NOISE CONTROL IN MECHANICAL SYSTEMS

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Week:03

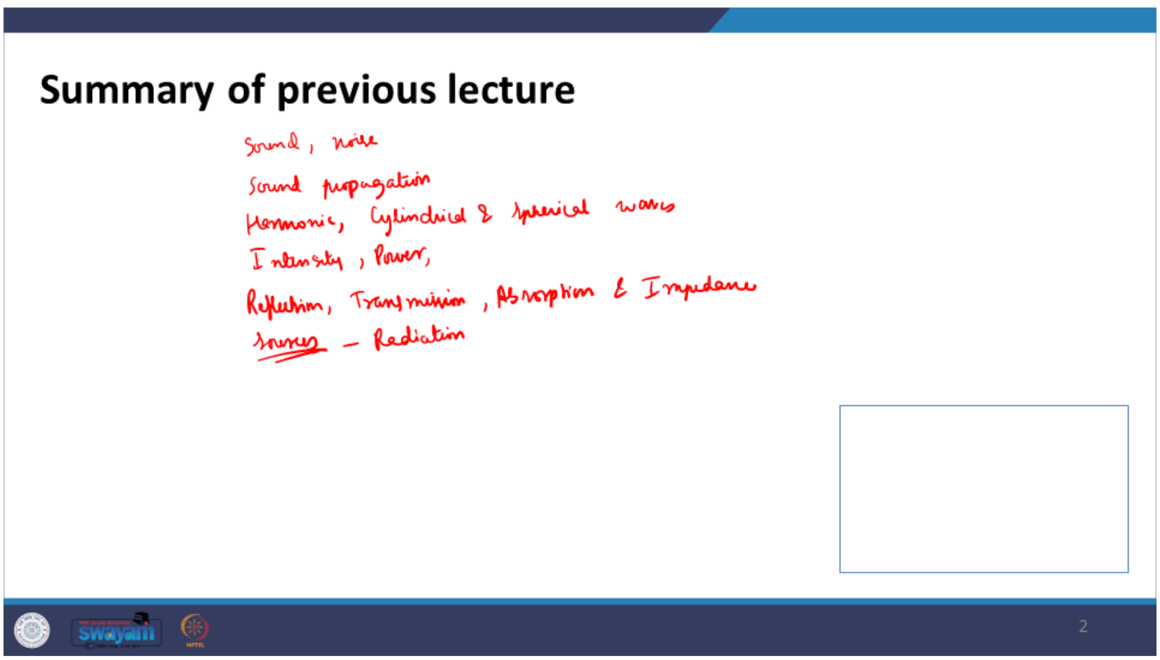
Lecture:13

Lecture 13: Time domain analysis of sound signals

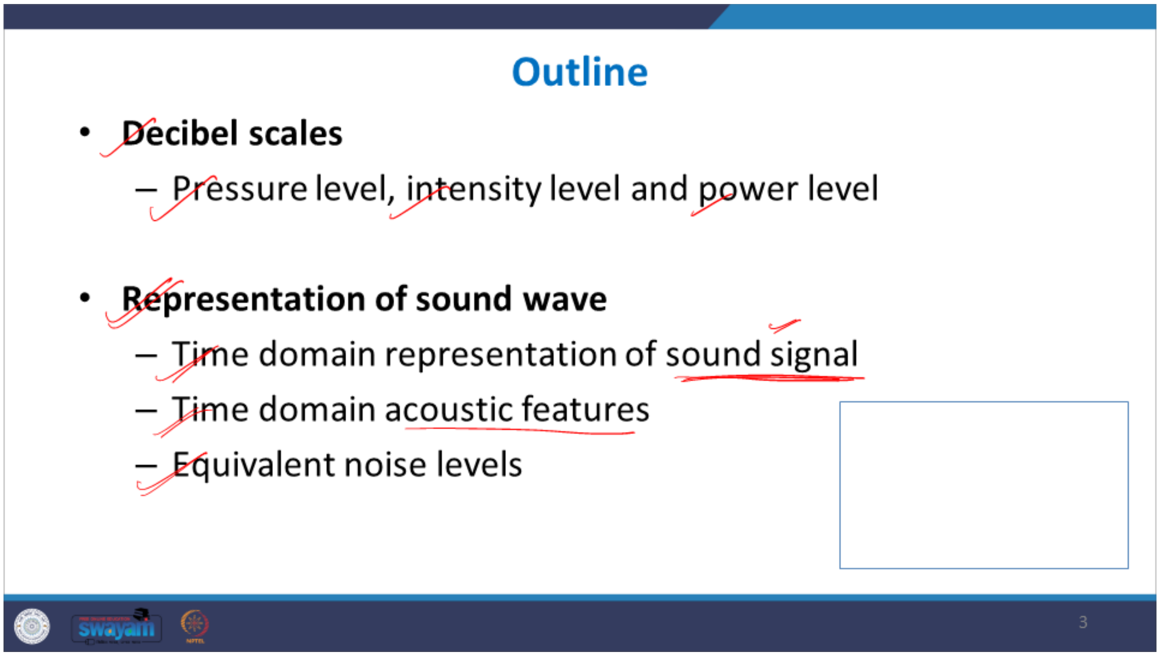
Hello and welcome to the lecture 13 in the series on noise control in mechanical system. This is Professor Sneha Singh from IIT, Roorkee. So, in this lecture 13, we will deal with time domain analysis of sound signals. So, over the couple of weeks and these lectures forward, we will be discussing about the module named sound signal analysis. where we see time domain and frequency domain analysis.

So, to firstly summarize what we have done so far. So, far we have learnt about the concept such as what is sound, noise and then sound propagation. Then we have found out the solutions to sound propagation for harmonic, cylindrical and spherical waves. Then some concepts like intensity, power and you know reflection, transmission, absorption. and impedance.

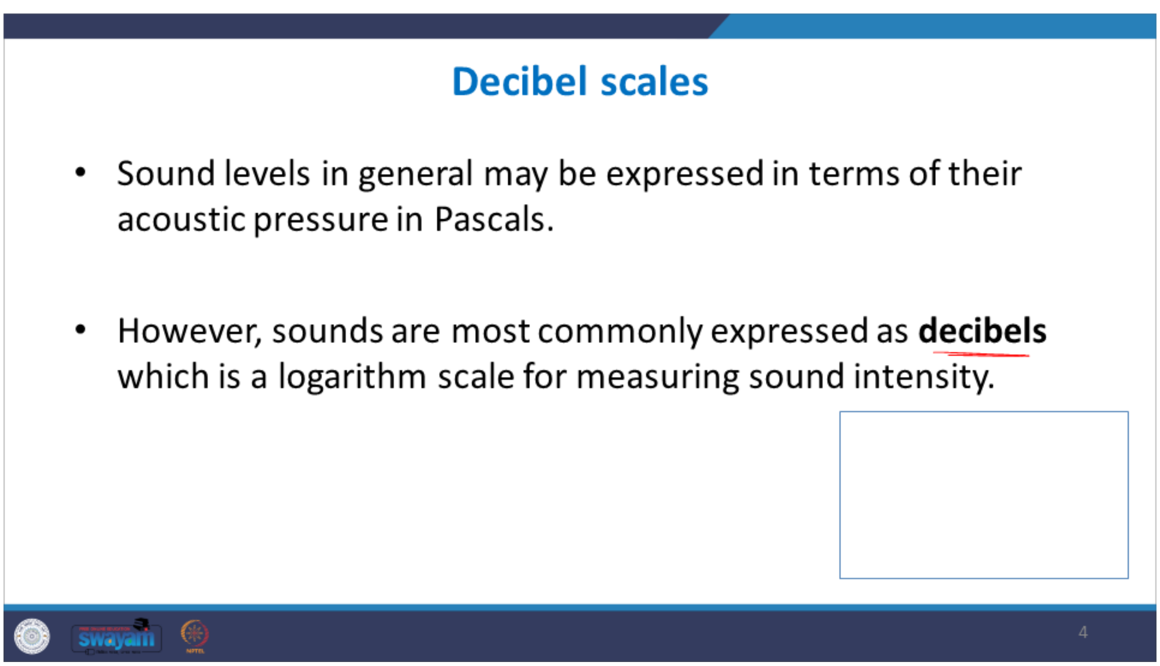
So, basically overall to sum up you know and some acoustical sources as well and their radiation we have discussed. So, overall how sound wave propagates? So, we have discussed some acoustic fundamentals and the physical phenomenon of sound wave propagation. Now, we come to the engineering part of it which is we see how we analyze the sound waves and you know represented in some you know interpretable numbers which can be used by the noise control engineers.



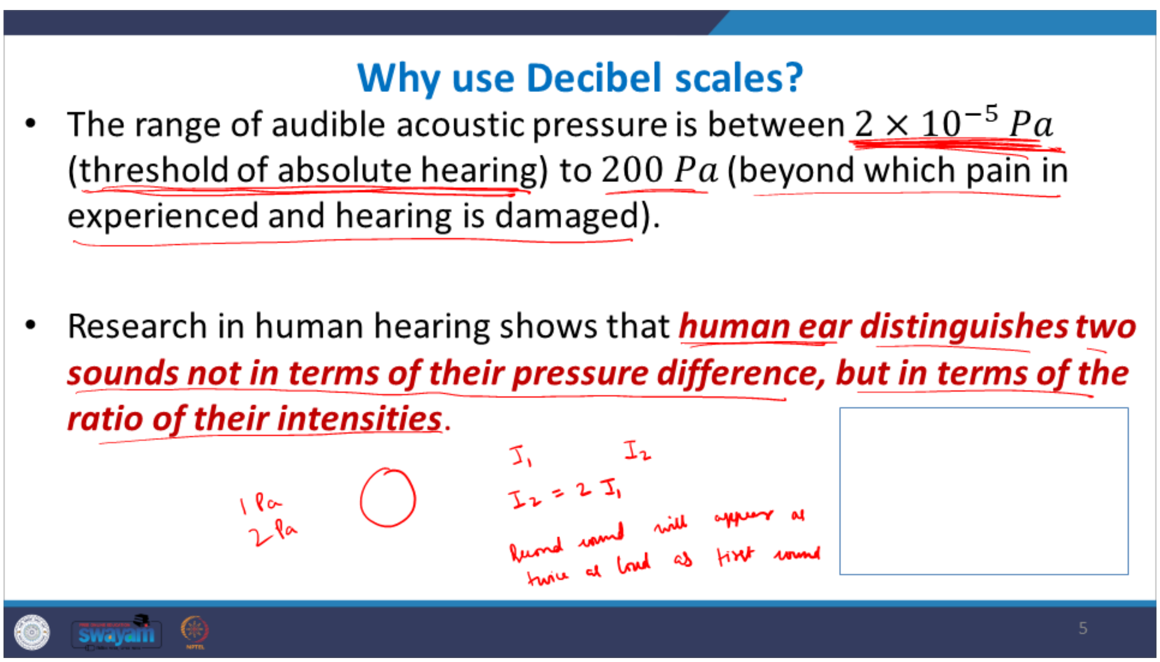
So, we will introduce decibel scales okay and within the decibel scales we have three important measures which is the sound pressure level, sound intensity level and sound power level and how to represent the sound wave in a time domain as a sound signal. So, from now on before that I was considering sound as a wave and as a physical phenomenon, but now this is a signal that has been measured by us and given to us and how we do the engineering and we obtain some interpretable numbers. Equivalent noise levels concept will also be introduced and various other acoustic features of sound signal.



So, decibel scales, decibel scales were introduced to measure the level of sound now you would think that we already know that sound is basically a variation in the pressures that fluctuates and carries forward in space and time so the level of sound can very easily be measured by the acoustic pressure or the acoustic amplitude of the sound so what why do we need a separate scale for that so the need is because You know,



the range of audible acoustic pressure that we can hear is somewhere between 20 micropascals, which is actually the threshold of absolute hearing, to around 200 pascals beyond which it causes pain in the ear. ear and hearing gets damaged. So, we have a huge range if you think about it from 10^{-5} to 10^{2}. It is extremely large numbers and very large variations in the magnitude. And it has also been found that from the various studies done on human hearing that when suppose a human hears a sound of let us say 1 pascals and a sound of 2 pascals. okay then it if you hear a sound of one pascal and then a sound of two pascal the ear is not able to distinguish and say that it is a sound whose intensity or whose pressure has doubled so it doesn't feel like that the level has doubled in fact from the various research into human hearing and listening studies what has been found is that The human ear distinguishes two sounds not in terms of their pressure difference, but rather in terms of the ratio of the intensities, which means that suppose there is a sound I_1 and there is another sound I_2 such that I_2 is twice of I_1. Here I_1 and I_2 are the acoustic intensities of two separate waveforms of sound. Then to a human ear this would be you know the second sound will appear as twice as loud as the first sound. So, here the pressure has not doubled, but rather it is the intensity. So, the human ear can actually distinguish between the ratios of intensities and not the pressures in a linear scale. okay



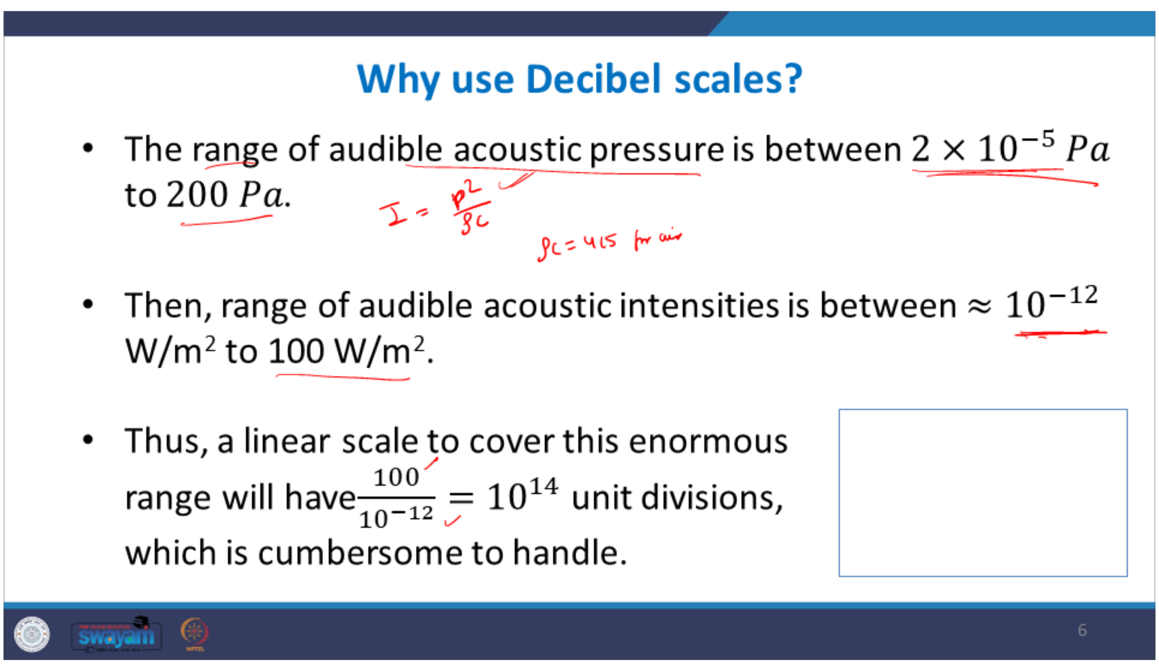
Now, if you think about this, the range of audible acoustic pressure is somewhere between this to this. Now, over here if this is the range in the pressure then I is what the acoustic intensity which is

$$I = \frac{p2}{\rho oc}$$

for a generic wave front. Then in the far field then the range of audible acoustic intensity is considering rho c as 415 for the air. You will find it is approximately between 10^{-12} to 100 watts per meter square. for this. So, if you see here suppose we had some scale because human ear distinguishes between intensities rather than pressure.

So, if we had some scale where we could measure these intensities then the highest number is 100 the lowest is 10^{-12}. So, it is almost 10^{14}. So, if it is a linear scale we would need 10^{14} unit divisions minimum the minimum 10^{14} unit divisions which is a very cumbersome number to handle. And that is why to make things easier to handle for engineering purposes what has happened we have introduced the logarithmic scale which is also called as the decibel scale.

So, here instead of having linear numbers we will have logarithmic numbers.



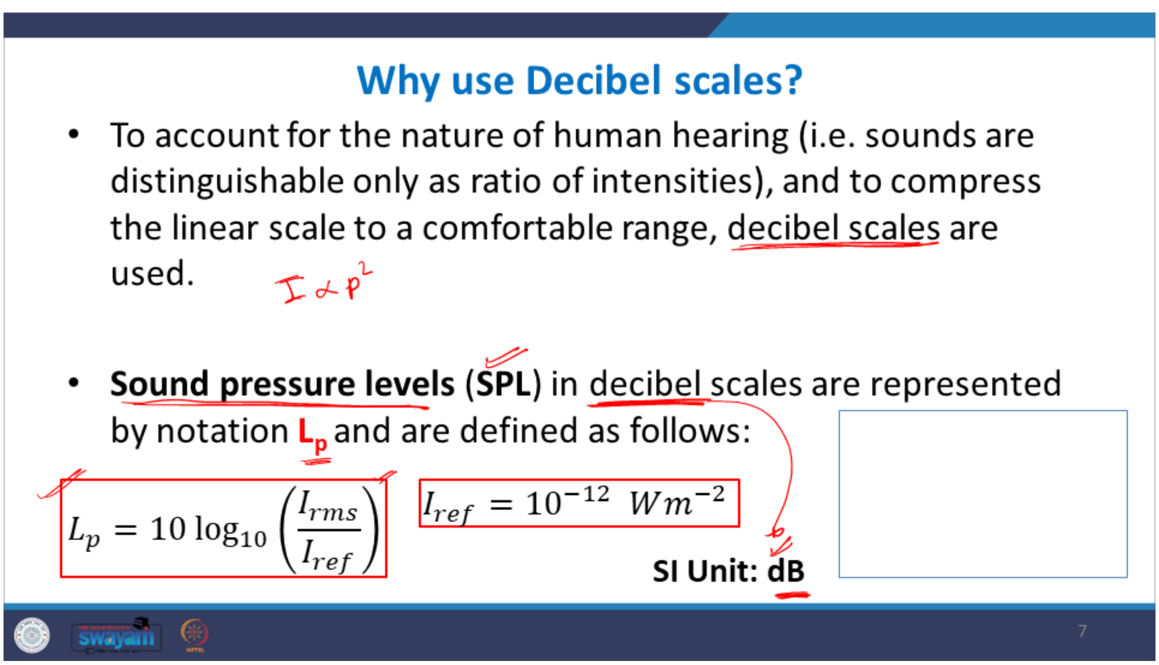
So, in the logarithmic scale we first calculate the sound pressure level this is an important term which in noise control comes like various times it is very important for noise control. So, the decibel scale. Here SPL is also represented by L_P which means the level of the pressure. So, the sound pressure level is given as

$$\mathbf{L_p = 10\log_{10}\left(\frac{I_{rms}}{I_{ref}}\right)}$$

So, what we are doing is that we are taking the reference intensity and what is that reference intensity? It is the intensity which is minimum audible ok. So, over here which corresponds to the threshold of absolute hearing this is the pressure of the threshold of absolute hearing and if you convert it using this formulation into the intensity it is approximately 10^{-12}. So that becomes your reference intensity because you start hearing from that level and then you compare the ratio of the actual intensity with the reference level and you do 10 log 10. So obviously where will this scale begin?

This scale will begin when? So, the absolute hearing is at I reference. So, if you see here 10 log 10 of $I_{reference}$ by $I_{reference}$ would be 0. It is log 10 of 1. So, the scale starts from 0.

So, SPL at 0 means it is the beginning of your hearing. It is the threshold at which hearing begins for a healthy human ear. And then various other levels you get. you represent the units of the sound pressure level as dB which is like a short form of decibel okay.



Okay so another way to represent the same sound pressure level which from now on let us refer to that as SPL so one way is that we know that the intensity is always if you are considering the same medium intensity is directly proportional to p square.

So, this equation the ratio of the intensities we can represent in terms of the ratio of the square of the pressures where p reference becomes again what was introduced earlier which is the threshold of absolute hearing. So, the corresponding p reference comes here over here and p_{rms} of the given sound which whose sound pressure level we want to measure. So, for gases most of the gases including air the $p_{reference}$ is 20 micropascals whereas for liquids we have 1 micro Pascal ok.

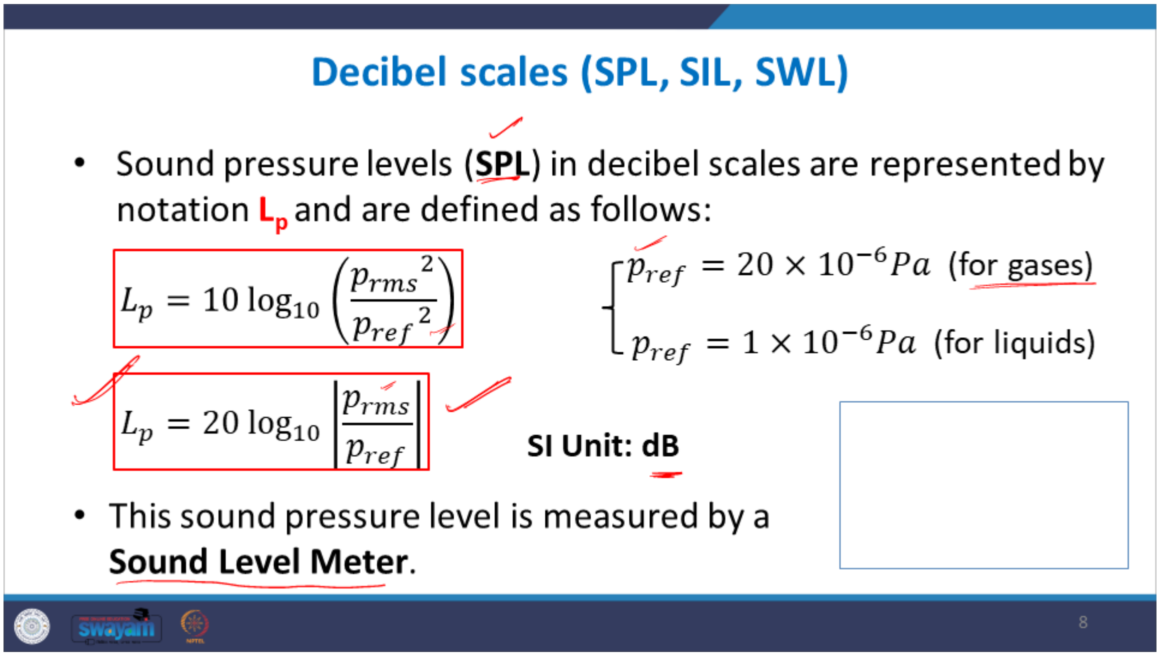
So, over here if you see so this is

$$\mathbf{L_p = 10\log_{10}\left(\frac{{p_{rms}}^2}{{p_{ref}}^2}\right)}$$

So, by the property of log this quantity whole square it comes outside and gets multiplied.

$$\mathbf{L_p = 20\log_{10}\left|\frac{p_{rms}}{p_{ref}}\right|}$$

again an important relationship and the units is as usual the decibels. So, these become the two important relationship Usually, we measure this using sound level meter. We will have a separate lecture on instrumentations used in sound signal analysis where you will see how it is measured.



Some other measures other than sound pressure level is the sound intensity level SIL the same way as we had here so you see that the sound pressure level and the sound intensity level they end up having the same definition one is expressed in intensity the other is expressed in pressure and as you can see from this derivation that the same thing when derived like this will end up getting this particular equation so this equation

$$\mathbf{L_I = 10\log_{10}\left(\frac{I_{rms}}{I_{ref}}\right)}$$

And this equation

$$\mathbf{L_p = 20\log_{10}\left|\frac{p_{rms}}{p_{ref}}\right|}$$

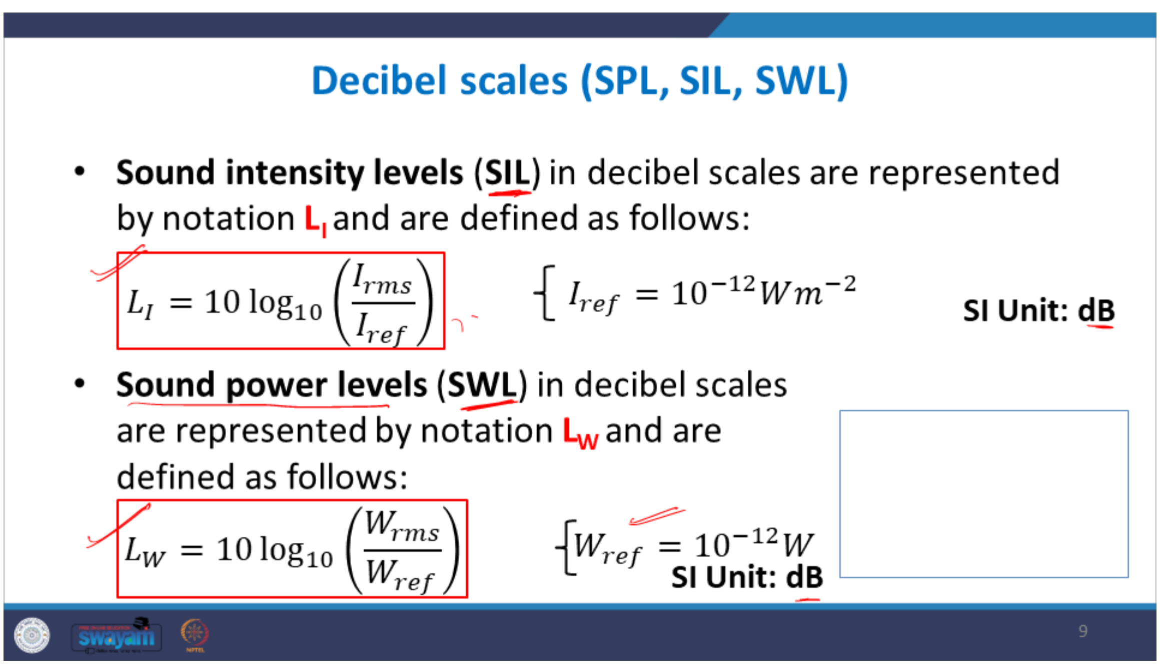
are essentially the same equations so SPL and SIL are essentially same okay now there's something called sound power level now here i would like to say and recall what i have told you in some of the previous lectures is that the intensity level or the pressure of the intensity and the pressure of the sound especially if it's a spherical wave front or a cylindrical wave front Here the, you know, it is dependent on the point of measurement. So, at what point you measure the intensity or the pressure.

So, the pressure and intensity are dependent on the location in the space. For spherical wave front, usually the intensity is inversely proportional to the distance from source whole square. okay and for spherical it is inversely proportional to the distance from the source so both of these quantities SPL and SIL will then depend on where you are measuring so if i have for example some machinery is running next to me if i measure the level of the machinery here where i am standing it would give me certain decibels if I go further away the decibel levels will reduce and which is obviously apparent from your hearing as well as you keep moving away from some machinery source your hearing also so you feel that the level is reducing you feel that the sound is going down as you are moving away from the noise source any kind of machinery source so both these SPL and SIL they are dependent on the location But sound power level is actually a characteristic of the source that is producing it.

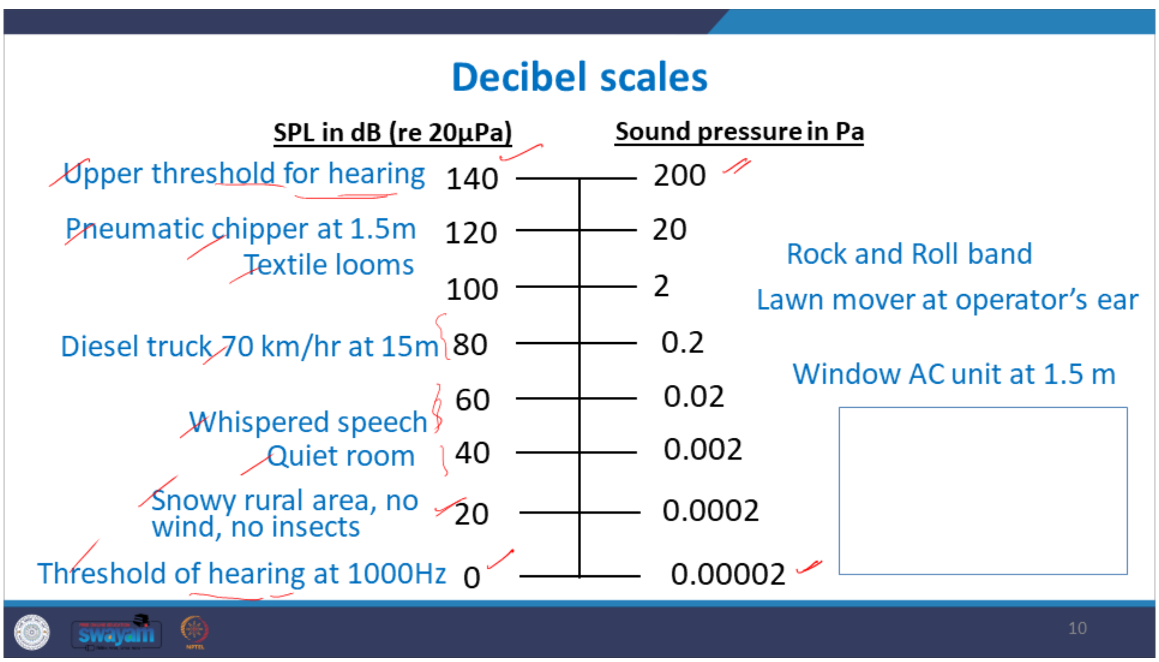
So, it is the amount of its as the terminology suggest it is the level of the sound power and sound power is independent of the measurement location as already discussed in previous lectures, it is a characteristic of the source. So, here it is defined like this

$$\mathbf{L_W = 10\log_{10}\left(\frac{W_{rms}}{W_{ref}}\right)}$$

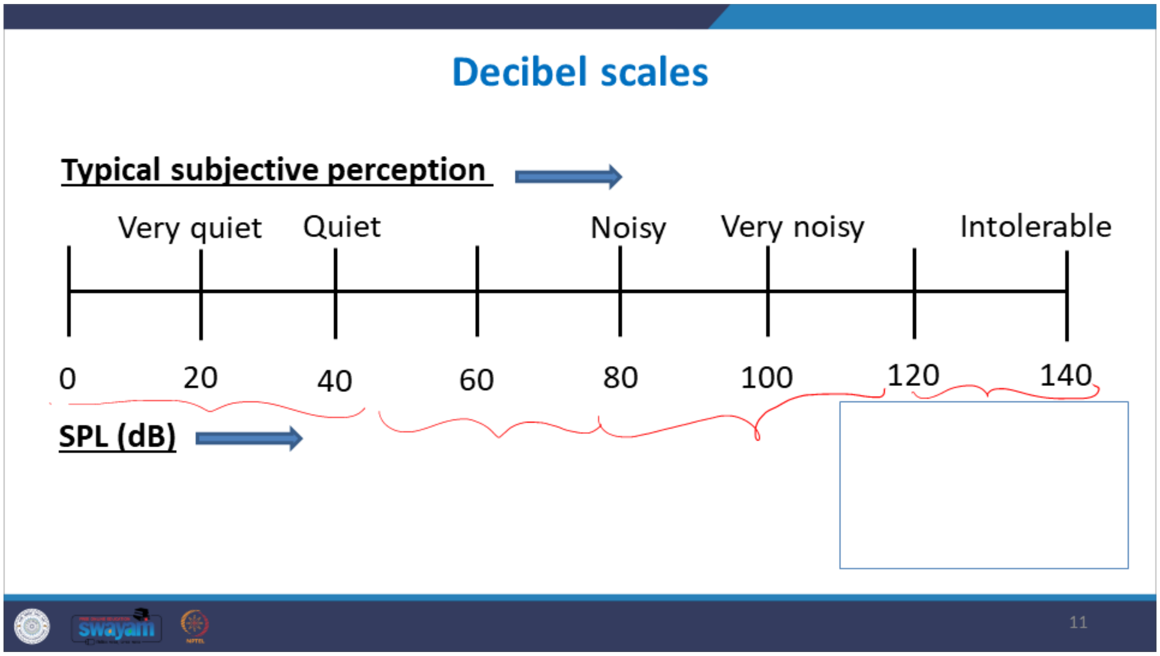
where this is the reference and all of these because they are logarithmic scales they are have the unit as dB.



This shows a typical comparison if you see here, this is the minimum pascals, this is the maximum pascals within which the ear can handle and if you convert this. So, you see a huge variation right from 10^{-5}, we have come to 10^{2}. So, we have covered a large range of magnitude and here when you convert it to decibel scales from 0 till 140. So, you get such easily manageable range of numbers and that is the beauty of using the decibel scales. This shows some of the typical decibel scales of machineries and sources which are found in the daily life. where this belongs to the threshold of hearing and this belongs to this is the threshold of hearing this is the upper threshold beyond which this ear damage permanent damage to the ear then things like pneumatic chipper Very heavy noise. Textile looms also producing very heavy noise. Then the diesel trucks somewhere. Diesel trucks and large vehicles stand here. The speech and all that stands here. Quiet room over here. And then extremely low snowy rural area. No wind, no insects, nothing. almost heary silence can be up to here. Ideally, it's very difficult to actually have 0 decibels anywhere. You need extremely controlled environment to get 0 decibels. It does not usually exist in the real life surroundings.



Typical subjective perception, if you can say, so till you know 0 to 40 it has been found that people perceive it to be quiet sounds and from here the moderate level starts and then from here onwards it starts getting noisy to very noisy and finally, it becomes intolerable over here ok.



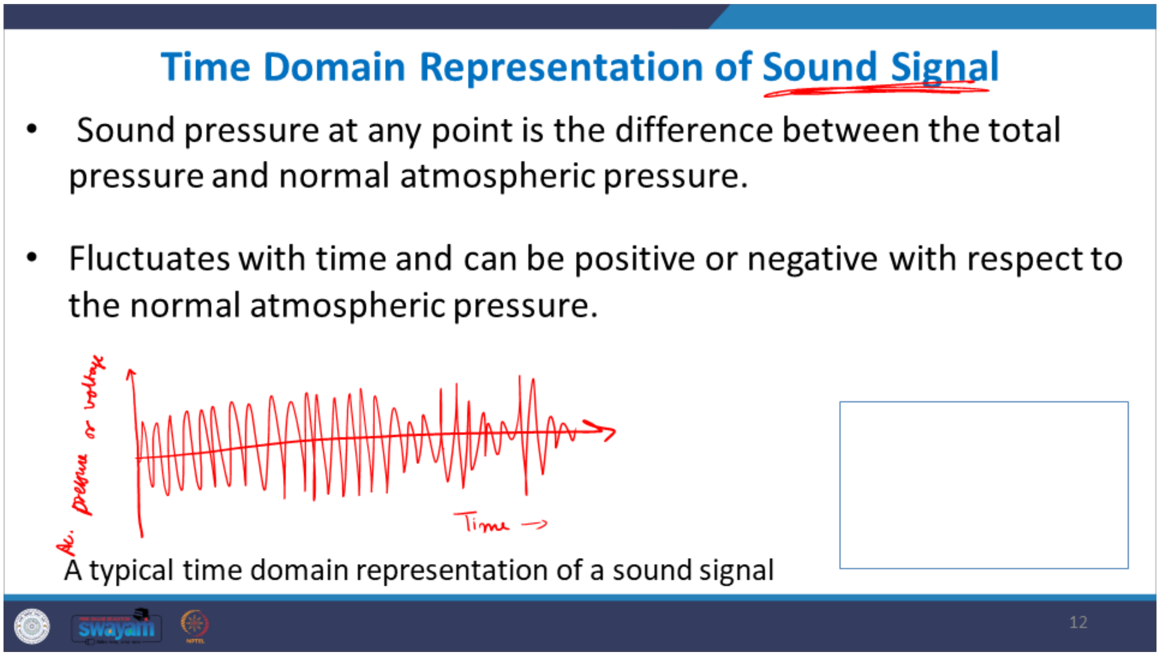
So, time domain representation. Fine, so till now we discussed about the sound pressure as a variation, sorry we discussed about the sound as a variation in the pressure over time and space, okay.

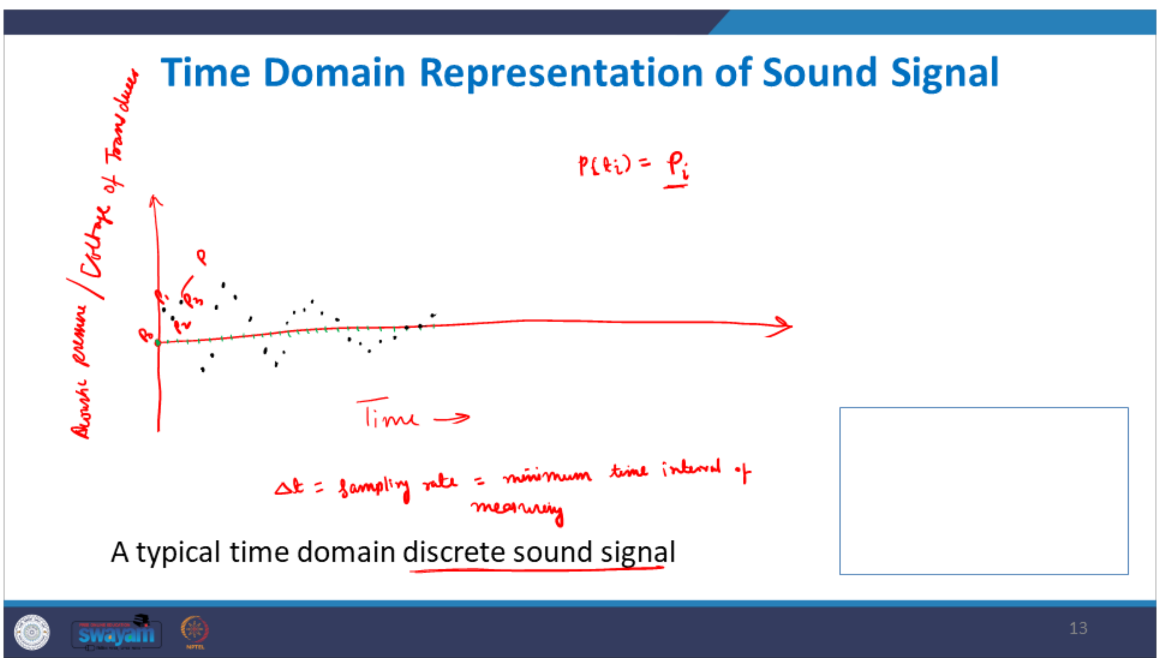
Now, if we want, what do you mean by sound signal? What do you mean by that? So, you measure the sound. You use a typical instrument like a microphone where the diaphragm of the microphone vibrates because of the longitudinal sound waves pushing against it. It vibrates and that vibration has some piezoelectric crystals behind it which convert this vibratory motion into corresponding electrical voltage which is directly proportional to the amplitude of the vibrations. okay So, this electrical voltage is then measured and they are represented. So, that is called as the sound signal which is the signal which you have obtained from some kind of instrument or a transducer which converts the vibrations of the particles into an interpretable fluctuation in the electrical voltage. So, this is how it can be. So, let us say this is your let's say I'm giving you a time domain representation this is your time scale and this is the pressure you can either say the pressure or you can say the voltage acoustic pressure or the volt corresponding voltage of the transducer so it would be like a fluctuating signal over time something like this depending on what the source is like. So, this is a typical sound signal. okay but you know that you know essentially the sound is continuous it is not like at every point in time you have some sound pressure available but when we measure it from any instrument because at the end of the day as an engineer we measure the sound levels so we use certain instrument like a microphone or any form of transducer we measure it and convert it into these voltage signals and then we do various analysis on these measured signals.

So, the measurement there is always a limitation you cannot measure at fractions you cannot measure at literally every point in time ok every measurement is done at certain time intervals even though it may be very small but it is discrete in nature so analog data most of the available instrumentation is actually collecting the signals as a digital data so the same as a discrete sound signal would be like this you have time and you have the acoustic pressure or the corresponding voltage of the transducer that has measured this and a certain fluctuations. So, let us say Δt is the sampling rate which is the minimum time interval of measuring ok.

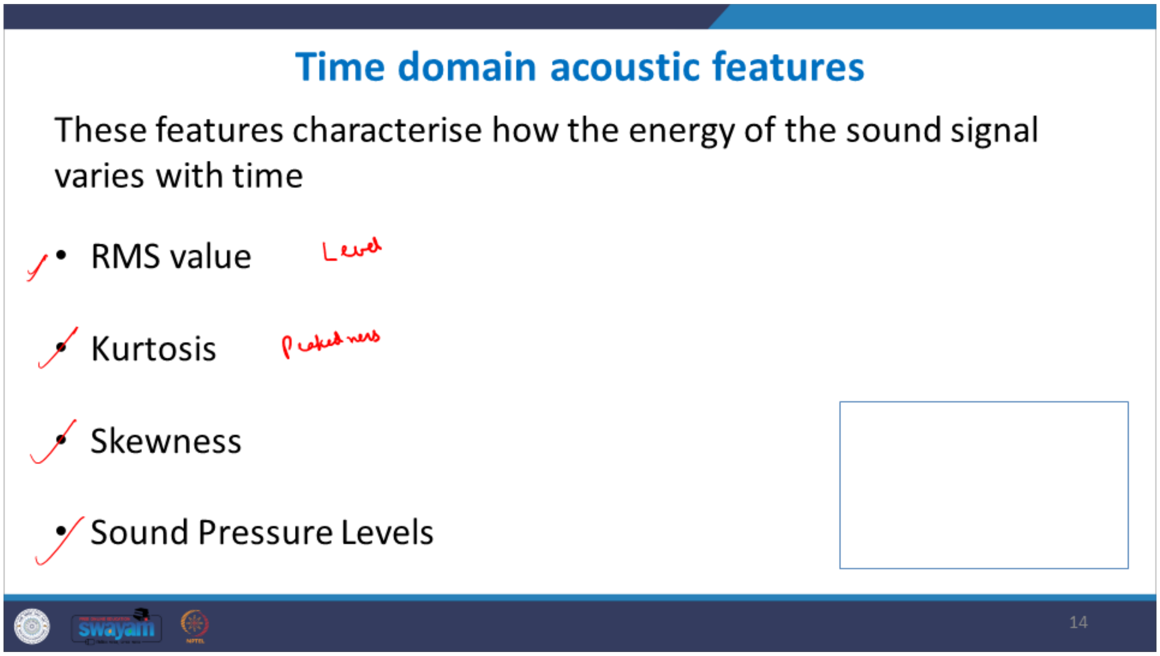
So, here what is happening that after every Δt we are making one measurement. So, let us say we started from some instance t equals to 0, then at certain other instance. So, I will indicate all the measurement instances using this green. So, let us say this is one instance then we took a measurement here, here, here, here. So, this is how these are the various time instances at which our instrument measured the sound signal right.

So, the same thing over here would be like this may be some pressure we got here, some here, here, here, here, here. So I am giving a very elongated view. So at every you know instant of measurement we are getting a certain sound pressure level okay. So from this so this is the data we are getting discrete points where each point represents a particular this is you know a collection of various it is like a collection of various pressure at various time instance so pressure at some time instance ti in short we can write it as pi so these are a collection of various pressure points p0, p1, p2, p3 and so on. So, the pressure at the various time instant. So, these are a collection of various p_i's where i varies from 0 till whatever you know number of points or the number of samples you are collecting from the instrument. So, this becomes like a typical time domain representation where we have discrete time, we have got discrete points over different time intervals. Now, let us see if this kind of discrete signal we are obtaining. So, the same waveform now is appearing as a collection of points each taken after a Δt interval right. So, these collection of points or p_i's the various acoustic pressures we can do various form of analysis because this is essentially if you treat this as a collection of points or a distribution of discrete points then various analysis can be done.





Like you can find out the rms value of whatever points you have measured you can find out the kurtosis skewness and sound pressure level for this discrete sound pressure data

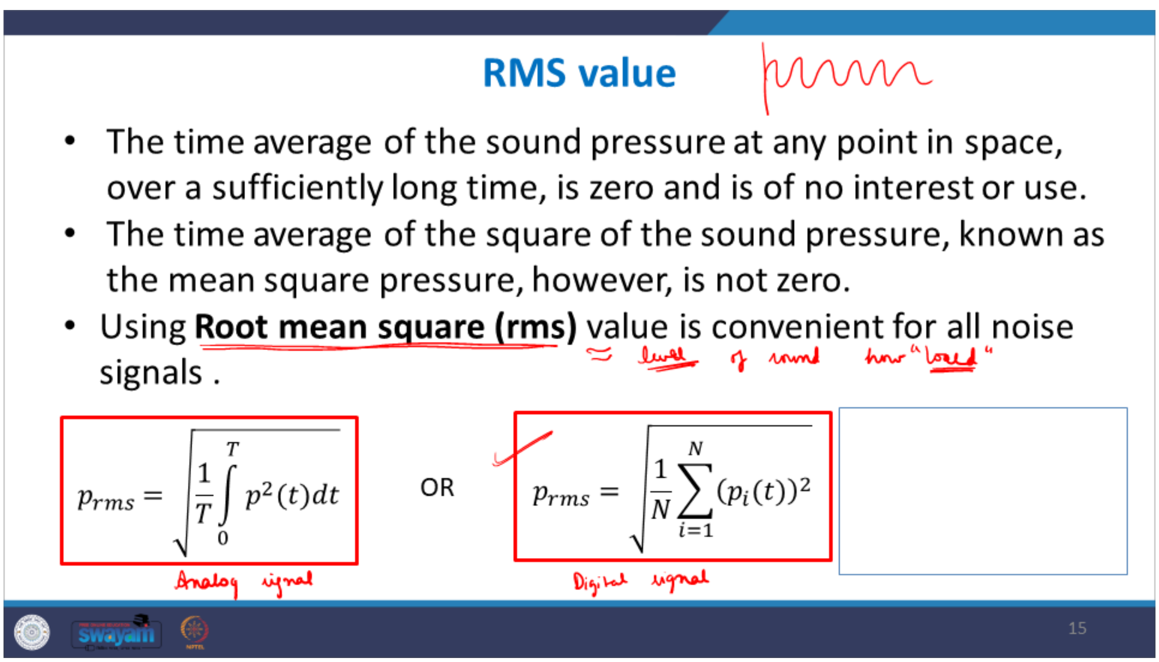


So let us see how you define the rms value Now, for any waveform you know that sound signal is essentially a collection of various sinusoidal waveforms. okay So, various it is a collection of various harmonic waves. So, all of them have a waveform like this. So, over a long time interval their average would automatically become close to 0 or fully 0. So the average of sound pressure over a long time interval for any signal or any kind of source would automatically come out to be 0. So the average does not make sense because it is essentially a fluctuation that we are measuring. So its average would be nearly 0. So, that is of no interest. So, we actually measure it using root mean square to find out what is the level. So, root mean square. So, how do you do it? Just the same way as you use root mean square in statistics. So, in statistics you have a collection of points and you are finding out the RMS in the same way. Here also these pressure are what the you are having a collection of pressure as points. or you have a collection of pressure as discrete values and you find out the root mean square of those values.

So, essentially if it was a continuous data for analog signal it would be 1 by t where t is your measurement duration which is a long time and then integrated from 0 to t p square dt, but this is the more familiar kind of equation for the discrete signal or the digital signal where what you have is root mean square. So, first you do the square and find out the mean of the sum of square and then you root over. So, the summation of the squares of these pressure values and then dividing by n to get the mean and then root over.

$$\mathbf{p_{rms}} = \sqrt{\frac{\mathbf{1}}{\mathbf{T}}\int_{\mathbf{0}}^{\mathbf{T}} \mathbf{p^2(t)dt}}$$

$$\boldsymbol{p_{rms}} = \sqrt{\frac{\mathbf{1}}{\boldsymbol{N}}\sum_{\boldsymbol{i=1}}^{\boldsymbol{N}} (\boldsymbol{p_i(t)})^2}$$

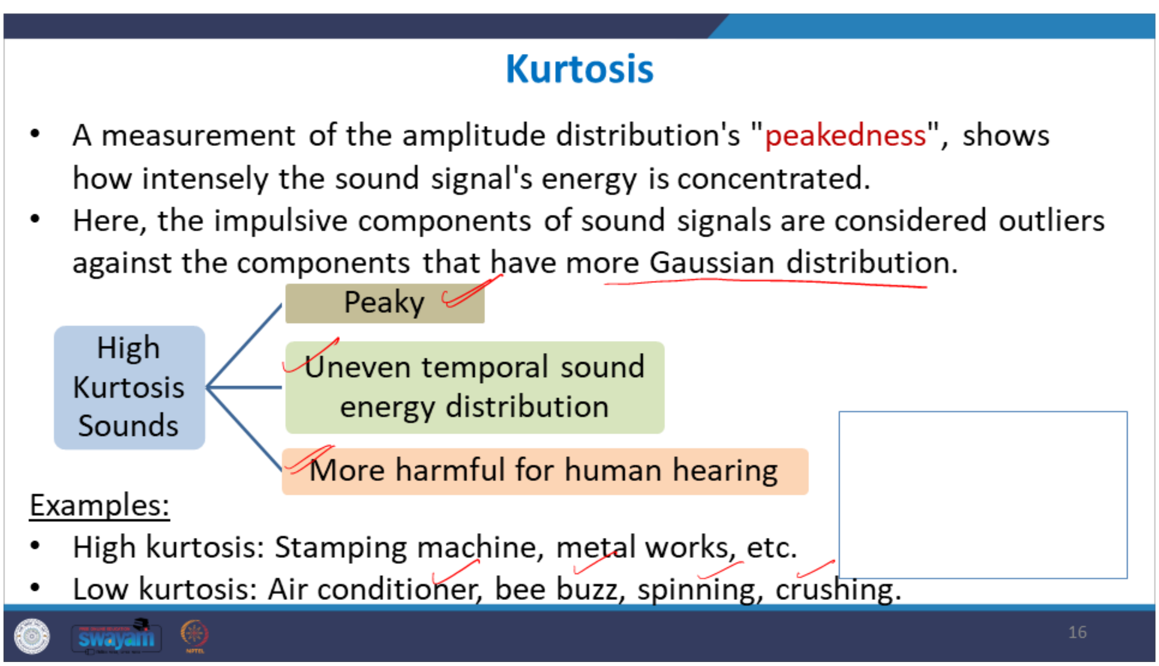


Then next kind of acoustic feature in time domain is kurtosis.

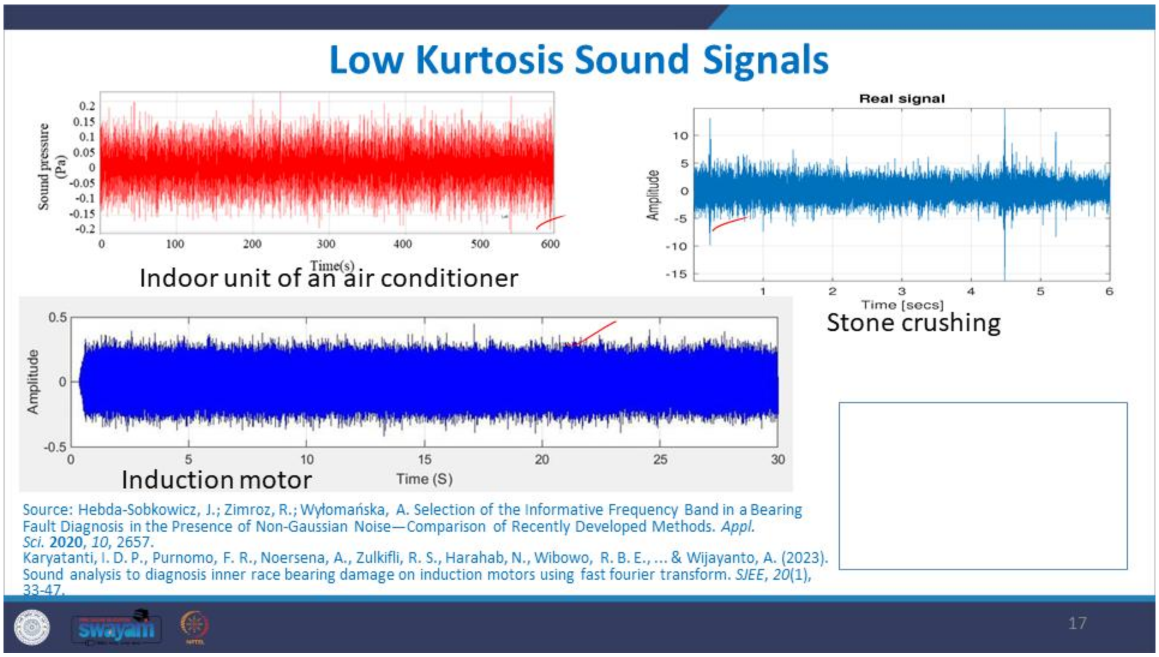
So, RMS typically represents you know the level it corresponds directly to the level of the sound or how loud the it is. So, it corresponds to the level of the sound or how loud or strong that sound is. And you can see here as well in the sound pressure level, we had this equation where it is calculated, it is proportional to the P_{rms}. And so, this corresponds to the level, this corresponds to the peakedness.

What does that mean? So, it says that you know in that sound signal how is the data how is the pressure data distributed how peaky it is which means that how you know peakedness means that is the distribution smooth or are there certain short events or transient events we have a certain high intensity or a certain low intensity event so how far away it is from the gaussian distribution that gives you the kurtosis of a sound signal So, the high kurtosis sounds would appear more peaky in their kind of signals that you obtain. It would have more uneven temporal sound energy distribution and it would and they are also more harmful for human hearing.

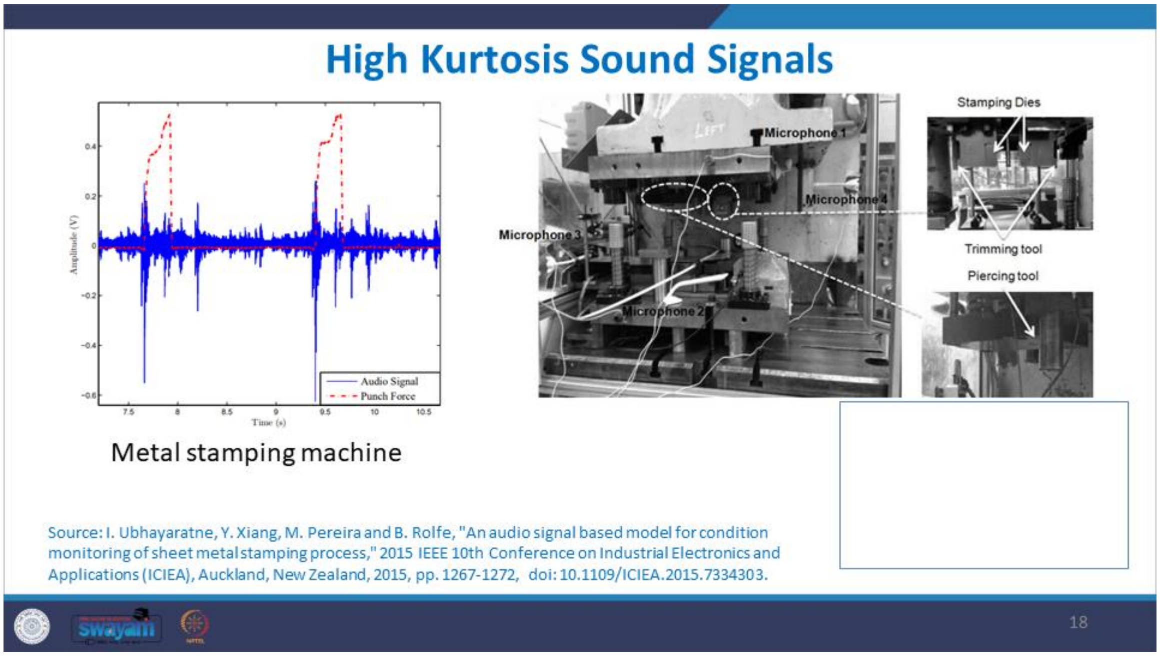
So, some of the examples of high kurtosis signals are stamping machine, metal works, etc. Especially in a signals where you know you have I mean these are more irritating to human ear. more of these impulsive sounds are there sudden high intensity bursts and all that whereas low kurtosis sound could be you know some of the machinery which is continuously running in a steady state so any steady state machinery running continuously like air conditioner the buzzing of the bee is Very much like a Gaussian distribution spinning and crushing.



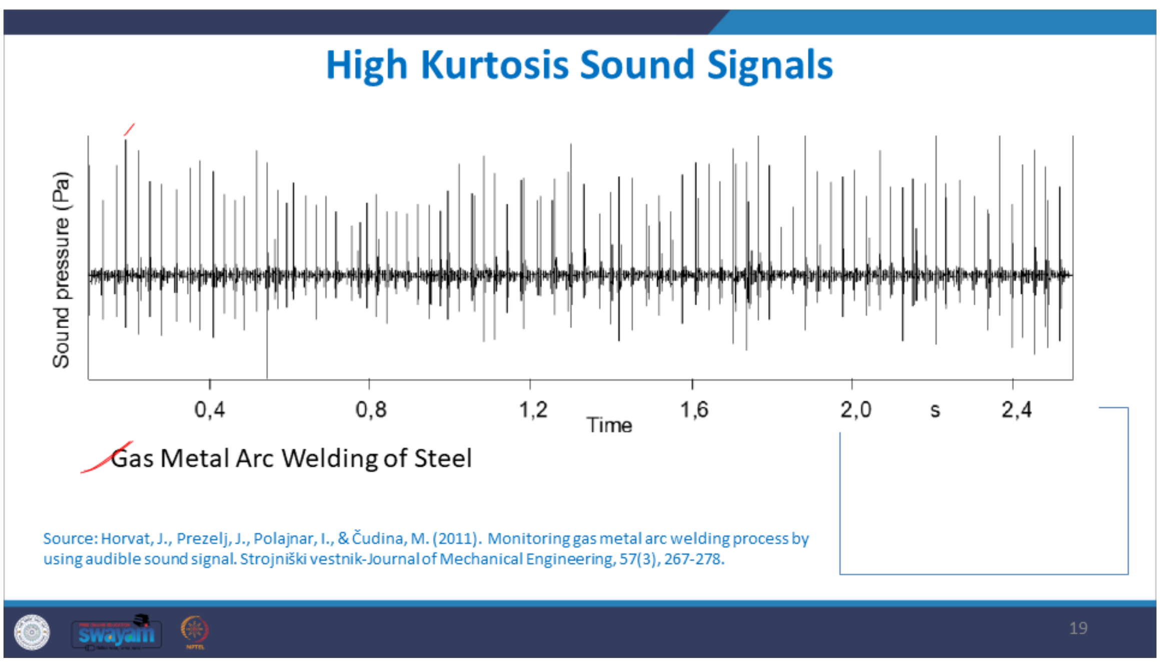
So if you see here the waveforms of these low kurtosis sound signal this is how it looks like looks more or less uniform distribution



whereas if you look at the high kurtosis sound signals. This is of a stamping machine where you know the microphone is placed quite next to the stamping tool. You see that if compared to these kind of signals, here you see suddenly low sound and then sharp peaks in between. So very peaky in nature. So that is a high kurtosis sound signal which is more dangerous for human hearing.



Again, another example is the sound pressure when this gas metal arc welding of the steel is going on. Again, you see very very peaky waveform, you know these sharp narrow peaks unevenly distributed and in between this very low intensity event and sudden sharp intensity event. So, you can see the peakiness in the signal.



Now, how do you, so this is the sort of description of what do you mean by kurtosis but how do you calculate it because we engineer are interested in values we want to get one number so that we can do all the manipulation we can make comparisons etc so how to calculate it We calculate it as the fourth central moment of the distribution here.

$$\mathbf{K_x = \frac{M_4}{\sigma_x^4}}$$

The distribution is the distribution of pressure because pressure, we have various pressure values. That is our distribution. And this is the standard deviation and mean. So, essentially, this is the kurtosis of a pressure distribution.

$$\mathbf{K_p = \frac{\sum_{i=1}^{N}\left(p_i - \mu_p\right)^4}{N\sigma_p^4}}$$

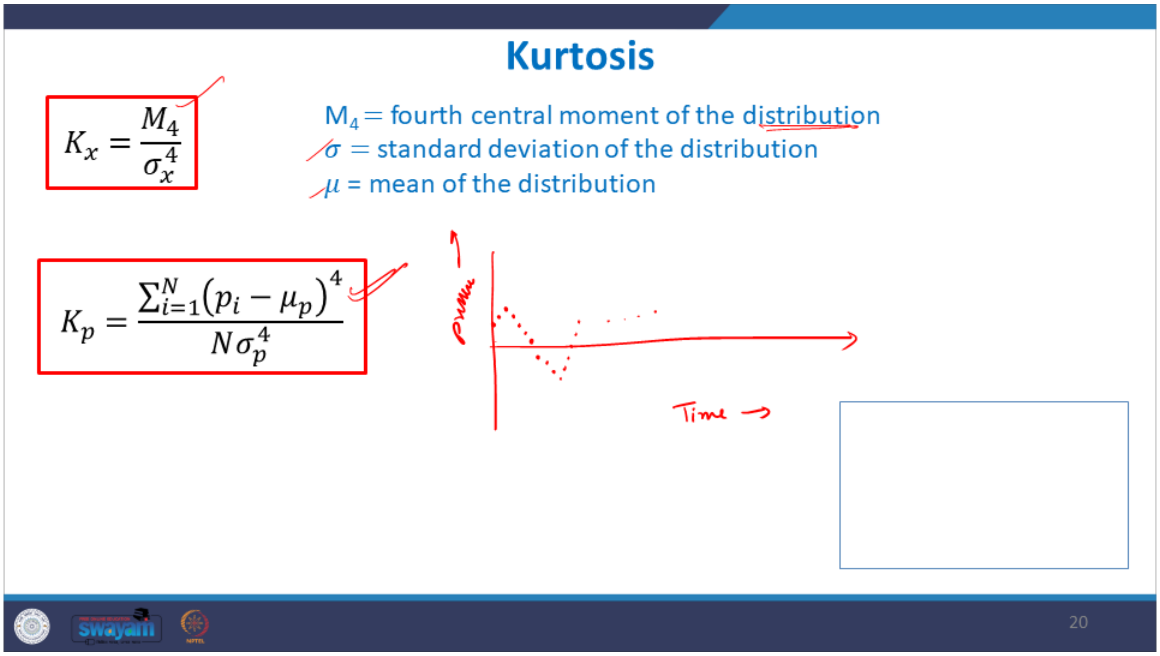
So, again, what is these pi's?

You have this time signal. You have the pressure. and because it is collected through some instrument we have used certain instrument to measure it. So, we get it as a collection of points at the intervals of Δt whatever those points are. So, and so on.

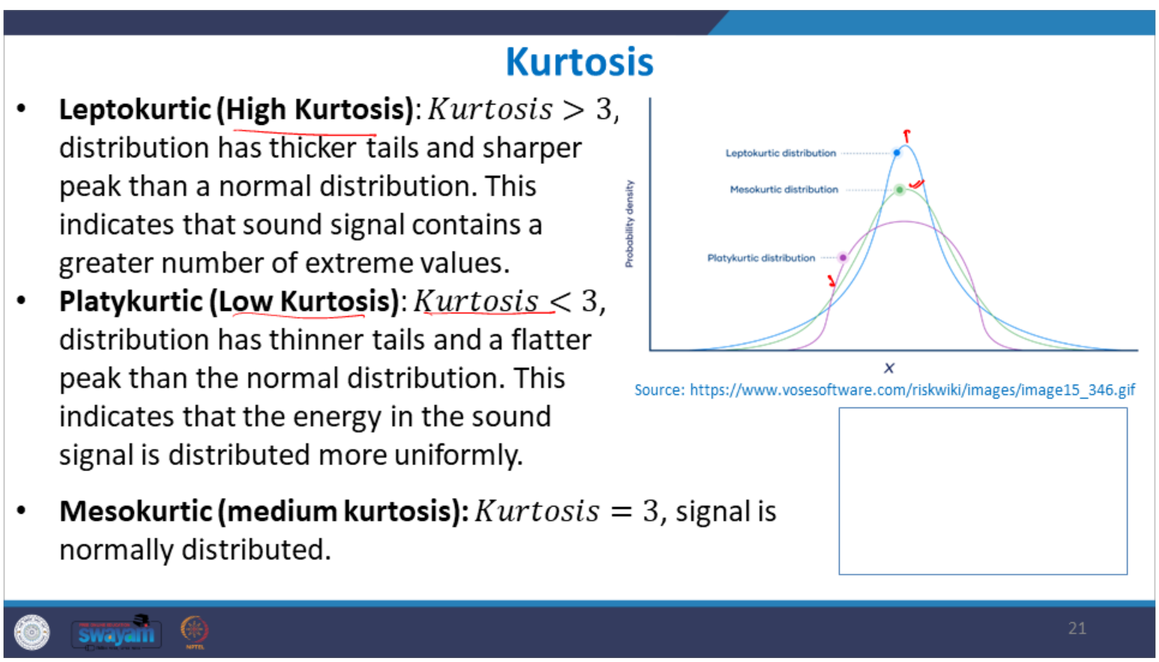
So, it is a collection of these p points. So, these and so you will find out the mean and then it is

$$\left(\boldsymbol{p_i - \mu_p}\right)^{\mathbf{4}} / \mathbf{n}$$

which is the number of points you are taking and σ_p being the standard deviation of that distribution so what does it mean



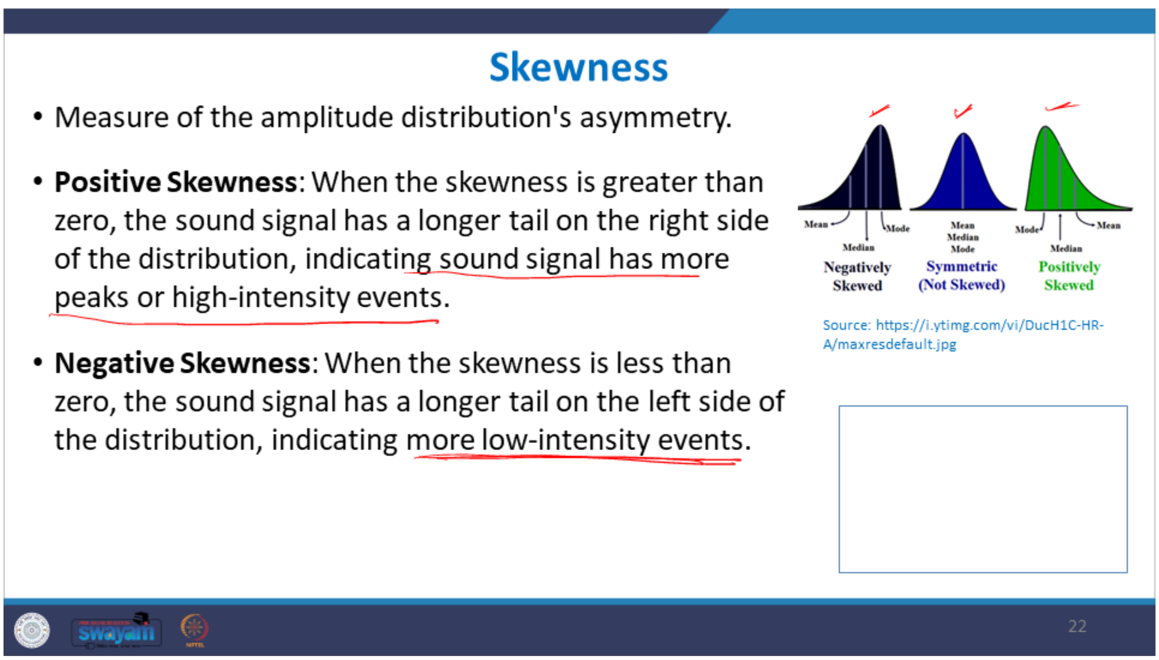
So when you find out the kurtosis using this equation you will see that the signals that have kurtosis more than 3, they appear they are sort of called as high kurtosis signals, they are more they have more thicker tails. So, they have suppose this green is the standard distribution, then these signals are much more sharper, this is lesser. And if the kurtosis value is less than 3, that means that it is a low kurtosis signal, more even distribution of energy, more uniform distribution and at exact 3, you have a normally distributed pressure.



The other acoustic feature is the skewness. So skewness again just like in statistics skewness means the same thing. So you are deriving all these from the these are all statistical features which we are using to describe a collection of pressure points. okay

So positive skewness means that the sound this is a normal distribution. This is the negatively skewed and positively skewed data. So, what it means in the case of sound signal is that that same sound signal has got more peaks or more high intensity events. Whereas, if it is a negative skewed data which means that it has more of a low intensity event. So, whatever is the mean pressure, so here mean is the rms.

So, whatever is the overall rms pressure. Most of the events are high events compared to the rms pressure whereas in the positive skewed data whereas in the negative one more of the events are low intensity events compared to the mean data.

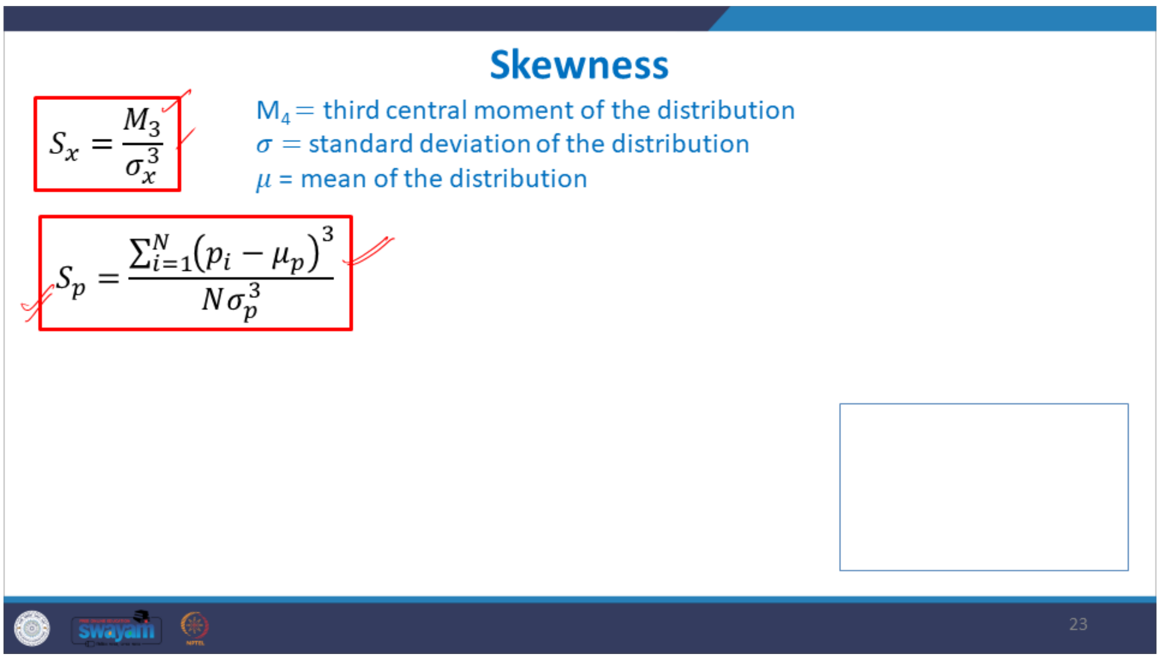


This is how you calculate the skewness which is the third central moment by σ_x whole cube.

$$S_x = \frac{M_3}{\sigma_x^3}$$

So, in terms of the pressure distribution it is this particular formulation you subtract every pressure value. from the mean of the distribution whole cube and sum it up divided by n which is the number of points you have taken and sigma p being the standard deviation of that pressure distribution ok.

$$\mathbf{S_p} = \frac{\sum_{i=1}^{N}(\mathbf{p_i} - \boldsymbol{\mu}_\mathbf{p})^3}{\mathbf{N}\boldsymbol{\sigma}_\mathbf{p}^3}$$



There is another acoustic feature which is usually you know very much used. So, already I have described what you mean by sound pressure level. So, over there you see that it is

$$\mathbf{L_p} = \mathbf{20}\,\mathbf{log_{10}}\left|\frac{\mathbf{p_{rms}}}{\mathbf{p_{ref}}}\right|$$

Then there is one way variation to that is equivalent continuous sound pressure level or the total equivalent sound pressure level. This is the most important use measure for environmental monitoring.

$$\mathbf{L_{eq} = 10log_{10}\left[\frac{1}{T}\int_0^T \left(\frac{p(t)}{p_{ref}}\right)^2 dt\right]}$$

$$\boldsymbol{L_{eq} = 10log_{10}\left[\frac{1}{T}\int_0^T 10^{L(t)/10} dt\right]}$$

So what it means is that suppose I go as a noise control engineer to a factory setting. Now a factory setting or a factory workshop is a very dynamic environment. Sometimes there are some workers talking, sometimes there is some machinery going and then there is a break time. So it is very dynamically changing. So, okay and suppose I go there I monitor that environment for 24 hours.

So, what should be the sound pressure level? So, in that case you know SPL which is the P_{rms}. So, in that case we are not if it is not practical to measure the sound levels for the entire 24 hours' duration. Suppose we want to if it is a very dynamic environment with a lot of entities involved So, if you want to monitor its noise levels, it is not practical to you know measure it for 24 hours, get such a huge amount of data and then find out its rms and then finally, get a one unique SPL using that formula that is

$$\mathbf{L_p = 20\,log_{10}\left|\frac{p_{rms}}{p_{ref}}\right|}$$

So, it is not practical for monitoring environments where you have to monitor it for a continued period of time or for a much larger period of time. So, what do the noise engineers do?

Equivalent Sound Pressure Level

- **Total equivalent sound pressure level**: or **Equivalent Continuous Sound Pressure Level** is the average energy level in decibels of the sound over a period of time.
- Most common measure for **environmental monitoring**.

$$L_{eq} = 10log_{10}\left[\frac{1}{T}\int_0^T \left(\frac{p(t)}{p_{ref}}\right)^2 dt\right]$$

L(t)=sound pressure level at time t

T= a long period of time

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$$L_{eq} = 10log_{10}\left[\frac{1}{T}\int_0^T 10^{L(t)/10} dt\right]$$

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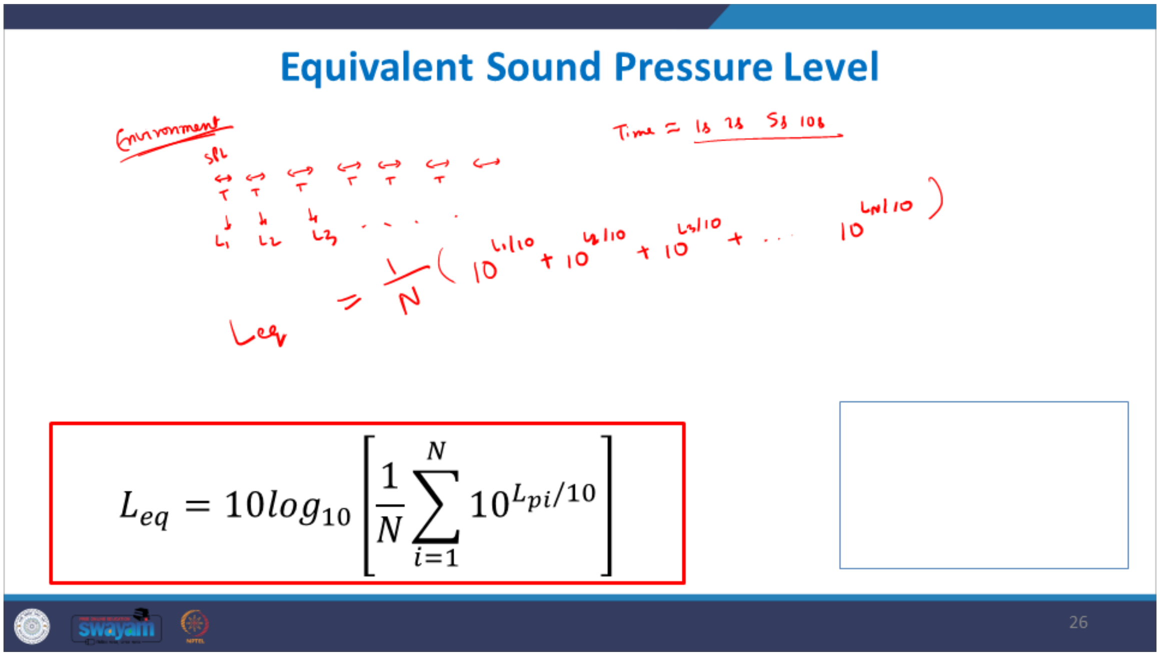
Let me get to the discrete data. What do they do is that they go there they let us say they measure it at fixed intervals in time which means that Suppose, I go to certain environment, it could be a factory environment, some park, some theater, anything and at different points in time, some time period you can say, some fixed time durations, I am measuring the time signals and at each time I am getting a certain SPL value. Let us say at the first measurement I got L_1 and the second time L_2, L_3 and so on. So, because I am going there at different points in a day and for a fixed duration. So, this duration has to be fixed for each measurement. and at those fixed duration I measured the overall sound pressure level because I can't obviously I don't need huge amount of data.

So, I measure it for like 1 second or 2 second that's enough to give me a lot of data of what's happening at that particular time. So, I just take a measurement of usual the usually you know the measurement is usually done some of the usual values are 1 seconds, 2 seconds, 5 seconds, 10 seconds. like that not more than a minute so these measurements are done and SPL at the different point in time are calculated so we get these different levels all measured for a fixed duration of time and then we use this formula where what we do is we sum up them so we do a logarithmic summation Let us say we got n such measurement points. We did the logarithmic summation of these points. Then we took a mean out of it to get one overall sound pressure level which we call as the equivalent continuous sound pressure level.

$$\mathbf{L_{eq} = 10\log_{10}\left[\frac{1}{N}\sum_{i=1}^{N} 10^{L_{pi}/10}\right]}$$

So, it is used for environmental monitoring for very dynamic environment. So, suppose for a factory I say that in a 24 hours of monitoring of the noise of a factory let us say I have gotten the equivalent level as 80 decibels. That does not mean that it was 80 decibels throughout, right?

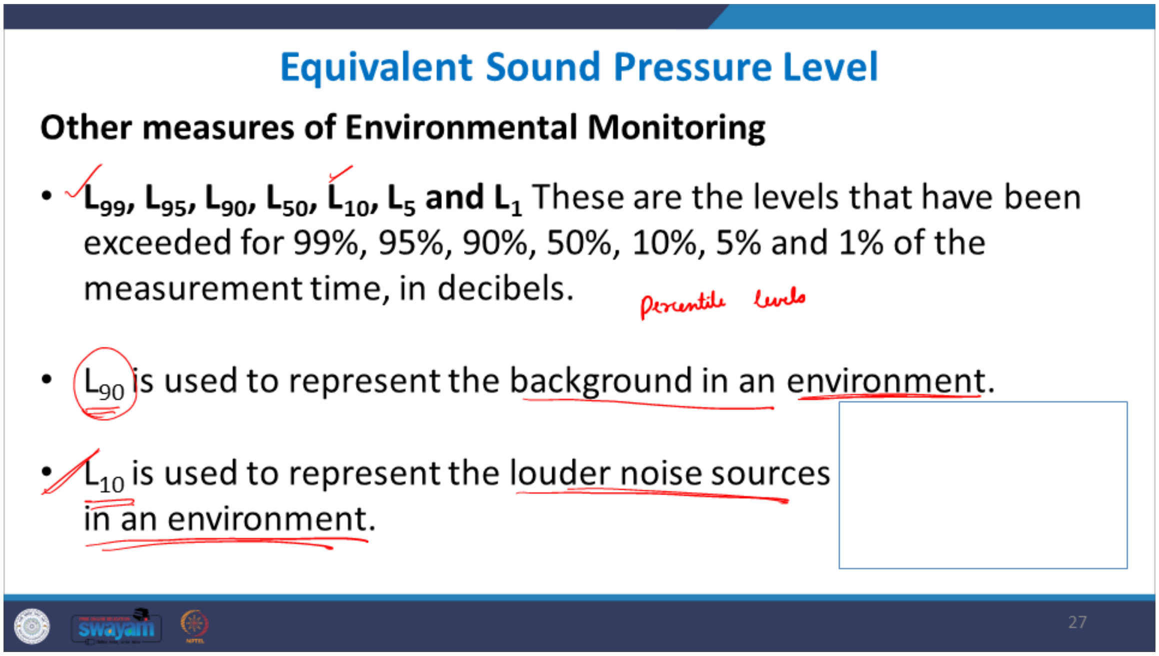
It could be that at some point in time when there was a peak workload, the levels could be as high as 95 or 100, whereas at certain points when it is break time or it is almost shut down, it could be up to 50. but overall 24 hours monitoring I am getting one unique kind of SPL for that environment.



Other measures, so in the same way like we have the $L_{equivalent}$, we have got various other you know percentile levels of sound pressure level ok. These are called as percentile levels. So, as the name suggests L_{99} means it is what

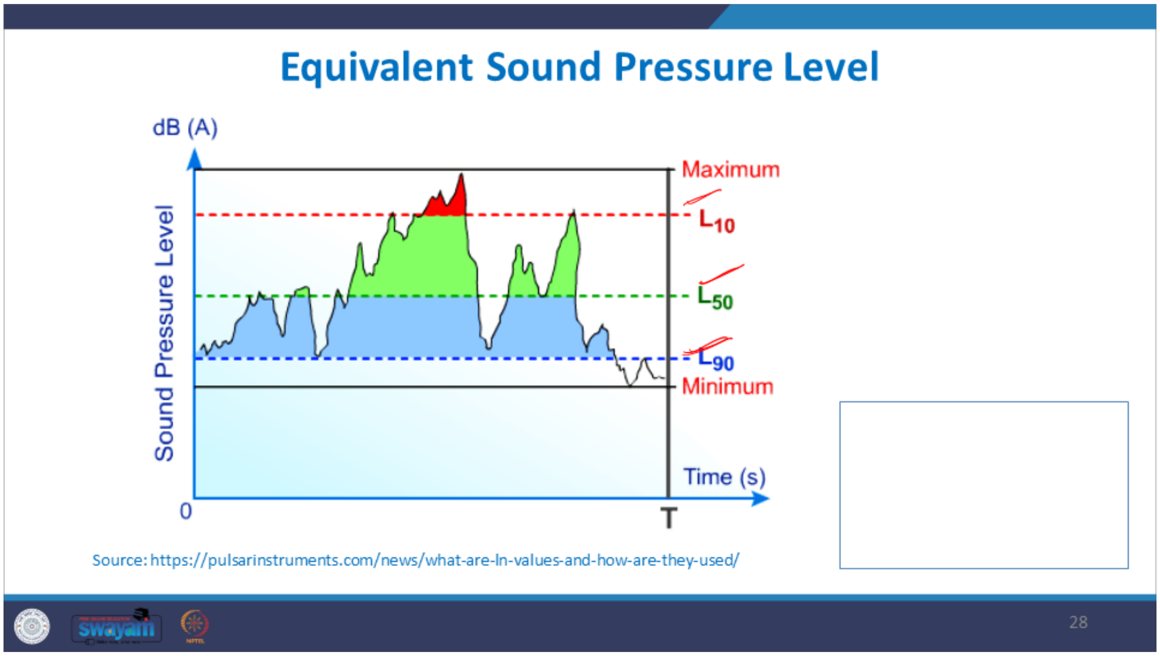
you know the level it means that you know 99% so you suppose measured some sound waveform suppose you measured in certain environment again this is also used for environmental monitoring so in certain environment you did your measurement for a long some significant amount of time so the level beyond which so 99% of the data exists which means that 99% of the time the noise pressure then the sound pressure was above that level that becomes your L_{99}. Similarly, L_{95} is that 95 percent of the time during which you made the measurement the level was beyond that level the sound pressure was beyond that level and so on L_{10} which means 10 percent of all the measurement time 10 percent of the time the level was beyond this level. So, basically you know L_{90} would mean what? That almost 90 percent of the time the level was above this particular level.

So, suppose I have gone to some environment which is very dynamic, some park I have gone to and sometimes the vehicles are going, people are passing by, children are crying, so many things are happening and then in the middle we have certain small quiet periods. So, 90 percent of the time if the waveform is above it, if the pressure is above it which means that this corresponds to the lowest. So, it is typically taken to represent the background of that environment. So, this is the background level and when almost no source is present or typical no nearby source is present. Then this is the background level of that environment and similarly L_{10} which means only 10% of the time the pressure was higher than this level means it represents more louder noise sources or dominant noise sources of that environment.



This shows a typical sound pressure level distribution. Here L_{90} is like corresponding to the background. This was the overall background and then in the middle whenever there were nearby sources then this increased. Otherwise there were no nearby sources that could be seen then this was the level that was maintained. This L_{90} over here.

Then these are some of the common measures L_{50} and L_{10} which correspond to more acoustically intensive events.



Okay, so with this I would like to close this lecture which was the first lecture on time domain representation of sound signal. So, thank you for listening.