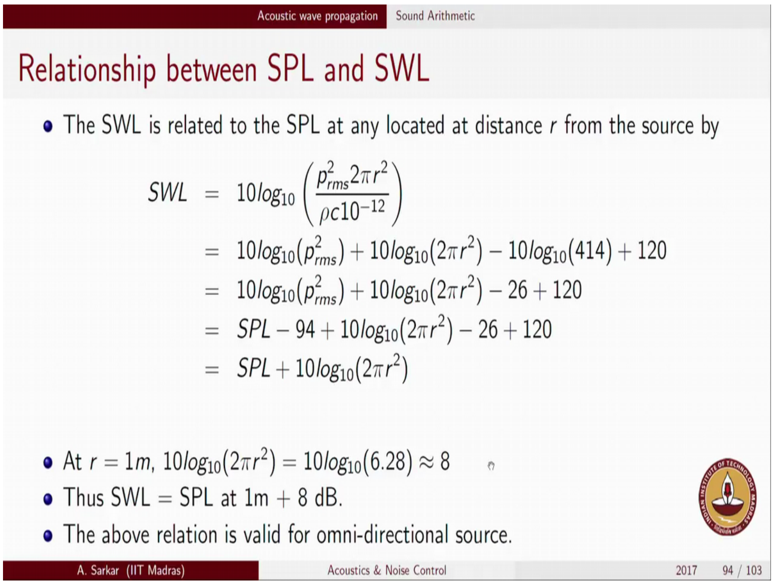

Now it so happens that you can actually relate the SPL and SWL. So, let us see how you can do that. So, the SWL can be related to SPL at any desired location from the source and these are some very useful ways in which we can turn from one sort of a scale to the other scale.

So, you will recall that the intensity at a point if you have a plane wave assumption would be given by P square by 2 rho c; where P was the amplitude associated with the plane wave, but then P rms we are understand is P by square root 2. So, P square by 2 rho c is effectively P rms square by rho c. So, P rms square by rho c is in fact the intensity at that point. So, this group of terms P rms square by rho c is the intensity associated with the point of interest. And that gets multiplied with 2 pi r square; what does that imply? It is a hemisphere.

So, as we will see that in most cases sound measurement would be done in a room and obviously the room will have a flooring. So, if you have a sound source hear what we will do is, we will enclose will take measurements only on the part which is above the floor with there is no way in which we can go below the floor and take the measurements.

Therefore, the area is like a hemisphere which is enclosing the sound source. And that is why these rooms wherein these sound testing is done is called semi any quick or hemi any quick chamber will come to those things when we do take up the experimental methods in acoustics. But at this point I would just like to say that 2 pi r square is the area of the hemisphere which encloses the sound source, like that multiplied with the intensity gives the total sound power and that has to be divided with the reference value of 10 to the power minus 12.

So, this is the formula for sound power level for the given source. And then if you just do routine simplifications using the properties of logarithm. So, what I have done in the next step is that I have broken down this product of the numerator as two terms; addition of these two terms and then you get a negative of 10 log 10 rho c. So, that comes rho c for air is c is about 340 and rho is about 1.2, so that product gives about 414; rho c are the characteristic impedance in SI units is going to be 414.

So, this is the denominator associated with rho c with it comes with a minus sign. And lastly there is a 10 to the power minus 12 and you have to take minus 10 log 10 of 10 log base 10 of 10 to the power minus 12 and that is why you get a 120 as the last term. And in the next step what I have done is just I have replace 10 log 10 414 it is approximately 26. So, that part is easy. You will recall that the SPL can be related in terms of P rms; I mean 10 log 10 P rms square plus 94 was the SPL which is what we did in our introduction itself. So, here we are just converting the formula for 10 log 10 P rms square to SPL minus 94; that part also should be easy. So, we have SPL minus 94 minus 26 plus 120 plus 10 log 10 2 pi r square.

So in fact, what you see is 94 and 26 is also giving you 120. So, these two numbers will get cancelled with 120. As a result what you have a sound power level is the SPL value plus 10 log 10 into 2 pi r square. What is R? R is the radius of the hemisphere over which the sound values have been recorded. And if you say that you will record it at 1 meter which is again as per certain standards you are supposed to record it at 1 meter. So, at R equals to 1 meter if you do this calculation 2 pi of 1 is 6.28. And if you take log and multiply with 10 this is approximately going to be 8.

So, as a result the very handy formula for converting between SWL and SPL is going to be SWL equals to SPL at 1 meter plus 8 dB. But please remember here there are lots of assumptions that have been used in this derivation of this handy formula. One, we have taken a plane wave approximation for the intensity. Second, we are actually ignoring the directivity of the sound- we are saying that all the directions are having the same value of sound pressure; that is also not quite true.

So, SWL equals to SPL at 1 meter plus 8 dB please you will keep hearing these lots of industrial standards and industrial practices, but please be well informed that this formula comes with a lot of pinches of salt. So, it is not that rigorously define, it is just sort of an approximate thumb rule you may wish to locate look at it that way. But this is a very useful manner in which you could shift between SWL and SPL.

As I said the above this relation is valid for Omni directional source and distance quite far from the sources such that you have the plane wave condition getting satisfied.

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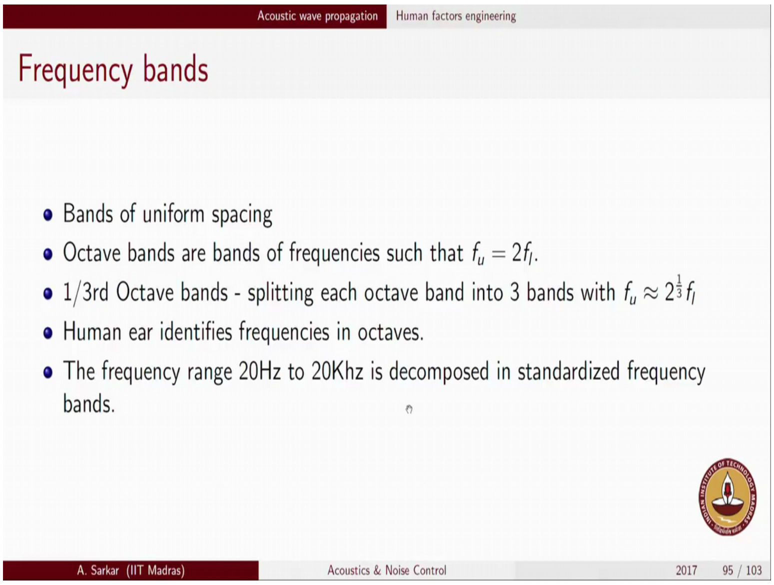

Till now we have been looking at the sound pressure level and the associated quantities use which is the x axis of our acoustic plot; sorry the y axis of our acoustic plot. The x axis is generally the frequency axis. So, we will sort of calibrate this frequency axis also in a little different fashion. And towards that end we will have to define frequency bands.

So, frequency bands the way they originate is I mean what we are going towards is basically the octave bands. So, these frequency bands will be bands of uniform spacing. Sir, you could say that you wish to define, let me just make myself clear. So, on the frequency axis you could either define bands of uniform spacing. So, all these are bands of uniform spacing. All the deltas associated here will be same. So, these are bands of uniform spacing.

Maybe when you take your data through a digital acquisition system data acquisition system you take it at the frequency interval of 1 hertz sometimes 0.5 hertz, but at a uniforms spacing. So, you could choose the frequency bands at a uniform spacing, but also you could another practice especially very applicable in acoustics is the practice of choosing octave bands. So, in the octave band choice the bands are not uniformly spaced, but the bands are chosen in the following fashion where the upper limit is equal to twice the value of the lower frequency limit.

So, let me just illustrate that again to you. So, in the octave band situation what we will do is in this fashion. So, here let us say we have the lower limit as 1 kilohertz. The upper limit associated with this band will be 2 kilohertz. So, this band is of spacing 1 kilohertz, the next band will be from 2 kilohertz to 4 kilohertz.

That means, this spacing is going to be 2 kilohertz. What about the band preceding this one. So, the preceding band will start from 500 hertz. So, accordingly the band will be off size 500 hertz. So, at times there is good reasons why and I will tell you the reasons in a moment that there are reasons why we would like to choose bands of this kind where the upper limit and the lower limit adjust the; sorry the upper limit is double the value of the lower limit. That essentially means that you will no longer get a uniform band gap or in uniform and bandwidth in the frequency axis rather these band which will be in a geometric progression.

One way in which you can convince yourself that at least the acoustic perception depends very much on these type of scale, this geometric progression scale rather than a linear scale is that if you have played harmonium are probably any similar instrument I would not know much about piano, but at least for the harmonium the Indian music if you recall. So, in the Indian music system it so happens that we have the seven Sur's- sa re ga ma pa dha ni sa; and what is that, that is precisely the scale in the harmonic.

So, within the scale you will have sa re ga ma pa dha ni sa all the Sur's defined. So, this when you sing a song or when you sing this rag or whatever you call it sa re ga ma pa dha ni sa it happens to fit one octave or a band of this kind where the upper limit and the lower limit is just the related by a integer factor of 2. So, when you say that you are going higher up in the scale you are actually traversing the next band.

So, when you play harmonium sa re ga ma pa dha ni sa in the first scale you are possibly traversing this frequency band, and when you when you are saying that you are shifting to a higher scale you are possibly traversing the next frequency band. So, the point is this that the human perception of frequency is such that it tends to perceive; the differences in frequencies are perceive more in a geometrical progression scale than in a linear scale.

Because you can distinctly make out at least for music you can distinctly make out that whether you are singing in a scale between 1 kilohertz to 2 kilohertz or from 2 kilohertz to 4 kilohertz or between 500 hertz to 1 kilohertz. It is actually easy to distinguish between 500 hertz and let us say 700 hertz, but it is very difficult to distinguish between 3000 hertz and 3700 hertz, because you are now at a higher band and in that band the bandwidth is higher. So, this 700 hertz difference or which was very much perceptible in the lower band will now been no longer very perceptible in the higher band. So, a beginner like me when we make a mistake of putting the frequency in our vocal cords that difference may actually may unnoticed at least for the novice here if it is played at out there higher scales. But it will definitely get notice if it is at a lower scale.

So, it is actually very convincing to me this is one of the reasons probably I would say that how I do convince myself that we actually here in octave bands, we do not hear in terms of linear frequency axis. As I said you can play with mat lab command sound and do this for yourself. You play out a sound of let say 600 hertz, you play out a sound of let say 900 hertz and ask yourself or maybe you should do it with a friend of yours without telling him that what is the different frequencies that you are playing. And ask him whether he is able to discriminate between these two sounds.

You will see in the lower end he will be able to discriminate, but in the higher end with the same difference of frequency this perception will sort of fade out, because in the higher end of the frequency axis our perception is over a broader band of frequency; only if it changes by one octave we distinctly perceive. The difference between the starting sa and the ending sa of sa re ga ma pa dha ni sa is actually double, and that is what is like a clearly perceptible difference in frequency.

Whereas, the other works by the way re ga ma pa dha ni sa are all in geometric progression they are never in arithmetic progression. So, within this scale of f l and f u u you will need to put all the seven Sur’s. So, that is exactly how you can construct your artificial simulated piano if you want to. So, this is the starting point sa, next will be re. next will be ga. And so on, and finally here you will end with the next sa. So, all the seven frequencies will be fitted within the same octave band. So, that is one way in which we could relate our at least Indian music system to these ideas.

So, octave bands are these bands which exactly rely on this perception of sound. And hear in the human factor in the acoustic engineering comes in. So, octave bands a sort of more tune towards the human perception of sound, whereas this bands of uniform spacing as you will measure out using a data acquisition system is data acquisition systems will just measure in its pure return form the acoustic signal at uniform bandwidth; usually 1 hertz sometimes 0.5 hertz also would be employed.

So, that is the difference between these two types of bands. And since we hopefully understand these bands of uniform spacing I will quickly elaborate octave bands and also one-third octave bands in the remainder. So, what happens in one-third octave bands is that we split each octave band into three sub bands. And again the splitting is not linear the splitting is in geometric progression. So, f l and f u this time will be related in 2 to the power 1 by 3, because remember three of these one-third octave bands must be collated to get the entire octave band.

So, each octave band; so an octave band will comprise of a scale between f l- the lower frequency limit and 2 f l. So, what will be done now is that we will split this into three parts. So, this will be 2 to the power 1 by 3 f l, this will be 2 to the power 2 by 3 f l, and finally this is 2 f l. So, the widths are obviously different because these are in geometric progression. So, the frequency bandwidth of octave band one-third octave one-eighth octave or whatever you can say is going to be different. And I shall quickly show you these octave band values also.

As I said through the analogy of music and the perception of music we tend to hear in an identify frequencies in octaves rather than this linear frequency scale. So, our perception of frequency our ability to distinguish two different frequencies seems to be different across the frequency scale. And therefore, octaves are sort of more suited to the human perception of (Refer Time: 34:23).

So, what is done is that as per as standardization the entire audible range from 20 hertz to 20 kilohertz will be decomposed into these standardized frequency bands. So, f l and f u are not like any arbitrary number which you and we can define, it has been already standardized and this is how the standard picture looks like.

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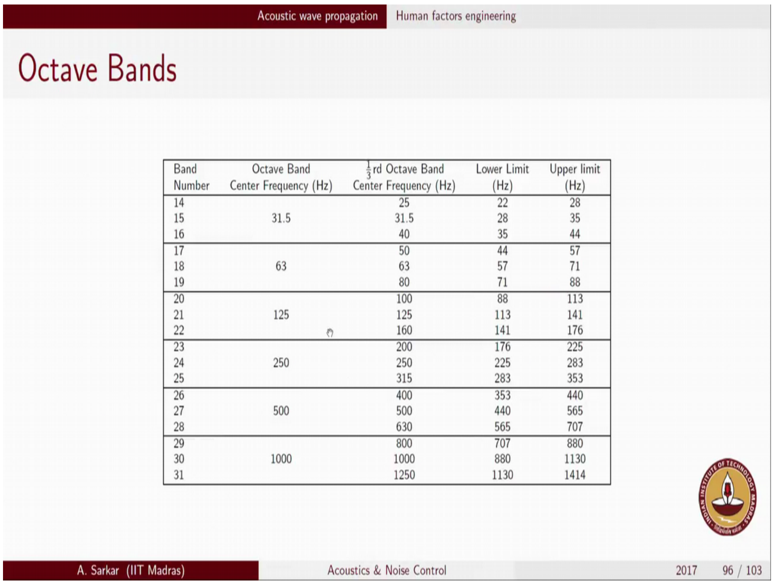

Band Number	Octave Band Center Frequency (Hz)	$\frac{1}{3}$rd Octave Band Center Frequency (Hz)	Lower Limit (Hz)	Upper limit (Hz)
14		25	22	28
15	31.5	31.5	28	35
16		40	35	44
17		50	44	57
18	63	63	57	71
19		80	71	88
20		100	88	113
21	125	125	113	141
22		160	141	176
23		200	176	225
24	250	250	225	283
25		315	283	353
26		400	353	440
27	500	500	440	565
28		630	565	707
29		800	707	880
30	1000	1000	880	1130
31		1250	1130	1414

So, I will just take one random band. So, band number 30 is octave band number 30 is identified with an octave band which has a center frequency from thousand numbers the center frequency if intern means a lower limit of 707 and an upper limit of 1414. Please note that the upper limit and the lower limit is going to be related by a factor of 2; 707 and 1414 is the band for band number 30. When you look at this table try to identify the bands please look at 30 if you are identifying the octave bands you should look at the numbers which are divisible by 3.

So, 30, 27, 24, 21, 18, 15, these are the band numbers which are associated with the octave bands. The intermediate once are the one-third octaves. So, 29 band number is a one-third octave band which starts at 707 and ends at 880. If you strictly do this calculation you will say that 2 to the power 1 by 3 multiplied by 707 is not exactly 880, but it is approximately 880.

We do not want to complicate this table by including decimal numbers and look making it look pliancy rather, we would like to have numbers which are nice integers. And towards that end which slightly tweak that formula or rather approximate that formula with 2 to the power 1 by 3. But you can verify that all these numbers do follow that rule that roughly for example here: in the band number 27 you will see 353 and 707.

So, 353 into 2 should actually be 706, so it is 707. The reason why it is 707 not 706 is because the next band starts at 707. So, if you had given 706 here then the complication would have been that what happens to the frequency band between 706 to 707. So, little bit of tweaking twinning has been done from this very god given formula that octave band has to be like the upper limit has to be twice the lower limit. With the little bit of tweaking you are shown this table which is as per the standard please remember this table is not constructed by as it is as per the standard.

And therefore, in the standard practices people will actually referred to terminologies such as octave band number 30 or one-third octave band number 30. You should also make this difference when use refer to octave band 30 as I said it is 707 to 1414. But when you say band number 30 in the one-third octave band then it refers to 880 to 130. This clearly shows that within an octave band there are three one-third octaves. And please note the duration or the widths of these bands are all different it is not of the same width. So, this is about octave band.

So, as I said it starts at around 20 hertz and you have it still even 202 kilohertz.

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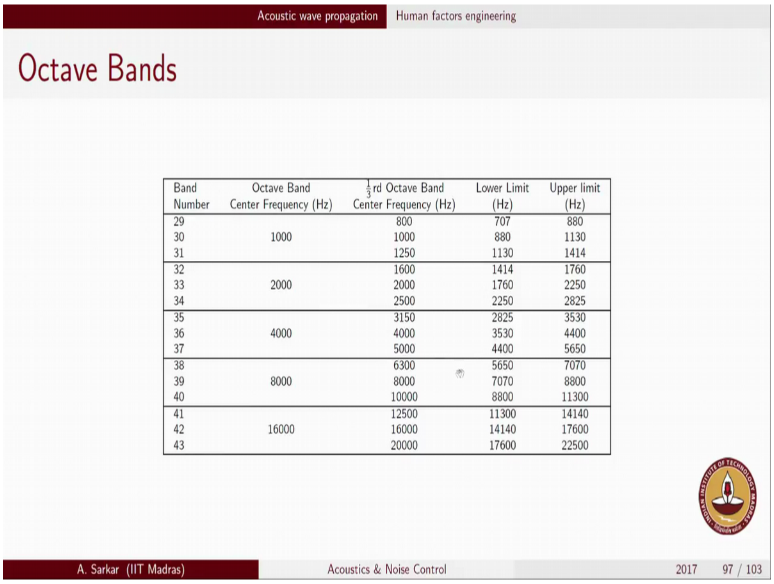

Band Number	Octave Band Center Frequency (Hz)	$\frac{1}{3}$rd Octave Band Center Frequency (Hz)	Lower Limit (Hz)	Upper limit (Hz)
29		800	707	880
30	1000	1000	880	1130
31		1250	1130	1414
32		1600	1414	1760
33	2000	2000	1760	2250
34		2500	2250	2825
35		3150	2825	3530
36	4000	4000	3530	4400
37		5000	4400	5650
38		6300	5650	7070
39	8000	8000	7070	8800
40		10000	8800	11300
41		12500	11300	14140
42	16000	16000	14140	17600
43		20000	17600	22500

So this is the last band; band number 42; there octave band number 42 which goes from 11.3 kilohertz to 20.5 kilohertz. So, this is as per the standards how the octave bands are defined.

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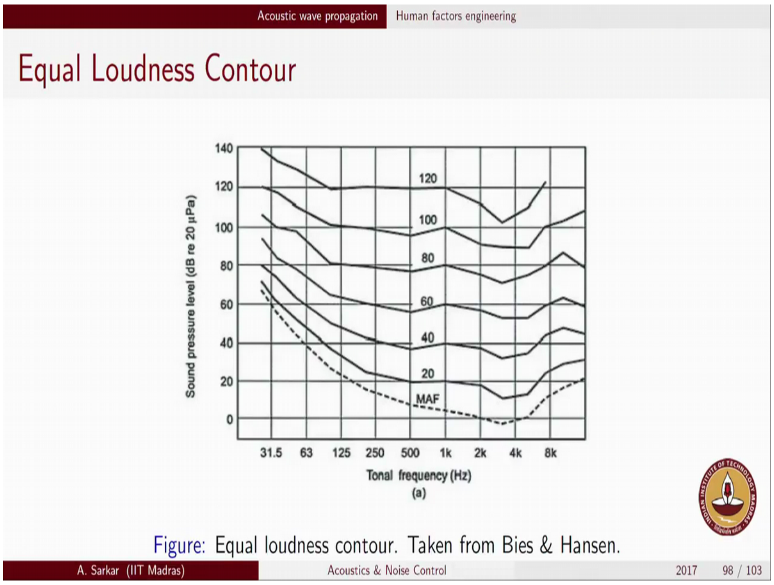

So, as we said that human perception of hearing is dependent on frequency. So, it turns out that even though we are saying that it is a sound pressure level which sort of matters the most, but again through extensive experimentation on various human subjects it has been found that the sound pressure level at various dBs, but a different frequencies are not perceived as equally loud. This graph is what is known as the equal loudness control. And again this is done in the literature through extensive experimentation and sort of standardized by now.

So, what each of these curve shows are there is the following that: a 20 decibel sound which is in this band of 500 to 1 kilohertz is perceived as equally as loud as let us say 60 decibel sound or little more than 60 decibel sound at this 31.5 hertz. So, this in turn means that we are not very sensitive in this low frequency. In this low frequency let us say about 125 hertz or so you see that there is actually a sharp a steep increase of these curves; which means that to excite your ear drums you actually need to have much higher sound pressure level.

Again if I have if I pick up one more a 40 decibel sound in this 1 kilohertz range is equally loud as let us say 75 decibel sound at 31.5 hertz. So, you cannot simply rely on this decibel base scaling to capture the effect of the perceived loudness, because sound pressure being the physical source by which our perception of hearing is triggered is all very fine, but due to our anatomical features and you do not various complications associated with the human factors involved in the perception of this audible sound there is more to it than meets the eye.

And therefore, just plain vanilla implementation of the decibel levels possibly does not incorporate the intricacies associated with human hearing which is as shown in this plot. Because, had it been true that sound at the same sound pressure level should have been equally loud. So, all these graphs in that case should have been flat, but that is not what it is shown. It is definitely showing a trend wherein the lower frequencies are not so very sensitive. The lower frequencies in other words even if you have a higher sound in the lower frequency zone I mean starting from 31 hertz you possibly would not have much of it the annoys associated with it.

In fact, as I told you when I did this experiment on myself I found that I am absolutely not sensitive to this below hundred hertz. So, I take it at this way that I do not hear anything below 100 hertz. So, the question therefore, that remains is how can we incorporate this idiosyncrasies associated with the human hearing.

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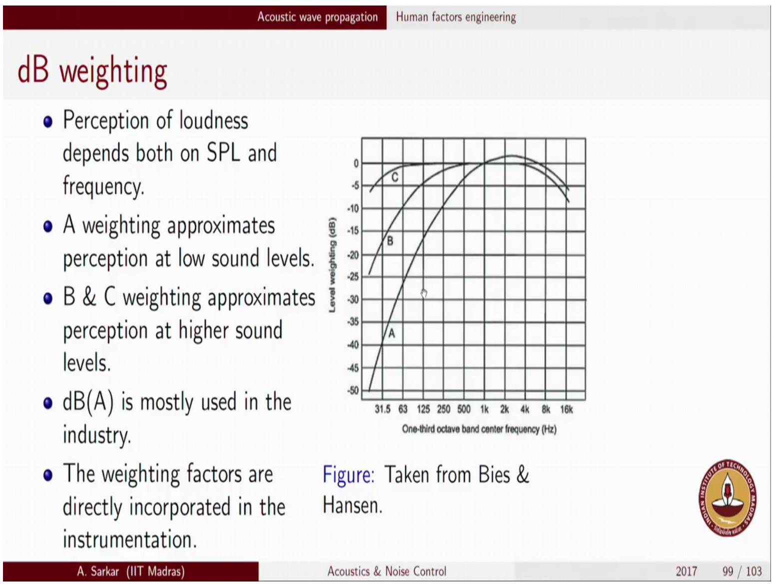

So, the answer is we will incorporate this through a waving factor. Now that we understand that the lower frequencies need to have an underweight, because the lower frequencies are not so very sensitive in triggering our perception. So, we evolve certain weighing factor. So, whatever is the decibel value we correct the decibel value based on this way in factor. And you will see there are three types of weighing factors which are possible A, B and C, but all of these weighing factors sort of down place the lower frequencies of the zone which is below 1 kilohertz is heavily downplayed.

The zone which is most sensitive is like 1 kilohertz to 4 kilohertz around here it is most sensitive. So, whatever is the decibel value you on top of that decibel value you make the correction factor if you are having a decibel of let us say 80 decibel at 31.5 hertz, you simply subtract minus 40 because you know the frequency associated is the with 31.5 is gives you a correction factor of minus 40.

So, whatever is the plain vanilla calculations showing for the decibel levels you need to subtract or correct that number with this correction factors given by these graphs. When you use each of these graphs A B or C you get a different scale which is call dB A dB B or dB C. Uut usually dB A is the one which is most commonly used and this is the units in which the pass by norms are expressed. If you recall in the first class we have seen the pass by norms they are all expressed in dB A.

So, more often than not you will encountered the unit as per standardization to be dB A rather than in dB itself or rather than dB B or dB C. These other two weighing factors are used in other applications possible in the aerospace industry there are some applications. And because these are sort of approximating the sound at the higher sound levels not at the lower sound levels. So, dB A as I said is most commonly used in the industry. And these weighing factors are actually incorporated you do not have to do anything when you take a sound level meter there is a button wherein you can say that I want the A weighted sound pressure level expressing decibels. So, this weighing factors will be automatically incorporated in the hardware or in the software if it is a data acquisition system, then you will directly be able to take the measurement in terms of dB A.

But please understand the units of sound can vary from simple Pascal's which is just the linear implementation. And then it can go to decibels, and their reason for a decibel implementation is to compress the range and also as we will see the perception of sound also seems to follow a logarithmic scale rather than a linear scale; just like the perception of frequency seems to follow a geometric progression scale rather than a linear scale.

And lastly this A weighting has been taken in to incorporate this feature of our human hearing wherein we tend to downplay the effect of lower frequencies. Will stop here and will continue in the next class.

Thank you.