NOISE CONTROL IN MECHANICAL SYSTEMS

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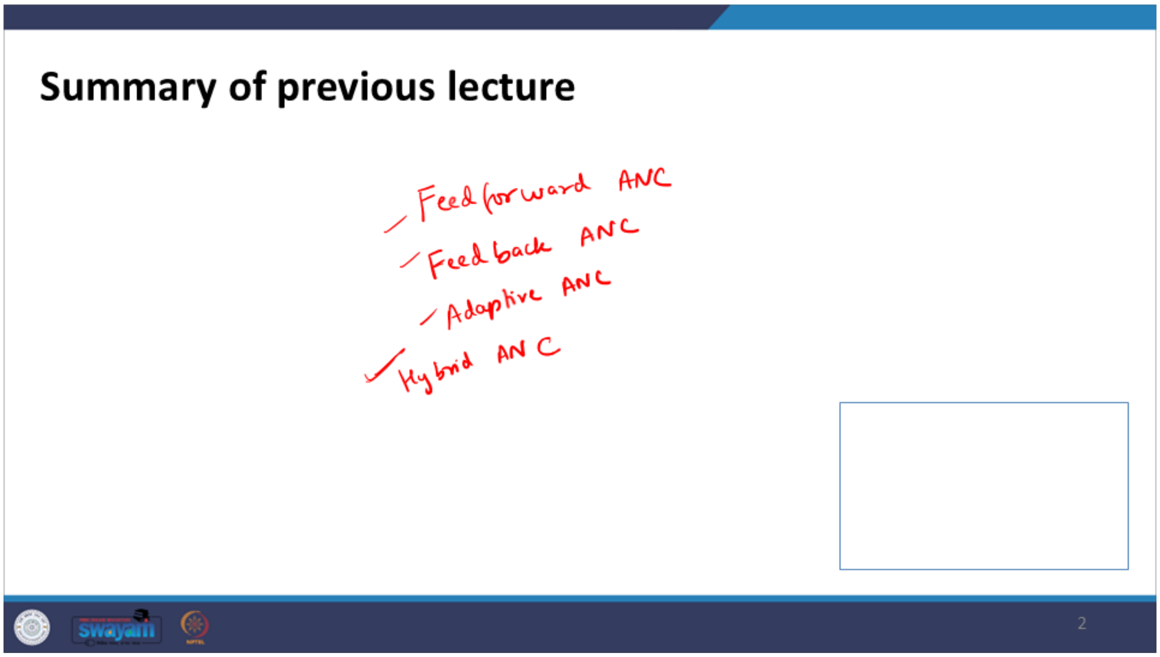
Week: 7

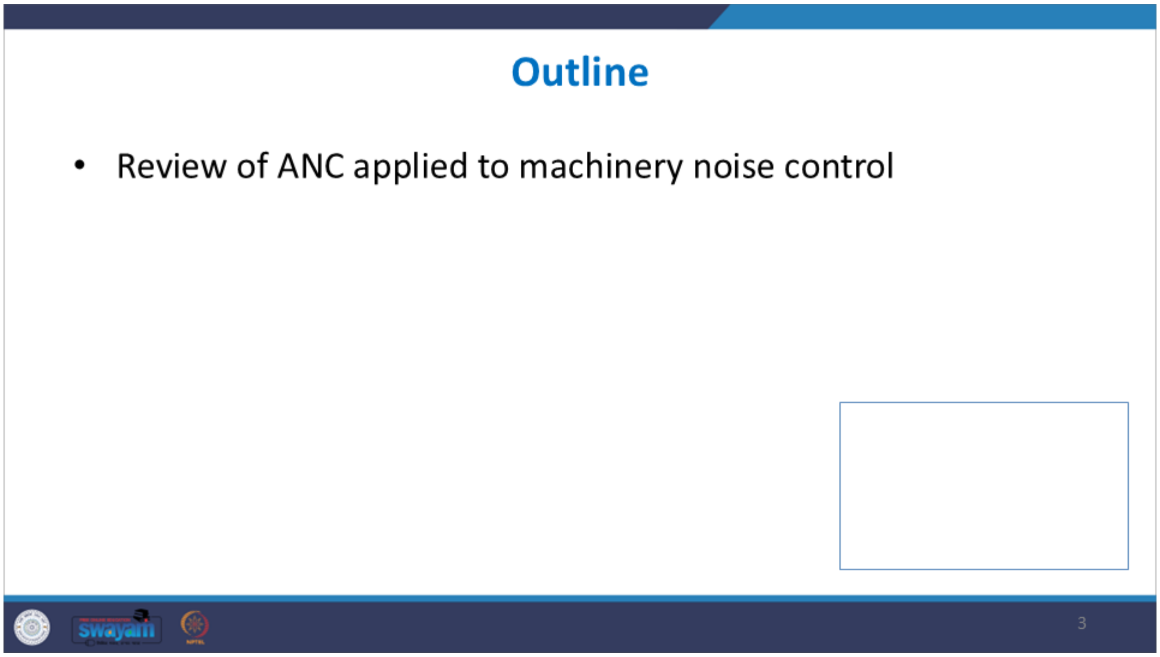
Lecture: 31

Lecture 31: Active noise control: 2

Hello and welcome to this lecture on active noise control in the course on noise control in mechanical systems. So, this is our continuation of the previous lecture where we began the strategy for active noise control, where active noise control was introduced, what it is, and what the basic architecture of an ANC system is, the basic components inside an ANC system, and some of the common types of ANC systems such as What do you mean by a feedforward ANC? Then a feedback ANC. Okay. And similarly, there are things such as adaptive ANC and hybrid ANC. Okay, so these kinds of things.



So, in today's outline, what we'll do is that in this lecture, we'll review a few of the ANC techniques that have been applied in the field of machinery noise control, you know, because this course is based around noise control in mechanical systems. So, let us see some case studies from the published papers.

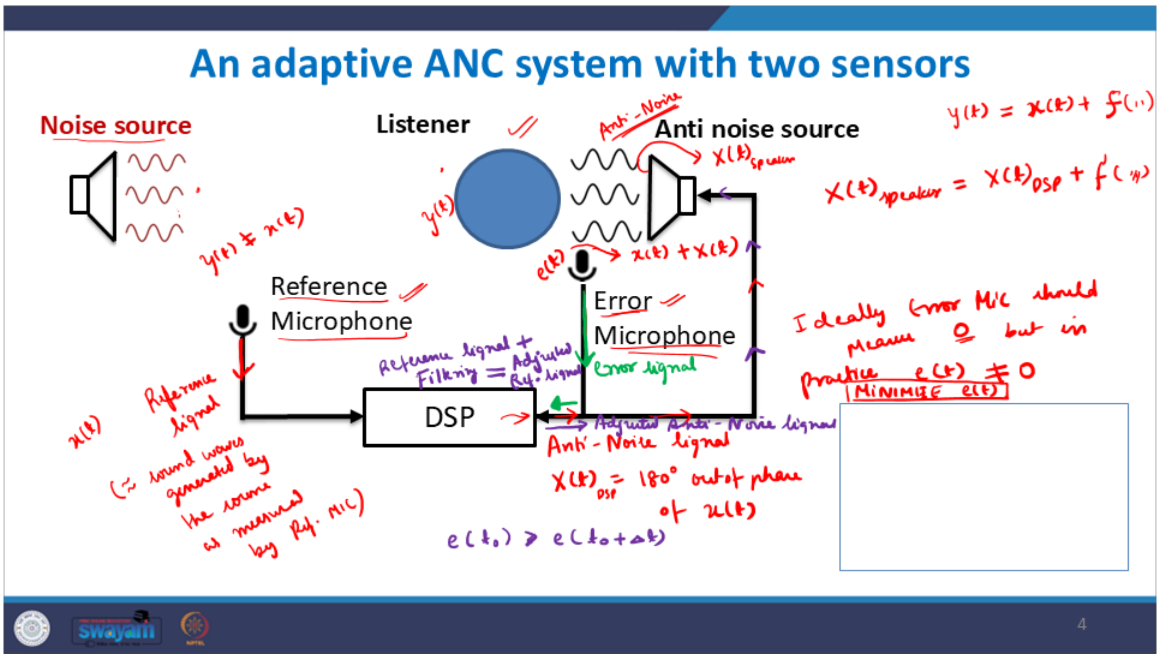


So, before that, we'll review two papers where they have used an adaptive ANC system. So, what is an adaptive ANC system? It is very similar to the hybrid ANC system which was introduced in the previous class, and what happens there is that there were two microphones. So, one microphone was facing the ambient environment near the source, which then gave the feedback. So, from the ambient, from the first microphone, a reference signal was generated and given to the DSP, which then provided for. So, this is a typical architecture of an adaptive ANC system that has two sensors. So, what happens here is that let us say we have some noise source, and it is emitting some sound waves. So, near the noise source, slightly away from the listener, there is a microphone that faces the noise source. And this microphone Know or the feed-forward microphone in the previous lecture, I had called it the feed-forward microphone, or you can call it the reference microphone. It first measures the sound signals generated by the source. Okay, so it is located near the source and then it creates a reference signal. So, this reference signal then means that, you know, it is sort of a representation of the noise measured near the source. This is what the reference signal is. It corresponds to the sound signal, the sound waves generated by the source as measured by the source as measured by the reference microphone. So, that becomes your reference signal, which then reaches the DSP or the digital signal processor. The digital signal processor, through its various kinds of electrical circuits and electronics, is able to create an anti-noise signal. So, finally, this generates an anti-noise signal. based on the reference signal that has been created. So, for, you know, a very simplistic setup, suppose we had some reference signal X(t), let us say, then it will create something which is antiphase of X(t) or simply, you know, whatever is X(t), you know, it would be antiphase of it or simply negative, given that if X(t) can be represented as a sine wave. It will give you an anti-phase signal, let us say capital X(t), which is, you know, 180 degrees out of phase of the reference signal X(t), and then this signal would be sent, you know, to the you know, the secondary loudspeaker or the anti-noise source, which will then create this reference signal out of phase and hopefully, these should cancel each other near the listener's ear, and there should be noise reduction at the listener's ear. But, you know, that does not always happen, you know. So, why do we need the error microphone? You know, first of all, whatever are the, whatever is the measurement here we can never be sure that the signal measured here is the same as what signal actually reaches the receiver's ear. Because, you know, the signal goes through so many distortions along the path it may undergo. First of all, you know, if it is a spherical sound source, anyways, as it goes away, if it is a spherical or a cylindrical sound source, as with the distance and the propagation, the amplitude anyways attenuates the amplitude decreases. At the same time, due to the various structural objects present in the pathway, there could be diffraction and scattering. So, due to these various path effects, diffraction, and scattering effects, the signal actually reaching the listeners here could be slightly distorted compared to what was actually measured by the reference microphone. So, the reference signal may not exactly. So, the signal reaching them, you know, the receiver, let us say this is y(t), this is the signal actually reaching the receiver. This y(t) could be some kind of x(t) reference signal plus some kind of, you know, additional factors. Let us say some factors, f, you know, which account for the various diffractions, distortions, attenuations, whatever it underwent while reaching the receivers here, ok. So, some factors which could be a function of various things, ok. So, this is not exactly x(t), but x(t) with slightly distorted elements. So, that is why; to make a more adaptive and accurate system, a second microphone is installed very close to the listener, that is called the error microphone.

What it does is, you know, it measures what is the sound actually reaching the receiver. So, it will not measure the sound actually reaching the receiver, what it will measure because by the time what happens, the reference microphone is measuring the sound, sending the reference signal to the DSP, and the DSP is generating the anti-noise signal which is being emitted here through this loudspeaker. You know, near the listener's ears, the cancellations have happened. So, near the listener's ear, what is it? It is x(t) plus capital X(t). OK, the two have now. If suppose there was perfect, you know, out of phase, if suppose there was no distortion, whatever signal was measured at the reference microphone, exactly that same signal reached the listener's ear. Then x(t) plus capital X(t), given it is everything can be represented as a sinusoidal wave, would ultimately cancel each other out because it is just the opposite of the previous signal, and we should get 0. So, ideally, the error microphone should measure, you know, 0, but situations are never ideal, so it is never 0 actually. But in practice, it is never 0. You know, the signal measured at the error microphone, let us call it y(t), okay? y(t) is the signal reaching the receiver, so let us call the signal at the error microphone as small e(t). This is small e(t), which is what?

$$\mathbf{e(t) = x(t) + X(t)}$$

So, in practice, this signal measured at the error microphone is never equal to 0, but the purpose is that for a good ANC system, it should try to minimize the error signal. So, this becomes sort of like the objective that you should make the error signal as less as possible, so that, you know, as near perfect cancellation can be achieved. Okay, so it measures this, and because of the various distortions that might happen or some other kind of, you know, noise over here, you know, it's never zero. So, this error microphone will sense what is the error, you know, between achieving a perfect cancellation and actually having something. So, whatever is the error it is measuring, you know, so this should be minimized to zero. So, this error microphone will then send up a new signal, which is the error signal. Okay, so this first microphone sends the reference signal. This is going to send you the error signal, okay? This will also go to the DSP, and now the digital signal processor has got the reference signal and the error signal. And if it is a very fast processing, you know, circuit, the time lag between the reference signal and the error signal should not be very much. If it is too huge, then obviously it does not work for the high-frequency components. Like in the previous lecture, we saw that for, you know, if it is feedback or this kind of a system, then for high frequencies where the fluctuations are very fast or in a highly rapid environment, then the time lag, the short time lag between the reference signal and error signal, will make an impact. But, in a more steadier environment, this very fraction of time lag between the different signal and the error signal does not matter. And then, when the DSP is all this information, what it does is once it receives the error signal, it does what? It takes the original reference signal and does some filtering to it in order to reach a new adjusted reference signal, and then from once then from the second time onwards, you know, a new adjusted anti-noise signal is sent to the, you know, speaker for, you know, creating. And then the speaker creates the, you know, after adjusting the reference signal, it creates a new, you know, adjusted reference signal, which is then emitted near the listener's ear. And once again, the error microphone will catch what is the new error signal, and hopefully, the error signal at the first-time instance, okay? This is like the assumption that the error signal at some time instance t should be greater than the error signal at the very next time cycle and so on. So, slowly and slowly, with each pass, the error signal gets minimized, and the system adapts to give as perfect cancellation as possible.



So, these are the terminologies I already discussed through the diagram: what is reference microphone, error microphone, the reference signal, and the secondary source or the secondary loudspeaker, also called as the anti-noise speaker.

Some important terminologies

❑ **Reference microphone:**

- The reference microphone is placed near the noise source. It picks up the unwanted noise (the "reference signal") before it reaches the listener.
- The signal from this microphone is used to analyze the characteristics of the noise.

❑ **Error microphone:**

- The error microphone is placed close to the listener or at the point where noise cancellation is needed.
- The signal from this microphone helps to refine and adjust the anti-noise signal in real time to improve the cancellation effect.

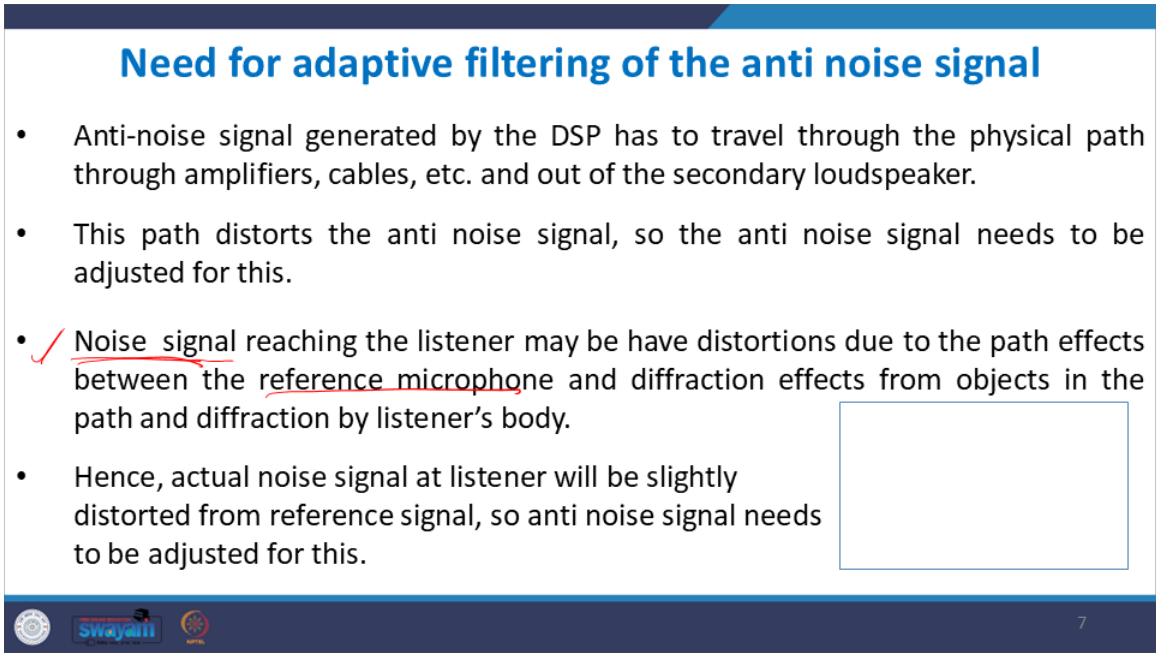
❑ **Reference signal:**

- The reference signal is the noise signal captured by the reference microphone.
- This signal serves as the input for the ANC system.

❑ **Secondary source or Secondary loudspeaker:**

- The secondary loudspeaker (or anti-noise speaker) is used by the ANC system to emit the anti-noise signal.
- This signal is generated to be the exact opposite of the reference signal, canceling out the unwanted noise when the two signals meet.

Now, I already told you that, you know, once we receive no reference signal, in the first pass, we just create an anti-noise signal. Then, in the second pass, once the error signal is received, we readjust our anti-noise signal and emit the new adjusted anti-noise signal, or the new sort of adapted anti-noise signal. So, we are doing some filtering to the already existing anti-noise signal. So, what is the need to do this kind of filtering? Why is the need to do this kind of adjusting? First of all, I already explained to you that, you know, there could be. So, the first So, the first reason could be that the noise signal that was measured at the reference microphone may not be exactly the same noise signal that actually reached the receiver. Because the anti-noise signal that was initially created is directly based on the reference signal, hence, perfect cancellation may not be achieved, you know. Whatever was measured initially will go through some distortions to finally reach the receiver's ear, and hence, the adjustment needs to be made to the anti-noise signal to get that cancellation effect, perfect cancellation. The second is that, you know, once the anti-noise signal is generated, if you can see here, so there is first the signal from here till here will have certain distortions, so y(t) would not be exactly the same as x(t) in the same way. You know, whatever adjusted anti-noise signal is sent from the DSP, okay? So, let us say this anti-noise signal is capital X(t), so this is the capital X(t) from the DSP, okay? And then there is some capital, and this is the actual capital X(t) which is emitted at the speaker. Okay. So, because these are all, you know, internal electrical components, they have their own resistance and their own transfer functions and frequency responses. So, obviously, based on, suppose, you know, if you are using very long cables or poor-quality cables, poor quality speakers, or amplifiers, what exactly the processor is generating, you know, this anti-noise signal X(t) at DSP. There could be some distortions while this wave or this signal is passing through the various cables, amplifiers, and to the speakers, and hence, the actual signal that is being emitted outside may still be distorted. So, X(t) at the speaker could again be, you know, the X(t) that was actually created at the DSP plus some distortions. So, some additional distortions that have distorted it. So, these are the various distortions, and hence the need for an error microphone and adapting based on every pass, okay. So, these are the different. So, that is why, you know, the anti-noise signal, some filters are required. So, that it can adapt in real time to the changes that the sound waves are undergoing.

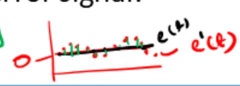


So, one of the very common, you know, filtering techniques that is used in these various ANC systems is the filtered X least square least mean square algorithm. So, what we want to do is that, you know, suppose we want our error signal, our, you know, anti-noise signal, expected anti-noise signal should completely Or, you know, the two signals should be exactly out of phase for perfect cancellation. But suppose, you know, our set standard is that, you know, there should be a flat response. But actually, the error signal that we are getting, let us say, this is our actual error signal that we measure at the reference microphone. And this is the zero level. Okay, we want it to be at zero level, but it is not. So, this is our error. This is our expected graph, but we are getting some error as e(t). So, this is, you know, the true value of e(t), and this is the actual e'(t), if you call it as the actual measured error signal, then you want to minimize the difference between, you know, the expected error signal, which should be 0, and the actual measured error signal, which is not 0, but some value. So, to minimize the difference between the two graphs, we are trying to do like a curve fitting, you know, the typical least mean square, which we use, you know, so that, you know, that the, because it is a digital signal processing. So, the, you know, the sum of the least mean squares means that the sum of the squares, you know, you have to minimize the, you know, the sum of the square of these distances, okay, these distances. Which means, so a typical LMS algorithm means that we minimize the whole square because, you know, sometimes it will be positive and negative. So, we are trying to minimize the square of the distance between the two graphs. So, this means the least mean square algorithm, but then, to top that, we have the filtered x. Again, in this course, I am not giving you the detailed algorithm because that would be out of scope for the current you know, course, but just a brief overview of, you know, just like we use, you know, how we minimize the root mean square errors here, we are minimizing the mean square errors between the two, you know, expected error signal and the actual measured error signal, and with each pass, are you know, whatever, we try to minimize this error or the square of the distance the square of the difference between the two signals, and then filtered x is an additional, you know, advancement to the LMS algorithm where, you know, something happens, some adaptive filters or weights are added, you know, these weights could be dependent on the frequencies and the time. So, this is also added. So, on top of the fact that we have to minimize this, we also add sudden weights to the, you know, sort of received the weights to sort of this, you know, error signal, sorry, the weights to the anti-noise signal so that we can minimize the error, ok.

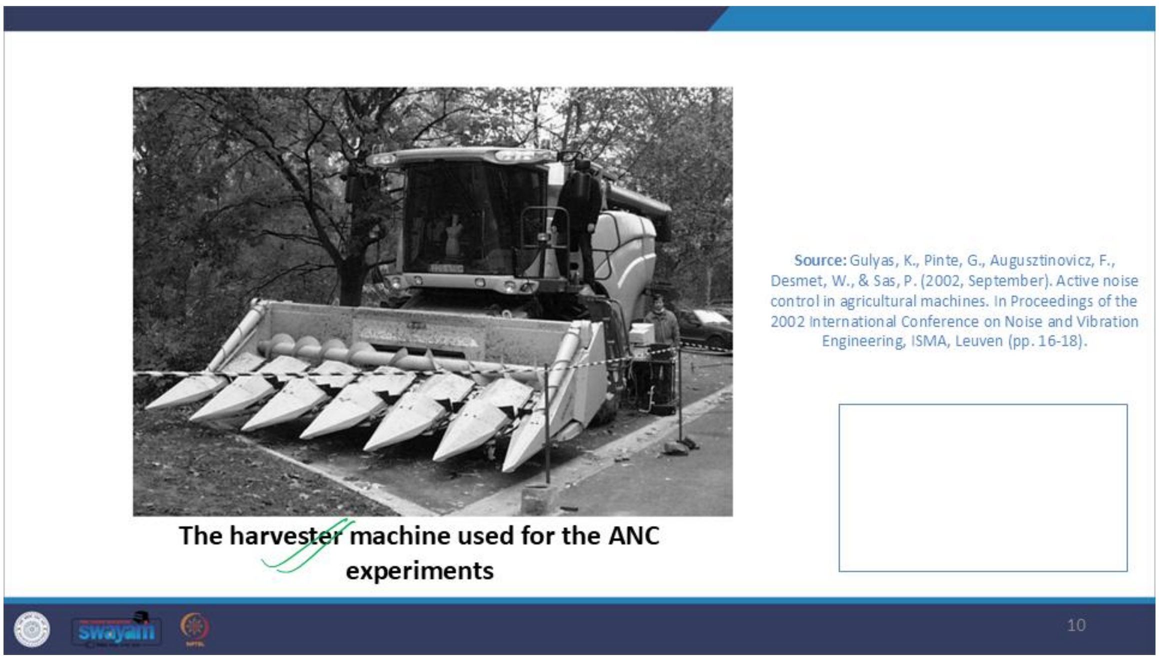
Filtered-X least-mean-square algorithm

- The **Filtered-X Least Mean Squares (Filtered-X LMS)** algorithm is an adaptive filtering technique widely used in **active noise control (ANC)** systems.
- It extends the traditional LMS algorithm by compensating for the distortions by the **secondary path**—the physical path through which the anti noise signal travels.
- It uses an estimate of the secondary path to filter the reference signal before updating the adaptive filter's weights.
- It further filter the signal based on error signal.

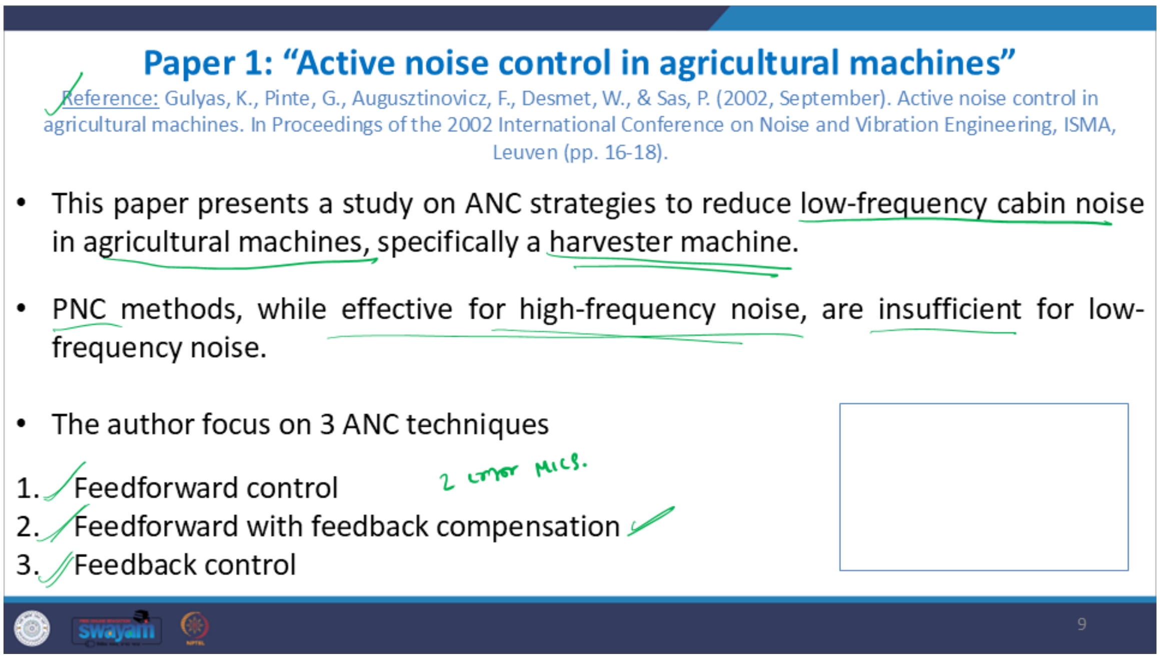
$w(f,t)$ + MINIMISE $E[(e'(t) - e(t))^2]$



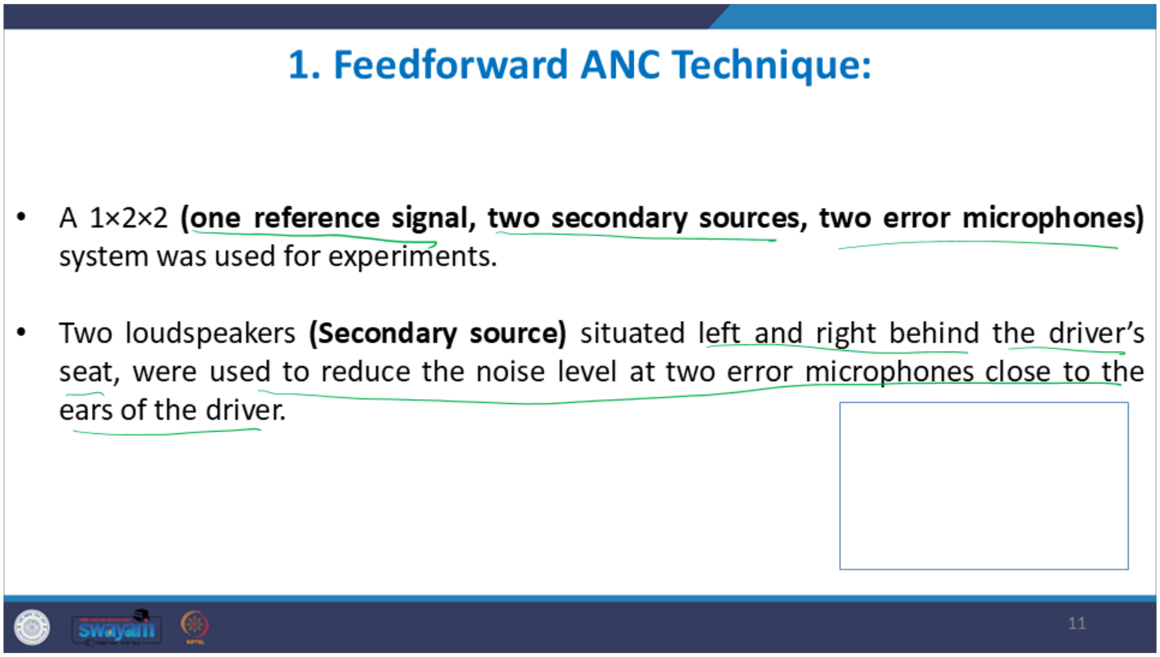
So, with this, you know, let us see, you know, two papers, one is the active noise control in agricultural machines. published in 2002, and this is the reference of the paper here. So, what they have done in this paper is that they are adapting, you know, some ANC strategies to reduce low-frequency cabin noise in agricultural machines, and then they have taken the case study of a harvester machine, which I can show you here.



This is the harvester machine where ANC was applied. It generates a lot of noise, and obviously, you know, here the objective is that whatever is the operator who is working on these machines, near the operator, we should be able to reduce the noise, which is mostly low frequency. So, because the PNC methods, they are ineffective for, they are only effective in the high-frequency zone, they were not, they were found to be inefficient to reduce the harvester machine noise. So, ANC strategy was adopted. The authors have seen how three different types of ANC strategies are working for this particular case study. So, first they used the feedforward control, then the feedforward with feedback compensation, and then the feedback control. So, here the feedforward control and the feedback control are the same as what I discussed in the previous lecture, where you know you just use one microphone in feedforward, which is the reference microphone, and generate your signal in the feedback, where you only use the error microphone. So, instead of having a microphone near the source, you have a microphone near the listener's ear. Okay, and only that signal is used, and once you combine both of them together, you get this second kind of technique where two microphones are essentially used, both near the source and near the receiver, in order to adapt. So, all three versions were applied.



Okay, so here, for the feedforward ANC technique, a 1x2x2 system was used for the experiment. It was tested using one reference signal, two secondary sources, and two error microphones. Two loudspeakers were used, situated on the left and right behind the driver's seat, and they were used to reduce the noise level at the two error microphones close to the ears. So, the advancement they made is that instead of having a single error microphone, they are using two error microphones in order to minimize and make the system as accurate as possible, okay.

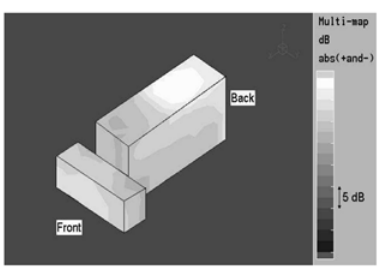


And the filtered-x LMS algorithm was used for this, you know, to adapt and create the anti-noise signals and make it adapt in real-time to the changing sound signals, okay. So, what was happening is that, you know, here, the quality of the reference signal determines a lot of the success of the feedforward control. And hence, good coherence and consequently good reference signals can be found only on certain parts of the machine because the main challenge was to obtain a good quality reference signal, as it is a complex machinery, the harvester machine. So, at different points of the harvester machine, different kinds of sound signals could be present, as in the sounds. The SPL distribution across the machinery may be quite uneven, and the source could be very directive in nature. Hence, where should we measure the reference signal? That is what the authors faced as a challenge, to create a good quality or more representative reference signal, okay.

- The well-known **filtered-X Least mean squares algorithm** was used as adaptive algorithm.
- This algorithm processes the reference signal, coherent with the noise at the error microphones, and the signals from the error microphones to calculate the appropriate anti-noise signal for the secondary loudspeakers.
- The quality of the reference signal determines the success of the feedforward control.
- Good coherence and consequently good reference signals can be found on those parts of the machine, which are responsible for large parts of the noise spectra inside the cabin.

So, to find out the location where we should measure these reference signals, they had done sound intensity scans, or simply we call it noise mapping, to find out the distribution around the harvester machine and the zones of high sound intensities. Based on that, they selected the zones or the places where they would measure the reference signal.

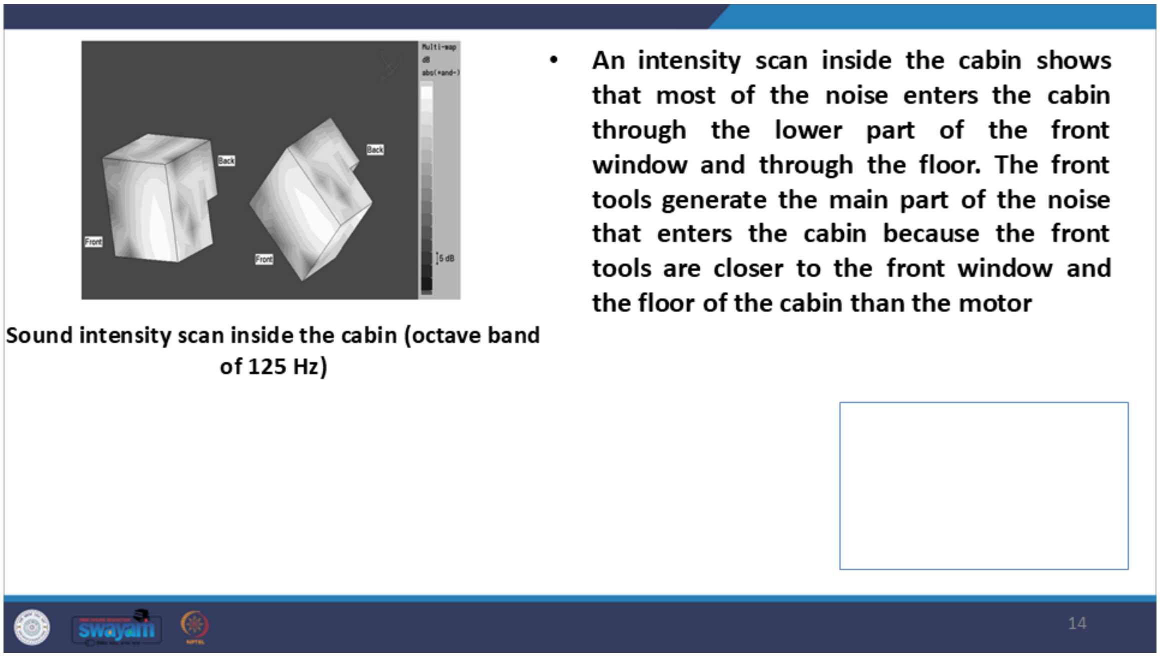
- These main noise sources were searched on the basis of sound intensity scans around the whole harvester machine and measurements of the coherence between the radiated noise and the noise inside the cabin.



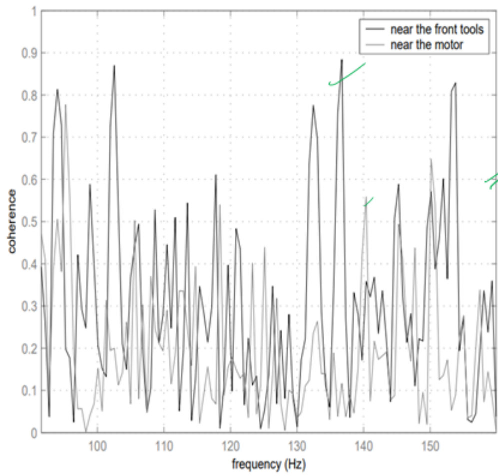
Sound intensity scan around the whole machine (octave band of 125 Hz)

- **The high sound radiation at the front and the back of the machine indicates that the front tools and the motor are the main sources of the low-frequency noise, generated by the machine.**

This shows the intensity scans of the various compartments within the machine.



This shows the measurement of the coherence near the front tools and the coherence at different places in this particular figure. So, this is near the motor, and this is near the front tools. So, good coherence was found near the front tools. At various kinds of frequencies, which indicates that these front tools are responsible for the main part of the noise inside the cabin.

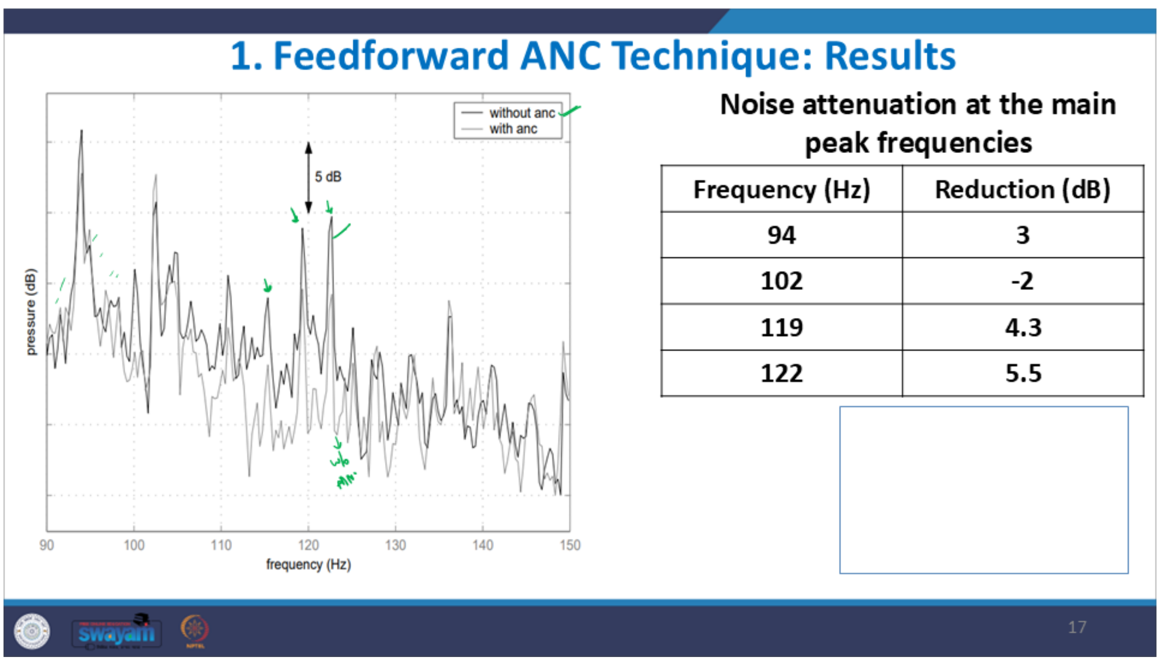


- **A measurement of the coherence near the front tools and the coherence at a different place (near the motor) is represented in figure.**
- **The reasonably good coherence near the front tools, especially at some important peak frequencies of the noise spectrum in the cabin, indicates that the front tools are responsible for the main part of the noise inside the cabin.**

So, once the main noise source was identified, the reference signal was collected, and this algorithm was implemented. So, the reference signal was collected where they found the best coherency, and they selected that, as you know, they chose that as the location for collecting the reference signal.

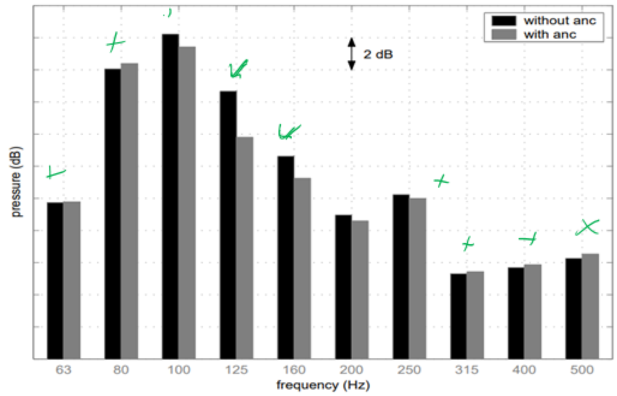
- Because the signal from the microphone under the floor had the best coherence, it was chosen as the reference signal for the feedforward control algorithm.
- This reference signal was filtered by a high (> 70 Hz) and a low (< 300 Hz) pass filter to avoid control effort in the non-problematic frequency range.
- The convergence factor of the filtered-X LMS algorithm was set to 0.008

So, this shows, you know, the result of adapting the feed-forward ANC technique. So, here the solid line, if you can see, is the SPL or dB versus frequency distribution when no ANC was working. So, you had a high, you know, dB versus, you know, frequency, but when ANC was adopted, these lighter lines, okay, this one indicates without the ANC. So, as you can see, you know, the sort of performance of this ANC was very frequency-dependent. So, at the low frequencies, you do not see at many of these frequencies much difference between the two algorithms, but in certain frequencies like these, there is a high difference, so a high level of you know, the ANC is found to be very effective in certain frequencies. So, this shows, you know, typical frequencies and the reduction achieved in decibels, okay.



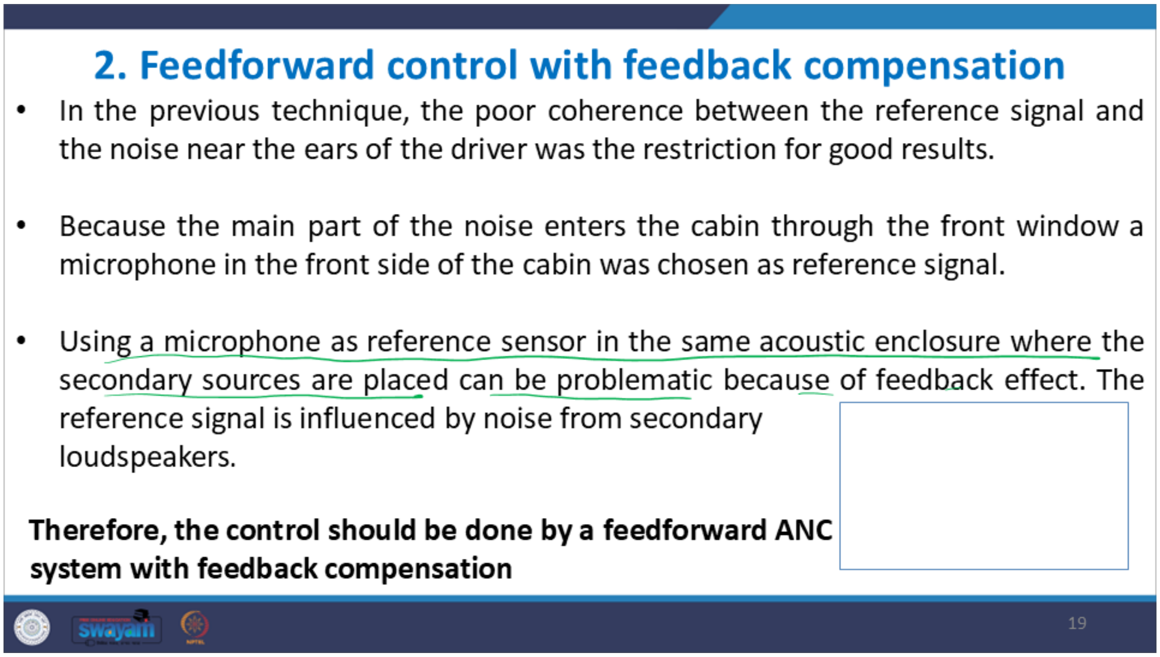
Frequency (Hz)	Reduction (dB)
94	3
102	-2
119	4.3
122	5.5

This is another representation of the same result; you can see that here you have more effectiveness, okay, in these two areas, whereas in the rest of them, the feed-forward was not as effective.

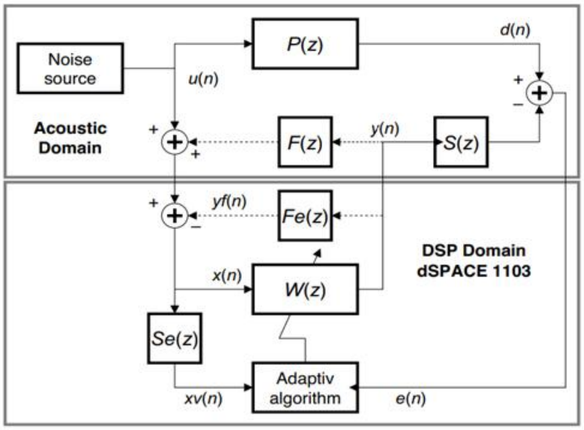


Noise level in 1/3 octave bands with and without feedforward control (microphone under the cabin as reference signal)

So, then feed-forward control with feedback compensation was, you know, implemented. So, what is it in the previous technique? Because you know the results were not up to the mark, as only in some frequencies, a couple of the frequencies, some good amount of decibel reduction was found, and then for the rest of them, effectively, there was no reduction in levels. So, the new technique was implemented. So, here, you know, using a microphone as a reference sensor in the same acoustic enclosure where the secondary sources are placed could become problematic, you know, because of the feedback effect.



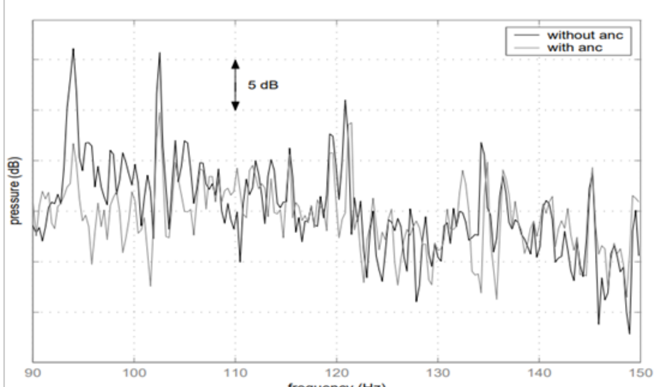
Anyways, this shows the feed-forward ANC system with feedback neutralization because what they found was that wherever they were measuring the reference signal, they had an acoustic enclosure. You know, that was distorting their result, and that is why some compensation had to be made in order to make the results better. Again, I will not go into the details of this method but rather just focus on what algorithm worked for them.



Feedforward ANC system with feedback neutralization

So, what they found was that once they adopted some additional compensation for the acoustic feedback that is being received from the acoustic enclosures where, you know, in the cabin where the speakers were being played. So, once that compensation was done, there was a remarkable improvement overall in the reduction.

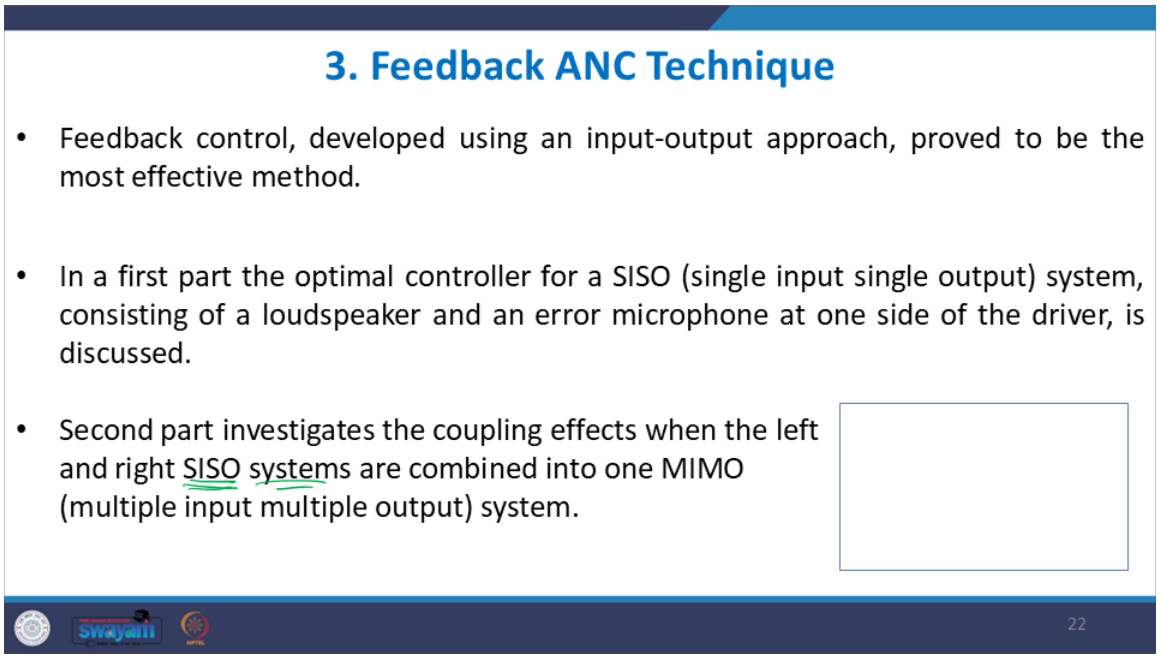
2. Feedforward control with feedback compensation



Noise attenuation at the main peak frequencies

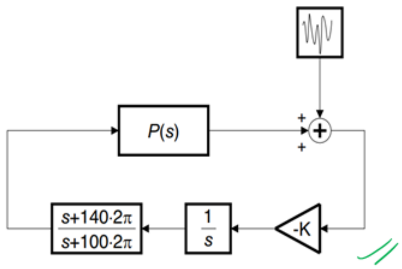
Frequency (Hz)	Reduction (dB)
94	10.1
102	5.9
119	1.9
122	2.3

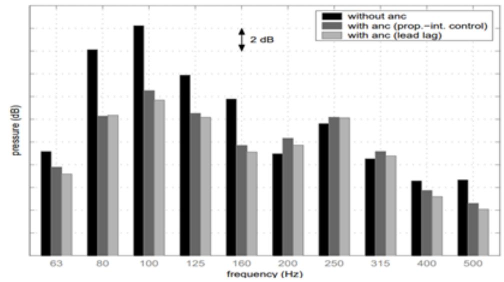
Then, feedback ANC was used with single-channel, single-input, single-output systems like this. And then, multi-channel systems were also used in this, you know. So, with increasing the number of channels for receiving the signals as well as processing and sending the anti-noise signal.



So, see, the single-input, single-output simply means that, you know, sort of, the entire ANC or the entire thing is being done just across one channel. So, you have one reference signal, one error signal like that, and you are doing the adaptation.

SISO system:





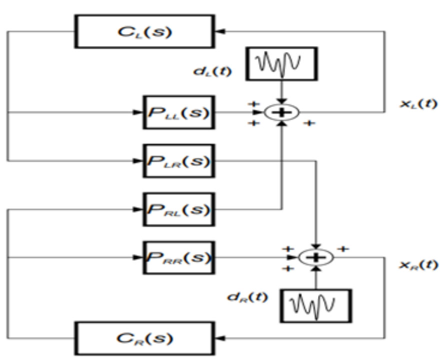
Block diagram of the SISO feedback system with a proportional-integral controller and a lead lag compensator

- **The noise level can be reduced with a maximum of almost 7 dB in the low-frequency octave bands.**

But in the multiple-input, multiple-output systems, what you have is multiple channels doing the same things. So, from various other locations, you are getting various kinds of reference signals, various kinds of error signals, you are doing all of that, and you are getting.

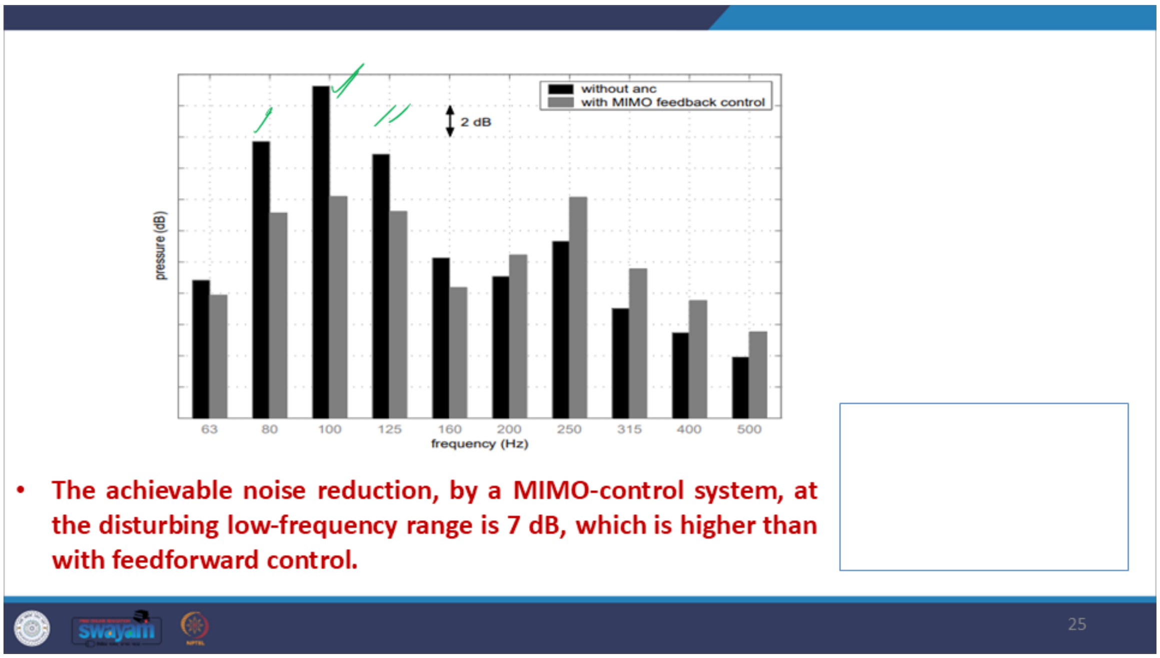
MIMO system:

- In this part, the two controllers of the right and the left ear SISO-systems are combined into one controller of the total MIMO-system



Block diagram of the MIMO feedback system

So, when you increase the number of channels for processing, you know the ANC algorithm, so you receive more data from the reference microphone. So, you have more than one reference microphone; you have multiple reference microphones and multiple error microphones. Then, the DSP processes the signals of the corresponding reference and error microphone pair in the multiple channels, and it generates the ANC, the anti-noise signal. Then, in that case, the you know, performance is much better. That is what they had found.



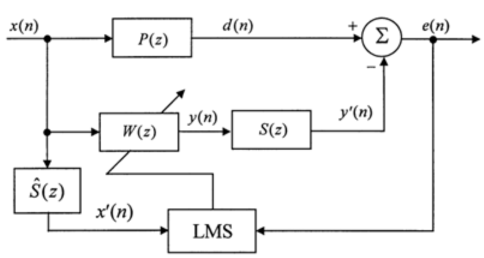
So, finally, the second paper here, the applications of adaptive feedback active noise control system, is being done, okay. This is the reference of the paper, you know. Here, the single-channel broadband feedforward ANC system was used, okay, with the same filtered XLMS algorithm. If you are interested to know how this algorithm works, you can obviously do a separate reading and see, but we cannot discuss it within this course. So, just we are giving a brief overview of it and then. So, this single-channel adaptive feedback active noise control system was used, and it was used to reduce the industrial machine noise in some of the large manufacturing plants. So, that was the goal for it.

Paper 2: "Applications of Adaptive Feedback Active Noise Control System"

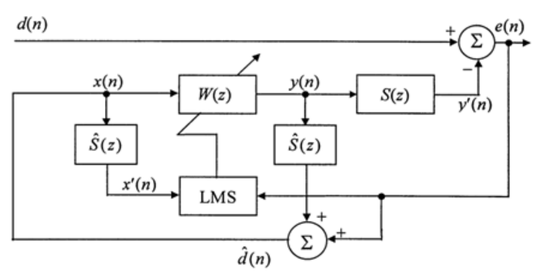
Reference: Kuo, S. M., Kong, X., & Gan, W. S. (2003). Applications of adaptive feedback active noise control system. IEEE transactions on control systems technology, 11(2), 216-220.

- The components of a **single-channel broad-band feedforward ANC system**, is described including the reference sensor, secondary source, error sensor, and the use of the **filtered-X least-mean-square** (**FXLMS**) algorithm for noise cancellation.
- Also described a **single-channel adaptive feedback active noise control** (AFANC) system for reducing industrial machine noise in large manufacturing plants.
- Performance of AFANC was experimentally verified.

This shows the various circuitry, which again we will not be discussing within this course.



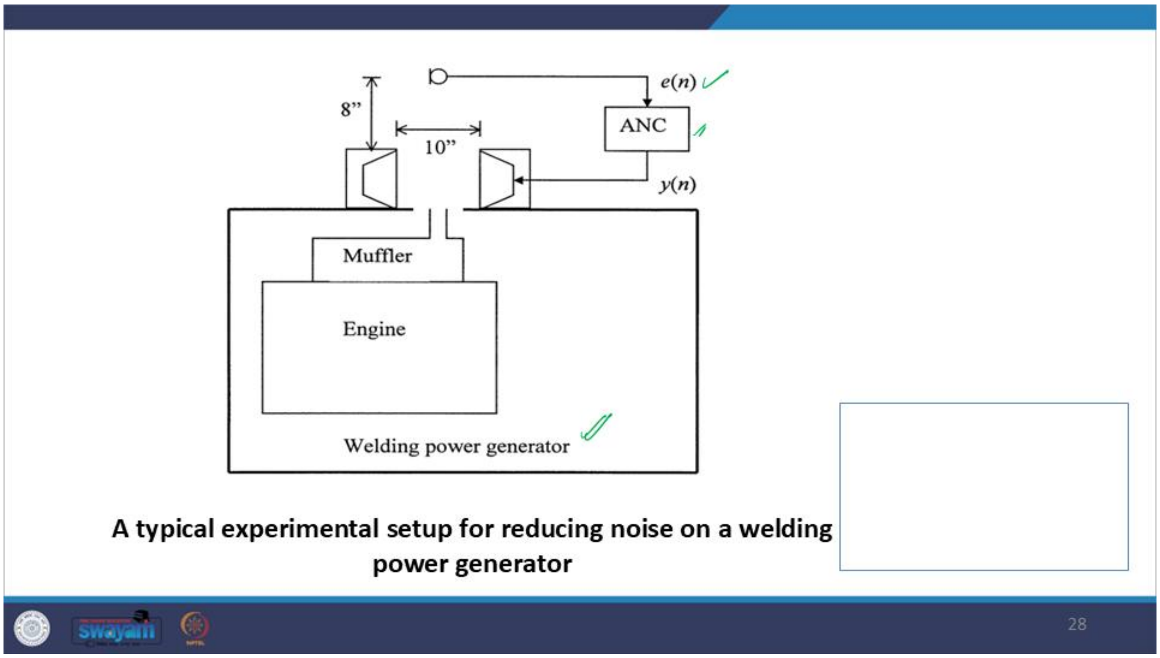
Block diagram of single-channel broad-band feedforward ANC system.



Block diagram of adaptive feedback ANC system.

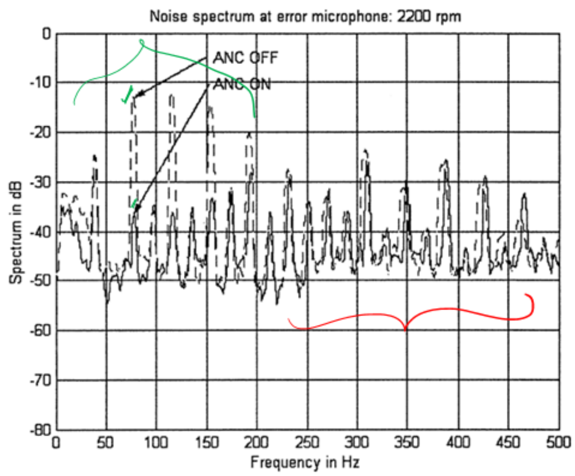
Source: Kuo, S. M., Kong, X., & Gan, W. S. (2003). Applications of adaptive feedback active noise control system. IEEE transactions on control systems technology, 11(2), 216-220.

So, an experiment was done where, you know, you had the engine within a welding power generator machine. So, this was taken as the case study, and a muffler was implemented. Then, there was, you know, the ANC system. This shows the error signal and the ANC system that was implemented.



A typical experimental setup for reducing noise on a welding power generator

This shows, so basically, in a welding generator machine, which creates, you know, a high amount of noise. the ANC was implemented, and a comparison shows this is when, you know, ANC is off, and this is when ANC is on. You can see marked improvement in some of the low-frequency ranges smaller than 200 Hz, so here you had a good performance. And then, in the high-frequency ranges, the performance is not; it is almost making no difference at all, but all the high peaks got attenuated at these various kinds of harmonics or peaks. So, whatever the peak harmonics were, they got attenuated by the ANC system.



Performance of AFANC algorithm for the reducing noise from the welding power generator running at 2200 rpm

- Dotted lines: ANC Off
- Solid Lines: ANC On
- The dominant harmonics at 76, 114, 152, and 190 Hz were attenuated at the broad-band noise level.

So, with this, I would like to close this lecture, which briefly discusses the case studies of where ANC has been applied in machinery noise control. Thank you.

Thank You