NOISE CONTROL IN MECHANICAL SYSTEMS

Prof. Sneha Singh

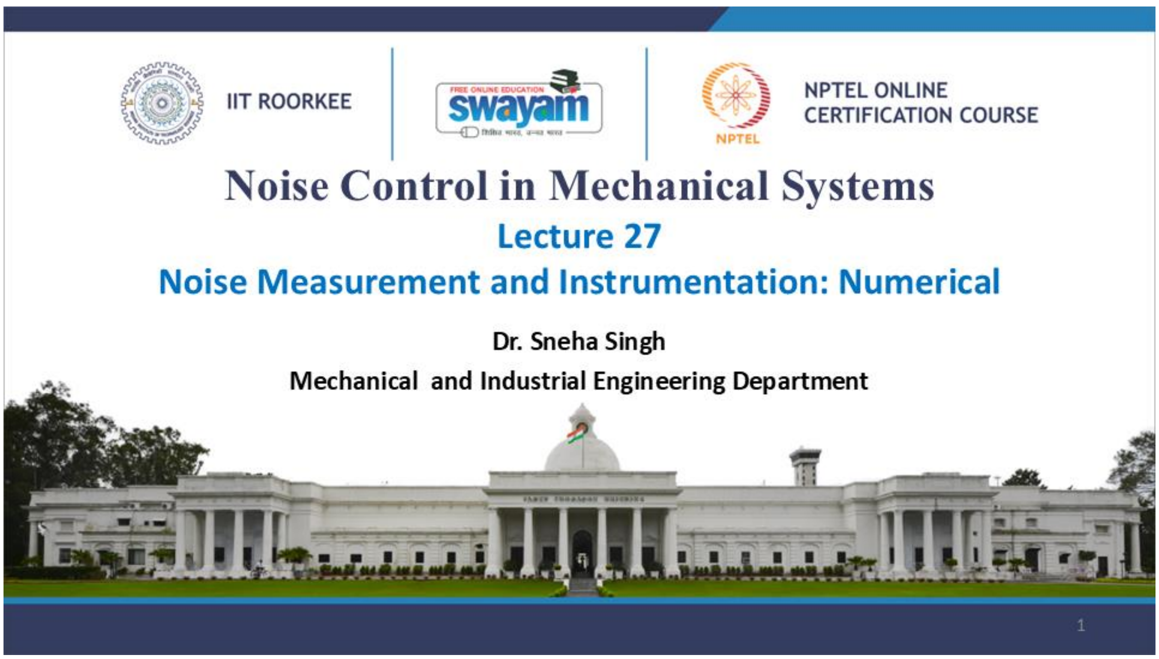
Department of Mechanical and Industrial Engineering

IIT Roorkee

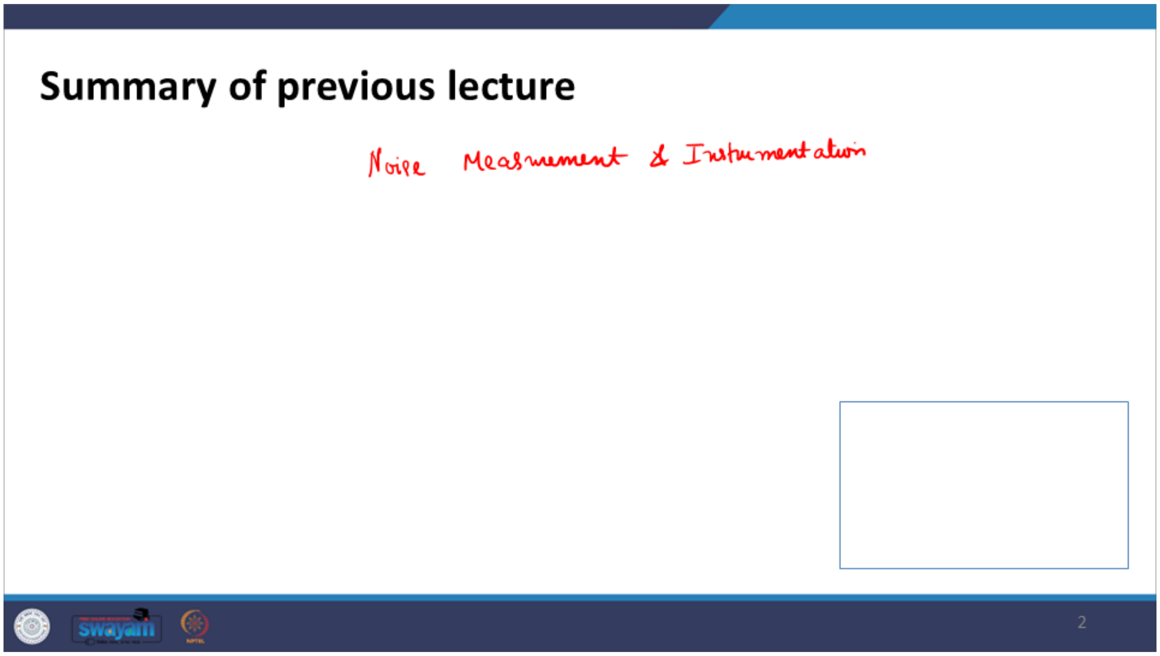
Week:6

Lecture:27

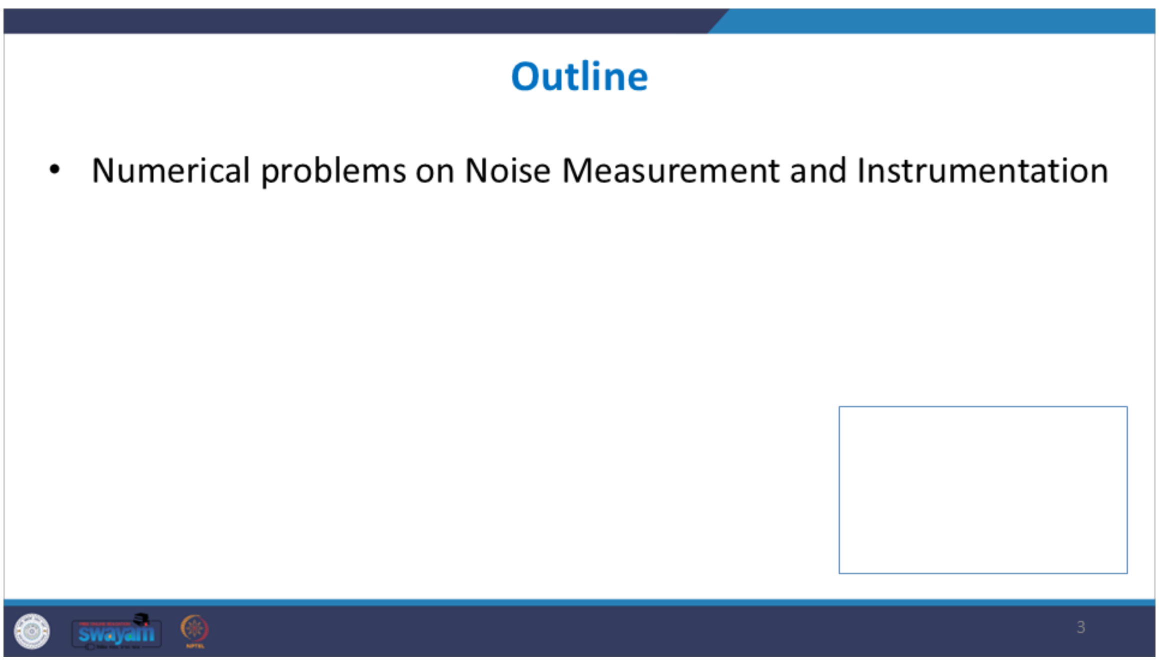
Lecture 27: Noise measurement and instrumentation: Numerical



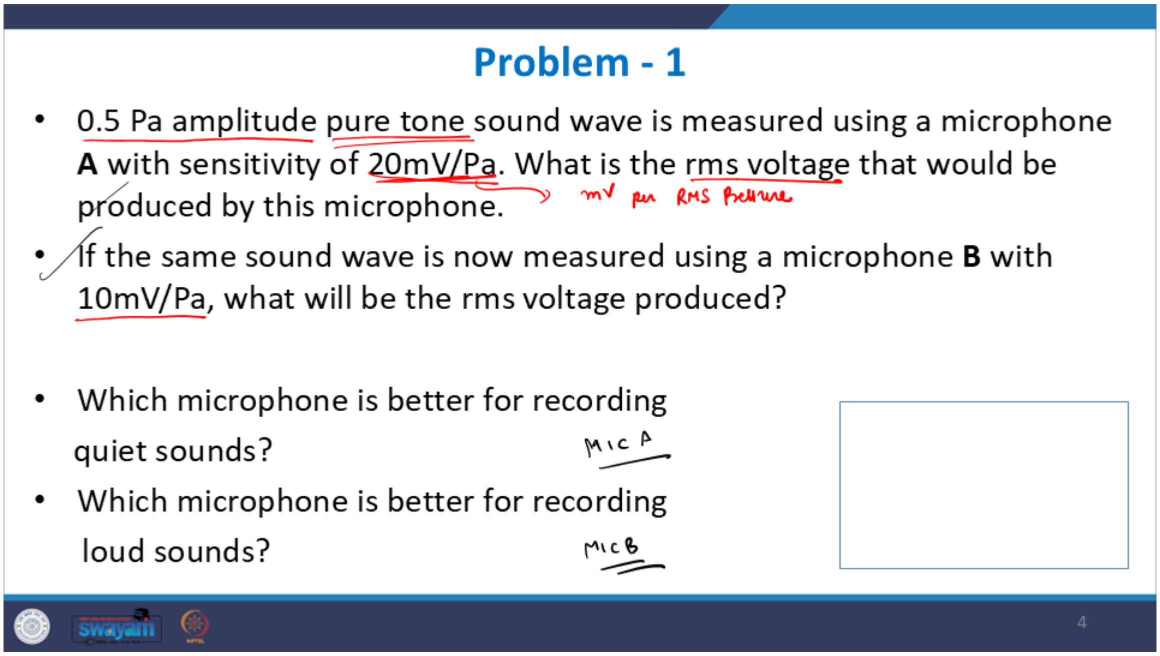
Hello and welcome to the course on noise control in mechanical systems. I am Professor Sneha Singh from the Department of Mechanical and Industrial Engineering at IIT Roorkee. So far, we have been doing the module on noise measurement and instrumentation, and we have covered various aspects related to that, such as the typical acoustical devices that are used, the characteristics of these devices, the sampling rate, dynamic range, as well as the frequency range and various other characteristics of these acoustical devices. At the same time, we have also seen the measurement standards and the guidelines for typical acoustical measurements in the field of noise control engineering. In this lecture, based on the understanding of the previous three lectures in this particular



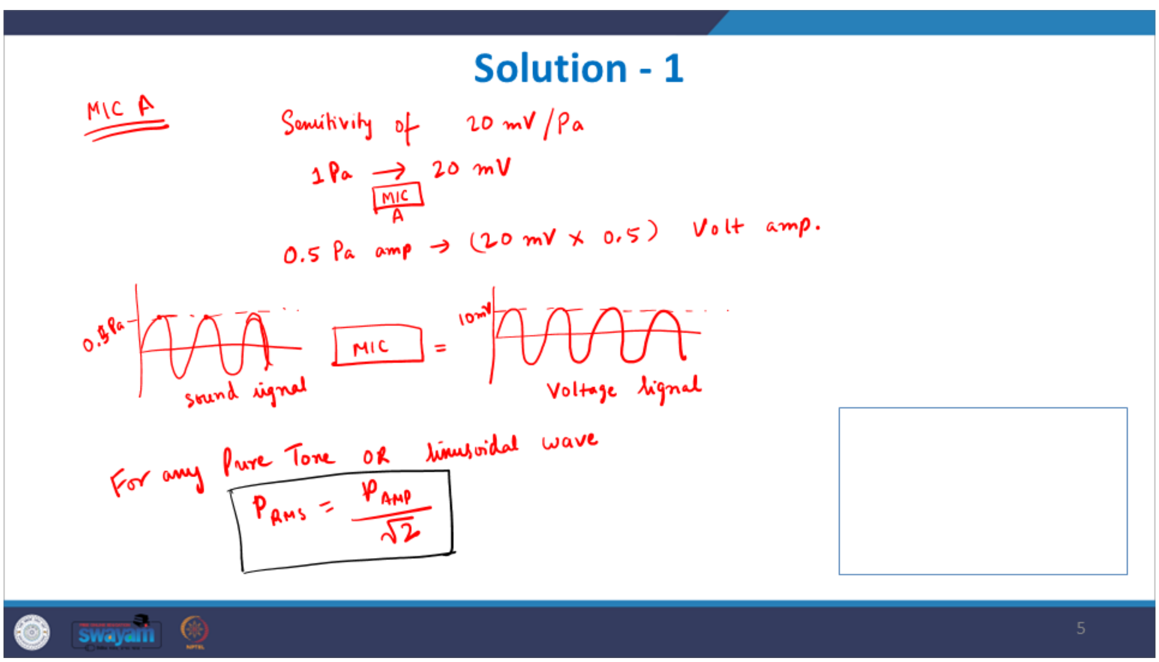
module, we will see some numerical problems on noise measurement and instrumentation.



let us see the first problem. Here it is given that a 0.5 Pascal amplitude pure tone sound wave is measured using a microphone A with a sensitivity of 20 microvolts per Pascal. What is the RMS voltage that would be produced by this microphone? let us first solve



the first part of this problem. So over here, it is given that the microphone here 0.5 Pascals amplitude is being passed to it, and the sensitivity is 20 microvolts per Pascal. for the mic A, what it essentially means is that all these sensitivities are measured for microvolts per RMS pressure that is passed to it.



Here let us see it has a sensitivity of 20 microvolts per Pascal. which means that if 1 Pascal of sound pressure is passed, be it an RMS pressure and amplitude, just the value of 1 Pascal of acoustic pressure, it will create 20 microvolts of voltage when it is passed through this mic A. 1 Pascal input, 20 microvolts is the output. In that case, if you think about it, 0.5 Pascals amplitude should give 20 microvolts multiplied by 0.5 as the voltage amplitude.

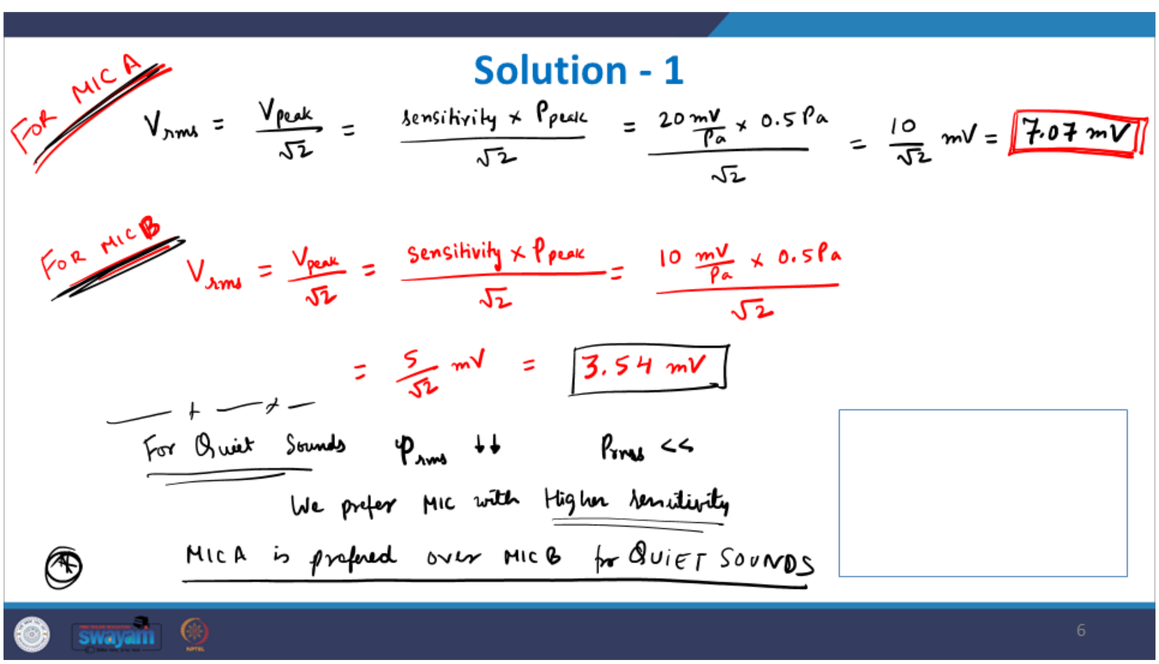
$$0.5\ Pa\ amp\ \rightarrow (20\ mV\ \times 0.5)\ volt\ amp.$$

This should be the amplitude-to-amplitude conversion. Let us say, for example, here we had some kind of acoustic pressure, which had, 0.5 pascals is the amplitude of this acoustic pressure, and once it passes through the mic, you are getting the voltage. And here, each of these 0.5 pascal values, when they are converted, they are converted into 20 into 0.5 microvolts, which is 10 microvolts. This is the sound signal, and this is the corresponding voltage signal generated by this microphone. The 0.5 microvolts, if you multiply it with the sensitivity, what you get is 10 microvolts. So here, the question asks, what should be the RMS voltage? for a sinusoidal, it shows that it's a pure tone sine wave.

For a sinusoidal wave, for any pure tone or sinusoidal wave, whatever the RMS value is, it is obtained by dividing the amplitude or the peak value by root 2.

$$P_{rms} = \frac{P_{amp}}{\sqrt{2}}$$

This is how you obtain the RMS value for this sinusoidal wave. So let us use this and find out.



What you get ultimately is that the RMS voltage that is obtained would be the peak voltage by root 2,

$$V_{rms} = \frac{V_{peak}}{\sqrt{2}} = \frac{sensitivity \times P_{peak}}{\sqrt{2}} = \frac{10}{\sqrt{2}} mV = 7.07\, mV$$

This is the answer to our very first part. This is for mic A.

Now, let us see what happens if the same sound wave, which is a 0.5 pascals amplitude, was now measured using a different microphone that had a sensitivity of 10 microvolts per pascal, then what would be the RMS voltage produced again using the same kind of, logic and the same kind of calculations for mic B.

The V_{RMS} produced would be because a sinusoidal, it is a pure tone wave. the pure tone. wave generates a corresponding sinusoidal voltage signal.

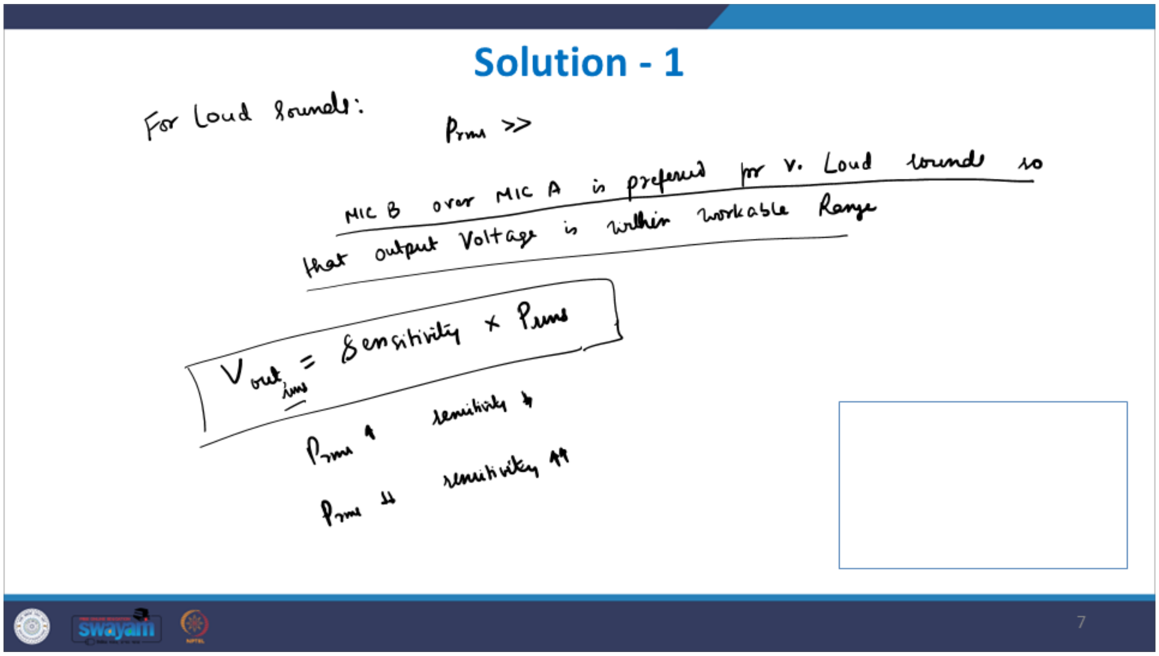
$$V_{rms} = \frac{V_{peak}}{\sqrt{2}} = \frac{sensitivity \times P_{peak}}{\sqrt{2}} = \frac{5}{\sqrt{2}} mV = 5.54\, mV$$

and the peak voltage can be obtained directly,

$$V_{peak} = sensitivity \times P_{peak}$$

And the peak value is given to us as 0.5. this here sensitivity is reducing by half. obviously, the overall answer should reduce by half. This is the answer for the second part for mic B.

It is also asked which microphone is better for recording quiet sounds and which microphone is better for recording loud sounds. Now, if you think about it, if the sound is very quiet, which means that for quiet sounds, usually it means that the pressure, the pressure value is quite low, or simply the P_{RMS} in general is a low value, and we would like to obtain a readable signal. We do not want a very weak or faint signal, so therefore for quiet sounds, we prefer a mic with higher sensitivity because already the RMS pressure is very low, and we would like to convert this very low acoustic pressure into a readable kind of voltage signal. The higher the sensitivity, the better, amplified signal we will obtain. Hence, for quiet sounds, mic A is preferred because it has a higher sensitivity over mic B for quiet sounds. Mic A is preferred over mic B for the quiet sounds recording. In the same way, with the same logic, for loud sounds If it is very loud, the P_{RMS} value could be very large. And in the last lecture, I told you that the microphones themselves have a dynamic



range, which means that there is a limit to the amount of sound they can record or the highest decibels of sound they can record. And with the increase in the decibel levels, ultimately, there is structural damage to the microphone itself, and they have a safe range. just to be on the safer side, both mic A and mic B could be used here because both are going to produce a good amount of signal, but just for the safety reason, we can choose mic B over mic A for very loud sounds so that the output voltage is within workable range, because whatever the voltage output ultimately is

$$V_{rms} = sensitivity \times P_{rms}$$

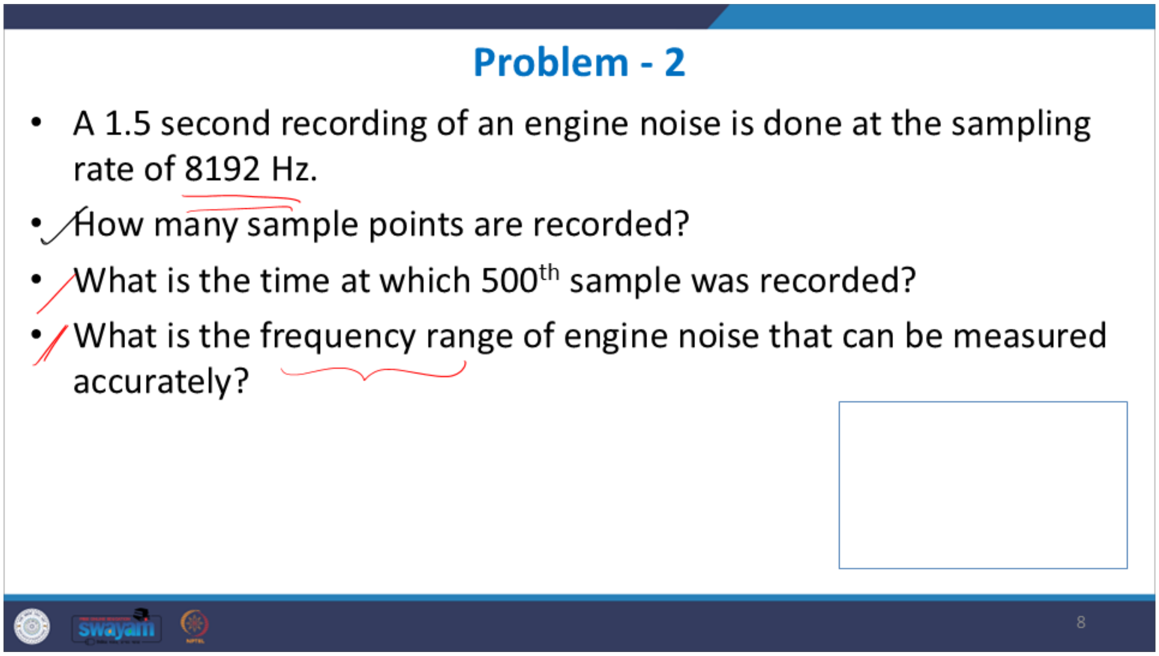
for loud sounds, if P_{RMS} is high, okay. we can do away with less sensitivity also, to keep the voltage within workable range, and if the P_{RMS} seems to be very low, then we need a more sensitive microphone to obtain not weak but slightly strong and readable signals. So here again, mic B is preferred over mic A for very loud sounds, and mic A would be preferred over mic B for quiet sounds.

After this, let us see another example. We have a 1.5-second recording of an engine noise done by a DAQ which has a sampling rate of 8192 Hz. various questions are asked:

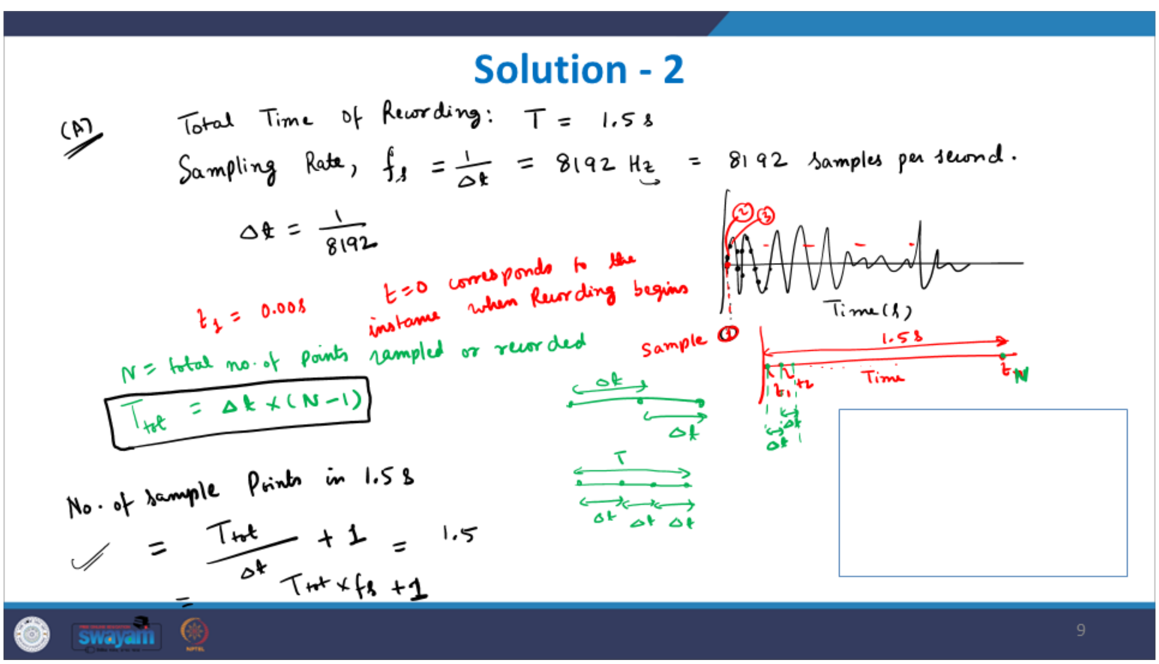
1) How many sample points are recorded using this particular setup?

2) At what time instant do we record the 500th sample?

3) And what is the frequency range of the engine noise that can be measured accurately?



let us go one by one and solve part A.



Here we are given that, the total time of recording t is 1.5 seconds, and we know that the sampling rate is inversely proportional to the sampling increment or the time interval between any two samples.

$$f_s = \frac{1}{\Delta t} = 8192\ Hz$$

what is Hz? We simply get 8192 samples per second. We are recording 8192 samples per second. Δt you can obtain using this equation; it would be

$$\frac{1}{8192} = \Delta t$$

for this recording, which means that suppose we have got a certain sound signal, and some signal is there. Then, between any two points, we have the increment, which is Δt , and that recording is being done like this.

let us say, first, the question is: how many samples are recorded? Here, let us see, this is our sample point one. Sample 1, and here the recording begins with the recording of the first sample. The time of the first sample is 0. from the 0, that is corresponds to the instance. T equals to 0 corresponds to the instance. When recording begins or data acquisition begins, sound recording. Then we have this as the sample point 2, this as the sample point 3, and so on. If you think about it, let us say, I think about it, this is just a time scale, and these are the times for the recording of sample 1. This is the time for the recording of sample 2, and so on. And finally, at some nth sample, which is the total number of samples, the recording stops. this is given to us as 1.5 seconds, the total recording time, and these intervals here, Δt the increment. You think about it Δt and so on, till the nth recording happens. How many Δt are there in these n samples? You have, you start from here, you begin your recording from here, here, and so on. there is a recording at the beginning and at the end. the number of points if you see about it, would be the total time here. The total time is divided into n minus 1 point.

so what we see is that the T total when n points are being recorded The t total when n points, where n is the total number of points that are sampled or that are recorded. Then, the total time scale is divided into n minus 1 division because we begin with the first sample and we end with the nth sample here. here, in between them, there are n divisions. For example, we are making a recording with only three sample points. If you see here, this is Δt, this is another Δt. The total time is composed of two such Δt. If there are four sample points then we have three Δt within that time scale, where this entire thing is total t, and the t total is divided by n minus 1 division.

$$T_{tot} = \Delta t \times (N - 1)$$

And now we have to find this n value.

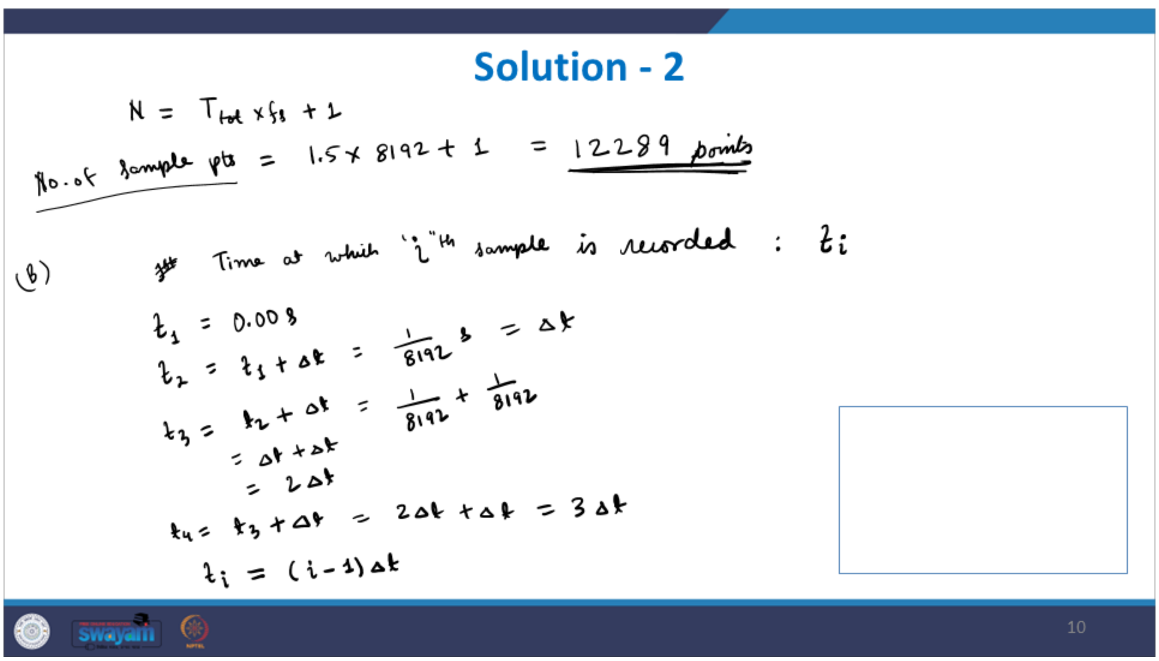
$$No. of\ sample\ point\ in\ 1.5\ s = \frac{T_{tot}}{\Delta t} + 1$$

if you do this

$$= \frac{T_{tot}}{\Delta t} + 1$$

1.5 seconds Divided by delta t and delta t is this 1 divided by you can also write it like this,

$$= T_{tot} \times f_s + 1$$



If you go here n is going to be t total Multiplied by F s plus 1 from this equation here 1 by delta t is Fs. The number of sample points if you solve this particular thing here the answer that you are going to get is.

$$No. of\ sample\ point = 1.5 \times 8192 + 1 = 12289\ points$$

these many points have been acquired.

In a duration of 1.5 seconds so that is our number of sample points.

let us solve the second part which is what is the time at which 500 sample was recorded so let us see here, first let us say time at which some ith sample This ith sample is recorded is denoted by the symbol ti. We know that the very first sample is recorded the moment we begin the recording and acquire our first data that is taken as the t equals to 0, it is the instance of the beginning of recording. The first recording happens at 0.00 seconds of the recording duration and then the increment between two sample points is the Δt. This would be whatever is the time instance of the first recording plus a Δt.

$$t_2 = t_1 + \Delta t = \frac{1}{8192}$$

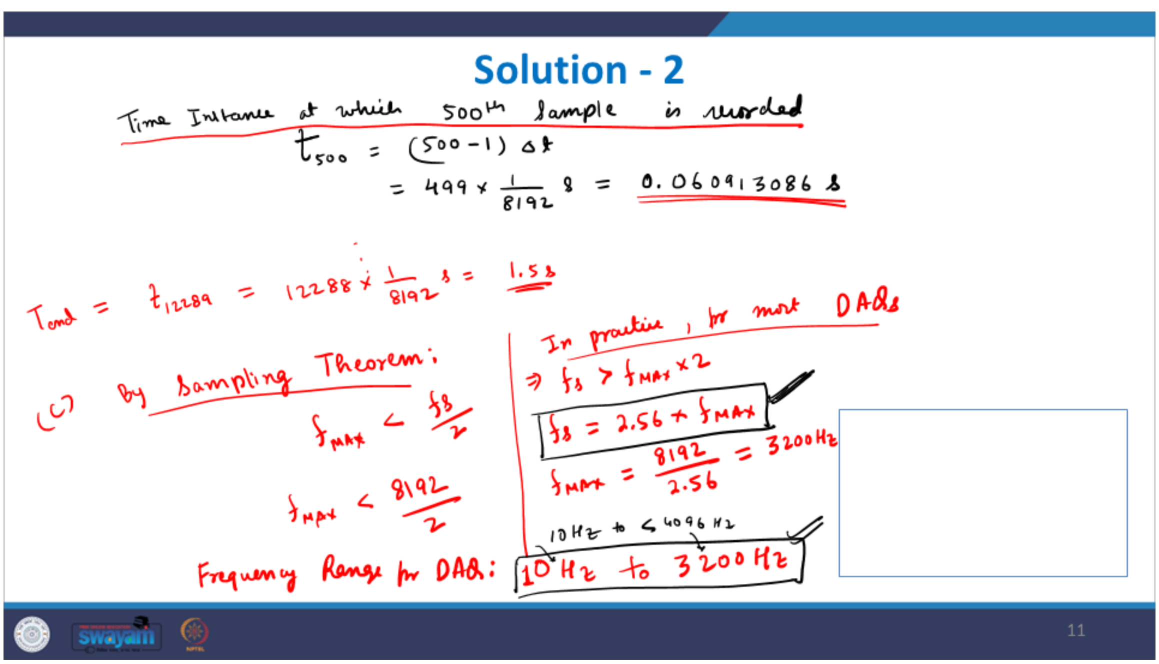
and t 3 would be again

whatever is the time at which the previous recording was done plus another increment. after another delta time there is a new recording.

$$t_3 = t_2 + \Delta t = \frac{1}{8192} + \frac{1}{8192}$$

Similarly, t4 would be t3 the time at which third sample was recorded plus the gap between the two which is Δt and third was recorded at $2\Delta t$ plus another Δt, this gives you $3\Delta t$. which means that for any ith recording, a generic observation is that if it is i^{th} sample

$$t_i = (i - 1) \times \Delta t$$



It gets recorded. You have to find out the time at which the 500th sample is recorded.

The time instance at which the 500th sample is recorded is simply

$$t_{500} = (500 - 1)\Delta t$$

$$= 499 \times \frac{1}{8192} = 0.0609$$

I am just using more significant digits because here Δt is very small. This becomes our answer for the second part of the question.

Now, if you just see by logic, suppose what is the time at which the last sample was recorded? We have t_{12289} the end time of the recording where the last sample gets recorded. this should be

$$12288 \times \frac{1}{8192} = 1.5\ sec$$

which gives you the total recording time.

Let us see the third part. What is the frequency range of engine noise that can be measured accurately? Here to find out the frequency range, we have a particular sampling rate, and we want to see what frequencies can be recorded.

Let us use the sampling theorem. As per the sampling theorem, which is also called the Shannon-Nyquist theorem, whatever is the maximum frequency that we need to register accurately, it should be smaller than fs by 2 or smaller than the Nyquist frequency,

$$f_{max} < \frac{f_s}{2}$$

which means that the maximum frequency that we can record accurately has to be smaller than 8192 by 2.

$$f_{max} < \frac{8192}{2}$$

Now, in practice for most of the DAQs, this must be smaller than fs by 2. This implies that fs has to be greater than f max multiplied by 2.

$$f_s > f_{max} \times 2$$

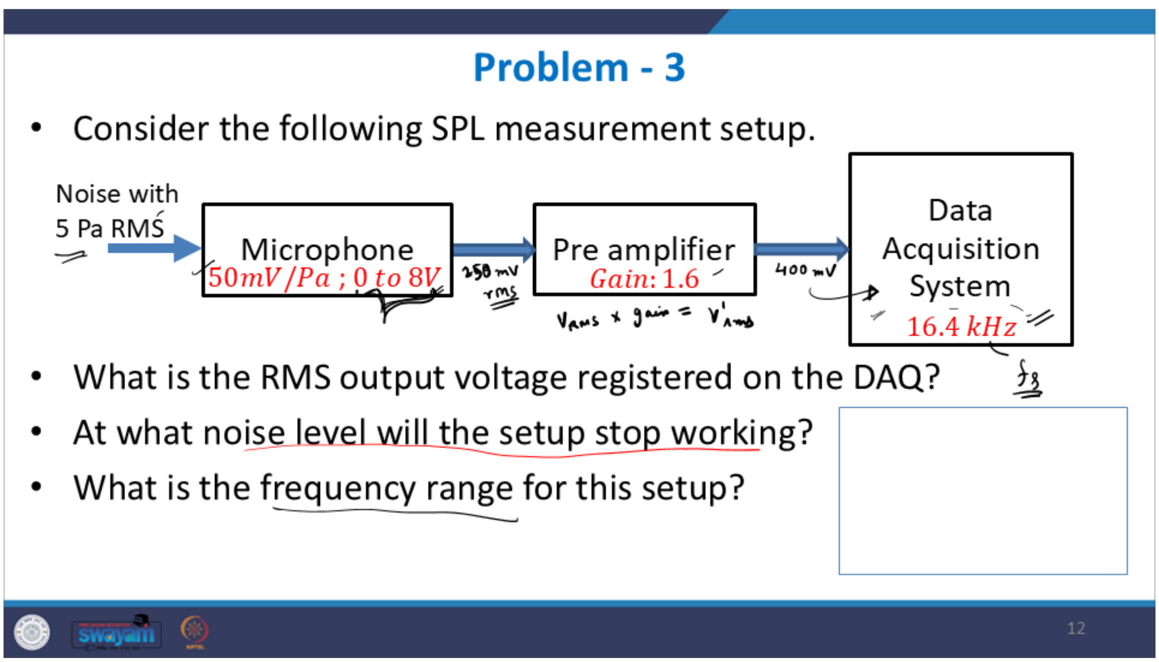
for most of the DAQs in practice, the fs is usually taken as 2.56 times the f max.

$$f_s = 2.56 \times f_{max}$$

This is the generic guideline that is used for most of the DAQs. They want to ensure that whatever is the frequency range of interest, the sampling rate is set so that it is greater than twice the frequency range of interest. Usually, we set it at around 2.56 times the maximum frequency of interest. If you think about it, using this thing, then the fmax using this setting should come out to be

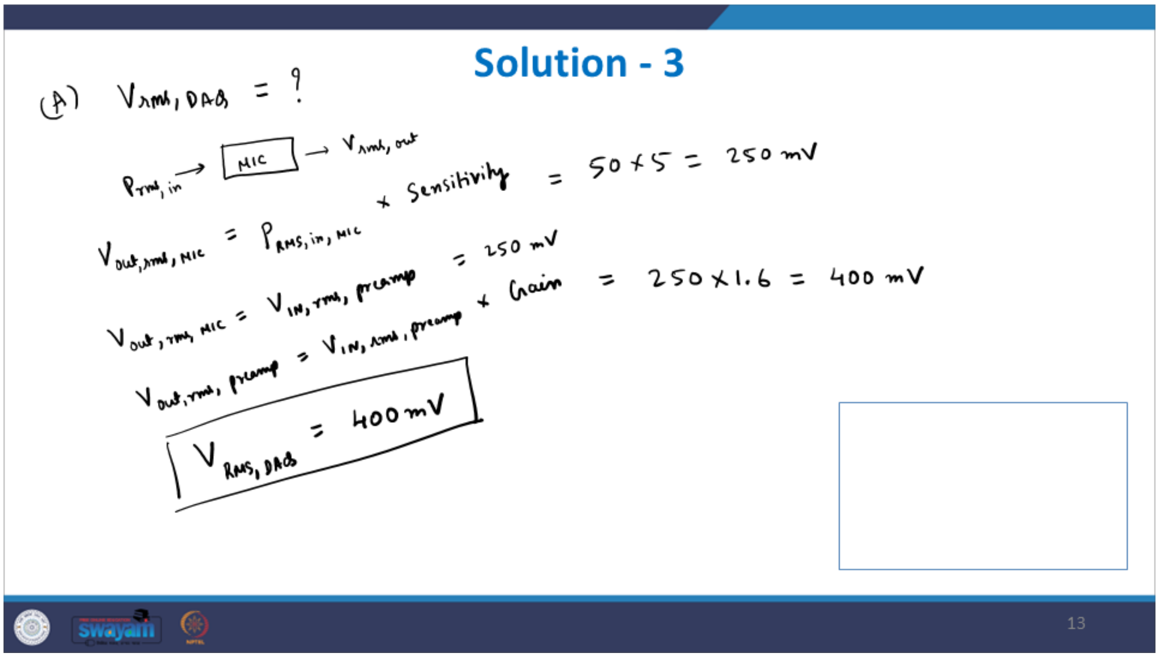
$$f_{max} = \frac{8192}{2.56} = 3200\ Hz$$

The range of frequency range for the given DAQ setup is going to be from typically zero Hz, and even at zero Hz, most of the DAQs don't function at zero Hz. We usually start at 10 Hz. That is again a generic setting for most of the DAQs. You can take 10 Hz to 3200 Hz with a setting of 2.56 factor. In general, what has to happen is that it should be between 10 Hz to less than 4096 Hz, but with the given setting, usually, this less than 4096 Hz is given as 3200 Hz with this particular formulation. This is something that is there for most of the DAQs, which gives us the frequency range.



So let us solve another problem. We are given some recording setup. We are recording the sound pressure level. some noise whose PRMS is 5 pascals is incident on a microphone, and some kind of values are given for this microphone.

Some characteristics are given in the red color. Then it goes through a pre-amplification stage, and some characteristics of the pre-amplifier are given. And then it goes to the DAQ, and again, some characteristics of the DAQ are given. you have to find here what is the RMS output voltage that is registered on this particular DAQ. Let us solve the very first part.



What we have to find is the VRMS at DAQ. here, what does the microphone do? The microphone converts this PRMS, whatever is input into the microphone, into some RMS voltage. Let us indicate the input and the output for the microphone using the subscript mic or the subscript mic. Then the V output RMS from the microphone should be simply whatever is the pressure.

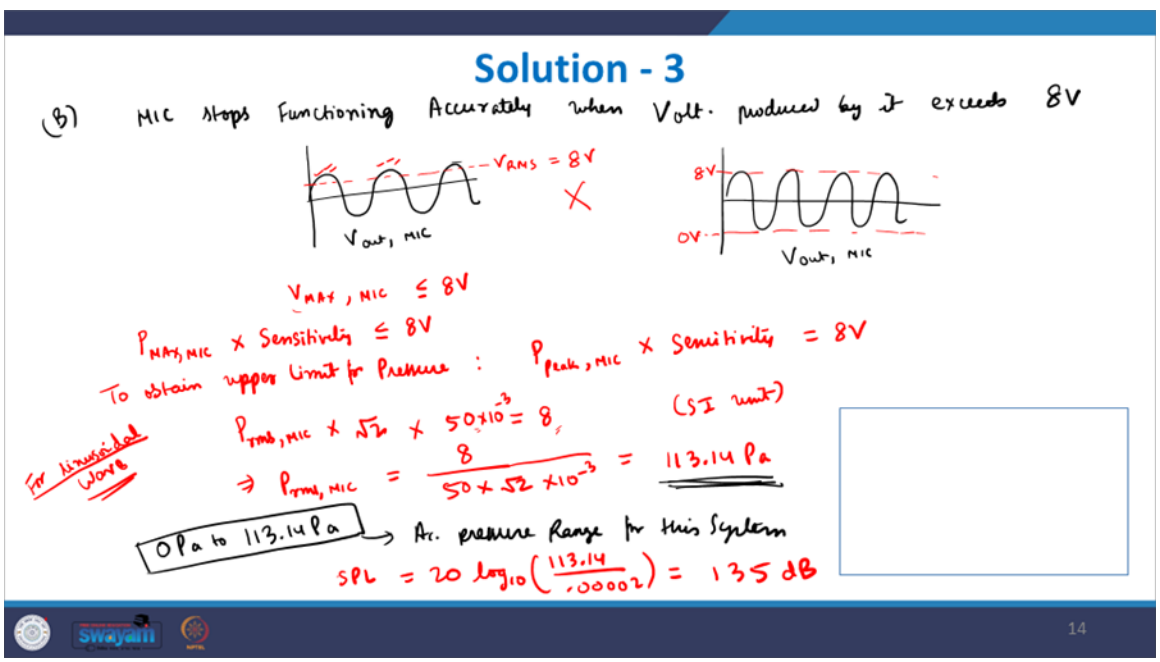
P_{RMS} that was input into the microphone, multiplied by the sensitivity, should give you the net output voltage that is coming out of the microphone. this will come out to be the sensitivity, which is given as 50 microvolts per Pascal. 1 Pascal gives 50 microvolts. 5 Pascals give 50 multiplied by 5, which is 250 microvolts. This is the output that is coming out here.

Here we are getting 250 microvolts as the RMS. Now, this is going through the preamplifier, and here the preamplifier has some gain value given. The purpose of a preamplifier is to amplify the signals. the gain factor indicates the factor by which each value in the signal is multiplied. here all the V_{RMS} that are coming inside will be multiplied by the gain to get a new voltage signal, which is amplified in nature. So here, if the gain is 1, the signal remains the same. If the gain value is more than 1, the signal gets amplified. If the gain is less than 1, then the signal is dimmed down or attenuated. What will the PRMS be? this V output RMS, which is coming out of the microphone, now becomes the input RMS for the preamplifier. this is what is going inside the preamplifier.

The output RMS from the preamplifier is that it will simply take this input, which is going into the preamplifier, and multiply those values with the gain to amplify the signal. now the 250 millivolts that was going into the preamp is amplified by this factor, and you get a certain value, which is 400 millivolts. This is now the V, the output coming out from this.

250 millivolts are now amplified to 400 millivolts, which are then received by the DAQ. The DAQ is doing nothing further; there is no other thing given for the data acquisition system. No further conditioning is done to this signal, and the signal is acquired at this rate. Overall, the V_{RMS} that is being registered at the DAQ will be 400 millivolts.

Now, let us see part 2, which is at what noise level the setup will stop working. Some noise levels are given. If you think about it, various characteristics are given. This is not going to give us the range of noise level, neither is this going to give us some range because here the range within which this microphone can work is given to us: 0 to 8 V. from here, we can work out the noise level range at which the system is going to stop functioning. The microphone will stop functioning once the voltage that is produced reaches 8 volts.



Here the mic stops functioning accurately when the voltage produced by it, exceeds 8 volts. That is the interpretation of this particular information here. 8 volts. Now, let us see if this is the voltage produced by the mic, the voltage signal. It says that the voltage cannot exceed 8 volts. Which means that not the RMS value but the max value because this is supposed RMS was supposed to be 8 volts. Then, suppose if the RMS, let us say, the first case. That we took that. That means that the RMS has to be 8 volts. RMS would be something like 1 by root 2 times the voltage. This is like the V_{RMS}. If you set this to be 8 volts, there are certain times where the instantaneous value is exceeding the 8 volts, so the microphone would damage, it will stop functioning. The maximum value has to be 8 volts, which means you have to set the peak voltage as 8 volts. This should not happen. What should actually happen is this: if this is the V output from the mic, this is your voltage signal, so this voltage signal should never exceed the 8 volts value. Which means that this is the maximum voltage that it can reach. This magnitude should be 8 volts. The peak amplitude should not exceed 8 volts. Given that, zero starts from here, okay? With this as a reference, the peak should not exceed 8 volts. when V max at the microphone at mic should be less than or equal to 8 volts. This means that, to obtain what is the pressure corresponding to the voltage max value, which means that the P max at the microphone multiplied by the sensitivity. V max will be obtained by multiplying the P max with the sensitivity of the microphone, and that should not exceed 8 volts. To obtain the range, to obtain the upper limit, we will use the equality for pressure Pmax incident on the microphone multiplied by the sensitivity is 8 volts. This is the upper limit of the P max or the P amplitude. I can see this as the amplitude. The amplitude of the pressure, so the amplitude of the pressure should not exceed, not exceed 8 volts. Now, let us relate this to RMS. the RMS of the pressure, given that it is a sinusoidal wave, this is our assumption. For a sinusoidal wave, and any generic waveform can, anyway, be represented as a summation of sinusoidal components. P RMS mic multiplied by root 2, this gives you the Ppeak.

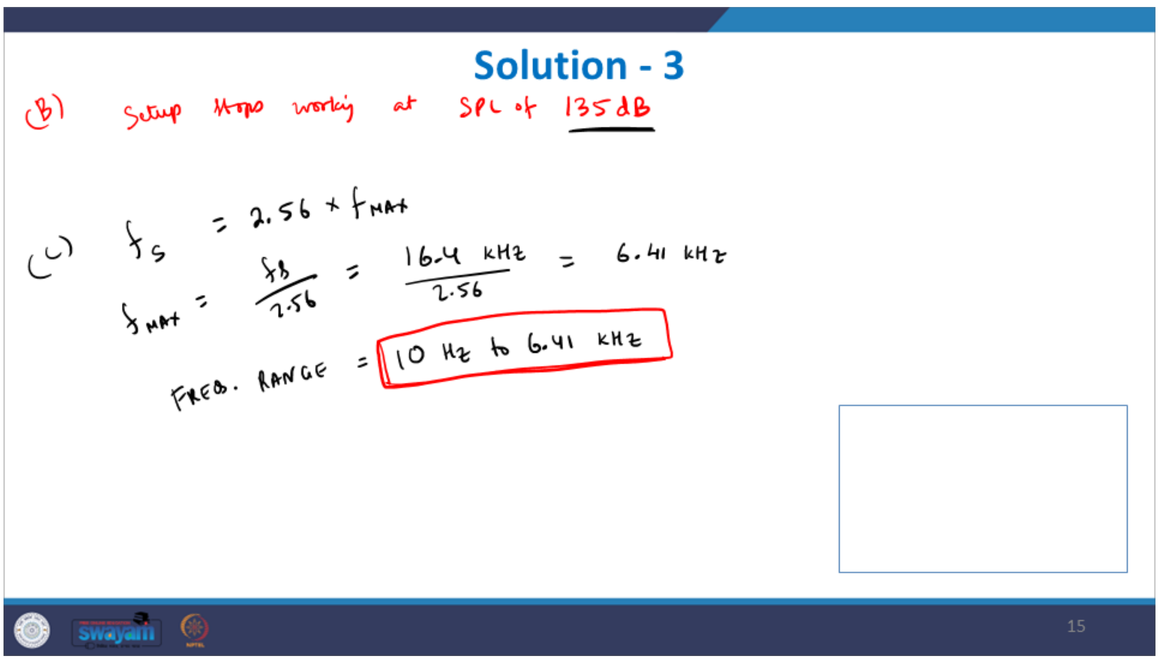
$$P_{rms.mic} \times \sqrt{2} \times 50 \times 10^{-3} = 8$$

$$P_{rms.mic} = \frac{8}{50 \times \sqrt{2}} = 113.14\,Pa$$

To get the pascals, so again, the value, if you solve it, comes out to be 113.14 pascals. this is the RMS pressure at which the microphone is going to stop functioning beyond which, so the upper limit for the decibels. You can either leave your answer like this, that this is the pressure range from 0 pascals to 113.14 pascals. The acoustic pressure range for this system, or you can convert it into decibel format. If you convert it into the decibel format, you have, the SPL for that would be

$$SPL = 20 \log_{10}\left(\frac{113.4}{0.00002}\right)$$

20 log 10 of PRMS, which is 113.14, divided by the reference pressure. What you will get when you convert it into decibels is 135 dB. You can also write this, usually it is written in dB, okay, the noise level where the setup stops working. We can say that for part B, the setup stops working. At SPL of 135 dB.



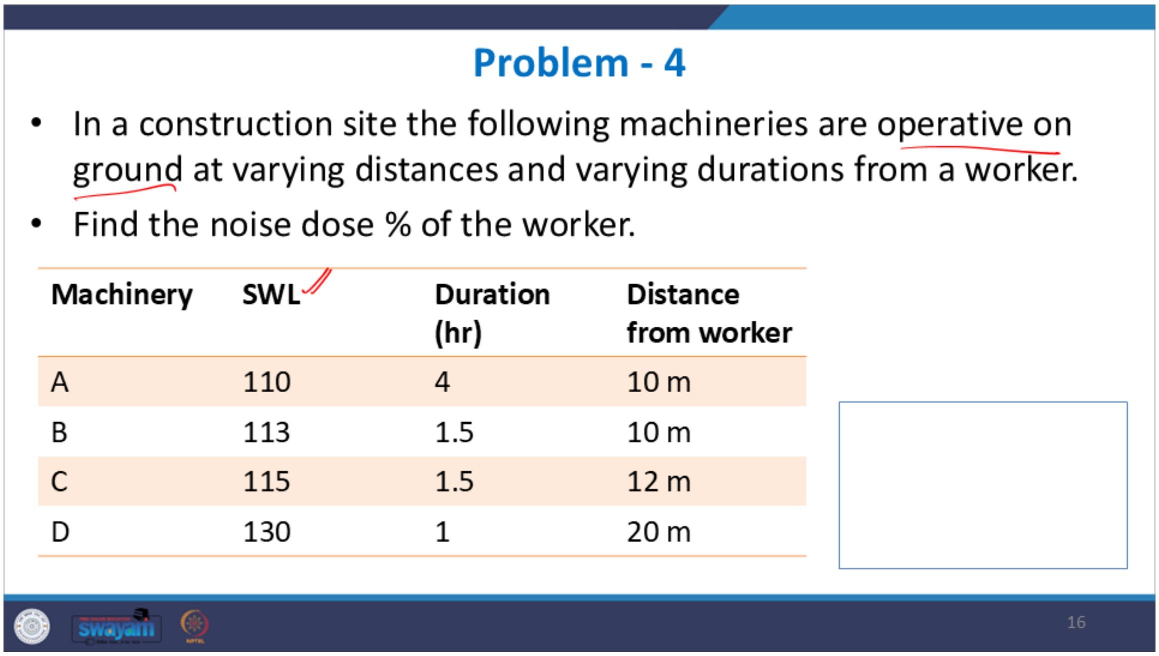
This becomes our dynamic range, okay. Now, let us see the last part of this equation.

What is the frequency range for this setup? Again, you can use the Nyquist theorem or the sampling theorem here. This is the fn. This is the sampling rate given for the data acquisition system, which is 16.4 kHz.

f max from the previous question, we know that the setting that is usually used is the fs needs to be greater than twice the fmax. The sampling rate is usually set as 2.56 times of the fmax.

$$f_{max} = \frac{f_s}{2.56} = \frac{16.4}{2.56} = 6.41\ kHz$$

The frequency range is going to be, let us say, typically from 10 Hz to 6.41 kHz for accurate recording. Usually, we do not start at 0. 10 Hz to 6.41 kHz. this solves all the questions.



Problem - 4

- In a construction site the following machineries are operative on ground at varying distances and varying durations from a worker.
- Find the noise dose % of the worker.

Machinery	SWL	Duration (hr)	Distance from worker
A	110	4	10 m
B	113	1.5	10 m
C	115	1.5	12 m
D	130	1	20 m

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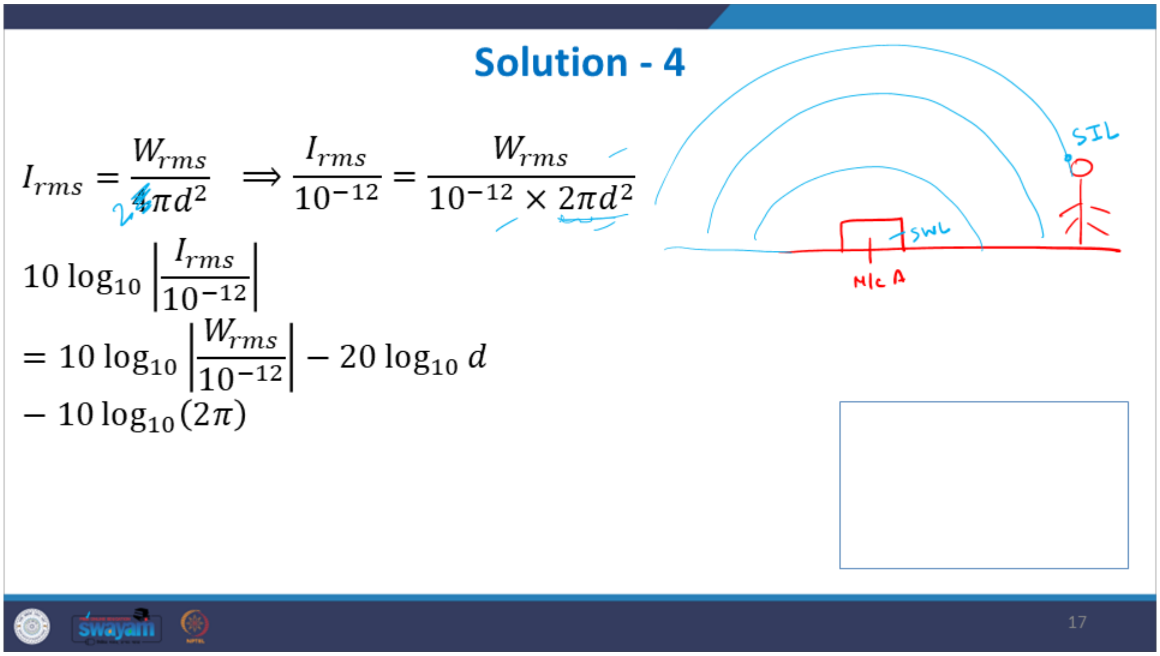
Now, we have a last problem in this particular topic.

In a construction site, the following machineries are operative at the ground at varying distances and varying durations from a worker. Find out the noise dose percentage of the worker. A table is given that these are the different machineries operating on a factory floor, and it is in an open field. Considering that it is a construction site, it is an open field environment, and the machineries are kept on the ground at varying distances from the worker. This is the sound power level of the various machineries. Let us deduce what should be the SPL that is heard at the worker location.

For that, if we take just the schematic of one simple machinery, suppose this is some machinery, any kind of machinery, and there is some worker here. This machinery here emits the sound waves in the free field. Given that it is kept on the ground, it will get a hemispherical wave front.

Not spherical but rather hemispherical wave front like this. Whatever this is the sound power level that it is emanating, and at the worker location, some sound intensity level is heard.



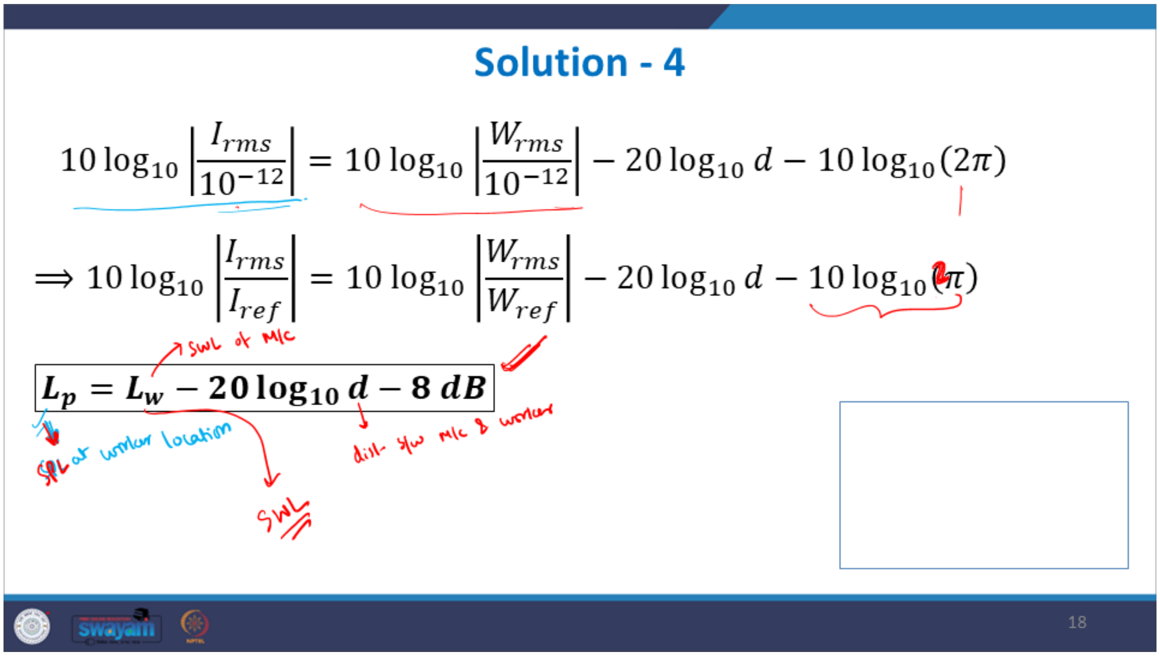
Let us relate the two. what you have is the intensity, the RMS intensity, which should be whatever is the power intensity divided by the area or the wave front area because it is kept on the ground. Here, we have a hemispheric, the hemispherical area, so we take the area of the hemisphere. This should be actually $2\pi d^2$. If suppose it was suspended somewhere in the air and acted as a point source, it would complete a full sphere, and it would be $4\pi d^2$. over here, we take $2\pi d^2$ square, okay? The power divided by the area, and then you do this calculation, and then you do the 10 log 10 of both sides.

What you get is this,

$$I_{rms} = \frac{W_{rms}}{4\pi d^2} \Longrightarrow \frac{I_{rms}}{10^{-12}} = \frac{W_{rms}}{10^{-12} \times 2\pi d^2}$$

$$10 \log_{10} \left|\frac{I_{rms}}{10^{-12}}\right| = 10 \log_{10} \left|\frac{W_{rms}}{10^{-12}}\right| - 20 \log_{10} d - 10 \log_{10}(2\pi)$$

If you see here, this is what we are getting; this becomes our SPL at the worker location. This is the SPL at the worker location, and this is the sound power level of that machinery, okay? And this is the distance between machinery and workers.



Once you solve it, these 10 logs 10 thing here, this becomes your Lp, this becomes your Lw or the sound power level minus

$$10\log_{10}\left|\frac{I_{rms}}{10^{-12}}\right| = 10\log_{10}\left|\frac{W_{rms}}{10^{-12}}\right| - 20\log_{10} d - 10\log_{10}(2\pi)$$

$$\Rightarrow 10\log_{10}\left|\frac{I_{rms}}{I_{ref}}\right| = 10\log_{10}\left|\frac{W_{rms}}{W_{ref}}\right| - 20\log_{10} d - 10\log_{10}(2\pi)$$

$$\boldsymbol{L_p = L_w - 20\log_{10} d - 8\ dB}$$

This is the equation that is deducted for this particular problem. using this equation now, we can obtain what is the SPL received at the worker location. with the same, we use this particular formulation; we put LW, which is the SWL. And we find out the SPL at the worker location; it comes out to be these values for the different machinery where this is the distance d. Now we have the SPL at the worker location; we can use the noise dose formula, which is given by this.

$$D = \left[\sum \frac{C_i}{T_i}\right] \times 100\ ;\ T_i = \frac{8}{2^{(L_i-85)/3}}$$

Where every Ti for the NIOSH. all of this we have covered in our previous lectures. now, let us see: This is the actual duration of exposure, this is the level at which these exposures

Solution - 4

Machinery	SWL	Duration (hr)	Distance from worker (d)	SPL at worker location
A	110	4	10 m	82 dB
B	113	1.5	10 m	85 dB
C	115	1.5	12 m	85.4 dB
D	130	1	20 m	96 dB

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Solution - 4

Noise Dose:

NIOSH

$$\boxed{D = \left[\sum \frac{C_i}{T_i}\right] \times 100\ ;\ T_i = \frac{8}{2^{(L_i - 85)/3}}}$$

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are being done, and by the NIOSH guidelines, this should be the maximum permissible durations for these particular levels.

Solution - 4

Machinery	Duration (hr)	SPL at worker location	Maximum exposure time
A	4	82 dB	16
B	1.5	85 dB	8
C	1.5	85.4 dB	7.29
D	1	96 dB	0.63

NIOSH

$$D = \left[\sum \frac{C_i}{T_i}\right] \times 100 \; ; \; T_i = \frac{8}{2^{(L_i - 85)/3}}$$

So the noise dose can be calculated by doing 4 by 16 plus 1. The actual duration should summation of the actual duration by the maximum allowed duration. So you see this, you compare this row with this row. And you see 4 by 16 plus 1.5 by 8 plus 1.5 by 7.29 plus 1 by 0.63 multiplied by 100. Overall you get a 223% noise dose, which means overexposure to hazardous noise levels.

$$D = \left[\frac{4}{16} + \frac{1.5}{8} + \frac{1.5}{7.29} + \frac{1}{0.63}\right] \times 100 = \mathbf{223}\%$$

Solution - 4

Machinery	Duration (hr)	SPL at worker location	Maximum exposure time
A	4	82 dB	16
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D	1	96 dB	0.63

$$D = \left[\frac{4}{16} + \frac{1.5}{8} + \frac{1.5}{7.29} + \frac{1}{0.63}\right] \times 100 = \mathbf{223}\%$$

Over Exposure to Hazardous Noise Levels

Okay, so this is how you can solve the various problems, and with this, I would like to close the lecture. Thank you.