**NOISE CONTROL IN MECHANICAL SYSTEMS**

**Prof. Sneha Singh**

**Department of Mechanical and Industrial Engineering**

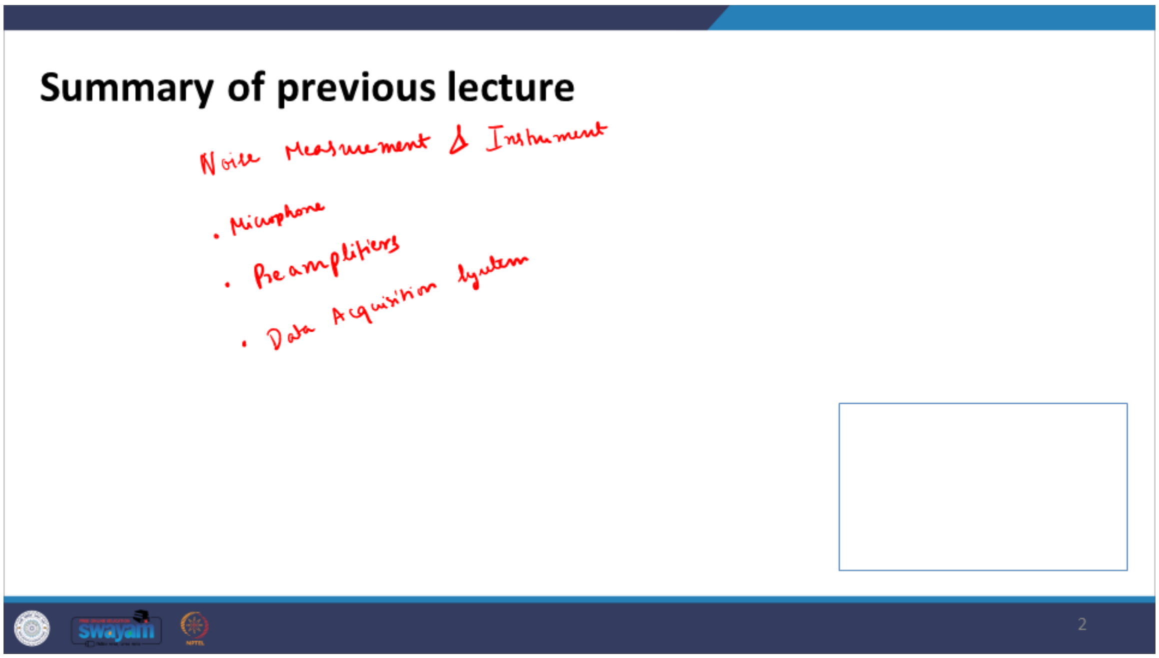
**IIT Roorkee**

**Week: 05**

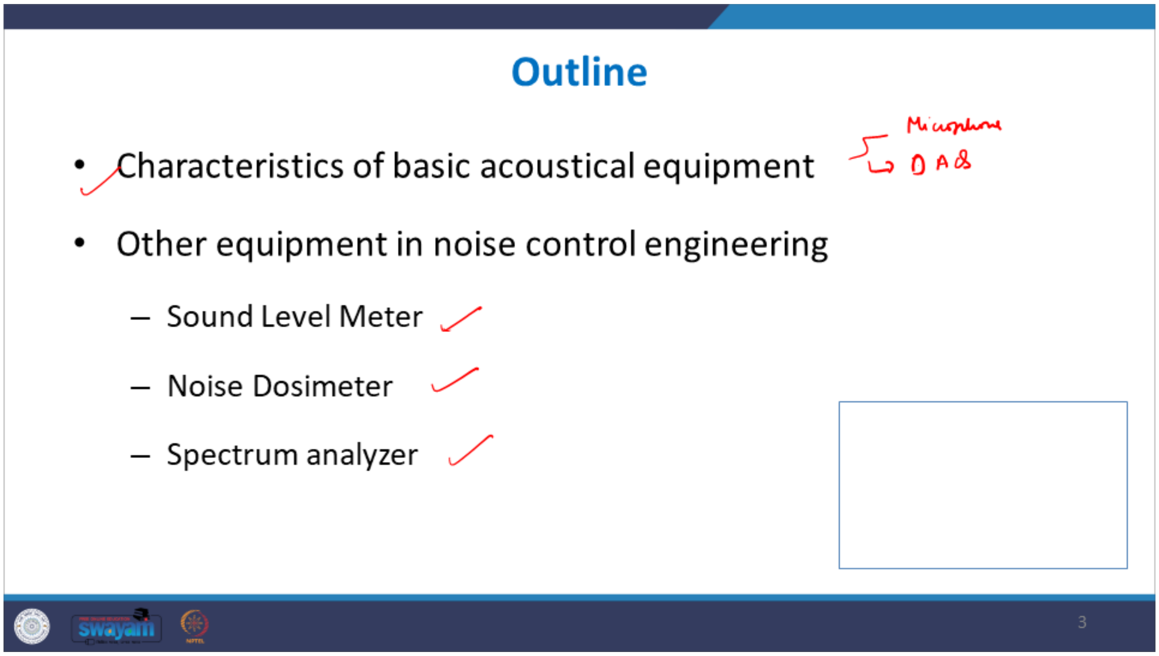
**Lecture: 25**

**Lecture 25: Equipment for noise control engineering: 2**


Hello and welcome to lecture number 25 in 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 in this lecture, we have been discussing, and we have started our new module on Noise Measurement and instrumentation. In the last lecture, we studied about microphones, the basic working principle behind them, and the types of microphones. Then we studied about preamplifiers, why preamplifiers are needed in any kind of acoustical measurement, and the types of preamplifiers as well. And then we closed the lecture with a discussion about the data acquisition system, which is an overall system that facilitates capturing acoustic signals, their processing, and storage.

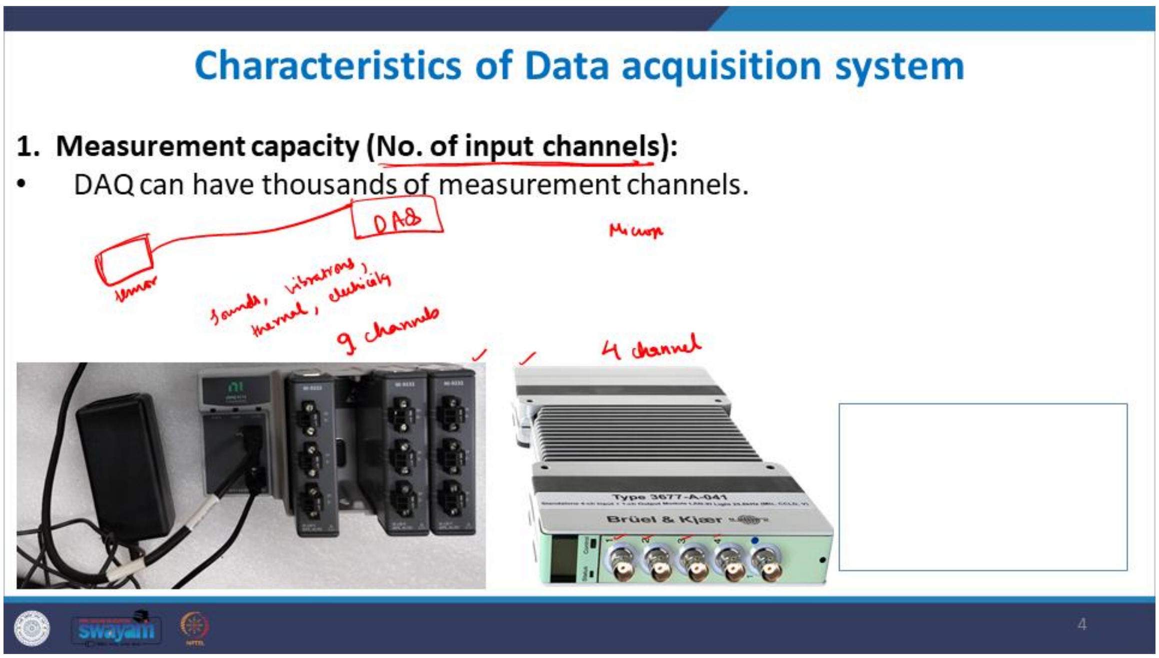



So, in this lecture, we will see some of the characteristics of these basic acoustical equipment, such as the microphone and the data acquisition system. We will have a look at their characteristics. And then some other advanced equipment that is being used nowadays in noise control engineering, such as sound level meters, noise dosimeters, and spectrum analyzers.




So, the data acquisition system is a major part of acquiring sound signals. In fact, the data acquisition system can be configured to acquire any form of data. It could be thermal signals, electrical signals, acoustic signals, and so on. So, some of the key characteristics of these data acquisition systems are, first of all, the measurement capacity or simply the number of input channels that the data acquisition system can support. So, here we have these two data acquisition systems that are available in our lab at IIT Roorkee. Okay, this is one data acquisition system. As you can see, it has a total of 9 ports. So, it has 9 channels. So, this means that simultaneously we can record data from 9 different sensors. Okay, so the sensor over here is a microphone in the case of acoustical measurements. So, the sensor is connected via a cable to the data acquisition device. So, 9 separate sensors can be simultaneously installed, and the data can be acquired. You need a greater number of channels in case of recording, such as, for example, when you are doing a recording in a complex machinery environment, such as when you are doing a recording in a railway vehicle or a big naval ship, etc. So, where you need to record a lot of things, maybe you need to record sounds, the sounds from various locations, the vibrations of various components, the thermal gradients, the electricity that is developed from various locations and components. So, a lot of data can be collected and simultaneously acquired if you have a greater number of input channels. In the same way here, as you can see the numbers 1, 2, 3, and 4. So, this has 4 channels.

This is a 4-channel DAQ, and this is a 9-channel DAQ. So, obviously, the measurement capacity increases with an increase in the number of channels, and hence the cost, the processor, and various other requirements also increase. So, in the same way, I was talking about how, if you have multiple channels, you can record any kind of physical phenomenon you want to, whether it's not necessarily an acoustic pressure that you want to measure. You can also measure, depending upon what kind of sensor you are connecting. If you are connecting a microphone, then from the microphone, you are getting over here.




I'll just say, suppose you have connected the microphone sensor, then you are able to get the acoustic pressure data. If, instead of that, you had accelerometers, which are devices that measure the vibration of a surface. So, if this was your sensor that was attached at the other end of a channel, then you would get your vibration data. In the same way, you had some, let us say, some kind of, you know, thermocouple or something like that, you would get, you know, the thermal gradients. So, basically, you are just limited by your imagination. Data acquisition systems ensure that you know you are able to capture some phenomenon. So, depending upon the sensor that is used at the end of this data acquisition device, depending on that, you can capture different types of signals and simultaneously, you know, sort of analyze them and process to see how they are varying. So, let us say, for example, in any kind of machinery, you want to see, you know, how much the vibration is impacting the overall sound radiation. You can have, you know, a sensor which is an accelerometer attached to a data acquisition device, and the other channel could be a microphone. You can place the microphone at the point of interest, which could be the receiver location, and you can place the accelerometer attached on a component which is vibrating on the machinery. Then, you try to correlate the vibration data with the acoustic pressures that you are getting. You can see whether their frequencies are matching, you know, the levels are matching, and in this form, you can correlate. In the same way, suppose you think that in some way, you know, the pressure drop in a machinery is doing that. So, you can measure the pressure drop and correlate it with the sound data. In the same way, other things also you can do. So, it allows you to, you know, sort of get, sorry, the various kind of varieties, you know. So, you can record various types of things such as acoustic pressure, thermal gradients, and voltage, depending on what is the sensor that you are attaching The other characteristic, I think, is the number of output channels. Over here, the output channels is 0, whereas, over here, the output channel is 1, this being the output channel. So, sometimes, you know, in the data acquisition device, you can also have an output channel. What it means is that it can be used for function signal generation. So, let us say you want to—it is usually used in the field of room acoustics. So, let us say you want to see the reverberation in the room. So, you can have a setup where, you know, you have a data acquisition system which has got an output channel as well as some input channels. So, in the output channel, you can have some signal generator that is creating a sine wave or, you know, broadband noise. So, whatever is your kind of signal of interest, a function signal generator is creating that noise, and through the output channel, you are blasting that noise into the room in an omnidirectional way. And after that, from the output channel, a typical signal that you want to create has been created and blasted into the room. From the input channel, then you are measuring the sound pressure level at the point of interest, and you see how quickly it decays to get the reverberation in the room.

## Characteristics of Data acquisition system

**2. No. of Output channels:**

- DAQ can have a few output channels for function signal generation.

Microphone sensor → Acoustic pressure
Accelerometer sensor → Vibration
Thermo couple → Thermal gradients
$\boxed{\text{sensor}} \rightarrow \boxed{\text{DAQ}}$
Output channels = 0

Output channel = 1




So, in these kinds of applications, you might also need an output channel, which can also be provided and I have already discussed, you know, the measurement variety. Different physical phenomena can be simultaneously measured one at a time or simultaneously.

## Characteristics of Data acquisition system

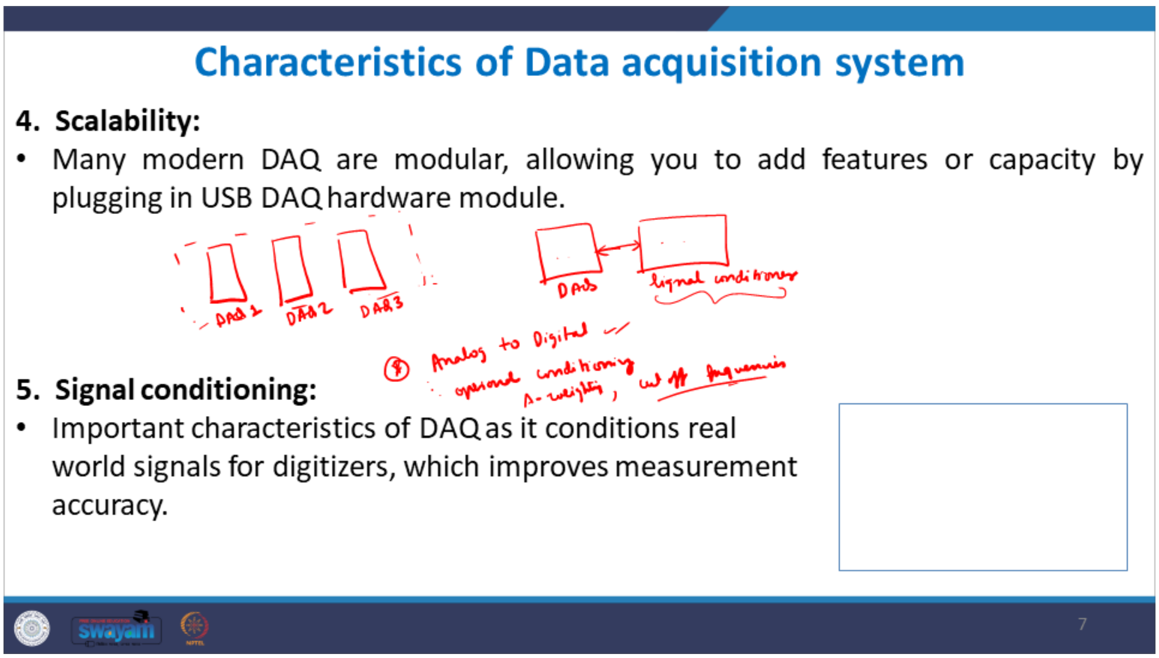
**3. Measurement variety:**

- DAQ can measure different physical phenomena, one at a time or simultaneously, such as acoustic pressure, thermal gradients, voltage, etc.

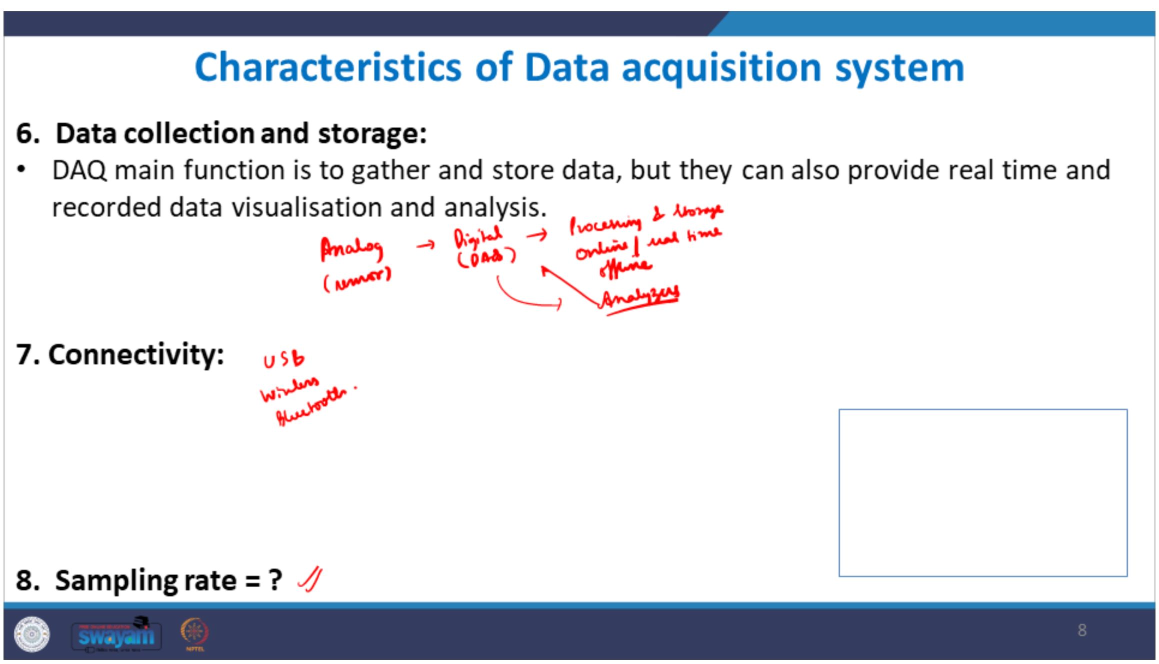
4 channel = 12 channel
chassis
Modular ✓
$\boxed{\text{sensor}}$ = ?

Modular ✗

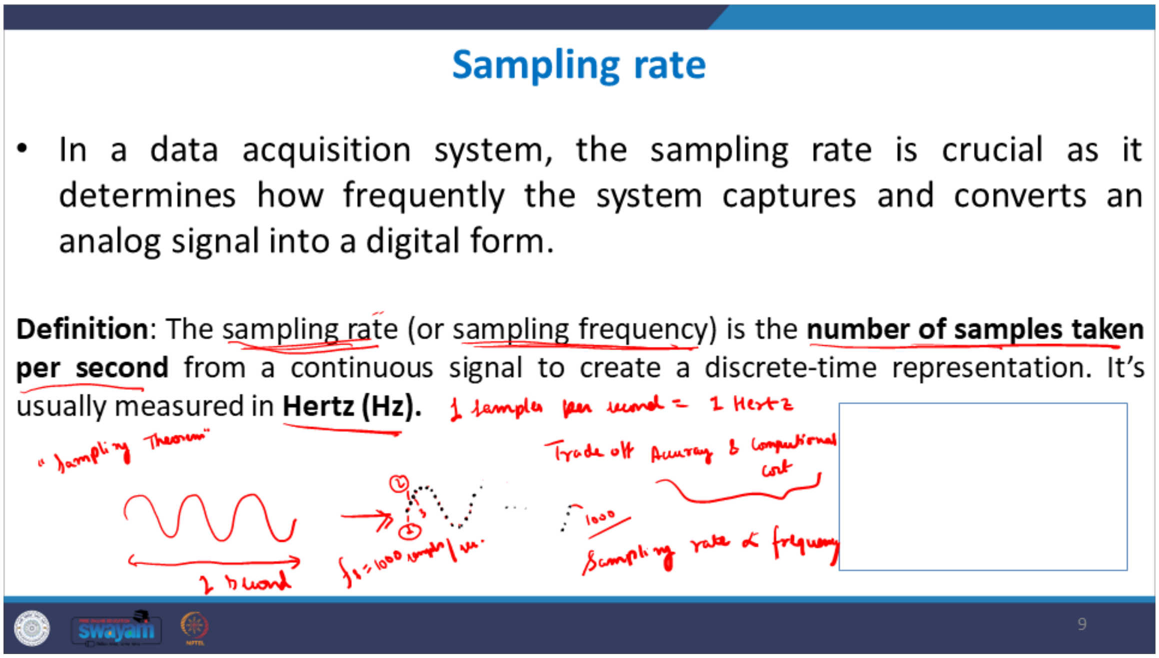
Then, you know, scalability is yet another characteristic of a data acquisition system. Many modern DAQs are made to be modular. And what it means is that it allows you to add features or capacity by plugging in the USB DAQ hardware module. So, you may have purchased one DAQ, and suddenly you think that, you know, I have purchased a DAQ, but I would like to upgrade it. Maybe I would like to increase the number of channels further. So, initially, you didn't have the kind of money or the funds to buy it, so maybe you purchased a three or four-channel DAQ. Later, you decided you got some more money, and you would like to upgrade and increase the number of channels. Then, with some of the modular options, there are some DAQs which are modular in nature, so it is possible you can add different modules, combine them to increase the number of channels, and each of these modules and each of the additions works independently of each other, but they can be combined to work together as a single hardware. In the same way, you can have USB connections through which you can connect your DAQ to some other functionality. For example, you might think that my DAQ is working fine; it is acquiring the signals, but I would like to add some additional filters to the signals that are being acquired. You can plug in through the USB connectivity to another hardware, which could be a signal conditioner, and you can get the functionality. So, you can either have multiple DAQs combined together. Suppose you want to increase the number of channels. So, your system can be upgraded like that. So, you do not have to. So, this ensures that you do not have to discard your old DAQ. It is not going to waste. You simply add them together or stack them together, and they function simultaneously. In the same way, you can have a DAQ which is being added with some kind of signal conditioner, additional or some frequency weighting functions, etc. So, various kinds of other functionalities you can add in. So, here, let us say for example in this one. So, this is not modular. So, this is not modular, whereas this is a modular example. So, what has happened is that I had purchased individually these three DAQs. So, here it is a three-channel DAQ purchased individually with a chassis. So, this is an external chassis over which each of these DAQs can be plugged in to work together. So, as you can notice here, right now it is a nine-channel DAQ, and it still has a slot left. So, I have the option that if I have additional funds, I can upgrade it from a nine-channel to a 12-channel DAQ by adding in a similar, you know, DAQ here like that. So, this is an example of modular functionality and the scalability of a data acquisition system Signal conditioning: some DAQs, you can either connect the signal conditioner hardware outside. And in many of these DAQs, sometimes the signal conditioning is inbuilt within the DAQ itself, which improves the signal quality. So, essentially, you know, the most essential work of a DAQ, the first kind of conditioning that it does, is it converts the analog data to digital. So, this every DAQ performs. And then there are some optional conditioning, such as, you know, A-weighting, then some kind of, you know, cut-off frequencies to remove the unwanted frequencies. Let us say whatever you only take the data of your frequency range of interest, like that. So, these kinds of options are also available in some modern DAQs.



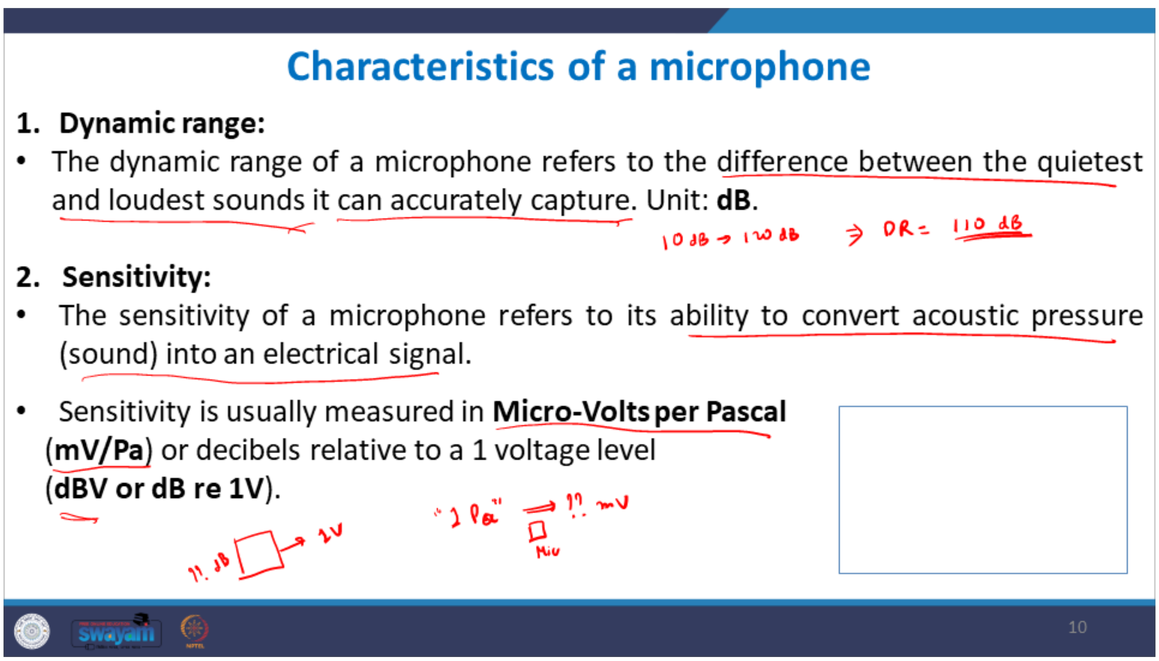

And obviously, an important characteristic of a data acquisition system is that it should be collecting the data and storing it as well. So, it is not just about acquiring the data. So, whatever is the analog data that you are receiving from the sensors, that analog data is being converted into digital data inside the data acquisition system because you cannot store the analog information for further processing. You need a finite number of points. So, the analog data from the sensor that is being received is first converted into digital data. So, first of all, analog from the sensor is converted into digital data. within the DAQ. Now, this data can be either stored. The DAQ can have its own internal memory, or it can be directly connected to a computer, and the data can be stored and visualized in real time, as well as processed. So, whatever processing and storage is there, it can be either in the online mode or the real-time mode. Or it can be offline, where the data is stored and processed when you are away from the factory setup. So, usually, the offline mode is more handy in case you are working in some kind of harsh environment and you don't want to wait too much. You just want to collect the data and go outside. And later, in your comfortable office environment, you want to process the data and see how it looks. What kind of spectrum the data has, and various other processing that you would like to do, like one-third octave and so on? You can do it in the offline mode where the DAQ data is connected to the computer and is stored for offline processing. So, it goes without saying that the DAQs will have some kind of analyzers which are configured with this DAQ to further analyze and process the data. In the same way, various kinds of connectivity options are also available, like USB connectivity, wireless connectivity, Bluetooth, and so on. So, not necessarily every DAQ has to be wired. Wireless technology in noise control engineering is now gaining a boost so that we can remotely monitor various things without the need for physical cables and various kinds of setups. Then, the sampling rate is one of the chief characteristics of a data acquisition system. What is the sampling rate?



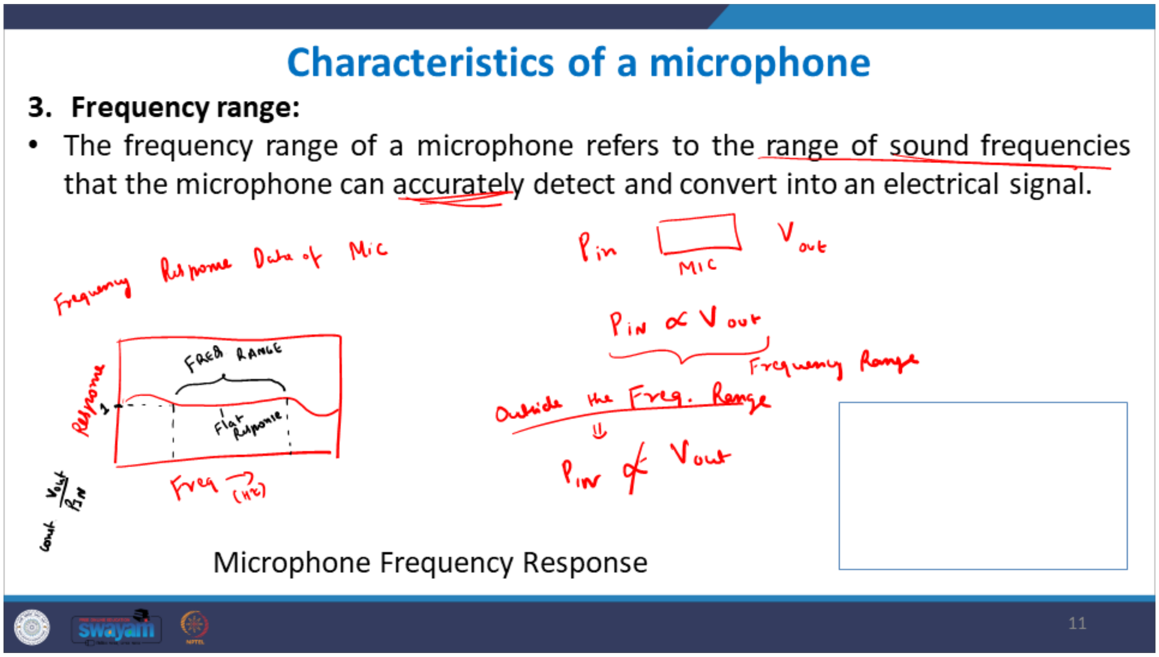
So, in a data acquisition system, the sampling rate is crucial because it determines how frequently the system captures and converts an analog signal into a digital form. So, as already stated, you know the sensors will be capturing the data continuously in an analog form, but it is when the processing is done, when the data is acquired by the DAQ, that this data is converted into a finite number of points or discrete points, and it becomes digital data. So, what is the rate at which this digitization is being done, or what is the rate at which an analog data is being converted into the digital form? It gives you the sampling rate of the data acquisition system. So, how is it defined? The sampling rate, sometimes also called the sampling frequency, because it is the rate. So, the sampling rate or sampling frequency is the number of samples taken per second from a continuous signal to create a discrete-time representation. It is usually measured in Hertz; sometimes it can also be referred to in samples per second. So, one sample per second becomes one Hertz. Okay, so it is the rate at which the samples are being taken. So, let's say, for example, you have some analog signal like this, and suppose this is a one-second data. And now you are storing, acquiring, and storing it in the digital form through the data acquisition system. And let us say your sampling rate is something like 1000, which means 1000 Hertz or 1000 samples per second. So, which means this one-second data could be represented as, you know, 1000 points which are equally spaced like this. and so on. So, if this is point 1, point 2, let me write it this way. So, these are the points over which the data is being collected, and finally, we get to the end of it. So, if this is point 1, 2, this is point 3, and so on. So, this becomes your So, if it is 1000 samples per second, then obviously this becomes 1000. So, 1000 points are collected per second. So, obviously, you would see that the higher the sampling rate, the more accurate the representation will be, but then the greater the computational and processing requirements. So, it is always a trade-off, you know. There is always a trade-off between, you know, accuracy and computational cost. So, you can always decide that for this kind of measurement, I am happy with just having fewer points per second. And for this kind of data, I would need more points per second. In general, if you're recording a high-frequency signal, you would need more points per second to capture that high-frequency signal because it's a signal whose frequency is what? It's the number of cycles per second. So, the signal is varying at a high frequency Sound would be varying much faster than a low-frequency sound per second, and hence, to capture these very rapid variations, you would need more points; otherwise, you would not be able to accurately capture it, and hence, the sampling rate would essentially vary with the frequency of your interest, whatever the frequency of the data. And more on this, we will study in the next lecture, where we see what the sampling theorem is and how exactly we decide what the sampling rate should be in a DAQ for what kind of sound.



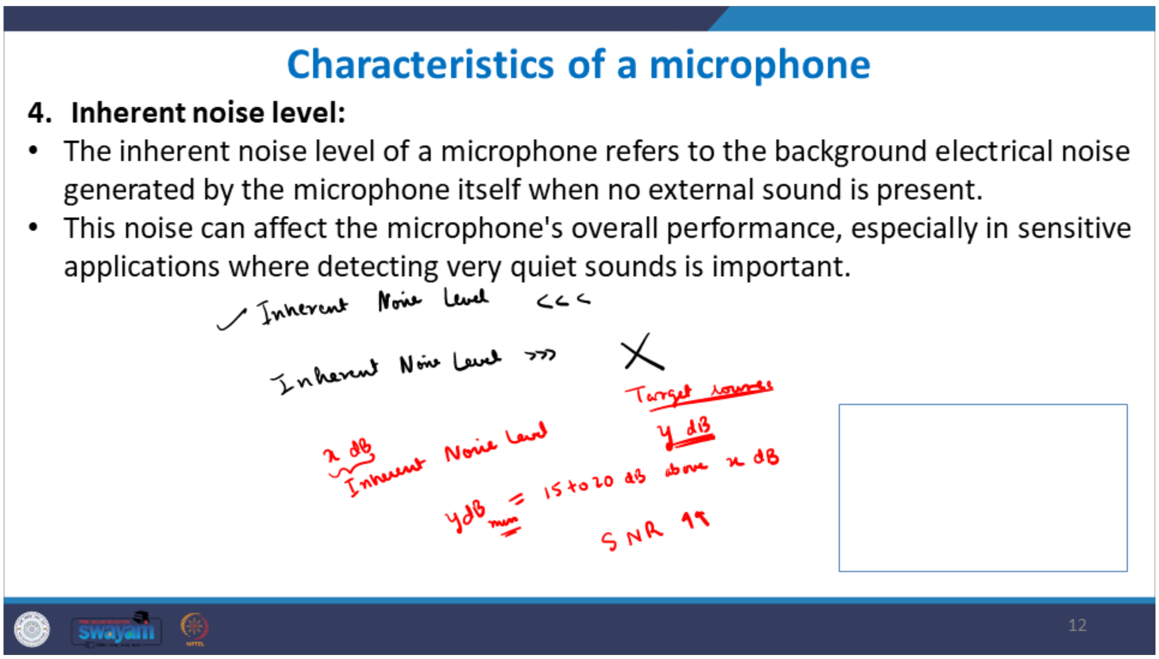

Okay, now let us see some of the characteristics of a microphone. So, you have dynamic range. It means that it is the difference between the quietest and the loudest sounds that it can actually capture. Which is measured in decibels. So, let us say the dynamic range, it is very straightforward. So, let us say the range within which a microphone can capture the sound is from 0 dB or let us say from 10 dB to let us say 120 dB. This is the range over which it can capture the sounds. Then obviously, the dynamic range becomes 110 decibels. Okay, it gives the range over which the microphone works properly because, see, microphones also have a capability, and for very loud or very low noise, they might not give the same kind of response, and their accuracy may drop down, or they may also suffer some physical damage. Then, sensitivity, this is yet another very important characteristic of a microphone. It refers to how sensitive, you know, how sensitively, you know, it is converting an acoustic pressure into the electrical signal or the ability to convert, what is its ability to convert the acoustic pressure into an electrical signal. It is usually measured in microvolts per Pascal, which means that per Pascal, which means that if suppose 1 Pascal of acoustic pressure it is measuring, then how many microvolts. it is generating to correspond to that 1 Pascal of variation because essentially, what does a microphone do? They convert the acoustic pressure into an electric signal. So, sensitivity is at what rate. So, per Pascal, how many voltages can they generate? Okay, sometimes the unit also used is, you know, dBV instead of microvolts per Pascal. So, I have already explained what is microvolts per Pascal. It means that if 1 Pascal of acoustic pressure was incident on the microphone, then how many microvolts it will create. In the same way, what is dBV? It simply means that Okay, so it simply here means that microvolts per pascal, and this is dBV, which means how many decibels of sound can create a difference of one voltage of pressure from the microphone. So, these are the two ways in which you can find out the sensitivity.



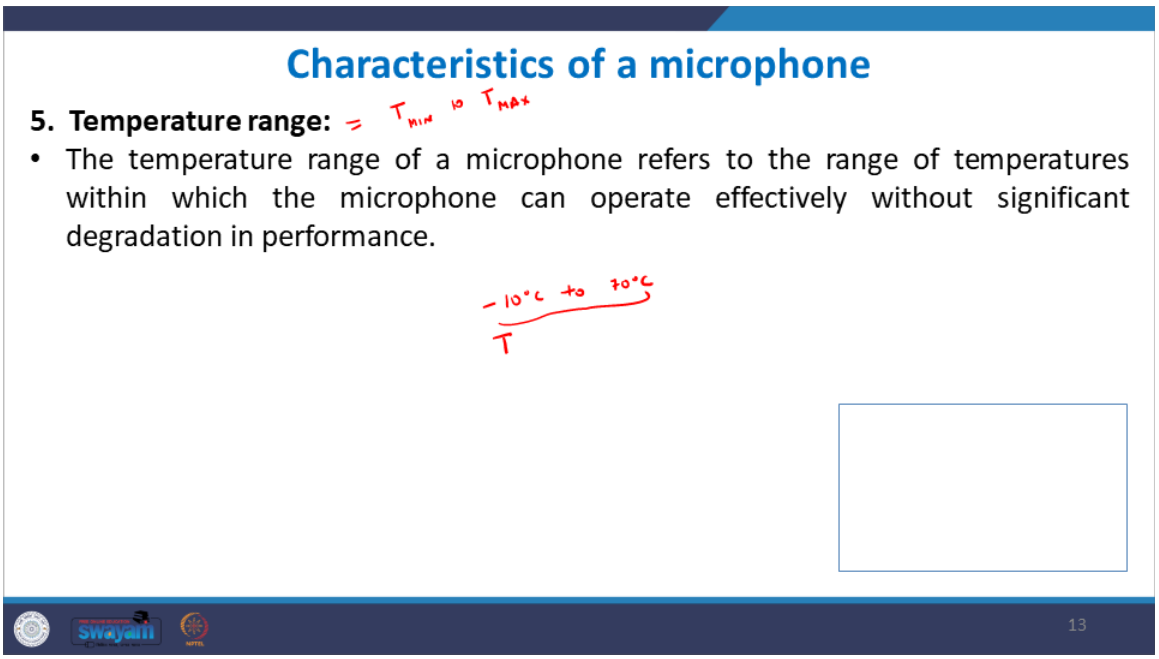

Frequency range usually, you know, microphones, the way they are made, they may be sensitive. They may have their own internal frequencies, their own natural frequencies, and vibration responses. Then, based on that, they may react to different frequencies in a different way because they themselves are some kind of mechanical structure at the end of the day. So, the range of sound frequencies within which it can accurately detect, or the range of sound frequencies where it is working accurately and very accurately converting it into an electrical signal, this becomes the frequency range of a microphone. So, let us say, you know, usually whenever a manufacturer is supplying you a microphone, they usually also provide the frequency response data. of the microphone. What is this frequency response data? It is something like this. You have the frequency scale in Hertz, let us say, and you have the response. So, a microphone should be able to give a flat response in the range of interest, which means that whatever is the input, the output is directly proportional to the input. So, whatever was the $P_{in}$ that is given to this mic, and some $V_{out}$ is coming out, right from input pressure to output voltage. So, there should be a one-to-one correspondence between the two. Only then the microphone is because we need to have a one-to-one correspondence. Whatever voltage is there; it should be directly proportional to the acoustic pressure. But this is only available in the frequency range of the microphone. Outside the frequency range, the $P_{in}$ may not be directly proportional to the $V_{output}$. So, the frequency response can look like this. Let us say if a microphone has this kind of frequency response, then you see where it is flat. So, what does response mean? It can simply be some kind of $V_{out}$ by $P_{in}$ and some kind of constant so that it is normalized to 1. This is the 1 value. So, some constant multiplied by $V_{out}$ by $P_{in}$, and this is normalized to 1 value. So, wherever this response is flat, that becomes our frequency range. Because here we are getting, you know, the $Pi_{nput}$ is directly proportional to the $V_{output}$; this becomes our frequency range. So, you know, the higher the frequency range, obviously, that kind of frequencies you can capture. If, you know, suppose the frequency range is very short, then you may not be able to capture, you know, the sound signals which have, you know, the frequencies outside the range. So essentially, what we see is a flat response. We should get a flat response curve that corresponds to a frequency range.



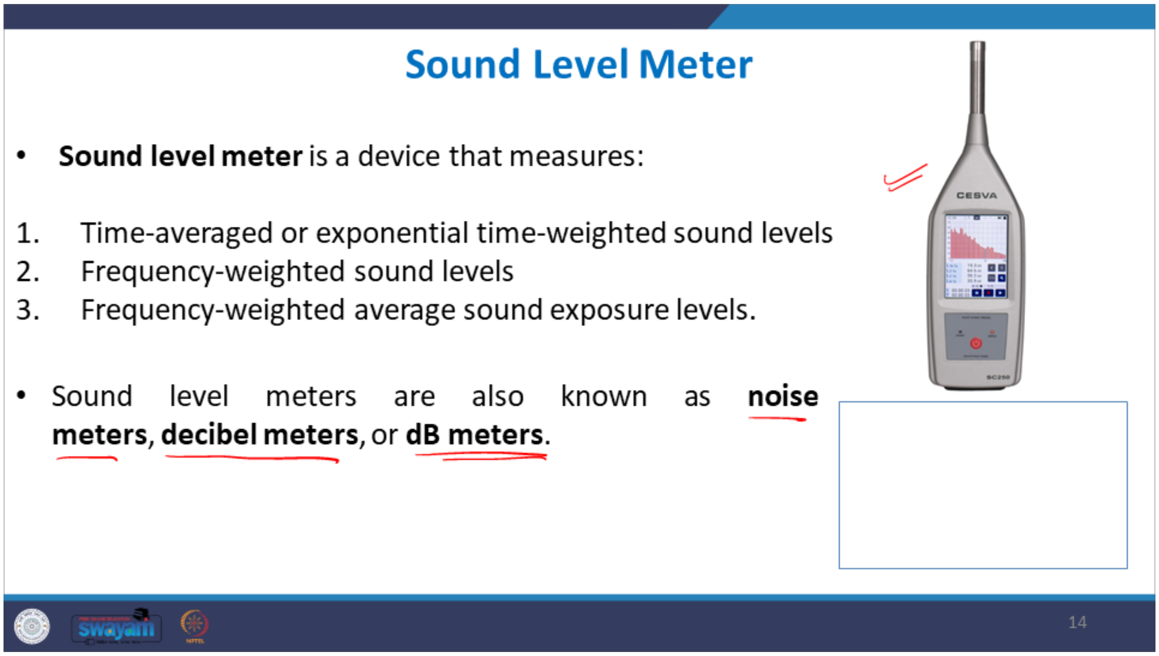

Then, the inherent noise level. So, microphones are obviously not completely devoid of any noise. So, any kind of measurement you do, any kind of digital signal processing, any kind of signal you are measuring, there is bound to be some amount of noise. It could be due to the measurement setup itself or due to the sensors. So, sometimes within the microphone, there could be, you know, some inherent noise level. There are various causes for this, such as, for example, the internal resistance of the material of the microphone. You know, when the pressure is being converted to electrical signals, the flow of the ions can generate some extra current. So that could be the reason, and there could be some thermal ionization that can happen. So, and then the resonance of the various elements within the microphone. So, there are various reasons. We still don't know what the various reasons for this are. But because of this, what happens is that there is some inherent noise level due to the buildup of the microphone itself. You cannot avoid it. So, the inherent noise level should be very low. Low as possible. If the inherent noise level increases, obviously that means that you cannot record a heavy sound. If it increases, this is not a desired scenario. Okay, so let us say, for example, typically, you know, during our measurements, in most of my experience, what I have found is that suppose the inherent noise level of the microphone is, let's say, some x dB. This is your inherent noise level. Noise level, then, and you want your target source alone to have, let us say, some y dB. Then you can accurately capture this y decibels source if, suppose, it is minimum, you know, 15 to 20 dB above the inherent noise level. 15 to 20 dB Above the inherent noise level. Okay, so obviously, suppose a microphone has an inherent noise level of 20 dB, and you have a very quiet 20 dB sound, you won't be able to accurately capture it because the noise and the signal are the same. You need to have a higher signal-to-noise ratio. And that is usually, in my experience, what we have seen is that it should be at least 15 to 20 dB minimum. I will say just minimum. It has to be a minimum of 15 to 20 dBs ahead of the noise level to be captured accurately.




Then, the temperature range, just like the frequency range, you know, the microphone is essentially some kind of mechanical structure. It would depend on the way the materials are used to make up that microphone, and what the dimensions, thickness, shapes, and sizes of these various structures within the microphone are. So, based on that, they will have a particular temperature range within which they will hold their shape and not change their dimensions. It is only within that range that the microphone is going to work properly. Suppose, after a certain time, what happens is that the microphone, let us say the membrane bends because of the high temperature or the material properties of the material itself changes. Then, obviously, the response is going to change. So, the microphone has a particular temperature range within which it performs. And beyond that, let us say some microphones can have a temperature range between, you know, minus 10 degrees Celsius to 70 degrees Celsius. So, the temperature range means that it is usually given by the temperature minimum. What is the temperature minimum to temperature maximum within which the structural The structural constitution of the microphone remains intact? There is no change in the shape, size, dimension, and the properties of the materials that make up the microphone. And that is why it will perform accurately within that range.



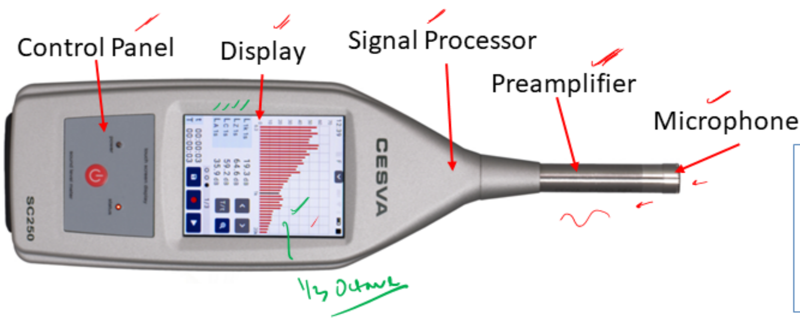

Okay, so the other kind of equipment that we are going to see is some advanced equipment like a sound level meter. It is a device that measures, you know, the time-averaged or exponential time-averaged sound levels, frequency-weighted sound levels, as well as various frequency-weighted average sound levels, okay? And sometimes they are also called noise meters or simply decibel meters or dB meters. So, you can see this is again a sound level meter that is available in our lab at IIT Roorkee.




So, essentially, what is a sound level meter? Now, imagine that before that, what we had was a microphone that was connected via cables to first a preamplifier, which was then connected via cables to a data acquisition system, which was then connected to some computer and signal processor. So, we had a lot of hardware and cables and a very elongated, cumbersome setup. Suppose in some conditions, you would like to record the sound, and you're okay not to record the detailed signals but just get a quick analysis of how the sound looks like. So, you're okay to not capture and use it offline but just to get some quick data, and you want some portable option. Then, a sound level meter becomes your option. So, it is like it has a microphone, a preamplifier followed by the signal processor, a display unit to display the data, some control panel, everything encased together in a very compact sort of device. So, that becomes your sound level meter. So, here the function, the functionalities of these various components that are within the sound level meter are the same.

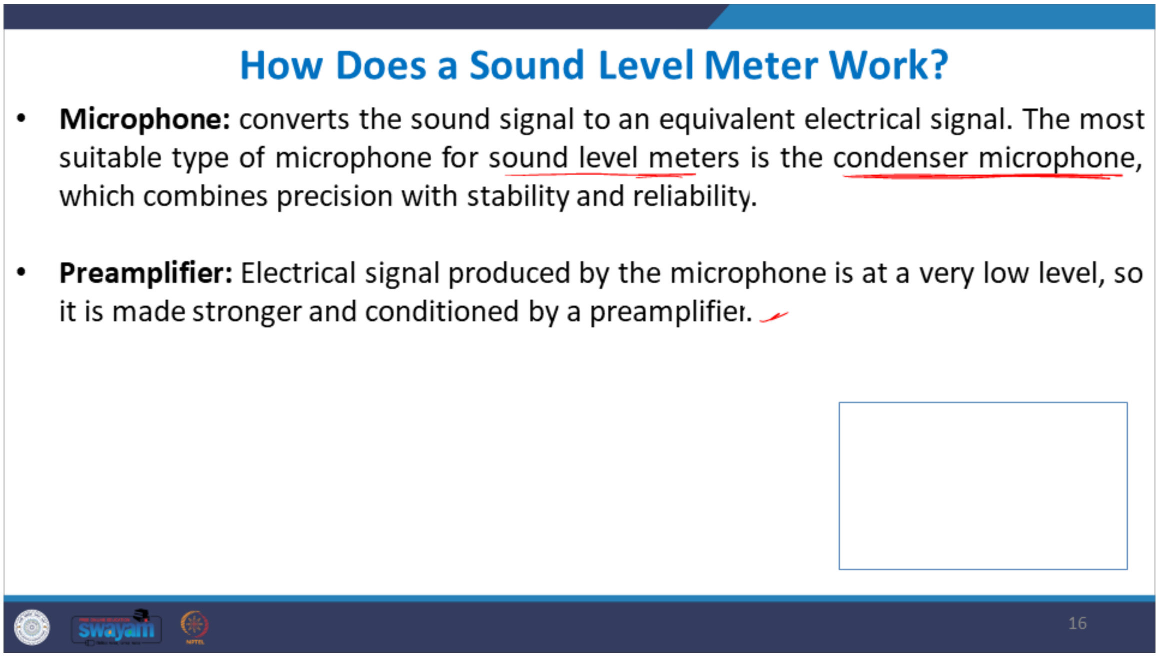
## How Does a Sound Level Meter Work?

- Sound level meters are composed of a **microphone**, a **preamplifier**, a **signal processor**, a **display** and a control panel.

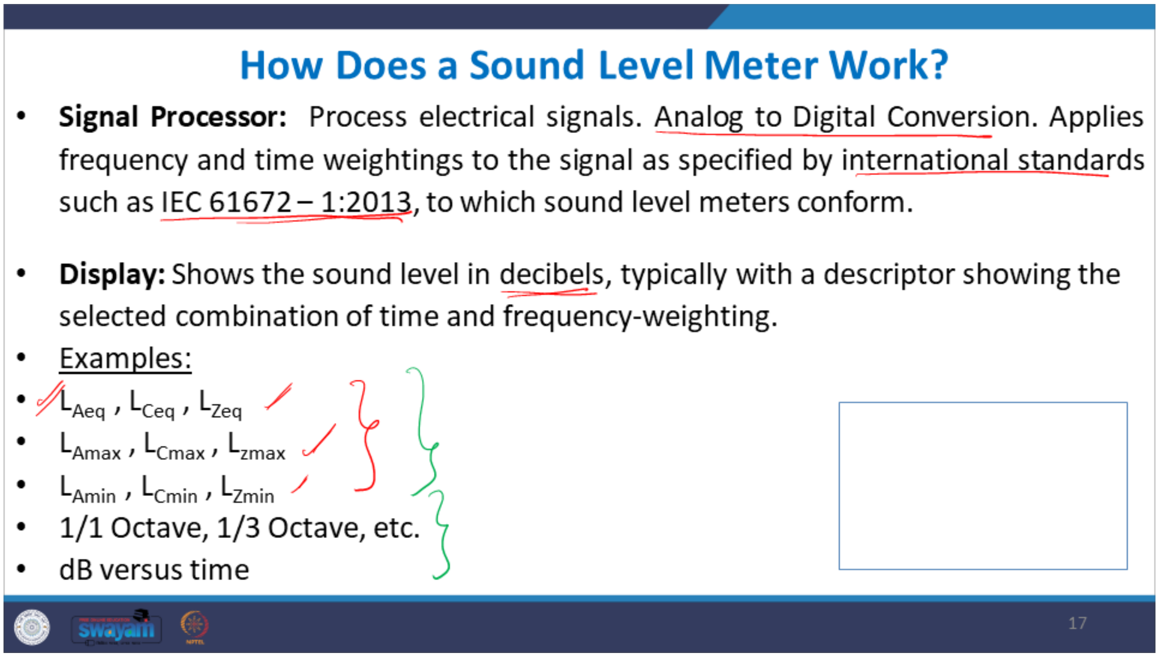




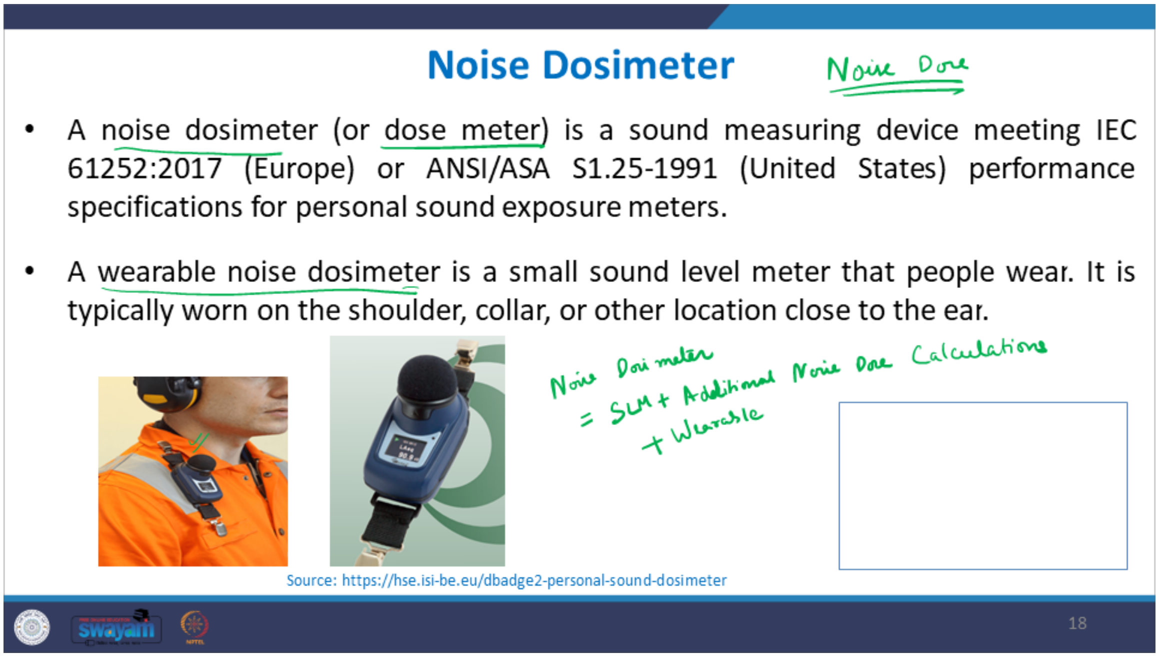

Like a microphone, it converts a sound signal to the equivalent electrical signals. And the most suitable type of microphones that are used for the sound level meters are the condenser microphone or the capacitor microphone because they combine both the stability and the reliability. Then we have some preamplifier because, obviously, why we need the preamplifier is because the acoustic signals are very minute pressure fluctuations. So, whatever electric voltage is generated is very, very low. So, it has to be made stronger and conditioned for further processing, which is done by the preamplifier, which is attached just behind the microphone over here. So, first, from, you know, pressure to electrical signals, Analog data, then the amplification of the electrical signals, then the processing of the signal,



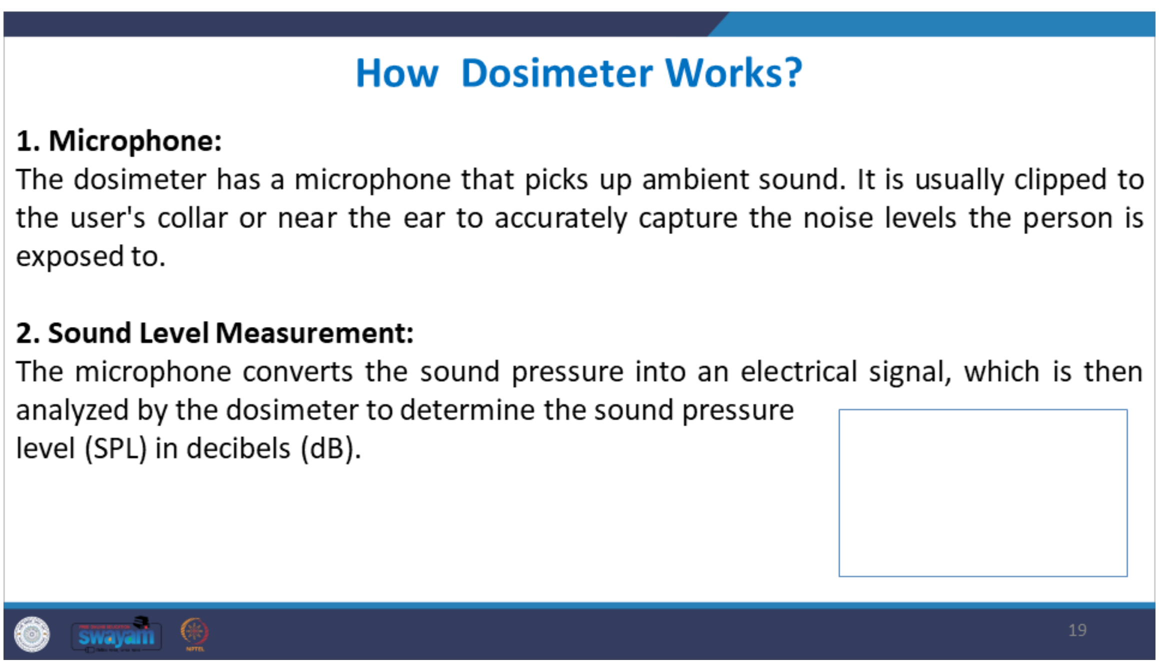

which means that digital-to-analog and analog-to-digital conversion happens to the amplified electrical signals, and then various kinds of frequency and time weightings are done. There are some international standards which sort of determine, you know, what kind of processing algorithms these sound level meters have to comply with. And based on these, you know, various kinds of standard algorithms available, the sound level meters calculate the decibels and directly display, you know, what the decibel levels are in the different frequency weightings, so they can display, you know, what the $L_{Aequivalent}$ is. Which is the equivalent sound level with the A filter, C-filter, and the Z-filter. These are some of the common filters used in the sound level meter. Then they also display the max values in these respective filters, the min values in these filters. Then the one-third, so usually, you know, most of the basic microphones are able to give these values, so you can directly go with this compact sound level meter and get an estimate of what the decibel levels are in a particular environment or particular factory setup, but if you purchase an advanced version, then some additional functionalities can also be incorporated. So, it can have both these numeric values as well as the spectral content. So, the one-third octave spectrum and the octave spectrum, then the decibel versus time, how the progress is. So, the time data, frequency data, all of this can also be seen. With your compact sound level meter, which is shown here,



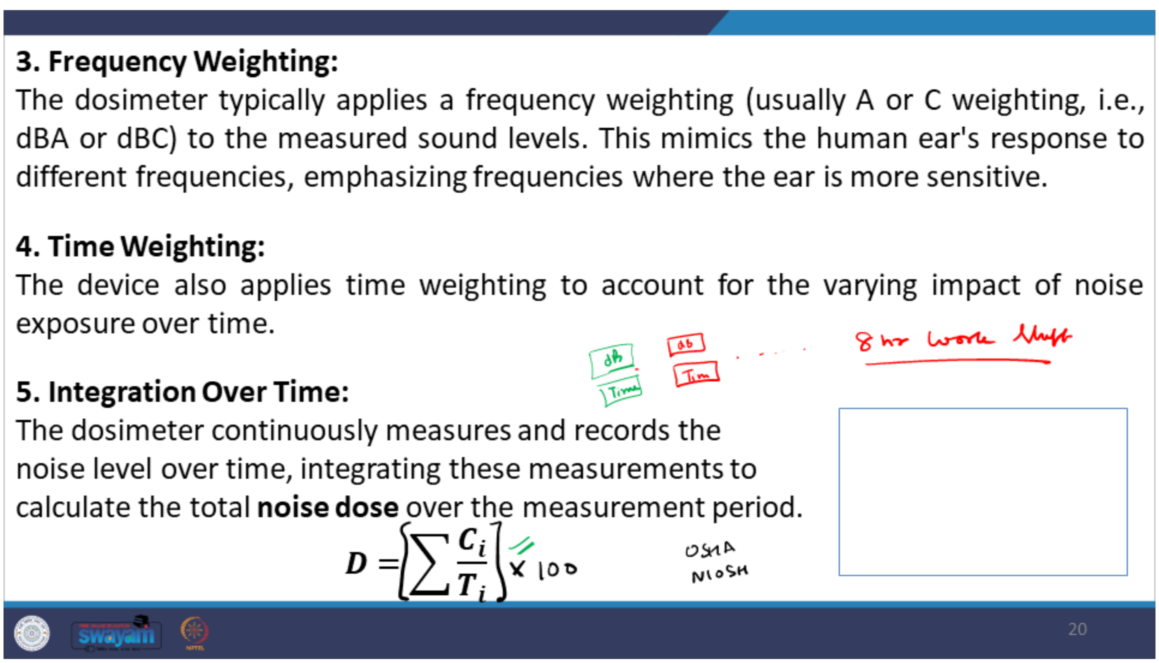

this shows a one-third octave spectrum here. So, you can see the one-third octave spectrum being displayed, and these various, you know, levels in decibels are being displayed. Then we have the last equipment that we will study in this lecture, which is the noise dosimeter. So, what is a noise dosimeter? Sometimes it is also called the dose meter. So, in some of the previous lectures, we discussed something called noise dose. This measures, you know, the exposure, the total noise exposure of a worker, the total daily exposure of a worker to noise in their occupational environment considering an 8-hour work shift. So, the noise dosimeter is something that calculates the noise dose of this worker. So, it calculates the daily noise dose. It is usually a wearable device. You know, the workers can wear it near their collars or shoulders or as an earpiece. So, it has to be somewhere near their ear so that accurate data can be obtained. So, this is, you know, a noise dosimeter. I can say a noise dosimeter is essentially, you know, a sound level meter plus some additional noise dose calculation. So, it is a kind of sound level meter that also does the noise dose calculations. Okay, and plus it has the functionality of being wearable directly on the workers' shoulders or near the ears so that wherever the worker is going from one factory to another factory or from one machinery setup to another machinery setup and they are working 8 hours a day, they can wear the noise dosimeter while they are working and they can, you know, do their work and the dosimeter at the end of the work shift, once the daily work shift is completed, will calculate and tell This is your total noise dose for the day and these are the typical decibel levels you were exposed to.




Okay, so this also has the microphone to measure the noise, then a sound level measurement or, you know, dB decibel values are calculated.



Then what happens is frequency weighting is done to convert it into A and C weighting that mimics more with the human ear's response. Then some kind of time weighting is also applied and integration is done. So, essentially, noise dose is calculated. So, they find out these are the decibel levels, and this is the time duration over which the decibel level was present. So, this was the dB level, and this was the total exposure time of this dB level. This was the second dB level, and this was the exposure time of the second dB level, and so on. So, throughout the 8-hour work shift, the noise dose works, and then using this formulation, that is the formulation of noise dose. It calculates the total dose, which is, you know, here $C_i$ is the duration of time over which a worker is being exposed to a certain decibel, and $T_i$ is the maximum exposure time for that decibel corresponding to either the OSHA guidelines or the NIOSH guidelines, what should be the maximum exposure time for that particular decibel level. So, all of this has already been covered in the previous lecture on how noise dose is calculated. So, within the device itself, the algorithm is built to calculate the noise dose, and once the noise dose is calculated, the data is shown and either the data can be downloaded for further analysis. The factory owners can use the data to see, okay, this is the noise dose of the different workers. And maybe I need to shift the job rotation so that the workers are not exposed to the hazardous level. So, they can later be used by the factory owners.



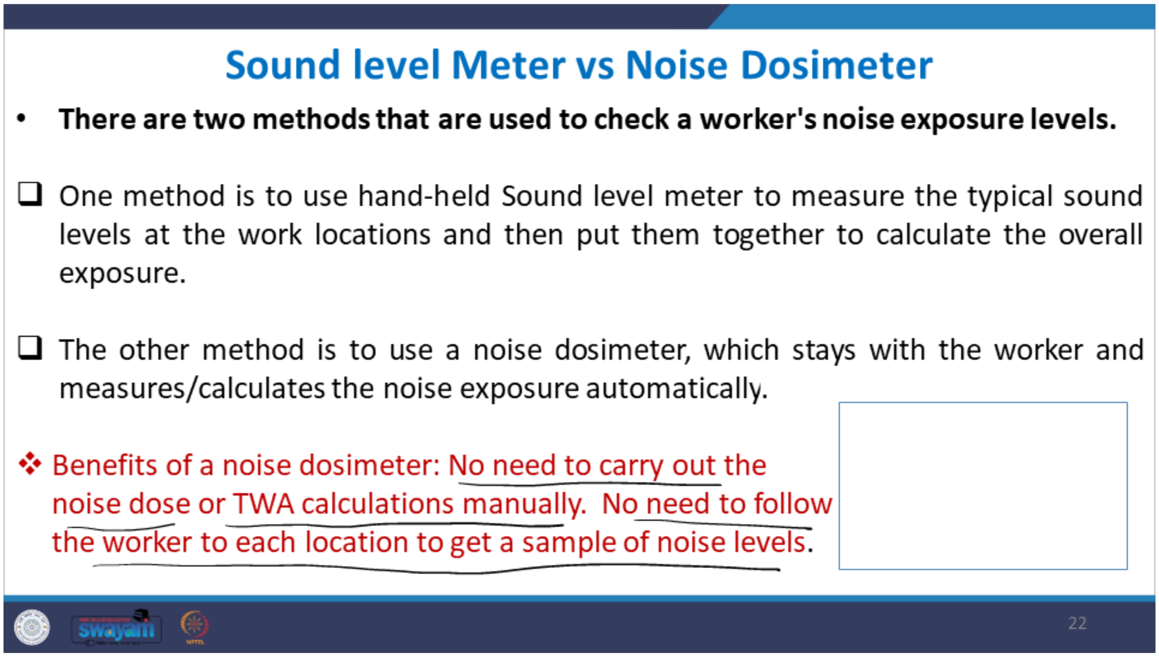
**6. Dose Calculation:**
As per OSHA or NIOSH recommended levels. A dose of 100% corresponds to the maximum allowed exposure. The dosimeter calculates the cumulative noise dose based on the intensity and duration of exposure.

**7. Display and Data Logging:**
The dosimeter displays the current sound level, accumulated noise dose, and sometimes other data like the time-weighted average (TWA) noise level. Many dosimeters also store this data for later analysis.

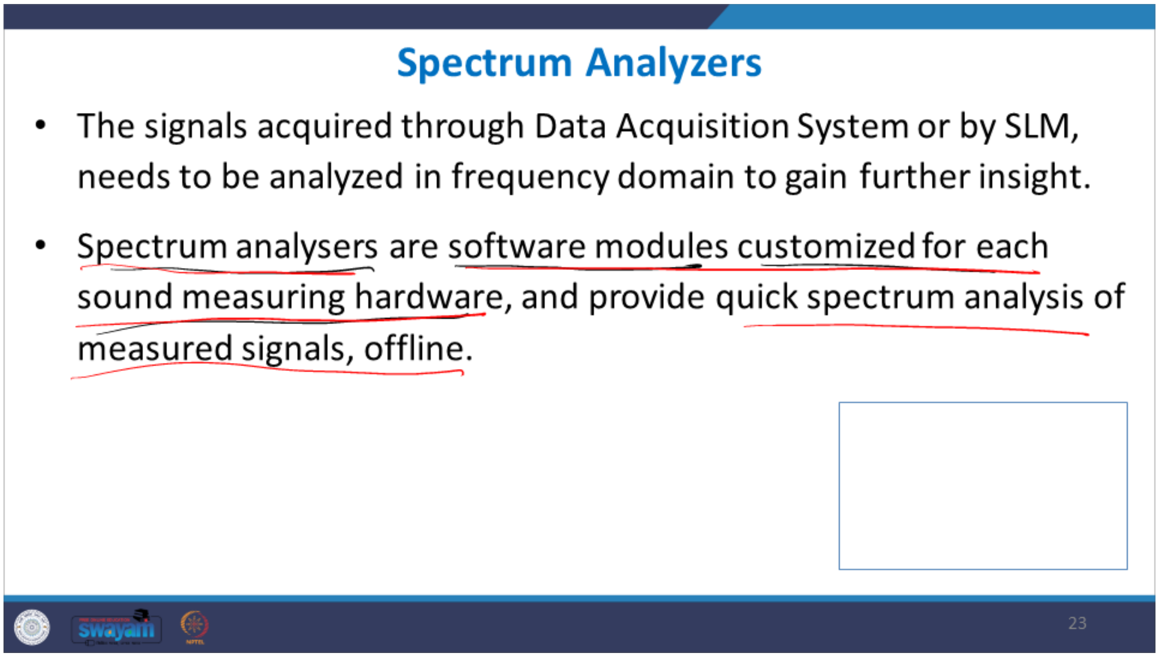
**8. Data Analysis:**
After the measurement period, the data can be downloaded to a computer for further analysis, helping to assess the risk to hearing damage of a worker, and to ensure compliance with noise exposure regulations.



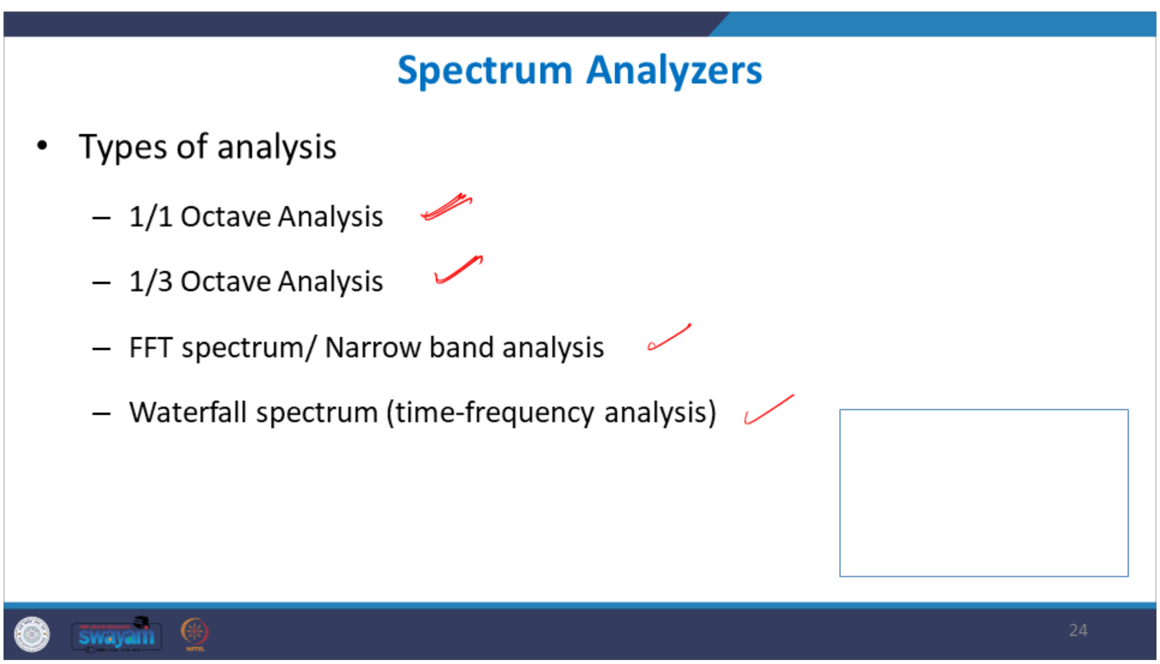
So, as I said, you know, a sound noise dosimeter is what? It's a sound level meter with some additional functionalities, okay? So, you know, what are the benefits of a noise dosimeter? There is no need to carry out the noise dose or TWA calculations separately. And there is no need to follow the worker to each location to get the noise samples. The worker can directly wear it, do their entire, you know, work in various environments. And at the end, you can switch off the device and give it to the researcher. They can analyze and see directly what the noise dose was. Okay.



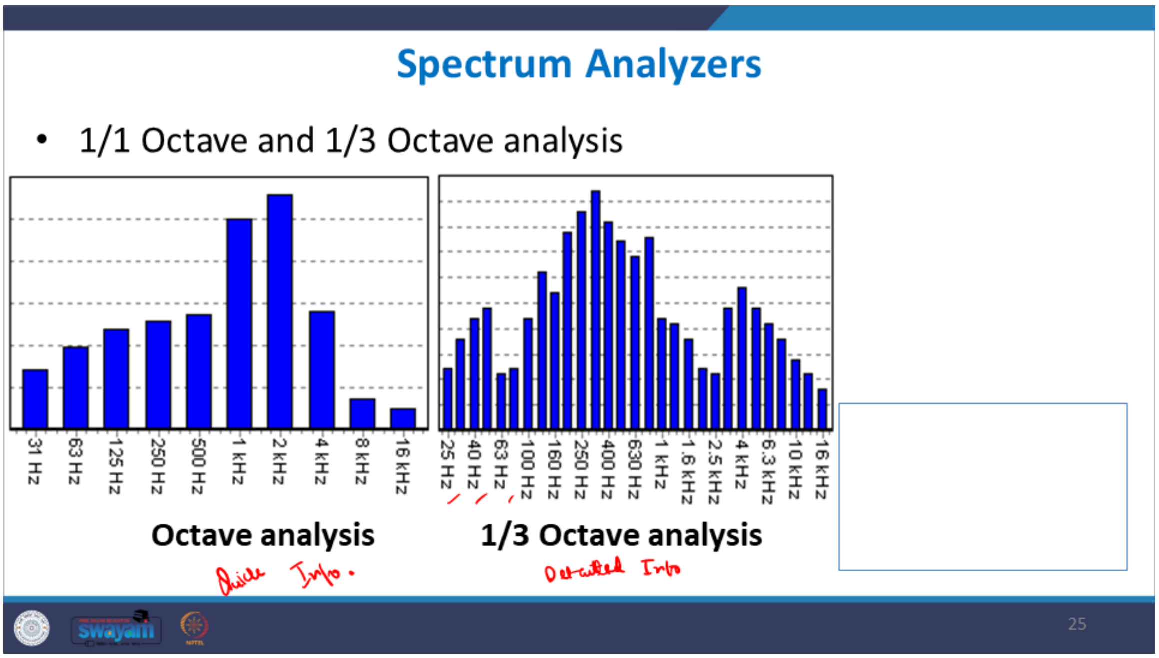

Okay. So, you know, okay. So, one more device we have is the spectrum analyzers. Okay. So, as I told you, the data acquisition system converts the digital to analog data and then analyzes it to get various insights. We can get beautiful visual representations, octave spectrums, FFT spectrums, and various kinds of things using the acquired data. So, usually, you know, there is a spectrum analyzer that is always a kind of software module customized for each kind of data acquisition system or sound measuring device. So, whatever your sound measuring device is, whether it is a sound level meter or a typical microphone plus the intact device. So, whatever your sound measuring setup is, at the end of the day, you are getting some digital data and then There is always a spectrum analyzer that is customized for each kind of sound measuring hardware. So, some quick spectrum analysis of these signals can be done offline.



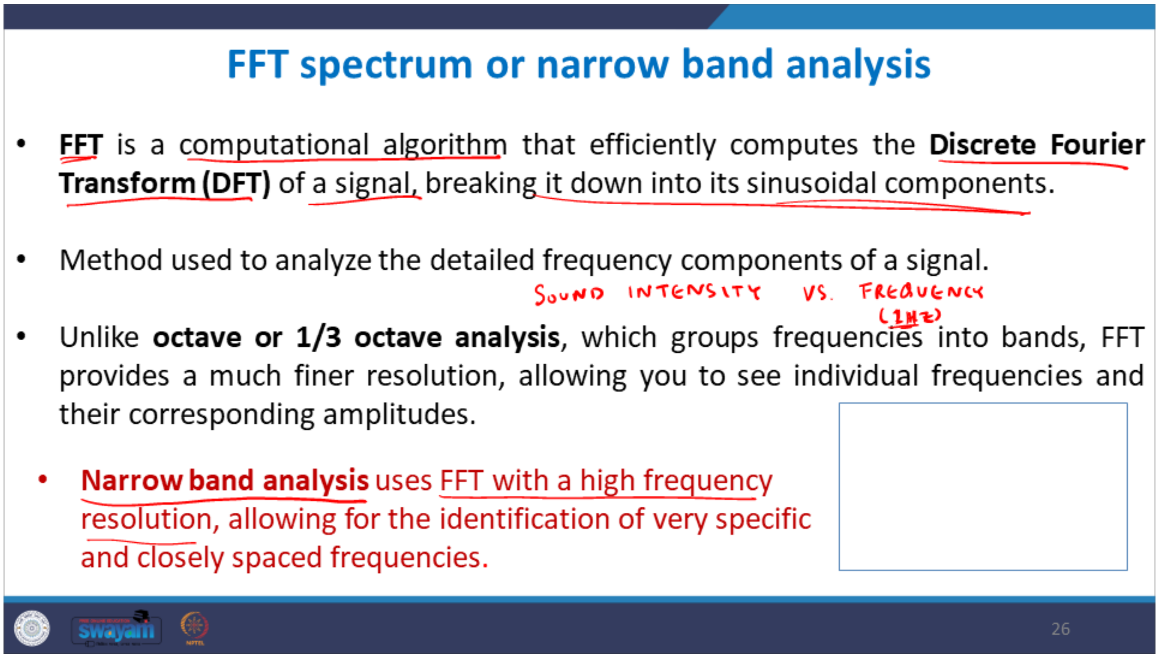

What are the typical kinds of analysis that are done? You have the one-by-one octave analysis, the one-third octave analysis, FFT spectrum or narrow band analysis, and the waterfall spectrum or the time-frequency analysis. We have already covered what you mean by the octave and the one-third octave analysis, okay.



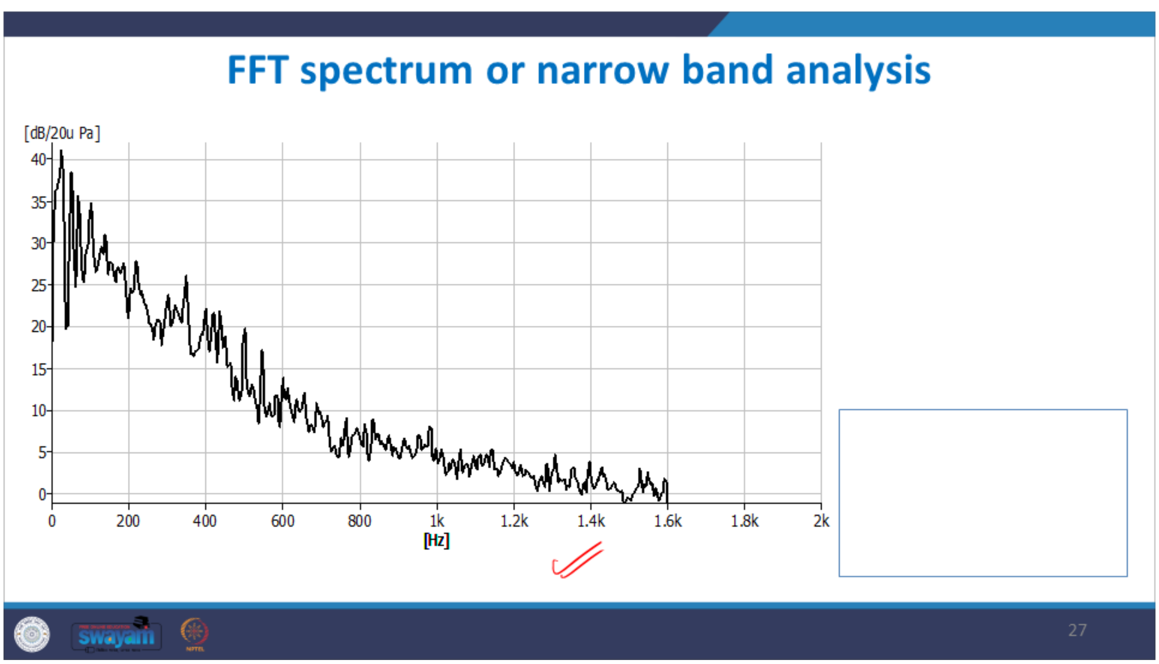
So, this can also be displayed using the spectrum analyzer of a particular hardware, and this is for more detailed information, this is for quick information about the frequency content, and this is for more detailed information about the frequency content.



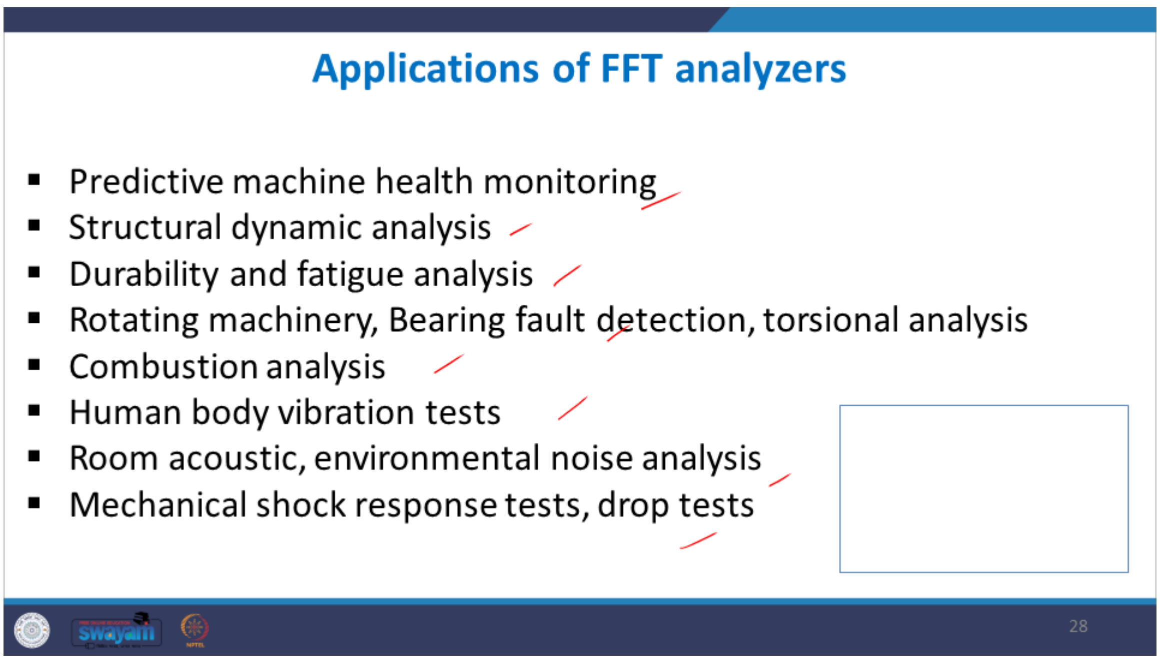

Then you have the FFT spectrum or narrow band analysis. In our lecture on spectrum analysis, I have covered what is meant by narrow band analysis. So, instead of dividing the frequencies into these individual bands, it is rather, you know, per Hertz, what is the frequency, what is the sort of sound intensity content per Hertz, and there is a continuous frequency scale. So, that gives you a frequency FFT spectrum or a narrow band spectrum. FFT is one of the most common computational algorithms that is used, which computes the discrete Fourier transform of a signal. Okay, it breaks down the various sinusoidal components and then generates what you want at the end of the day, which is, you know, sound intensity versus frequency. So, there is no frequency band, but usually, you know 1 Hz is taken as the bandwidth. So, the continuous frequency sound intensity distribution over this continuous frequency scale is obtained. So, narrowband analysis is sometimes used synonymously with FFT spectrum. So, whenever you are doing the FFT spectrum with a high frequency resolution, which means 1 Hz. So, per Hertz you are calculating, and a continuous spectrum you are generating, then that is also called the narrowband analysis.




So, this shows a typical narrowband analysis obtained from a spectrum analyzer. So, instead of having these big broad bands and discontinuous data, we are having a continuously varying intensity versus the frequency.



And they are used in various applications such as predictive health monitoring, structural dynamic analysis, durability and fatigue analysis, rotating machinery, machinery fault detection and torsional analysis, combustion analysis, even in human body vibration tests, room acoustic environmental noise analysis, mechanical shock response, and drop tests. Not just, you know, the FFT analysis, FFT spectrum analysis, or the narrowband analysis is not just used to see what is the content of a sound signal, what is the frequency content of a sound signal. It can be used to see the frequency content of the vibration signals and can be used for various kinds of, you know, vibration analysis as well.




So, with this, I would close this lecture. Thank you for listening.

**Thank You**