NOISE CONTROL IN MECHANICAL SYSTEMS

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Week:6

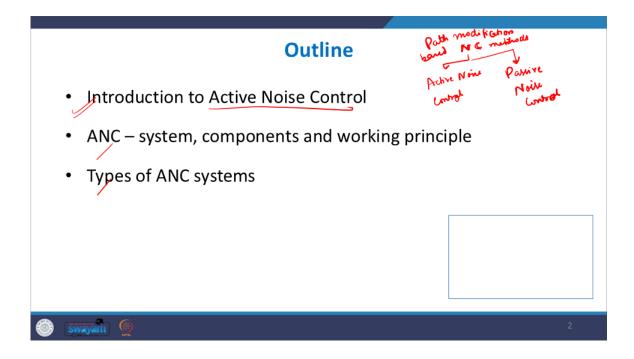
Lecture:030

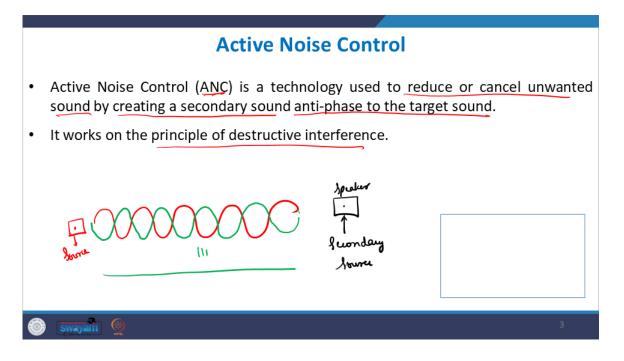
Lecture 030: Active noise control:1



Welcome to the course on noise control in mechanical systems with myself, Professor Sneha Singh from IIT Roorkee. In today's lecture, we will begin our discussion on active noise control. this is the brief outline. We will introduce active noise control, which is one of the major strategies in noise control. we know that in noise control, we have active noise control. These are the broad categorizations of path-based noise control, and then we have passive noise control. Both of them are path modification-based noise control methods.

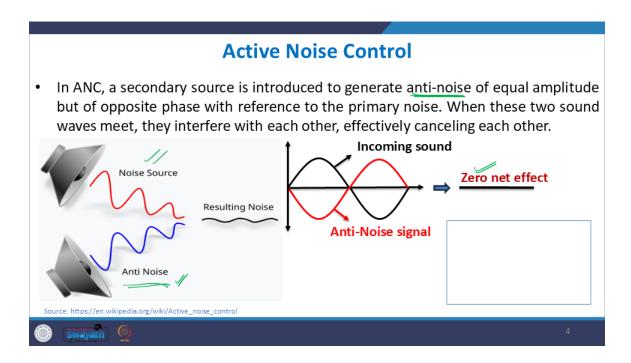
We will see what active noise control is, what the basic system components and working principles of an ANC system are, and what the types of ANC systems are.



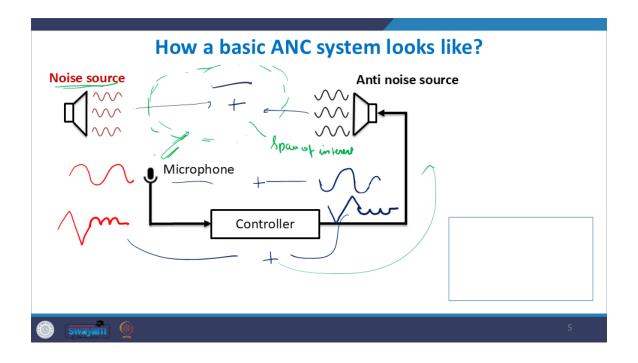


A slight introduction to active noise control was given in some of the previous lectures. What is it? We will call it ANC, the acronym for that. This is a technology where you try to reduce or cancel any unwanted sound by creating a secondary sound. If the sound is there, we create another sound which is anti-phase to the target sound, and it works on the principle of destructive interference. What happens? Suppose you have this sound that is

present in a space, and then this is your primary source which is creating the noise. Then what you do is you have a secondary source somewhere in that space. It could be a speaker or something like that, some kind of secondary source. And then the secondary source is such that it should always create the sound level which is anti-phase. a 180-degree phase shift results in exactly the same waveform, but with a 180-degree phase shift. what happens is the two waves will destructively interfere, causing a flat signal. This should be the end result. That is how it works.

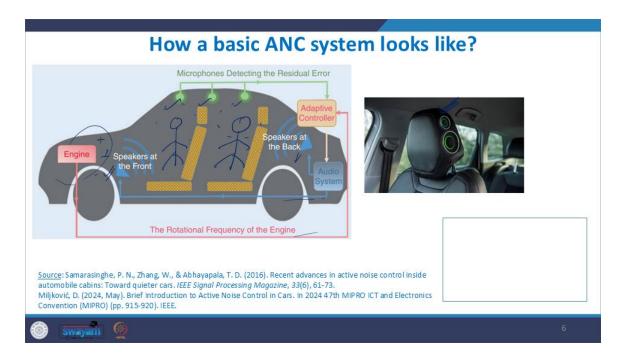


You have the noise source and an anti-noise source. Usually, this secondary source is also called the anti-noise source because it is generating a signal which is anti-phase to the already present noise signal, which is called the anti-noise signal. These are some of the terminologies used in active noise control. You generate the anti-noise, and the two are out of phase with each other so that when they combine, there is destructive interference, and the pressures cancel each other. Ideally, if everything was perfect, they should cancel perfectly. There should be perfect cancellation or perfect destructive interference, and the sound level should diminish and become zero. But because, in the real world, there are so many inherent random errors and variables, you do not always get perfect cancellation. In fact, in most cases, you do not get perfect cancellation. There may be some small level of imperfect cancellation, but either way, because it is anti-phase, the noise level reduces significantly. That is what happens.



This is how a basic ANC system would look like. The secondary source, or the anti-noise source, has to at every point in time, generate an anti-noise signal which is anti-phase with the original sound signal. that means there should be some mechanism through which the secondary source knows what sound signal to generate. The secondary source should know the current sound present in the environment and create a signal which is anti-phase with it. there should be some mechanism to measure the sound signal and give this feedback to the secondary source, which is the anti-noise source. That is what is happening. Whatever noise source is there, emitting the sound signals, the mechanism to measure it should be a microphone. Some microphone is present, usually closer to the sound source, to measure it. whatever is your space of interest. Let us say this is our space of interest, the space where we want to cancel the noise, and in the previous lectures, you saw that ANC is very space-dependent. It doesn't mean that if I have an ANC system, it is going to cancel the noise throughout a larger area. It is very spatial and localized. whatever is that location of interest, you have a microphone, and you measure the sound level in your region of interest due to the noise source. it measures the sound level in the region of interest, it gives it through some controller mechanism which drives the anti-noise source. based on whatever signal it is measuring, it will have some controller and actuator mechanism. That will drive the anti-noise source, and the anti-noise source would simply be whatever the measured

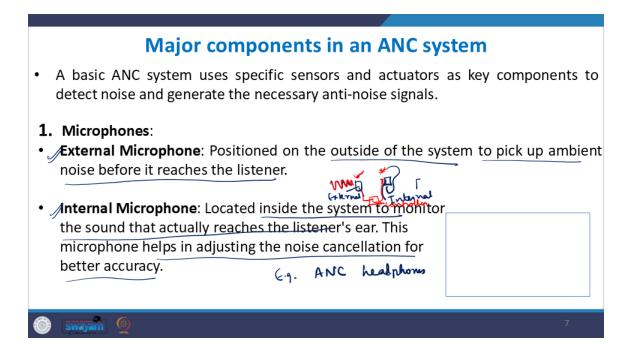
signal is, do a minus sign, convert it into an anti-phase signal, and keep generating the antinoise. The sound source, if it starts to vary from this to let us say this or like this, whatever then, the microphone will keep measuring, doing this controlling mechanism, and then this will also change its response based on whatever the measurement is. Suppose in the second scenario, the signal was something like this: lower frequency, higher amplitude. In some third case, you had some random signal like this, something like that. Based on whatever data has been measured, the anti-noise signal would be created, which is anti-phase to it. this will create something like this, if this is the sound wave that is coming up, it should be anti-phase of this, like this. In the same way, for this one, it would be some anti-phase signal again, Inverted in the other way. I will create a simple signal just for simplicity, for my own sake, and make the point clear. Something, I am just creating some patterns so that I make my point clear. Again, based on this, whatever the measured signal is, the controller will drive the secondary source so that it creates something anti-phase to the measured signal. Now in this case, what it will create is some sound source which is anti-phase to the measured. It would be something like this, like this. Whenever, these two are, whenever these two add up, In the first case, so whenever these two add up, they have a net zero effect. In the same way, whenever these two add up, they have a net zero. Always, it is a destructive interference. This is how the basic, ANC system is. You have a noise source, you have some microphone in your space of interest, you have a controller plus actuator mechanism to drive the secondary source, which then dynamically based on the measured signal, creates an anti-noise.



This shows a typical ANC setup within an automotive car. Usually in automotive cars these days, advancements are made because you have so much vehicle electronics embedded in an automotive environment. With that, you can connect your ANC system very well because there are already a lot of electrical and electronic components. This integration of the ANC system becomes very seamless within an automotive environment. This shows one such example where you have a speaker at the front. Your passengers sitting in these seats are of interest. You want to cancel the noise near the ears of these passengers, so whatever the noise sources are, let us say the engine and this part is creating a lot of noise. You can have some mechanism, and this is your space of interest, your space of interest is you are not bothered about what is the noise here or here. You are not bothered about where the source is and what kind of noise it is creating. You are bothered about what is the noise level at the space of interest, which is what is the noise level where the listener, your target listener, is deciding. you keep your microphone where you would like to cancel. you would like to cancel the noise where the passengers are sitting, and hence you have an area of microphones closer to the passenger's location. And they are measuring the sound signal and then passing it on. This reference signal is then passed on to the adaptive controller. The controller or some kind of mechanism is there which is going to give this, this is the measured sound signal. From these speakers, generate the kind of anti-noise signals in the same way. this will help the back passengers, and the same controller can then react.

Can also be connected to a speaker system on the front, so in the car, you already have the speakers in the front and the back. Embedding an ANC system becomes very seamless. You have the microphones near the passenger, you measure the sound levels, you have some controller mechanism that then finally drives and gives the signals to the already installed speaker systems of the vehicles. Then they create these anti-noise signals which automatically cancel out and dim down the noise near the passengers. This also shows, that sometimes you have the speaker systems directly installed at the seats, very close to the passengers.

What are the major components in an ANC system? You obviously have various kinds of microphones to measure what the sound level is because that would become your input for generating the anti-noise. In the microphones, you can have two types of microphones: external microphone and internal microphone in an ANC system. The external microphone is positioned on the outside of the ANC system, and it picks up what the ambient noise is just before it reaches the listener.



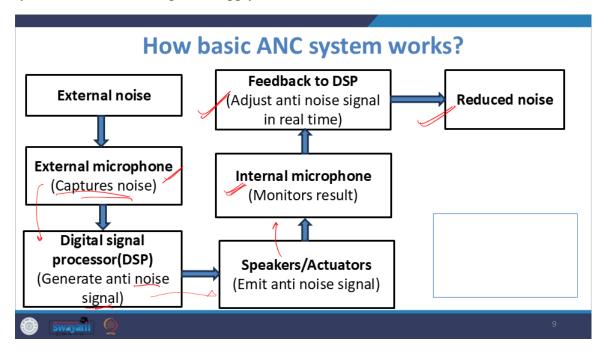
and the internal microphone is located very close to the listener's ear. Sometimes it is inside the listener's ear, in the case of, the listener is wearing headphones. You will see in the market; you have various ANC headphones that are available. So here, the microphones are located next to the ear in the earphones and all headphones. That becomes your internal microphone, which is located just at the listener's ear. This helps in adjusting the noise cancellation for better accuracy. So, what happens here is that, let us say, from one microphone, this is your listener's ear, and this is your secondary microphone. So, this is your external, this is your internal. Here, the two microphones work together. So mostly, you can also do the same thing with a single microphone, but for better accuracy and for better real-time adjustment, you use these external and internal microphones. External microphones measure the signal just before entering. What happens is that they measure that, this is the sound signal which is about to reach the speaker. Whatever the sound signal is. It is just about to reach the speaker. They would measure it and send it to the controller mechanism, and then the controller will immediately create an anti-noise and send it to the listener's ear so that the noise gets dimmed down. But, in the pathway from here, because the measurement to face here, so from here to here, the noise can have slight changes. You may not achieve perfect cancellation just based on what you measured earlier.

Over here, so you have an additional microphone just at the ear. When the cancelled noise comes in, this additional microphone records what is the cancelled noise at the ear. If it is zero dB or perfect cancellation, nothing happens. But if it is not zero dB, because whatever was measured before the ear, there could be some transformations from external to the ear, and perfect cancellation may not always be achieved. The internal one then measures whether perfect cancellation has been achieved at the ear or not. If perfect cancellation is achieved, then nothing is done. If it is imperfect, and there is still a noise level remaining, it will pass it as an error signal to the controller mechanism, which will tell you that some kind of adjustment needs to be done to the anti-noise so that perfect cancellation is achieved.

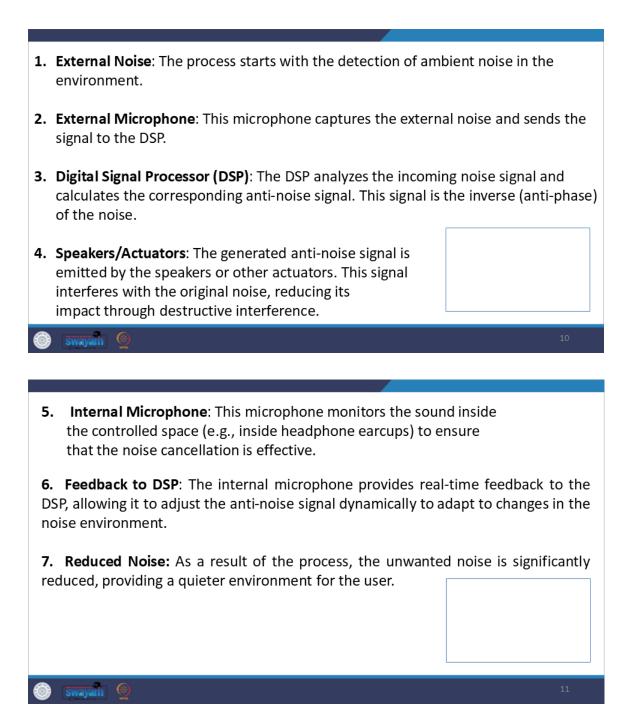
2. Digital Signal Processor (DSP): DSP is the brain of the ANC system. It analyzes the noise captured by the microphones and generates an anti-noise signal. This signal has the same amplitude but an inverted phase (anti-phase) relative to the detected noise. 3. Speakers: Speakers play the anti-noise signal to cancel out the unwanted sound through destructive interference. 4. Power Supply: The power supply, often a battery, provides the necessary energy to run the DSP, microphones, and speakers.

And you have, obviously, for all these processing and to create the various kinds of signals, you have a typical DSP, the digital signal processing, which detects the signals from the microphones and generates the anti-noise signals. It detects both the reference signal from the first microphone and the error signal from the second microphone, and based on this, it creates an anti-noise signal which is based on the feedback from both these microphones. And so on, then you have a speaker which will create the signal. The output signal that is passed, which is an anti-noise, is then passed on to these speakers. Which then create the anti-noise signal to cancel out the unwanted noise. You obviously, for all these systems to work, whether it is the microphone, your digital signal processors, your speakers, and all

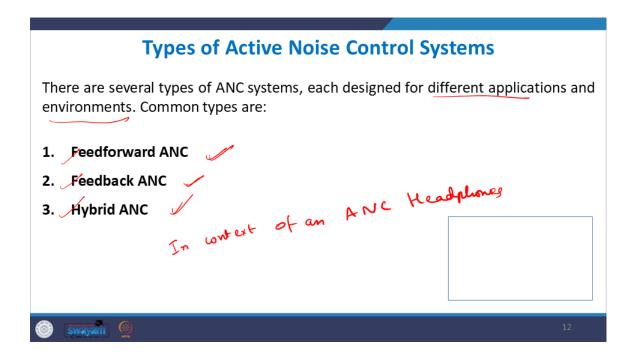
these, all these systems which work through electrical power, and a lot of circuits are there. you need some form of power supply, be it a battery or an external AC power source, but you need some kind of power supply.



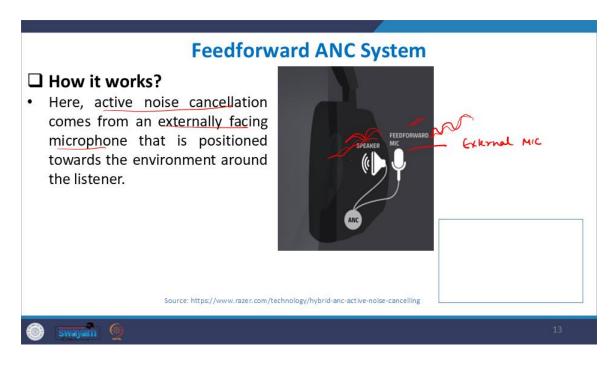
I have already explained this mechanism. Some external noise is there. Before reaching the listener here. You need the external microphone that, as soon as the noise begins, the listener hears a cancelled noise. Before reaching itself, you measure or capture the noise, and that is sent as a reference signal to the DSP, which creates an anti-noise based on whatever signal was measured by the external microphone. Then, this anti-noise signal is passed on to the speakers and the actuator mechanism, which then emit this anti-noise. Which reaches the listener's ear, but whatever noise is actually reaching the listener's ear, you have an additional component called the internal microphone. It monitors whether, whatever anti-noise was calculated based on these measurements, if due to some phenomenon, just the physical nature of this, space itself, there could be some diffractions, something which we might not have accounted for, due to which there is not perfect cancellation. It measures whether, a perfect cancellation is achieved at the ear or not. If it is not achieved, then it passes an error signal and feedback to this. Basically, it is passing this feedback again back to the DSP based on whether perfect cancellation is achieved or not. And then this DSP would readjust its anti-noise signal in real-time to get a perfect cancellation. All of this I have explained, what is the process like, and how the two microphones work in sync to reach perfect cancellation.



whenever we have a varying environment, because of these two microphones that are working in sync, whatever is the anti-noise, it can adjust, dynamically in time based on whatever is the error signal being passed from the internal microphone.



Now, there are various types of ANC systems. In fact, if you go into the literature of ANC, you would have countless various kinds of ANC systems, but we will just sort of decompose it down to three basic mechanisms which we will briefly see in this course. And based on what kind of application and what kind of environment your ANC is applied, you can take and pick one of these types of ANC systems. It is the feed-forward, feedback, and the hybrid ANC.



Let us see the feed-forward ANC. Let us discuss them in the context of an ANC headphone, ok. you have an active noise cancellation headphone which is quite common nowadays in the commercial markets, you have these headphones which cancel machinery noise or whatever it is because you use the headphones for listening to music, etc., and any other machinery noise, construction noise, etc., you would like to cancel it out. You can use these ANC headphones. Let us study these three systems in the context of this. the very first one is the feed forward. what happens here?

Active noise cancellation comes from an externally facing microphone. So here, what you have is I told you about an external microphone and an internal microphone. So here you only have this, feed forward mic or the external microphone. This is the microphone which is facing the ambient environment around the listener. It is not located exactly in the ear, inside the ear. This is just before the sound is about to reach the listener. Before that, it is present. It measures what is the sound which is approaching the listener. It measures the sound approaching the listener. And based on that, it creates an anti-noise signal which then plays through the speaker into the listener's ear. Whether a perfect cancellation has been achieved or not, you never know because there is nothing like an error signal. It just measures what is the sound approaching, based on that it creates an anti-noise which goes into the listener's ear, and then this sound which approaches, hopefully, it has canceled it perfectly or imperfectly, we don't know.

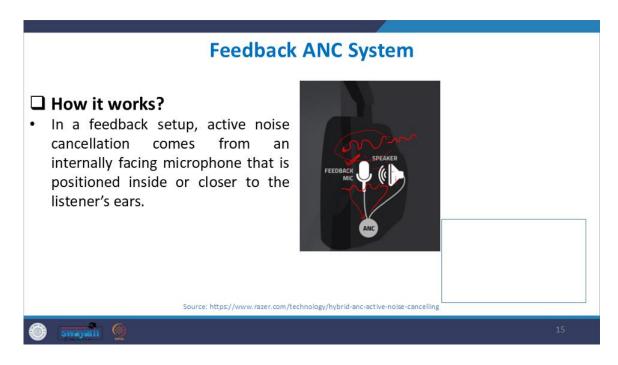
Feedforward ANC System

- Hence, the microphone hears the noise before the person does. The ANC then processes the noise and creates the anti-noise before sending the resulting signal to the headset speaker, preventing it from reaching the listener's ears.
- Advantages: Good at canceling high-frequency noises before they reach the ear.
- **Disadvantage:** Less effective with unpredictable or rapidly changing noises.



What are the advantages and disadvantages?

This feed-forward ANC system is good at canceling various high-frequency noises and less effective with unpredictable or rapidly changing noises. Suppose it is a rapidly changing noise, specifically, for example, in a location where the noise is varying, having a lot of spatial variations, let us say the noise which was approaching here due to some diffractions that are happening in the environment itself. Here, due to some diffraction, it might have changed. What is achieved here? It might have changed, so in that case, your perfect cancellation will not be achieved. In these unpredictable environments, it might not work.



The second type of ANC system is where you have only the internal microphone or the feedback mic. Active noise cancellation happens with this internally facing microphone. what happens is the noise approaches and, reaches the listener, and the microphone just next to the ear detects it and then it drives the DSP mechanism to create the anti-noise. And due to this, what happens is that in the feed-forward, suppose it is a very dynamic environment with a lot of structural elements and a lot of unpredictability in the sound wave variation. Perfect cancellation may not always be achieved because you never know that, the sound that was measured at time t equals to 0 and then it reaches the listener's ear at t equals to whatever, let's say t time. Whether that is the same and whether the reference signal, whatever, anti-noise signal is there, is able to cancel it perfectly, you never know what happens in the ear because you have no microphone there in the feed-forward. But in

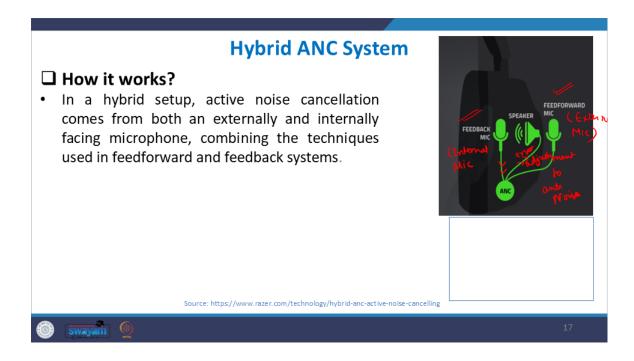
the feedback, you have the microphone right at the listener's ear. And it measures what is the noise actually present in the listener's ear and, based on that, it sort of creates the feedback mechanism and it drives the speaker to create an anti-noise. Now, if you see here, the noise has reached over here, and then what has happened? Then, based on that, some reference signal is generated to sort of create an anti-noise signal. What if, by the time this noise is reached here and the next set of wavefronts are something different? You measure this reference signal, you send it up, and a new signal is created, anti-noise to it, but by the time it is created and reproduced, a different form of signal has reached the ear. Which means that here there is always a lag. You first measure, create the signal, and generate the sound. There is always a time lag. If you do not have, a steady environment, then in that case, this lag is quite visible. Similarly, if you have high-frequency sounds,

Feedback ANC System

- The feedback microphone detects noise that has made it into the listener's ear, working with electronics to remove the noise from the signal while also adding a signal that cancels the noise that's getting into the ear.
- Advantage: Can correct errors in the anti-noise signal dynamically, improving effectiveness over a range of frequencies.
- **Disadvantage:** May have a slight delay in response and can sometimes struggle with high-frequency noise.



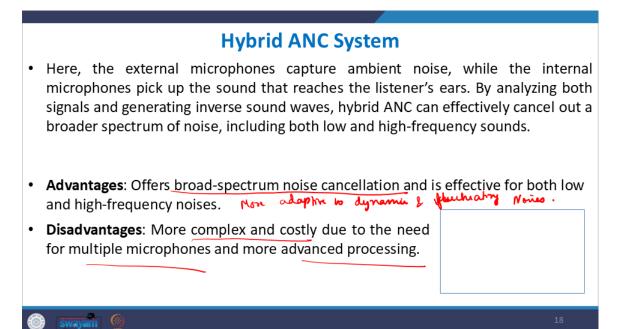
which means rapidly fluctuating sounds with respect to time, again, this lag from the measurement of the signal at the ear to the generation and the creation of the anti-noise, this lag becomes very evident at higher frequencies. Hence, this delay in the response is there, especially at the high frequencies. But suppose you have a low-frequency noise, which is very steady, especially when you have a low-frequency and very steady noise. Then this system will correct the errors and give you more perfect cancellation compared to the feed-forward system.



Then we have the hybrid system. when I discussed the working of the ANC system, I had mentioned both the internal and the external microphone, that is the mechanism used in the hybrid ANC system. You have both the feed-forward mic or the external mic and the feedback mic or the internal mic. This is the headphone, so it measures the noise just before, reaching the ear. What is the ambient noise around the ear? Then the refresh signal is created, and by the time that noise reaches the ear, cancellation happens. But there is also an additional mic present at the ear to see whether the perfect cancellation has happened or not. If it has not, it carries the error signal and adjusts the anti-noise signal dynamically by giving this feedback back to the NC system. Whatever is there and some kind of error signal and adjustment happens to the already created anti-noise signal. In the dynamically changing environment, and these environments, they can quickly adapt and change the anti-noise signal.

The advantages are that now they can work both for the low because they are combining both these features, both low frequency, high frequency, so a much broader spectrum of noise cancellation is achieved, and they are better, more adaptable to dynamic, and varying, fluctuating noises, dynamic environments, and fluctuating noises. But they are more complex and costly because they need multiple microphones, so more costly and more advanced processing. That is, obviously, the disadvantage, but they have better adaptability

and better results in almost all the frequency ranges. With this, I would like to close the lecture on ANC.



Thank you.

