

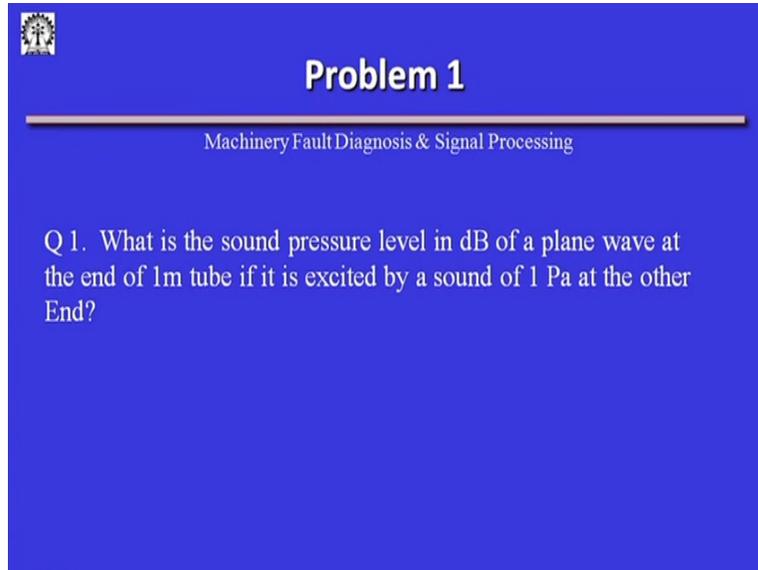
Machinery fault diagnosis and signal processing
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Module No # 05
Lecture No # 22
Numericals in Noise Vibration and Data Acquisition

In this class we are going to discuss about some numerical on noise vibration and data acquisition as by now we would have understood the fundamentals of vibration then little introduction into noise and of course we need to acquire data so that we can analyze them on the computer.

So in this class we are going to give you six examples is problems which we normally we encounter while doing machinery condition monitoring and this through this example I will tell you certain simple tricks or the trade which has to be used while doing machinery condition monitoring.

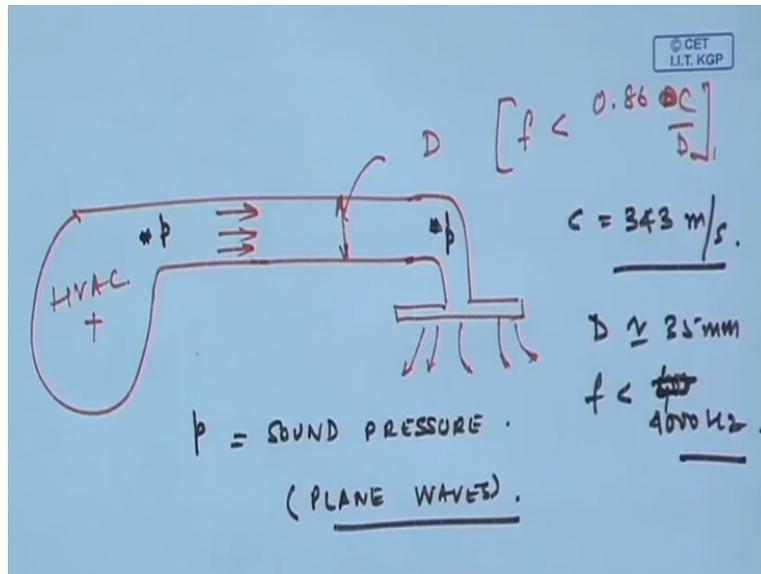
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The slide is blue with white text. It features a small logo in the top left corner. The title 'Problem 1' is centered at the top. Below it, the course name 'Machinery Fault Diagnosis & Signal Processing' is written in a smaller font. The main content is a question: 'Q 1. What is the sound pressure level in dB of a plane wave at the end of 1m tube if it is excited by a sound of 1 Pa at the other End?'

So the first problem is actually so very simple problem like what is the sound pressure level in decimal or DB at the plan wave at the end of 1 meter if it is excited by a sound of 1 Pascal at the other end.

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So let me explain you this problem in the sense we have a duct this a long duct it could be of any length so usually to give a source at a one end it is a plain wave source because the diameter of this duct is D such that the frequency below which the plain waves propagate or given by this equation. So $.86$ times speed of sound by the diameter of the duct for example the typical duct of diameter about 35 MM this close comes to about 4000 Hertz this is typical.

So that means I have a plain wave tube of duct diameter 35 MM so for any source whose frequency is less than 4000 I will be having plain waves travelling in this waveform. The plain waves I mean they are longitudinal they are one directional so pressure at any cross section is this is M . Any point at this cross section is this M any point in the cross section is this.

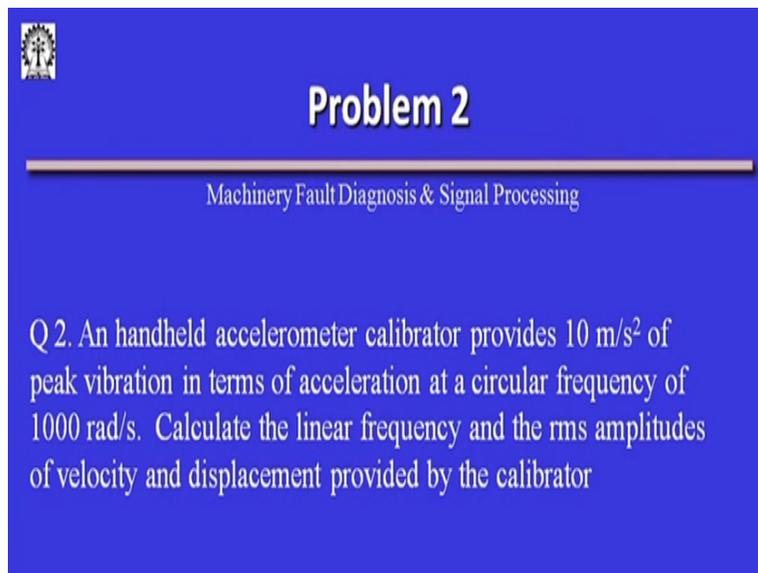
So this is a little tricky plan in the sense this sound wave, plane wave if it is propagating in a dug along the dug diameter satisfy this equation and the frequencies are less than 4000 hertz am going to have plain waves at the other end. So this is the other end and that the pressure will be this same of the inlet pressure = 1 Pascal and outlet pressure P outlet will be also = 1 Pascal .

So in order now to express Pascal in decimal we use the formula PL is nothing but 20 logarithms to base 10 P by P reference. So P reference is 2 into 10 the power -5 Pascal . For plug P as 1 Pascal in this equation I will come with the value of 94 decimal. This as practical significance in the sense that in any problem or any special duct application for example I have an HVAC system where I have blur giving an condition layer to a point so in sort of room and diffuser.

So in this diameter of this duct a waves is a greater than that plan waves propagate so any pressure in Pascal will be the same sound pressure and this P is the sound pressure. So we do not require any complication to get a formula to estimate the sound pressure level at the end of the duct as long as we know that it is a plane waves and plan wave conditions are satisfied. There is an I told you about the problem is in many times in industry will become cross section audio here noise is generated at one and then we have to guess or estimate the sound pressure level at the other end.

And if you just measure the diameter will ensure that the equations are abide C is the speed of sound in air at activity results is I will know in that this 35 MM the amount calculate as 4000 hertz so this can be calculated ok. So now we will move away to second problem which is again a very simple problem.

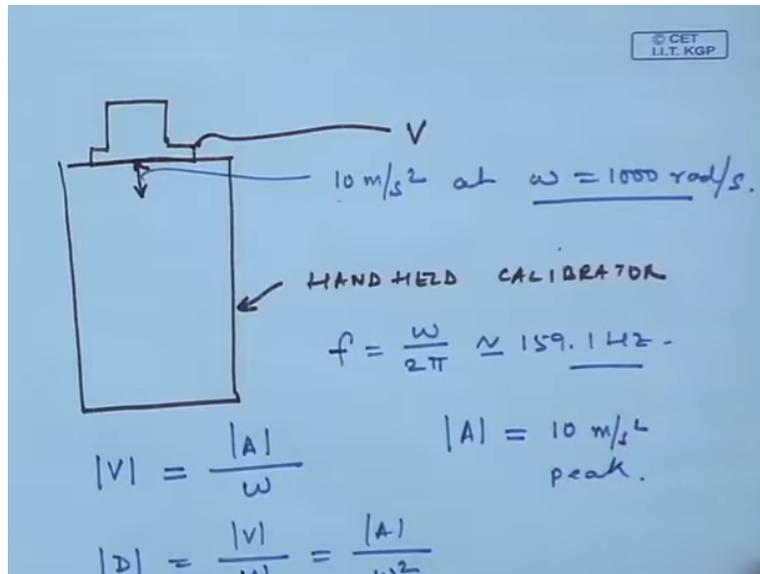
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This is and handheld accelerometer calibrator provide 10 meter per second square of peak vibration in terms s acceleration at a circular frequency of 1000 radians per second. Calculate the linear frequency and the RMS amplitudes of velocity and displacement provided by the calibrator.

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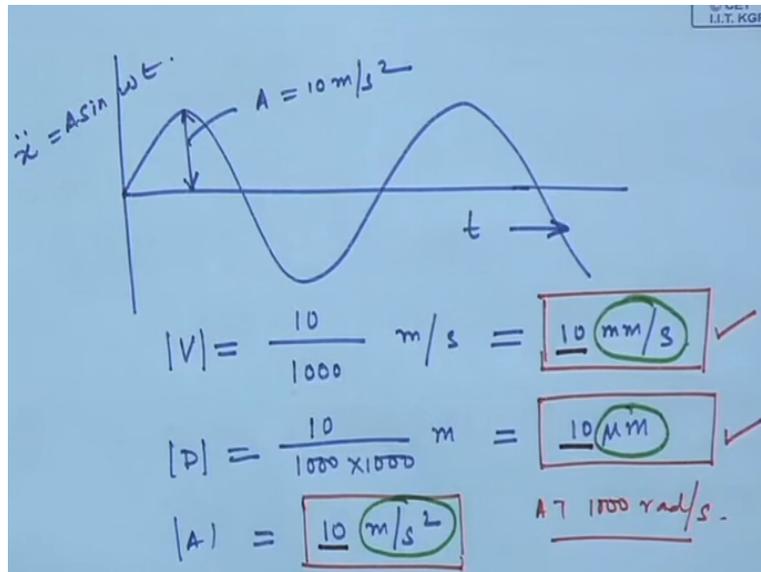


Usually to calibrate an accelerometer in the fuel it is actually mounted on a calibrator this is the hand held calibrator and this will give certain voltage or charge signal corresponding to this mechanical vibration subject to and this vibration is typically about 10 meters per Second Square at a circle of frequency of 1000 radians per second.

The question is what is the velocity and what is the acceleration as you know this is a fuel tone of 1000 meter per second its frequency of mega is sorry linear frequency F is equal to nothing but ω by 2 Pie and this comes down to 159 .1 hertz. But I know velocity magnitude for a harmonic wave is nothing but the acceleration magnitude divide by omega and this displacement magnitude is nothing but the velocity.

Magnitude divided by omega is equal to acceleration magnitude divided by omega square okay. So in this we have acceleration magnitude is 10 meter per second square and its say it is peak.

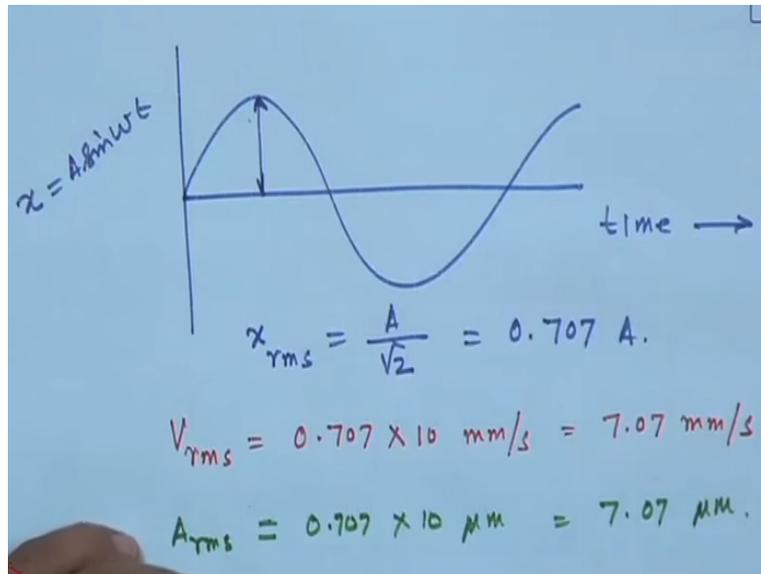
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So that means an follows to plot the time is such as signal so this value is nothing but 10 meter per second square is the acceleration wave form okay. So in this case the velocity will be nothing but 10 divided by omega is 1000 so this will come to meters per second so this will correspond to 10 millimeter per second so I am putting down to box to explain to you later and similarly the displacement magnitude is nothing but 10 divide by 1000 into 1000 meters that is equal to 10 micro meter.

So if my acceleration magnitude is 10 meter per second square for the same calibrator at 1000 radians per second I have velocity 10 millimeters per second displacement as micro meter. So to remember a good rule of thumb to a field is this is 10 this is 10 this is 10 only in the that we have to casual about the units one in the meter per second square others is in millimeters per second square and other is in micron.

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So this problem as a second problem as to calculate the RMS amplitude and we know for a harmonic wave any general $X = A \sin \omega t$ the RMS amplitude X_{RMS} is nothing but A by root 2 = $0.707A$. So to answer this second part of the problem to find out the amplitudes of velocity and displacement provided by the calibrator the V_{RMS} would be therefore .707 times time millimeter per second that is 7.07 millimeter per second.

And the displacement RMS $A_{RMS} =$ nothing but 0.707 times 10 micron that is 7.07 microns. These are certain handy numbers to remember in the fuel for example whenever we go to fuel this handheld calibrator which I am at shown in one of the previous classes as to how this calibrator is actually used to calibrate accelerometer in the fuel all we know is the mechanical signal of 10 meters per second square 1000 radians per second.

We will just see the voltage output from this accelerometer and so I am going to calibrate in the displacement mode or in the velocity mode or in the acceleration mode we will apply these actual mechanical value to the signal so as to calibrate what is the voltage level corresponding to or what is the mechanical value corresponding to particular voltage value we have obtained.

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Problem 3

Machinery Fault Diagnosis & Signal Processing

Q3. For a machinery having 90 dB of sound power level compute the sound pressure level in dB and Pa at a distance of 1 m in free field condition.

Now we will move over to the next problem question three which says a machinery having 90 decibel of sound power level compute the sound pressure level in DB and Pascal as a distance of 1 meter in free field condition okay.

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Q.3

SPL, SWL

$$SPL = SWL + 10 \log_{10} \left[\frac{Q_0}{4\pi r^2} + \frac{4}{R_c} \right]$$

Annotations:

- Q_0 : DIRECTIVITY
- r : DISTANCE FROM THE SOURCE
- R_c : ROOM CONSTANT.

Labels:

- SOUND PRESSURE LEVEL (pointing to SPL)
- SOUND POWER LEVEL (pointing to SWL)

Definitions:

$$R_c = \text{ROOM CONSTANT} = \frac{S \alpha_{av}}{1 - \alpha_{av}} \text{ m}^2$$

S = TOTAL SURFACE AREA OF THE ROOM.

α_{av} = AVERAGE SOUND ABSORPTION CO-EFFICIENT OF THE ROOM.

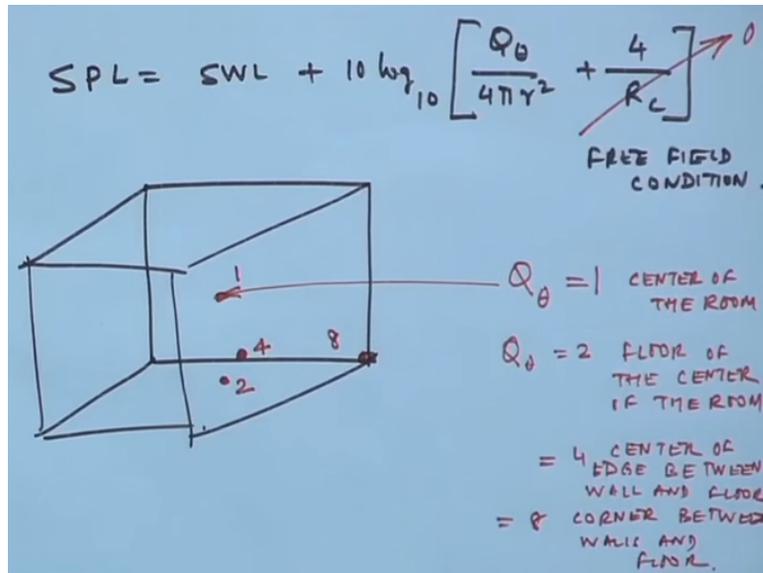
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Usually the sound pressure level is denoted by SPL and sound power level SWL and the general equation for the sound pressure level as with relation to sound power level is given by this equation that $SPL = SWL + 10 \log_{10} \left[\frac{Q \theta}{4 \pi r^2} + \frac{4}{R_c} \right]$ is a very important formula and I will explain you what this terms mean here this is the sound pressure level we all know that and this is the sound power level θ θ is a $Q \theta$ is

directivity or versus distance from the source R_C with the room constant which depends on the amount of sound absorption in the room.

So R_C is given by this formula $S \alpha_{\text{average}} / (1 - \alpha_{\text{average}})$ in meter square where S is the total surface area of the room α_{average} is the average sound absorption co-efficient of the room.

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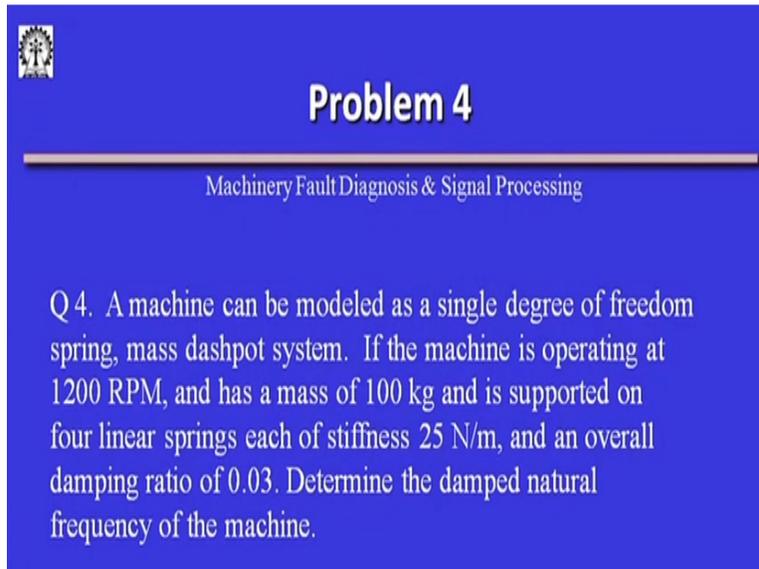
So in a room so far I take a wall and I divide them into small patches and each one as a different of Pie and area this is S_i . So the average α_{average} is nothing but $S_i \alpha_i / \sum_{i=1}^N S_i$ and then other total area. Now this could be of different material and they all could be of different α is for example this could be Gypsum this is could Fiber glass this could be jute of course now a days you know we have found out that Jute and its derivatives are good sound of this also.

So just to come back to the problem which we are discussing so we can have the room constant like this. So if the room was totally absorbing or there was a no sound which is the generated (0) (17:55) room and have a sound source and these are walls which are treated with sound absorbers and then we have amount of a absorbers in this walls. So and this was in the center of the room in typical case in this case you are absorption in the maximum. So your R_C is actually close to infinite.

So the second term $4 / RC$ because the room constant actually disappears in a case where it is absorbing that means there are no echo's or an unechoing condition have been obtained and this is essentially what is known as the free field condition. So only in the case of a free field condition this problem becomes so SPL will become $SWL + 10 \log_{10} Q_{\theta} / 4 \pi R^2 + 4 / RC$.

So this will become 0 in the case of free field condition okay and if it is in the center of the room the source in the center of the room if it is here the center of the room $Q_{\theta} = 1$ in the center of the room okay = 2 on the floor of the center of the room = 4 if it is on the center of the wall this is 2 this is 4 this is 1. Fourth is on the center of edge between wall and floor and this is equal to 8 as a corner between walls and floor.

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Problem 4

Machinery Fault Diagnosis & Signal Processing

Q 4. A machine can be modeled as a single degree of freedom spring, mass dashpot system. If the machine is operating at 1200 RPM, and has a mass of 100 kg and is supported on four linear springs each of stiffness 25 N/m, and an overall damping ratio of 0.03. Determine the damped natural frequency of the machine.

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$$SPL = SWL + 10 \log_{10} \left[\frac{Q_0}{4\pi R^2} + \frac{4}{R_c} \right]$$

FREE FIELD CONDITION, $Q_0 = 1$
 $V = 1 \text{ m}$

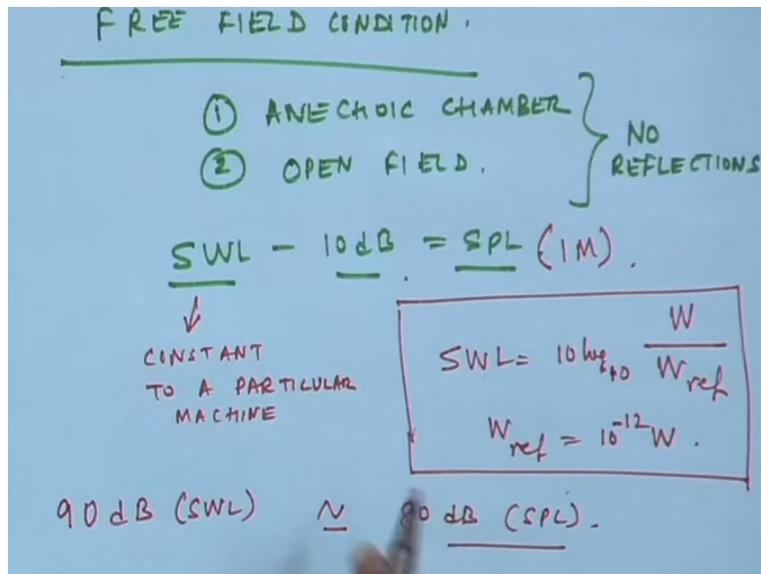
$$SPL = SWL - \underbrace{10 \log_{10} 4\pi}_{10 \text{ dB}}$$

So by this I mean this vice versa would also realized in the reciprocal relationship suppose you are a standing in sorry suppose you are standing in a room in a center of the room I will just draw one plane suppose I am standing here and I move to your loud speaker to this location here the directivity will increase here it to be louder then I put it in the middle of the room okay because of the directive issues on this is a you just explain.

So in other words if I come back to this problem in this case $SPL = SWL + 10 \log_{10} \frac{Q \theta}{4 \pi R^2} + \frac{4}{R_c}$ it has been told this free field condition. So free field condition mean this will vanish okay and the center of room if I take $Q \theta$ will be equal to 1 in the center of the room and at the distance $R = 1$ meter my expression will come up to $SPL = SWL - 10 \log_{10} 4 \pi$.

Now some of the important things we have observe from this case because here I have taken $Q \theta = 1$. So log of 1 is actually 0 so it will disappear and this value will be approximately = 10 decibel.

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Again I go to rule of thumb you should follow is in any free field condition now what are typical free field conditions one is Anechoic chamber other is here an open field in fact many times we do lot of noise testing in free field where in we do not have any reflection it has to be no reflections. So you can pretty much understand the SPL at 1 meter away from the source meter actually $SWL - 10 \text{ decibel} = SPL$.

So SPL is something constant to a particular machine is a inherent quality of the machine which is fixed so at one meter of away from the sound pressure level from the source sound pressure level will be whatever the value you have a SWL in decibels we have to subtract it by 10 DB okay of course we have to keep in mind the SWL DB and SPL DB have different differences and then something else we find out.

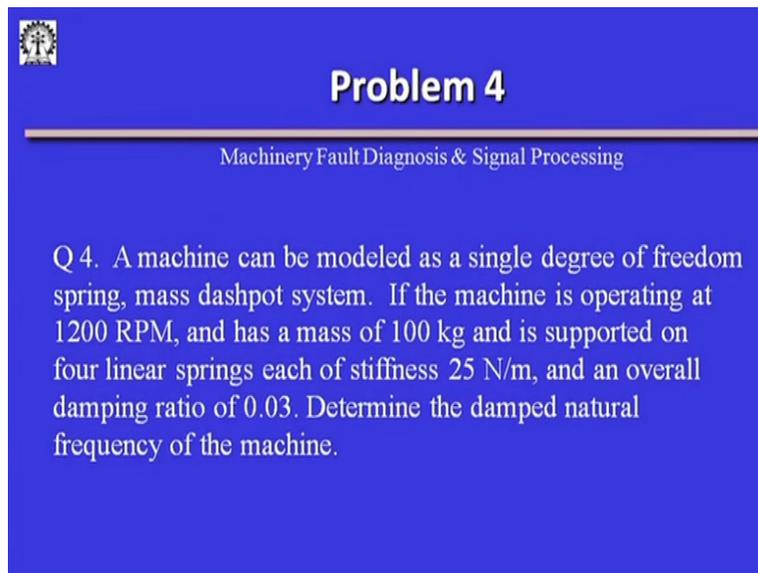
Say SWL is actually $10 \log_{10}$ of W by W reference where W reference 10^{-12} watts I have to keep this in mind. So in this problem the machinery was having 90 DB as SWL the SPL would be close to about 80 decibel in free field condition at the distance of 1 meter. So through this example I just introduced to few concept of room constant which is very important in machinery and (()) (26:43) to will be called to monitor the noise of machine which are having certain defects.

So we would be careful about the at what condition which we measure the sound pressure because at the environment changes this sound pressure level obviously change and like to saw it

depends on the directivity it depends on the room constant. So whenever we want to monitor the health of the machine by noise monitoring this is always recommended that we actually do the measurement in the same noise environment otherwise I will be known apple to apple orange to orange comparison unless we follow the simple rule.

And that is what through this example I just showed you how the room constant the distance on the source and the directivity does affect the sound pressure level which I have been measured okay.

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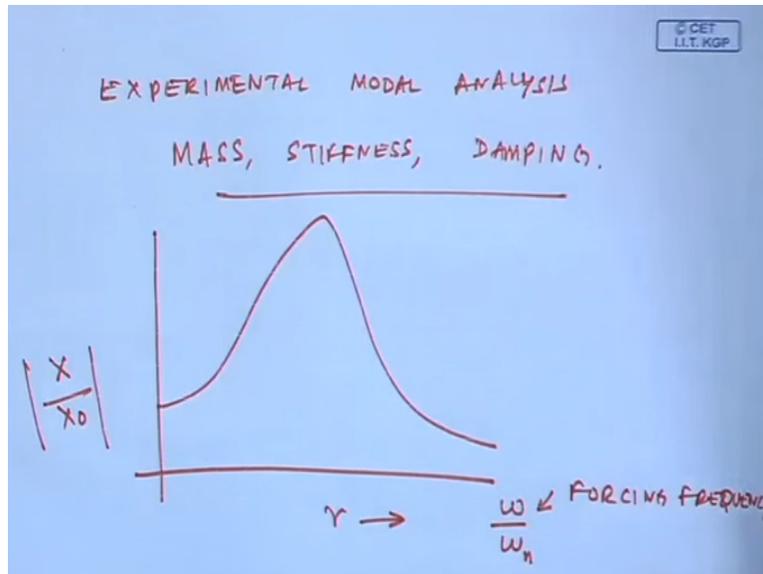
Problem 4

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Q 4. A machine can be modeled as a single degree of freedom spring, mass dashpot system. If the machine is operating at 1200 RPM, and has a mass of 100 kg and is supported on four linear springs each of stiffness 25 N/m, and an overall damping ratio of 0.03. Determine the damped natural frequency of the machine.

Now we will move to next problem this is again a simple problem on vibration that is question number four a machine can be modeled as a single degree of freedom spring mass dashpot system. If the machine is operated at 1200 RPM and has a mass of 100 KG and is supported on four linear springs each of stiffness 25 Newton per meter and an overall damping ratio of 0.03. Determine the damped natural frequency of the machine.

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See this is the machine mass supported on four strings am just showing to decline use am not showing the other two springs. So the affective springs there are four springs K effective is nothing but 4 times K and the value of K is hundred so this becomes 400 newton per meter and the natural frequency of this system $\omega_n = \sqrt{K/N}$ and that happens to be 400 and mass is also I just did a mistake here this will be actually case twenty five so this is 100.

So four times twenty five so this will be 100 divided by 100 so this will become 1 radial per second. So now the damped natural frequency of the system is given by this is expression so we are following the natural frequency is 1 radial per second damped natural frequency is $1 - 0.03$ square and this will be somewhat close to about 0.95 radial per second is a very simple example we were in just use the formula to find the natural frequency of the system and the damped natural frequency of the system.

Now many times what happens that of there were the fact that in this machine was running at 1200 RPM. So this machine is running at 1200 RPM the forcing frequency ω force is nothing but the frequency of forcing frequency is nothing but 1200 by 60 hertz 20 hertz and ω is $2\pi \times 20$ this is 40π radians many times just by estimating the mass of a machine and estimating the springs the stiffness of the springs.

We can get some idea as to some natural frequency of the machine particularly in many machinery condition monitoring situations the defects occur because unknowingly or knowingly

well out say unknowingly somebody is operating the machine at its resonant frequency or the designer did not take care of certain issues in mounting which would have been introduced so that the stiffness change the damping change and the natural frequency of the system happens to be close to the forcing frequency and then a condition of resonance occurs.

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The image shows handwritten mathematical formulas on a blue background. The first formula is $DMF = \frac{1}{\sqrt{(1-r^2)^2 + (2r)^2}}$. Below it is the definition of the damping ratio $r = \frac{\omega}{\omega_n}$. A horizontal line is drawn at the bottom of the image.

And because of resonance will have large motions and then there will be failure in the machine so a good way to estimate the natural frequency of a system is through experimental model analysis which we have discuss in which would have been discussed in these lectures but just by knowing the mass, stiffness and usually some gas on the damping.

We can estimate the natural frequency because vibration which is such a phenomenon on this three parameters and we cannot neglect it and the phenomena of vibration this actually very much frequency dependent this is the dynamic magnification factor X by X naught and R is nothing ω by ω N Where ω could be the forcing frequency and this is the natural frequency.

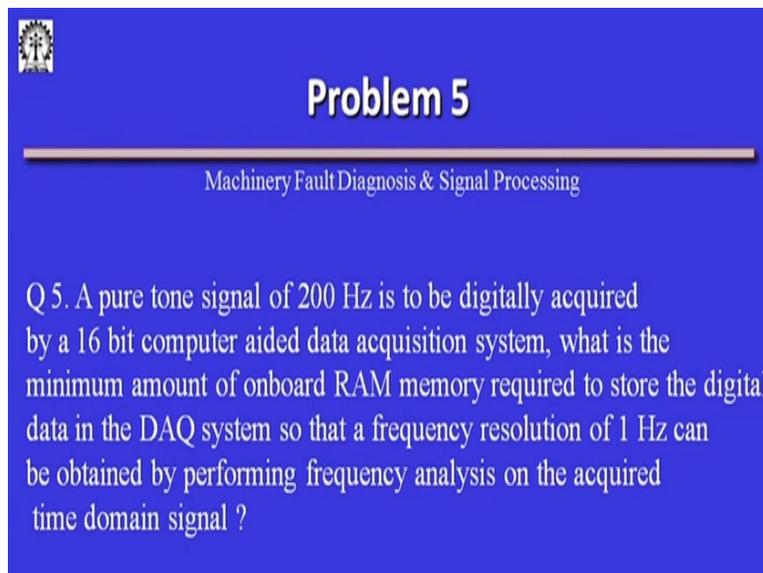
So the dynamic magnification factor is always given by 1 by $1 - R$ square whole square $+ 2$ Theta R square where theta is the damping ratio. In this case theta was given us 0.3 as you see the phenomena of damping is very much dependent on R and R is nothing but the forcing frequency ratio of forcing frequency and then natural frequency.

So these simple formula helps you to know the dynamic response of the machine knows helps you to estimate the damping natural frequency and we can also see how the machine is going to behave as a function of frequency by the plotting the dynamic frequency factors. So these are simple things which may needs to keep in mind while are doing machinery condition a monitoring or doing machinery fault detection.

So just to summarize we just sort two examples each from the classes of vibration and noise has to how simple equation gives us insight into handling few number beat in noise beat in vibration and these number can help us do a better estimation or better assessment of the machine health condition and these are very preliminary rudimentary problem which one needs to be absolutely sure about because later on in machinery condition monitoring we will be building on some of these very simple equations.

And if you remember on this will help you go long way and doing a actually do a machinery trouble shooting, Now we will come to two different other classes of problem which is this on computer aided data acquisition.

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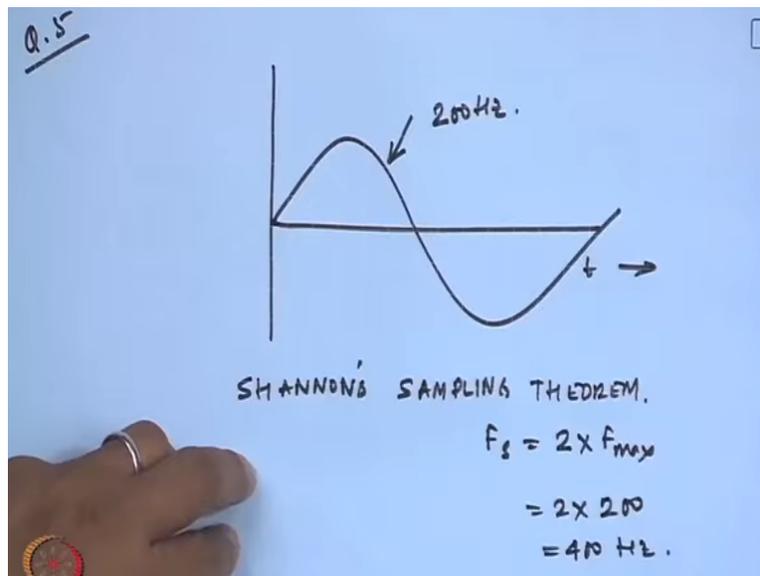
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So come to the fifth problem these a pure tone signal if 200 hertz is to be digitally acquired by a 16 bit computer aided data acquisition system. So what is the minimum amount of onboard Ram memory it is a Random access memory required to store the digital data in the digital data

acquisition system. So that the frequency resolution of 1 hertz can be obtained by performing frequency analysis on the acquired time domain signal.

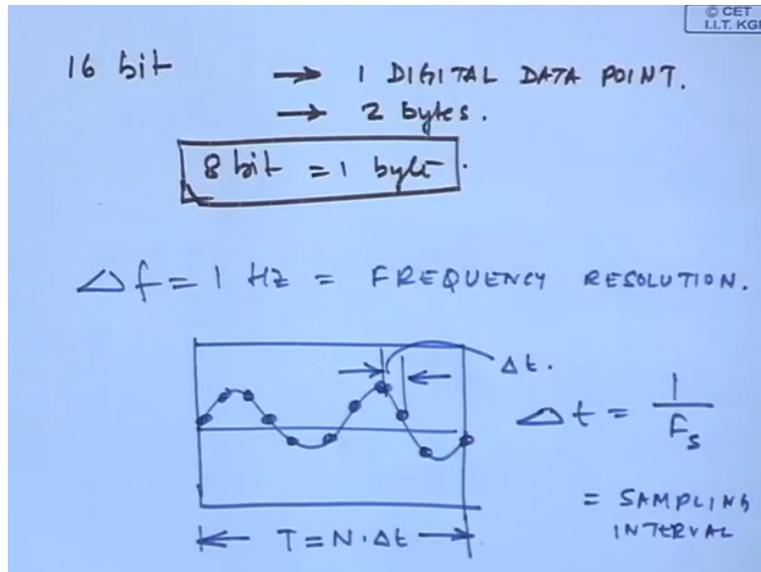
As you know this equation there are two three concept which are being in these problem we required two three concepts to solve this problem.

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For example I have a pure tone I mean a signal of a single frequency and this pure tone signals is of 200 hertz so by Shannon sampling theorem without worrying about the over sampling and all that my sampling frequency refers should be equal to at least twice of max. So in this case it is 400 hertz sampling frequency okay this is something I have to keep in mind.

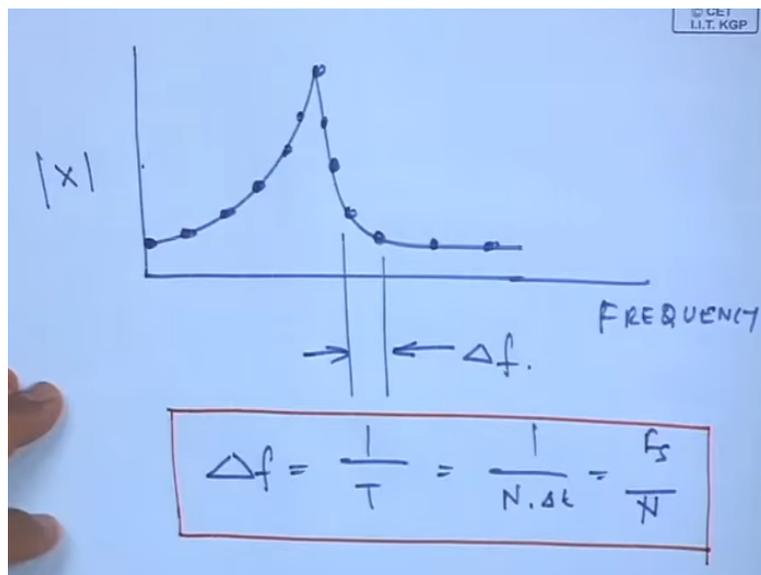
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And the next part of the system it is 16 bit computer that means one digital data point will be stored in 16 bit spaces so that it corresponds to 2 bytes because 8 bit = 1 byte okay. Now the question this signal pure tone signal which is sample will be processed digitally into its frequency domain so that the frequency domain resolution $\Delta F = 1$ hertz okay.

But we know there is a relationship between the ΔF and the log of time data which we have taken because this data which has been digitized (()) (38:19) digital data I take and data points is at this total time nothing but N times ΔT where ΔT is nothing by 1 of the one of the sampling frequency this is known as the sampling interval this value ΔT .

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So in the frequency resolution or in the frequency spectrum because this is a pure tone these values we come some digital value and then join them. So typical spacing between any two consecutive point is Δf some amplitude ok so there is a inverse relationship between the time and frequency and that is the most fundamental relationship this is a very important relationship.

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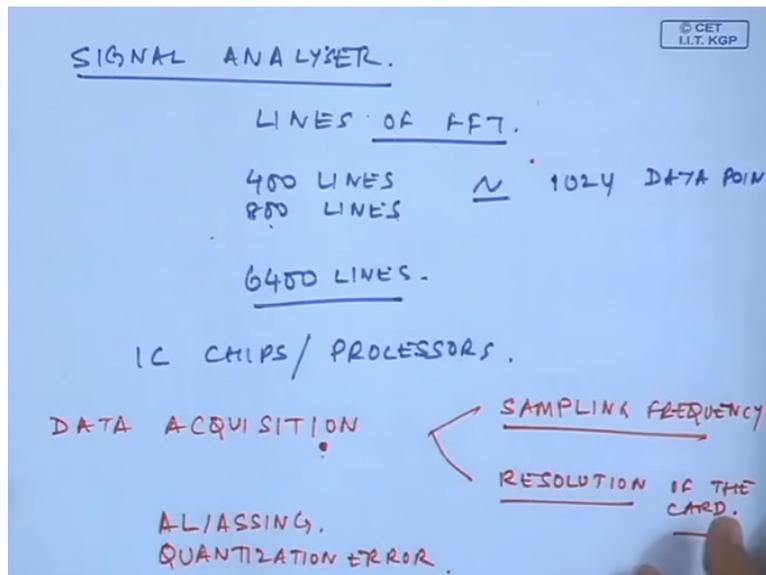
$\Delta f = \frac{F_s}{N}$; $\Delta f = 1 \text{ Hz}$
 $F_s = 400 \text{ kHz}$
 $N = \frac{F_s}{\Delta f} = \frac{400}{1} = 400 \text{ DATA POINTS.}$
 2 bytes
 TOTAL SPACE = $2 \times 400 = 800 \text{ BYTES.}$
 $2^{10} \approx 1024 \text{ BYTES} \approx 1 \text{ Kbytes.}$
 1 Kilo byte of RAM SPACE

So in this example we had the sampling frequency so Δf is nothing but F_s / N since $\Delta f = 1$ hertz F_s because of this sampling frequency requirement of (()) (40:25) this comes to be 4 hertz or N is nothing but $F_s / \Delta f$ that is $400 / 1$ that is $= 400$ data points. Here of course we have an assumption at which is normally not true that we have just ineffective on one block of data because no information are given by we are going to take more blocks of data and so on.

So just assuming that one block of data is being used we have to take 400 data points so each data points requires 2 bytes of space so total space total space = two times 400 = 800 bytes but as you know memories available in the power some power so the closest highest power will be actually 2 the power 12 it will be then 2 to the power 10 that will be 1024 bytes or which we known as 1 kilobyte.

So to analyze at the signal I will require 1 kilobyte of RAM space many times given as 800 bytes and I would normally prefer that a amps actually 1 kilobyte and not 1 not 800 byte but 1 kilo byte because you will not get 800 bytes of memory tool in the market. So this is how you can decide on the number of RAM requirement and you would have known the effective analysis.

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If we buy this signal analyzer they are sometimes they are specified by the lines of a FFT. So this is 400 lines, 800 lines this actually correspond to $400 / 2.56$ this corresponds to about 2.56 about 1024 data points. So points I mean more data points its more memories and you would have seen when you but of course now a days you know we have system with 6400 lines available in the present day systems.

So many times some of this FFT analyzers are actually on chips or on IC chips or processors because this kind of example helps to understand what is the numerical requirement in terms of the memories storage for data acquisition. As you know in acquisition system two most important parameters are the one is the sampling frequency and the resolution of the card in terms of whether it is 16 bit 12 bits etc.,

So these are very important I mean if you have a mistake in the sampling frequency have around sampling frequency you have the problem of signal aliasing and if you have do not have a get resolution and you have the quantization error. So looking back to the previous column so this if you want to have 1 hertz resolution we have to have more spaces.

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0.1 Hz.

$$N = \frac{F_s}{\Delta f} \approx \frac{400}{0.1} \approx 4000$$

Q.6

RESOLUTION.

$$\Delta V = \frac{\text{RANGE}}{2^n \leftarrow \text{bits}} = \frac{10}{2^{12}}$$

$$= 2.44 \times 10^{-3} \text{V}$$

$$= 2.44 \text{ mV.}$$

Next question is suppose I wanted to have .1 hertz .1 hertz you will see how you are memory a size will increase $N = F_s / \Delta f$ so in this case it will $400 / 0.1$. So this will increase by a about 4000 okay if you want better is resolution we have to have more a memory space so this is what is going to change is not it. So we have to keep that in mind so better resolution is more memories this and this are there related.

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Problem 6

Machinery Fault Diagnosis & Signal Processing

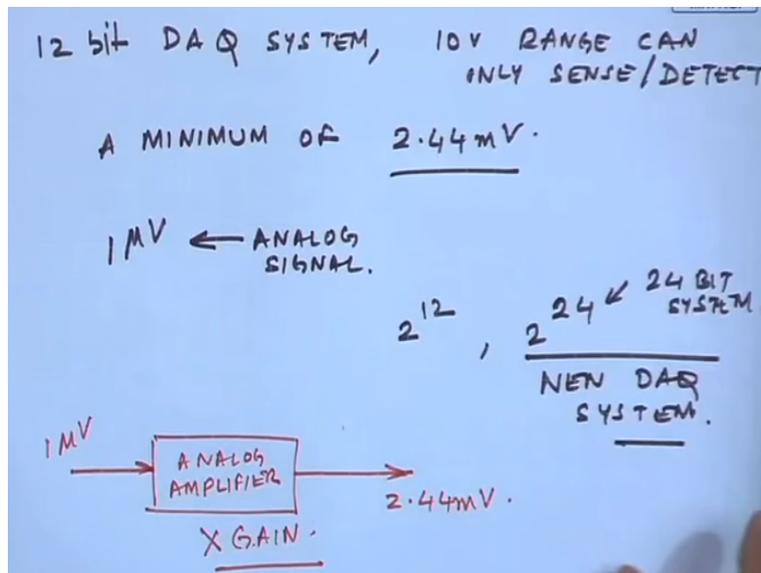
Q6. What is the gain in a signal conditioning analog amplifier required so that a 12 bit data acquisition system with a maximum input analog voltage range of 10 V can acquire a signal of 1 micro-volt amplitude ?

Now we will come to the next and the last problem on this topic that is what is the gain in a signal conditioning analog amplifier required so that a 12 bit data acquisition system with a maximum input analog voltage range of 10 volt can acquire a signal of 1 micro volt amplitude.

So this comes to problem because I have a machine by put a sensor and it gives me some analog voltage.

Now I have A to D system and then which gives me digital data which goes into a computer now obviously this A to D system as an minimum voltage in sense that is known as the resolution and this resolution is given by what is known as the maximum voltage is sense that the range by 2 to the power bit size where this is the bit size. So that is the lowest voltage it can sense so in this case if this range is 10 volts I have the bits as a 12 so 2 to the power 12 and this corresponds to about 2.44 and 10 to the power -3 volte that is 2.44 millivolt.

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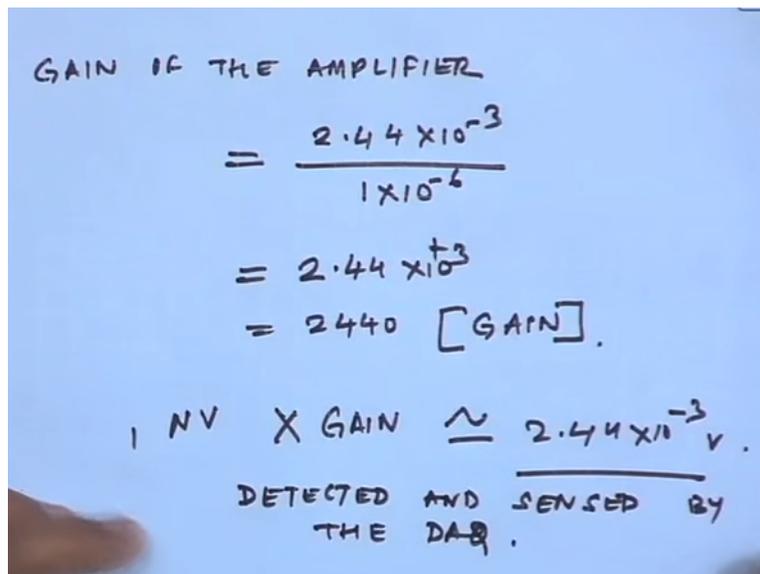
So in other words a 12 bit DAQ system with 10 volt range can only sense or detect or minimum of 2.44 mini volt so I have big problem in hand my (()) (48:27) asked to acquire 1 microvolt that is my actual analog system a very low temperature thermo couple signal will give you this come from value and suppose if you want to deduct such a system and you want to (()) (48:48) remotely or through a computer aided deduct the problem is your 12 bit system good enough some set.

Obviously you see because two 1 microvolt is much than 2.22 millivolt it cannot sense the couple of waves can do it. I can increase the bit size instead of 2 to the power 12 I can make it 2 to the power 24 in a 24 bit system that calls of course for a new hardware, new DAQ system.

Sometime this may not be possible but now a day 24 DAQ systems are available and so to have good enough amplitude resolution I have to always increase the bit size.

If that is now possible in between what I can that I can have a very nice analog amplifier with high signal to noise ratios to that I give my 1 microvolt and I make it at least 2.44 millivolt or more okay. So I have to amplify that I have to multiply with some gain so this analog amplifier the question is what is this gain so that or 12 volt system is going to register 1 microvolt.

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A handwritten calculation on a blue background showing the gain of an amplifier. The text is written in black ink. It starts with 'GAIN OF THE AMPLIFIER' followed by a fraction: $\frac{2.44 \times 10^{-3}}{1 \times 10^{-6}}$. This is simplified to 2.44×10^3 and then to 2440 [GAIN]. Below this, it shows $1 \text{ NV} \times \text{GAIN} \approx 2.44 \times 10^{-3} \text{ V}$, with a note 'DETECTED AND SENSED BY THE DAQ.'

Obviously so the gain of the amplifier would be nothing gain of the amplifier is nothing is 2.44 is 10 to power -3 volt divide by 1 microvolt 10 to power -6. So this is = 2.44 into 10 power 3 is 2440 so this is the gain that means if I have 1 microvolt signal I multiply with the with this value of gain I will get 2.44 into 10 to power -3 volt which can be detected and sensed divided by the DAQ okay.

So we have to keep this mind while we have to select on a certain data acquisition system okay so a in summary on this class we just looked into simple problems typical example problems a couple of them from noise couple of them from vibration and couple of them from DAQ system as to making us familiar on some of the important concepts on data acquisition similarly signally quantization error conversion of SPL to Pascal to decibels.

How do you relate compressor level to an estimate sound pressure level from sound power level what is the meaning of dynamic magnification factor how do you estimate natural frequency damp natural frequency. So with this kind of a understanding we are ready go to a field to do some actually trouble shooting and in the subsequent classes actually will be discussing about how a machine monitoring is done through vibration how for volt is deducted by vibration monitoring thank you.