NOISE CONTROL IN MECHANICAL SYSTEMS

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Week:6

Lecture:026

Lecture 026: Considerations in acoustical measurements

Hello and welcome to lecture number 26 in the course on noise control in mechanical systems. in this lecture, we will study about some considerations that we need to make when trying to measure or perform acoustical measurements. to summarize, in the previous class, we started discussing the characteristics of microphones and DAQ. And then we studied the sound level meter, the noise dosimeter, and various kinds of acoustical equipment. And then we began our discussion on spectrum analyzers, which are essentially not hardware but software components.

These are configured to work with these kinds of hardware equipment, okay? whatever your hardware setup is for measuring sound signals—be it the microphone and the DAQ

Summary of previous lecture

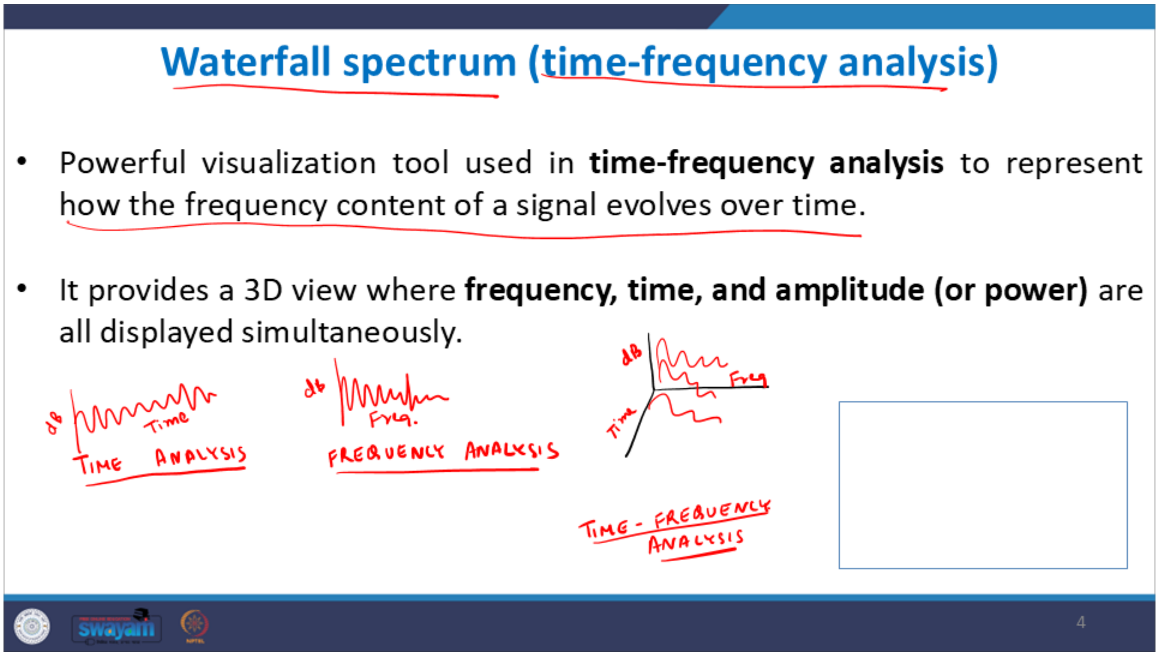
- Characteristics of MIC + DAQ
- SLM
- NOISE DOSIMETER
- SPECTRUM ANALYZER

system, the sound level meter, or the noise dosimeter—to this particular hardware, you have a customizable software module, which is the spectrum analyzer. It essentially processes the data obtained and creates valuable information about the sound signals being captured. We saw that spectrum analyzers generate 1/1 octave, 1/3 octave, FFT spectrums, narrow band spectrums, and the waterfall spectrum. this 1/1 octave and this 1/3 octave kind of analysis is typically done by a spectrum analyzer, as we already saw in the last lecture.

Outline

- Waterfall spectrum (Analysis by Spectrum analyzer)
  1/1 Octave, 1/3 octave,
  FFT Spectrum, Waterfall Spectrum
- Discrete signals
- Nyquist–Shannon sampling theorem
- Aliasing
- Measurement Standards and guidelines

We will continue our discussion and see the last type of analysis for this course, which is the waterfall spectrum analysis, which is both a time and frequency analysis.



This is also called time-frequency analysis because here the variation with respect to time and the variation with respect to frequency are both seen together. Let us say you have a time domain analysis. Suppose you have the dB levels; this is how the dB level varies with time. That becomes your time analysis or time domain analysis. In the various kinds of frequency analysis, be it the octave, 1/3 octave, or narrowband analysis, what you see is how dB varies with respect to frequency.

Now, if you want to see how the decibel levels or the intensity varies with respect to frequency and time, then that is called the time-frequency analysis. It is like a 3D variation. You have the decibel levels, how they vary with respect to time, and how they vary with respect to frequency. This kind of analysis we do is called time-frequency analysis, and the graph that we obtain or the visualization that we obtain is called the waterfall spectrum. here the x-axis corresponds to the variation with respect to time. The y-axis corresponds to the variation with respect to frequency, and the z-axis corresponds to either the sound intensity or the amplitude in decibels. you can do the waterfall spectrum not just for sound signals, but you can also do it for vibration signals, where the z-axis, for example, in the case of sound signals, corresponds to the decibel levels of the noise.

Waterfall spectrum (time-frequency analysis)

Components of a Waterfall Spectrum:

X-axis (Time)	Y-axis (Frequency)	Z-axis (Amplitude or Intensity)
Represents the time dimension, showing how the signal changes as time progresses.	Displays the various frequency components present in the signal.	Often represented by color or height, it shows the strength or power of each frequency at a given time.

In the case of vibration signals, it will correspond to the amplitude of the acceleration achieved due to the vibrating surface. The amplitude of the acceleration or the RMS value of the acceleration can also be expressed in decibels if you want. And why it is usually used? It is usually used to see how the frequency content evolves over time. How is it conducted? For example, a narrowband analysis is conducted using the algorithm called the Fast Fourier Transform or the FFT, and hence the name FFT spectrum.

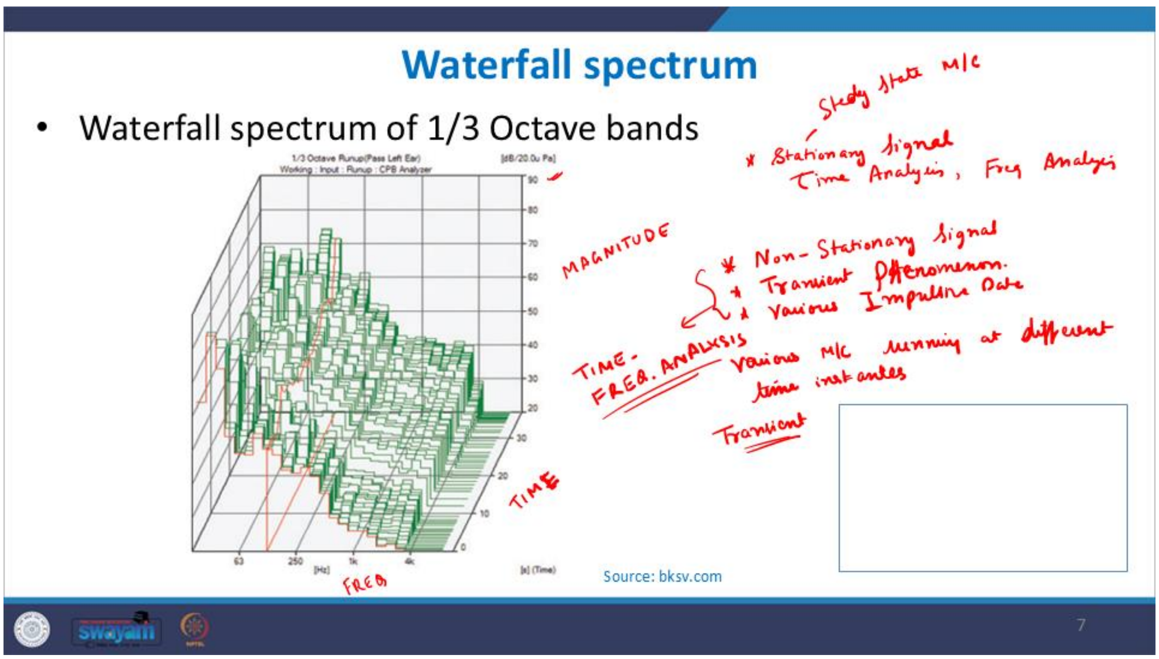
How Time - Frequency analysis works?

- The **time-frequency representation** divides the signal into short time segments, and then a **Fourier Transform** (usually via **Short-Time Fourier Transform**) is applied to each segment. This allows for tracking how the frequency spectrum changes over time.

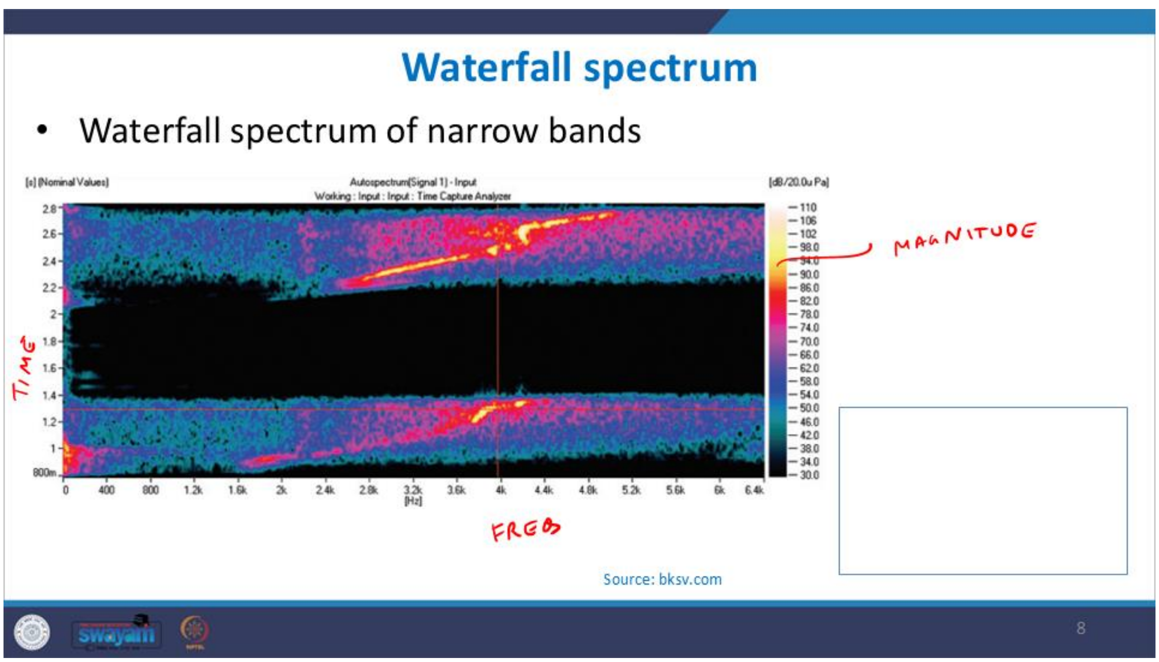
- The result is a series of frequency spectra that can be stacked to create the waterfall display. Each slice of the waterfall represents a frequency spectrum at a specific point in time.

In this case, what usually happens is that this is done using another form of Fourier transform, which is the short-time Fourier transform. What happens is suppose you have a 1-second or a 2-second recording. You divide it into small millisecond time segments, so small time segments you divide the entire data into, and for each time segment, you calculate the spectrum analysis. for this time segment, you did the spectrum analysis and got some frequency data. Then, for the next segment, again you do the FFT and you get the frequency data, and then again for the next time segment, you do the FFT separately and you get the frequency data, and so on. you are doing successive FFTs for the short time segments, and hence this algorithm is also called the short-time Fourier transform.

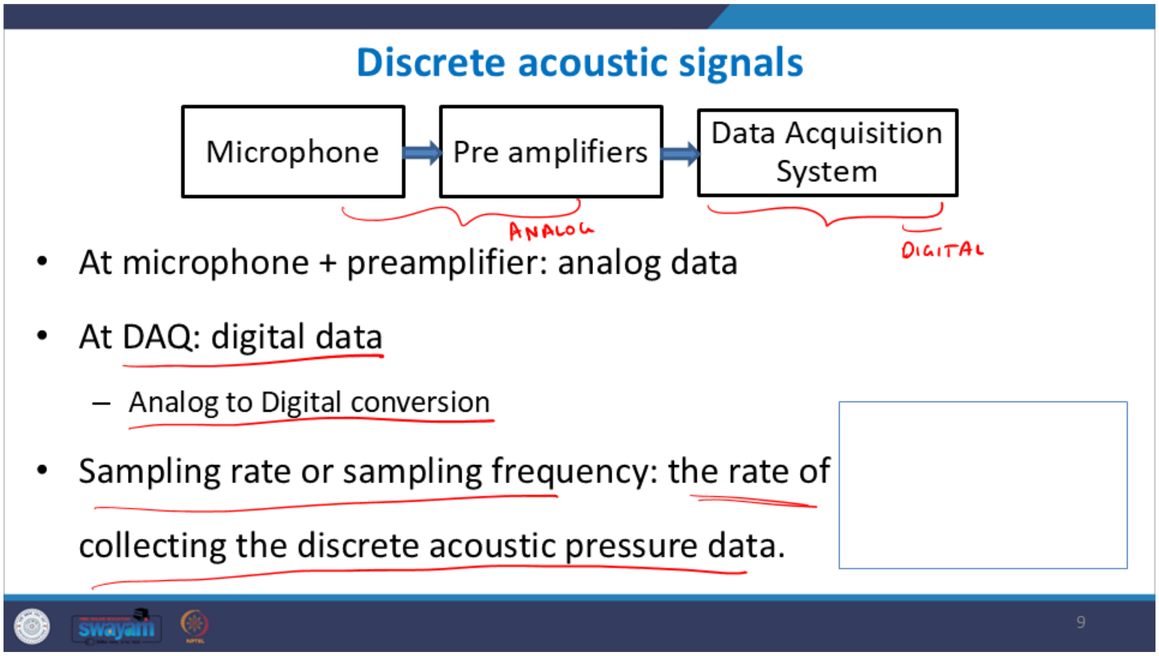
Where the entire signal is divided into small time fragments, and individual FFTs are done for these time fragments. Essentially, you get for the different time points and the frequency content. You get a waterfall kind of a beautiful waterfall display; it looks like this.



Suppose here you can see this is your decibels, this is your frequency, this is your time. From 0 seconds till whatever 40 seconds data this is showing, and this is the magnitude of the signal that you are measuring. You can see how the frequency content is changing over time, but usually what happens is that the waterfall spectrum is rarely used. Mostly, we see the time-domain signal separately and the frequency-domain signal separately because essentially, we get clear information. This looks a little bit more complicated. For example, you have got some kind of stationary signal. Which means that the signal is almost stationary or some kind of machinery running on a steady state, then you can just do the time analysis and the frequency analysis, and that itself gives you a lot of information. You need not go for a time-frequency analysis because it is computationally more costly. But suppose you have got a very non-stationary signal. Some kind of, transient information, transient phenomenon is happening, various impulsive phenomena are happening where you do not have a steady state running machinery; rather than having machinery running in a steady state, but instead of that, suppose you have got various machines running at different time instances. It is not like they are running together simultaneously, but they are running and stopping at different time intervals. Various machineries and various kinds of transient phenomena are also happening, such as, suddenly during the machinery run, one part broke down, or some workers came, some visitors came and just saw the machinery and went away, or suddenly there was some kind of siren that was blowing up, various kinds of transient activities are happening. Impulsive noise data is there; then there you would need to do these non-stationary signal analyses, or you would need to do the time-frequency analysis. So here, in this case, for these cases, this would give you more valuable information. When, there is a lot of you have a non-fluctuating environment, and there are a lot of dynamic things happening which are changing over time; then you would need to do a time-frequency analysis.



This again shows another waterfall spectrum. There it was one-third octave spectrum in a waterfall; here you have a typical variation over frequency. This is a narrow-band spectrum waterfall. you have the frequency, and this is your time data in minutes; it is shown, but this is the time data, and this color shows the magnitude. I introduced the concept of sampling rate to you. What is the sampling rate?

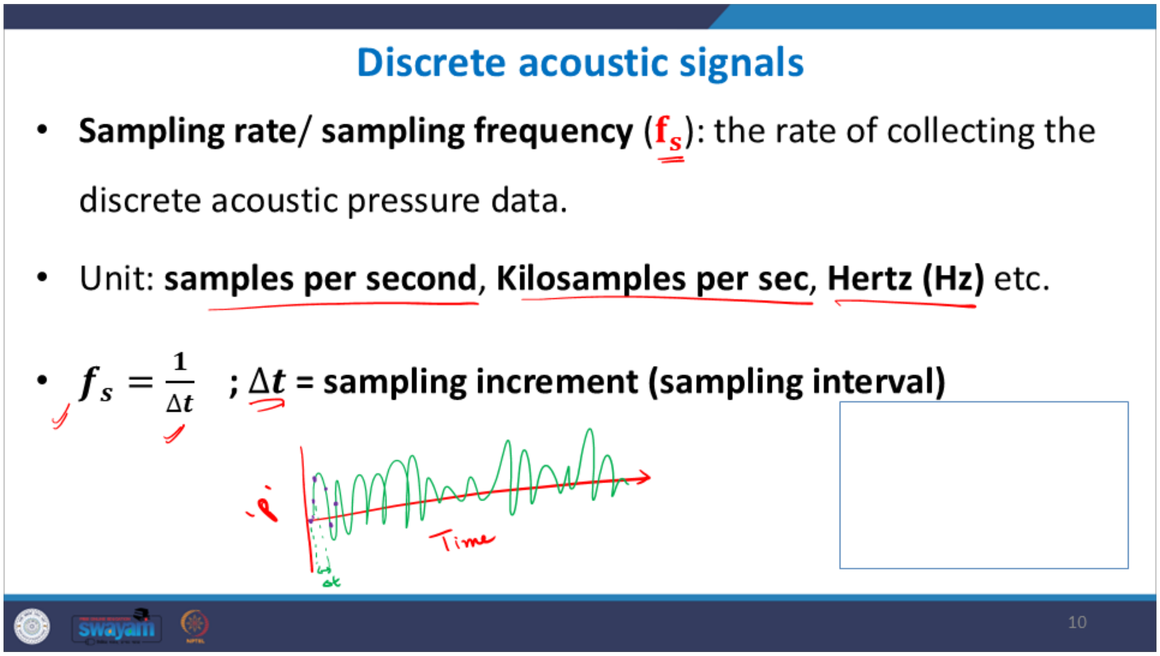


today we will see, how to decide the sampling rate of a data acquisition system, so that we can accurately convert analog data to digital data. Here at the microphone and preamplifier, till this level, we have the analog data. And then from here, the output is the digital data. the analog-to-digital conversion happens at the DAQ. We cannot store and process analog data. We can only visualize it in real-time, but we cannot store and process it further because it is an infinite number of data points; we have to digitize it.

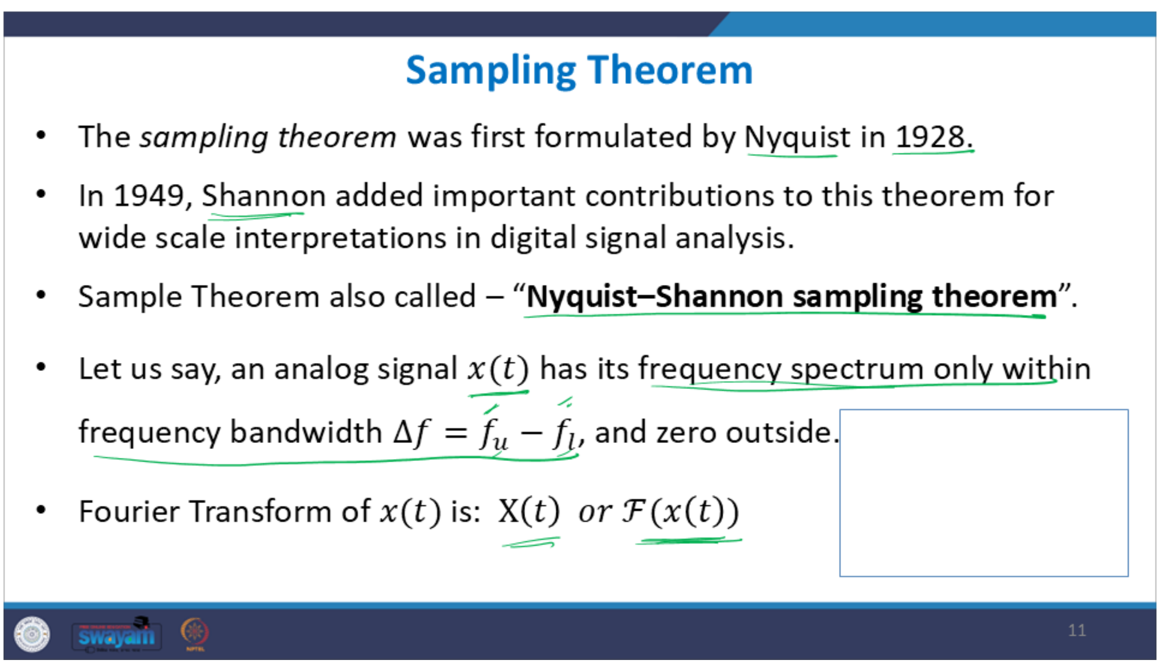
Sampling rate or Sampling frequency was introduced as the rate at which the discrete acoustic pressure data is collected. Usually, it is referred to as fs, a symbolic notation. It is the rate of collecting the discrete acoustic pressure data, and it is usually measured in Hz (Hz), or samples per second, kilo samples per second, and so on. One way to write this could also be that fs, or the sampling frequency, is

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

where Δt is called the sampling increment or simply the sampling interval.



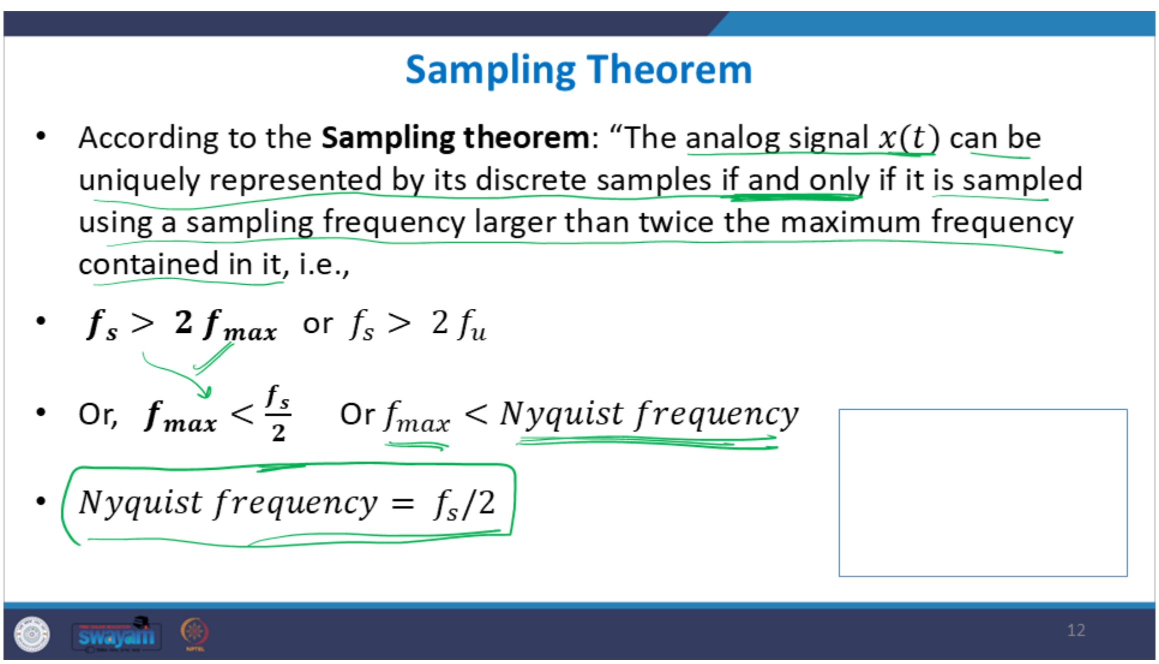
which means that suppose you had a certain signal, some kind of pressure is there, and this is your signal. you have some signal, some acoustic pressure variation with respect to time. what you can do is, at every Δt, you can record a point. The interval at which every new data point is collected becomes your sampling interval or sampling increment, and the sampling rate would obviously be the interval at which it is collected it would be $1/\Delta t$, or the rate of collection. What is the sampling theorem? This determines what should be your sampling rate for a particular data collection.



It was first formulated by the scientist Nyquist in the year 1928. Later, Shannon, another scientist, added important contributions to this particular theorem, and some wide-scale interpretations were made for digital signal analysis. In the current digital signal processing, this is known as the Nyquist-Shannon sampling theorem. let us say, for example, we have some analog signal x(t) and it is lying entirely within a frequency range of fu to fl, so this is the frequency spectrum. that signal has got this as the range within which this entire signal lies, and outside the frequency range, the x(t) is becoming 0. It is only within fu to fl, where this is, the upper frequency and the lower frequency content of the signal x(t). Then, the Fourier transform is capital X(t), or this is the symbol for it. according to the sampling theorem, this analog signal x(t) can be uniquely represented by its discrete samples if and only if it is sampled using a sampling frequency that is larger than twice the maximum frequency contained in it.

You want to accurately represent it using the digital points. You don't want to miss out on any critical information. Then, whatever is the maximum frequency content beyond which there is no further frequency content inside X(t). your sampling rate has to be greater than twice this Fmax. suppose here we have an X(t) which is lying between fl to fu. fu being the upper range of X(t). Then, FS would be greater than twice of fu.

$$f_s > 2\, f_{max} \quad \text{or} \quad f_s > 2\, f_u$$



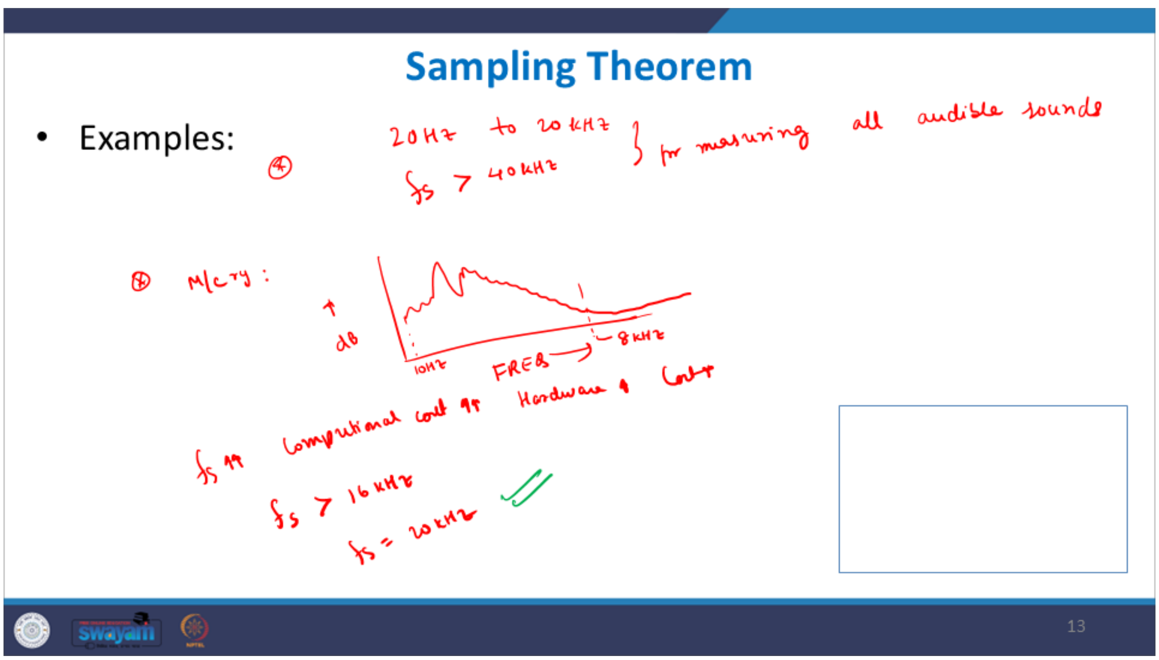
Whatever is your maximum frequency of interest, if you want to measure some sound signal and, you know that you want to measure it till this range, let us say 6 kHz, 8 kHz, or whatever. whatever is your maximum range of interest within which you want to measure a sound signal. It has to be then smaller than this fs by 2, directly from this equation.

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

Your maximum frequency content has to be smaller than fs/2,

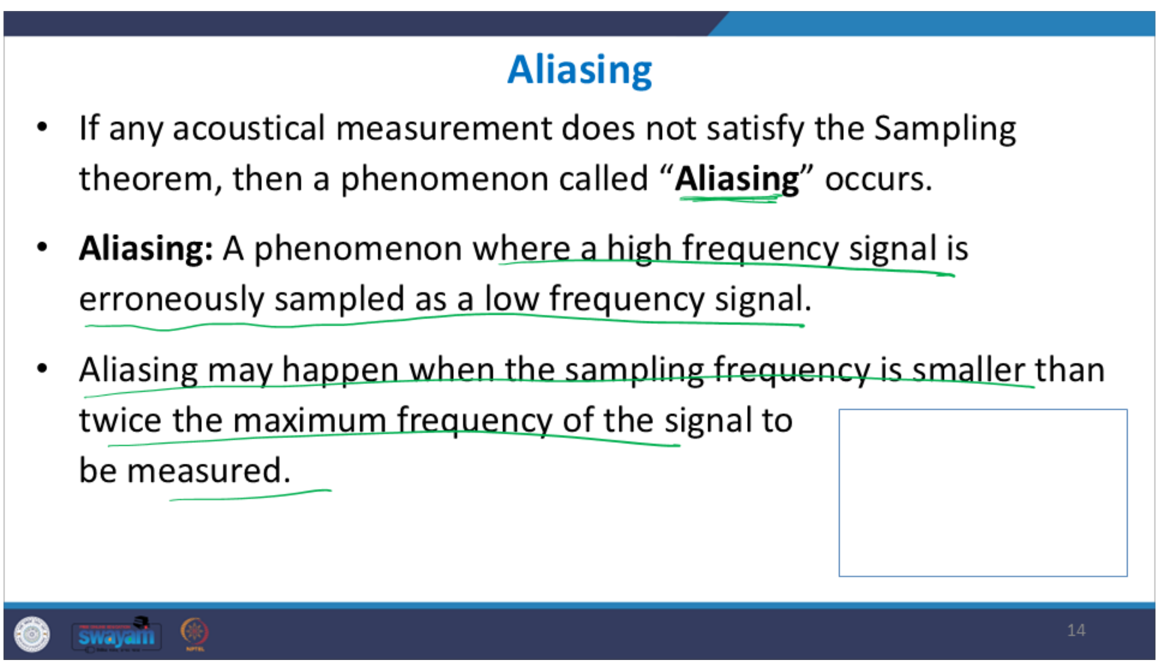
$$Nyquist\ frequency = \ f_s/2$$

which is the Nyquist frequency. We call this the Nyquist frequency. That becomes, the sampling theorem.

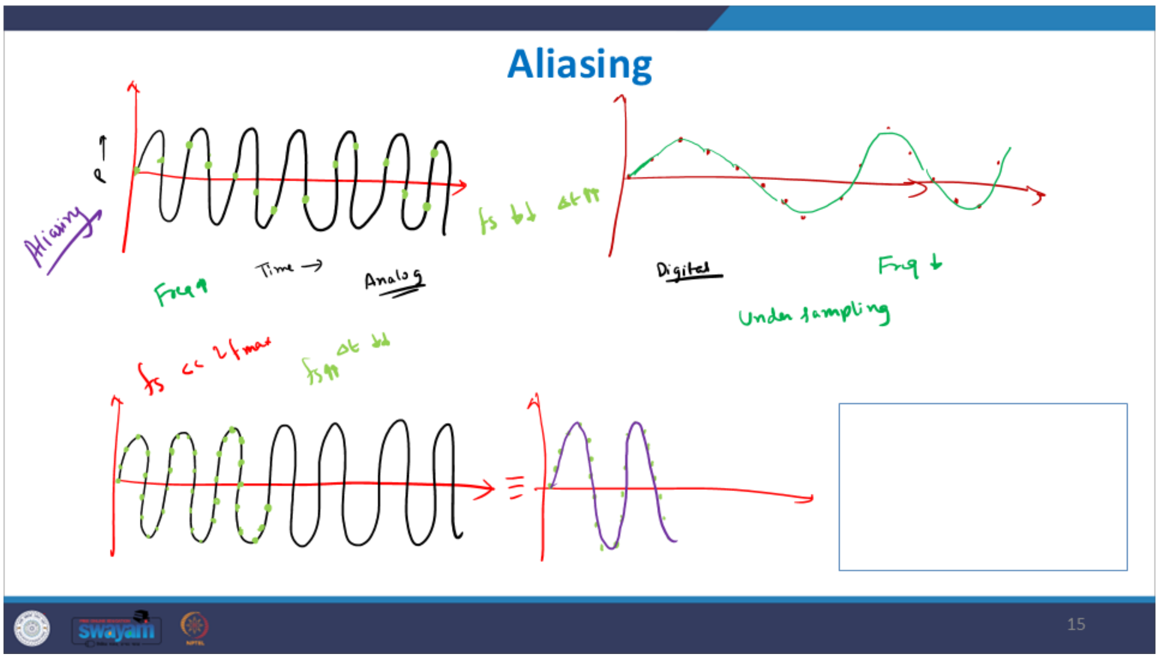


Some examples, let us say, for example, I am trying to record audible sounds, right? Usually, they are lying between 20 Hz to 20 kHz, which is our range of interest for most of the audible sounds. Then, in that case, if we want to record all the kinds of audible sounds in an environment, I can keep my sampling rate, which should be greater than twice this. (refer slide 13) it should be greater than 40 kHz, for measuring all audible sounds. In the same way, we have another example. Let us say I have gotten certain machinery, some kind of engine or some kind of mechanical machinery, and it is usually generating the data in the low frequency. I do a spectrum analysis or, out of experience, in the field of noise control, I know that this particular kind of machinery generates the data mostly in the low frequencies, and in the high frequencies, the magnitude goes down. Let us say, if this is your frequency and this is your dB, that the typical kind of variation is like this. And then it goes down like this. There would still be some noise present at all the various frequencies, but it drastically reduces beyond a certain point. Suppose I know that most of the sound is lying between 10 Hz and, let us say, beyond this point, let us say this is 8 kHz. This is the typical frequency content. I know, out of experience and measurement, that this is how the machinery measures. Then, in that case, why should I bother to keep a high sampling rate? Because, if the sampling rate is high, then my computational cost is going to be high. The computational cost will be high. The processing requirement, the hardware requirement would also be high, and the overall cost is going to increase, even processing would take a lot of time.

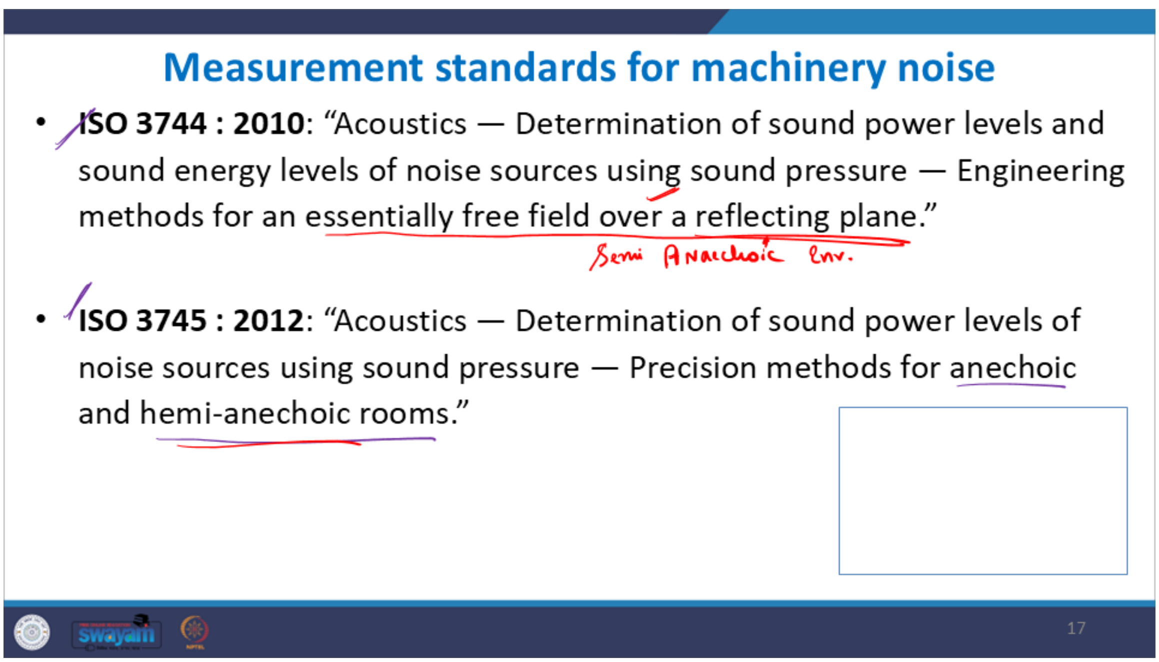
Why should I bother with it? If I know that the majority of it is within 8 kHz, where everything is lying, and beyond that, it is negligible compared to whatever is the content, it's negligible compared to the content at the other frequencies. Then, I can simply very safely set my sampling frequency greater than 16 kHz. I can keep it like, let us say, I can keep my sampling rate as 20 kHz, and it should be good enough to measure. I don't need to go to 60 kHz or 70 kHz like that. This is an engineering judgment that you need to make. That okay, this is my range of interest, and beyond which the magnitude is significantly low, then let me keep my sampling rate so that it satisfies this particular kind of theorem, and at the same time, my computational efficiency is achieved. that kind of judgment needs to be made.



Now, why do we have this sampling theorem? What happens if, suppose, you are not following this theorem and you are keeping a sampling rate quite lower compared to twice the maximum frequency? what happens if the signals are not satisfying? Then a phenomenon known as aliasing happens. What is this? It is a phenomenon where a high-frequency signal is erroneously sampled as a low-frequency signal. It may happen when the signals are not satisfying the sampling theorem. let us see one example of what is meant by aliasing. Let us say our original data is like this.

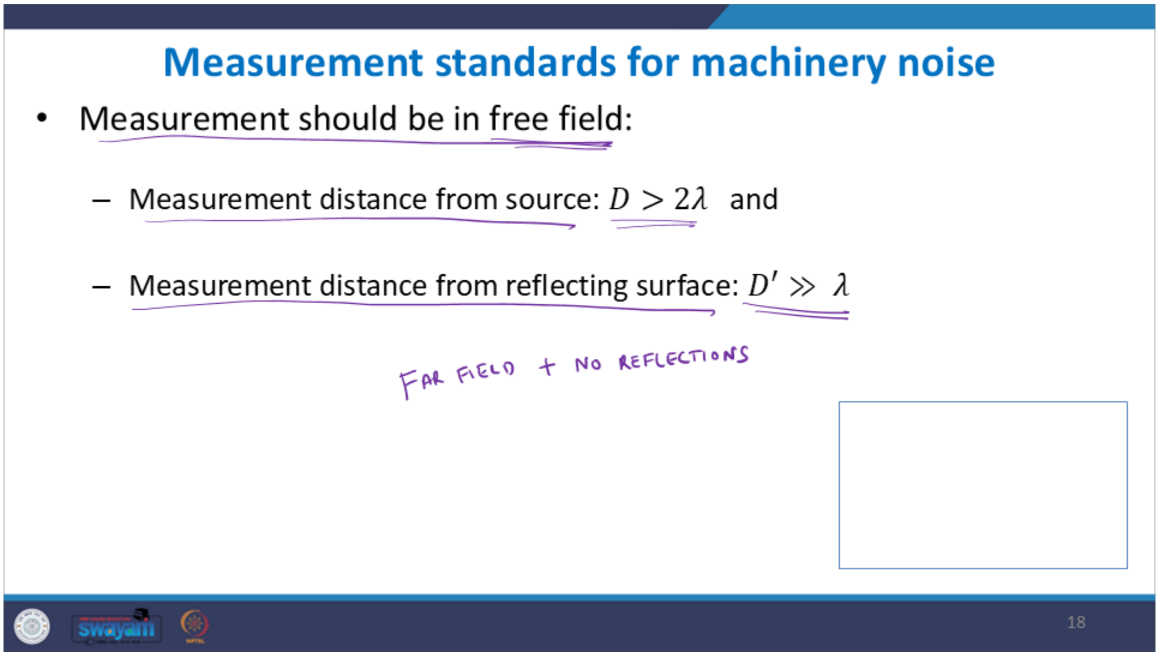


It is a high-frequency signal which is varying over time. This is your time scale. This is your pressure scale. And this is right now in analog data. And now I convert it into my digital form. let us say I have kept a sampling rate that is fs, which is much lower than this. You are keeping a lower sampling rate than what is actually required. What would happen? let us take an example of a very low sampling rate just to exaggerate the aliasing phenomenon and better understand it. I will use this symbol here. let us say I acquired my first data here. And then I acquired my second data here. My third data here, Fourth data somewhere here. I am keeping a very low sampling rate and I am acquiring the signal through these far-apart points. Let me just make these points bigger for better visibility. when I have acquired this signal, what will it look like? It looks like this, something like this. The processor is then storing it. when you have to further process this signal, whatever signal has been stored. Here, the computer is going to then recreate this as this signal. This is how it is stored. It can be interpreted as something like this. Here we had 1, 2, 3, 4, 5, 6, 7, 7 cycles in that time duration, but now we are having only 2 and a half cycles or so on. Here the frequency was higher per time, so many variations, but now we are getting the frequency as lower because of under-sampling. this is also called under-sampling. you are not sampling enough information to recreate the signal accurately, and you are missing out the higher frequency component, and that is why we take the maximum frequency of interest and based on that we decide a sampling frequency so that because lower frequencies we will anyways capture, but it is the higher frequency which will remain uncaptured like that, and suppose instead of this now we had. Our next scenario where our sampling theorem was satisfied, same signal like this. And suppose we had sampled it at a higher rate, complying with our sampling theorem. At a much smaller time increment, Δt here, because fs is higher, Δt is low here, fs was low, Δt was higher, a greater number of points were captured per second. And so on, we are able to get these points for the signals and so on. Essentially, we are retaining the information. It looks like it captures the higher frequency content. This is, the examples of aliasing. Here, aliasing has happened, and a high-frequency signal has been erroneously sampled as a low-frequency signal.



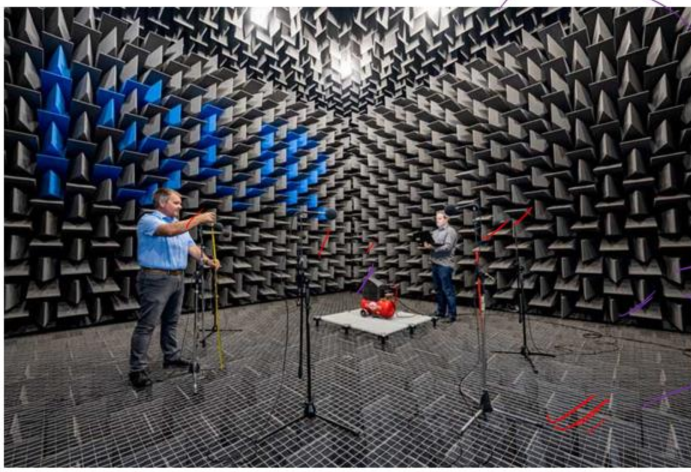
Let us now see some of the typical measurement standards that are used for machinery noise measurement. These can be certified by ISO 3744:2010. Which is the standard for the determination of sound power levels and sound energy levels.

And then there is another standard for the determination of sound power levels of noise sources using sound pressure and precision methods for anechoic and hemianechoic rooms. There are also additional standards. ASTM has its own standards and so on. But based on these standards, some of the, key guidelines I am going to elaborate on.



First of all, whenever you are measuring the machinery noise for noise control. The measurement has to be done always in the free field. In the previous lectures on acoustical sources, I discussed the types of radiation field and how the free field should be used because it is devoid of reflections, and it can give you the direct field information from the machinery source. In the free field environment, it means that the measurement distance from the source should be greater than twice the wavelength of interest, and the measurement distance from any reflecting bodies should also be much greater than lambda. Basically, you have to have far-field plus, if possible, low reflections or minimal reflections, then, from the standards that I already elaborated, the two ISO standards, anechoic chamber is quite preferred for most of the noise measurements. This shows a typical anechoic chamber, various kinds of these wedges are there, all these wedges are there surrounding, and here you have a wire grid at the floor. You have a wire grid at the floor, and you have the wedges on all the roofs and the ceilings. This is a typical anechoic chamber, and this machinery noise measurement is being conducted.

Measurement standards for machinery noise



- **Anechoic chamber** is recommended for measuring machinery sound field.

Wedges
Wedges
WIRE GRID

Source: https://isvr.co.uk/anechoic-chamber/

what is an anechoic chamber? It is a room that is designed to stop reflections or echoes of either sound any kind of reflections or echoes, they are completely stopped. How will that be possible? If the walls and the ceilings, whatever surfaces of the room are there, they are completely absorbing, the roof, walls, ceiling, completely, it completely absorbs the incident sound.

Measurement standards for machinery noise

- **Anechoic chamber**: It is a room designed to stop reflections or echoes of either sound or electromagnetic waves. They are also often isolated from any sound or electromagnetic energy entering from their surroundings.

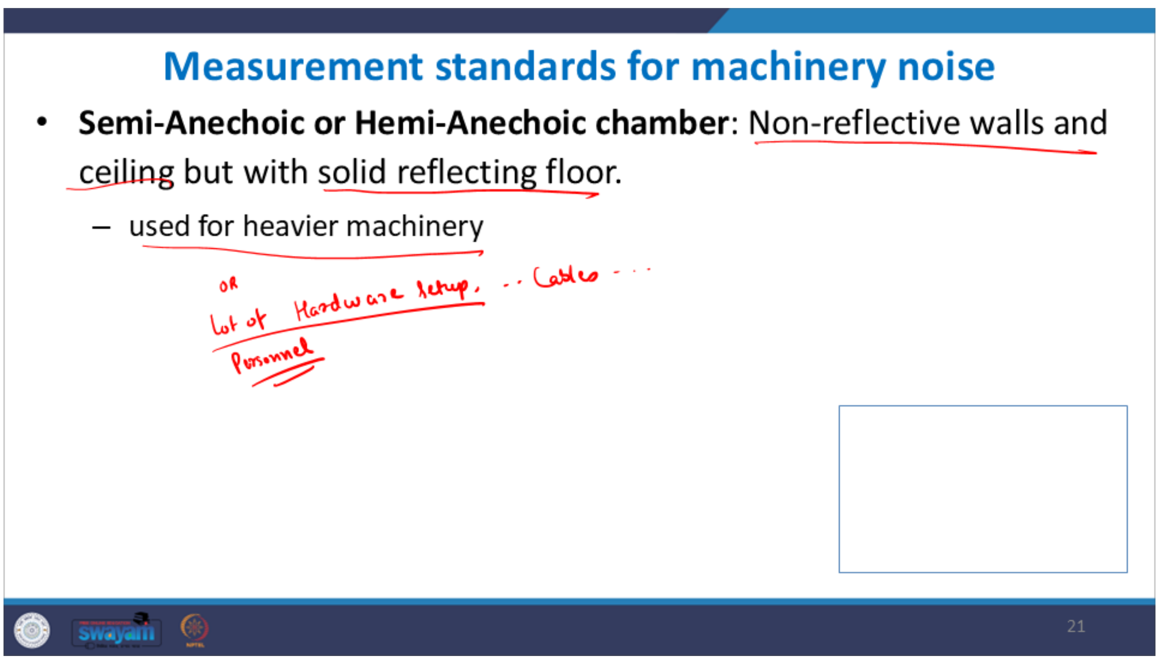
  Roof, Walls, Ceiling completely ABSORBS – the Incident SOUND
- This combination means that a listener exclusively hears direct sounds as if in a free field.

good Barriers

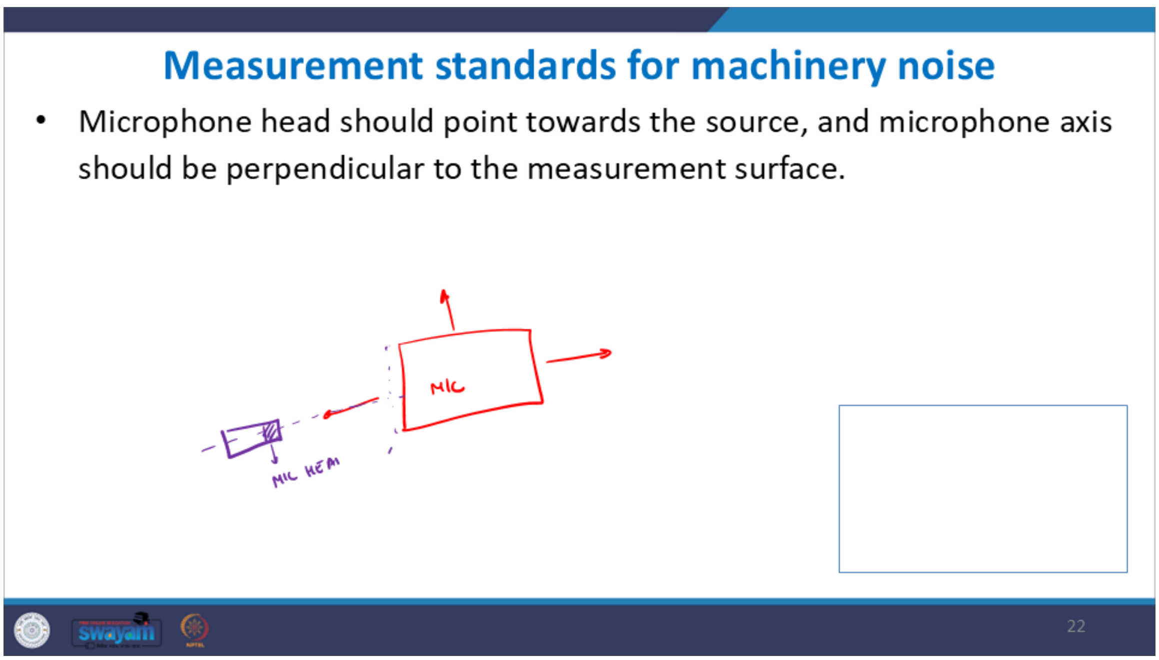
Roof, Ceiling, Walls, Other Surfaces

SAC ≈ 1

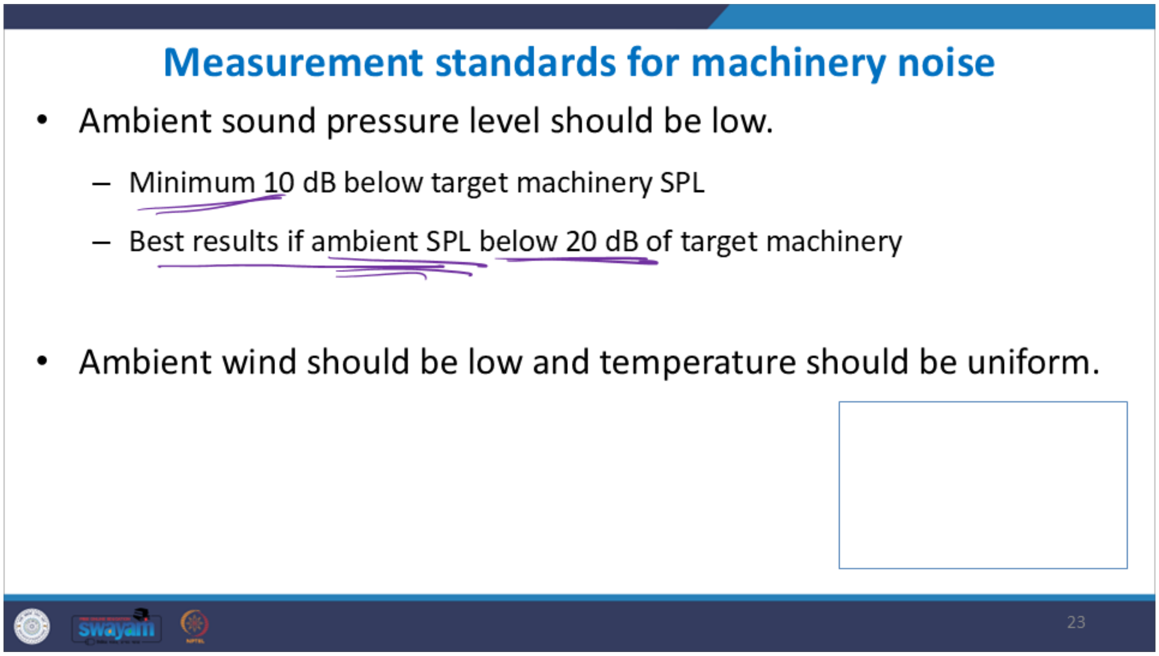
which means that the roof, ceiling, walls, and other surfaces which are present and are making up the room, they should have almost close to one sound absorption coefficient. when this kind of room is built, usually, not just the surfaces of the room are made almost completely absorbing to avoid any reflections, even, they are isolated from the surroundings. Whatever is the room inside, we have the wedges that are absorbing and avoid, but at the outside, we have these thick, good barrier material. which is stopping any sound from the outside to come inside. It is reflecting back; nothing is coming inside. The outside sound has been blocked, and inside, whatever machinery is creating, nothing is reflecting back. It looks like it is a free field, and at the same time, the outside noise has been cut down. The listener gets the feeling that they are directly listening to the sound in a free field. But most of the time, this anechoic chamber, usage may not always be convenient because here the floor is a wire grid.



You have to keep so many setups, you have to keep this machinery, people have to stand, measurement setups have to be kept. sometimes what happens is that a compromise is made, the walls and the ceilings are made as absorbing, whereas the floor which also should be absorbing, is actually made a hard floor. Then it becomes a semi-anechoic or hemi-anechoic chamber. the walls, ceilings, all of that are non-reflective in nature, but you have a solid reflecting floor, usually used when you use heavy machinery or, a lot of hardware setup is required, many personnel are required for measurement, people have to stand, a lot of hardware setup, cables, a lot of things are required, and you have heavy machinery. Then it is, in that case, it is not always feasible to extend, sort of suspend everything, and stand on the wire grid and do a measurement. in that case, you keep the floor as solid and reflecting, but then you use the walls and the ceiling, which are non-reflecting. These are the two types of environments that I used and the two ISO standards that I showed to you, Here the first standard shows essentially it is a free field over a reflecting plane. This standard is used for the semi-anechoic environment where everywhere else you have a free field environment, but the machinery is kept on a reflecting plane. when this is the condition, then semi-anechoic environment, this kind of standard can be used. These two standards have been developed precisely for that reason. If it is a fully anechoic room, then obviously, you can directly measure and get the sound power levels. But if it is over a reflecting plane, then some kinds of corrections are made, which are available in these standards.



So, another thing you have to ensure is that the microphone head should be pointing towards the source, and the axis should be perpendicular to the measurement surface. suppose you have machinery here which is radiating noise in whatever possible directions, you should ensure that your microphone is pointing like this. This is your microphone head. It is pointing directly towards the machinery, perpendicular, and the axis here is perpendicular to this measurement surface.



And obviously, the ambient sound pressure has to be low, at least 10 dB below the target machinery level, but in my experience, the best results are obtained if the ambient SPL is at least 20 dB below your target machinery level. The wind speed also has to be low, and the temperature condition should be uniform. Too much variation in the temperature and the fluid flow is going to change the sound speed characteristics.

With this, I would like to close this lecture. Thank you.