Active Noise Control

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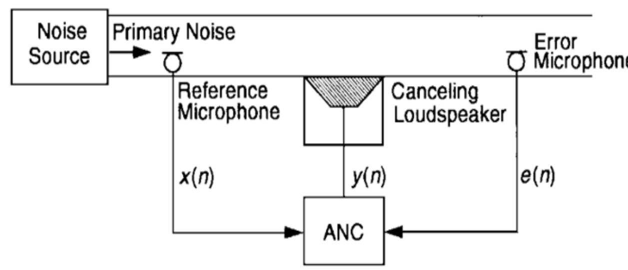
Overview

Acoustic noise creates a major problem in industrial equipment and automobiles. The passive techniques to control noise have proven to be expensive, take up a lot of space and are ineffective at low frequencies. This brings us to active noise control. Active noise control involves use of electroacoustic or electromechanical systems to cancel unwanted noise based on the principle of superposition. ANC fixes most of the shortcomings of the passive techniques. ANC systems are also cheaper and a lot less bulky. ANC systems must be adaptive in order to cope with variations in the noise.

Applications

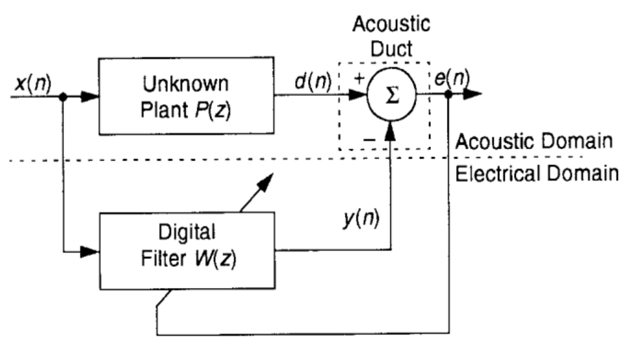
- Automobiles: Noise attenuation inside vehicles and electronic mufflers for exhaust and induction system
- Industrial: Canceling noise from almost any industrial appliance
- Appliances: Noise from daily appliances like ACs, exhausts, etc can also be controlled using ANC

I) Broadband feed-forward ANC



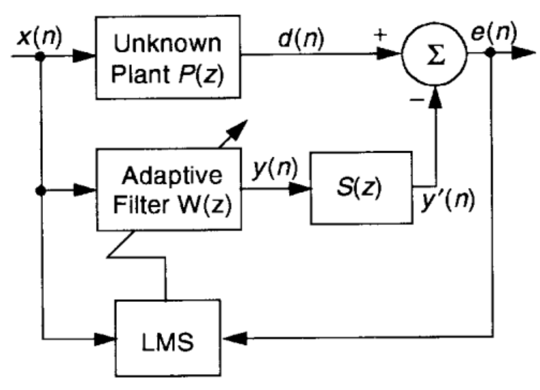
The main motive here is to minimize the acoustic noise. We are considering systems where there is a single reference sensor, single channel, single anti-noise source and a single error sensor. The performance of this system is monitored by the error sensor.

1) Basic model



Consider a basic broad-band ANC system, with an identification framework as shown in the figure. Here the primary path *P(z)* consists of the acoustic response reference sensor to the error sensor. *W(z)* is the adaptive filter, whose function is to minimize the residual error signal, *e(z)*. In the primary path, *d(n)* is the primary disturbance. Now this primary disturbance is acoustically combined with the output from the adaptive filter, *y(n).* This results in the residual signal error *e(z) = d(n)-y(n). As P(z)* converges to *W(z), E(z)* = 0, as *P(z) – W(z) = E(z).* This leads to *d(n)-y(n) = e(z)* = 0, which is the perfect cancellation of both the signals.

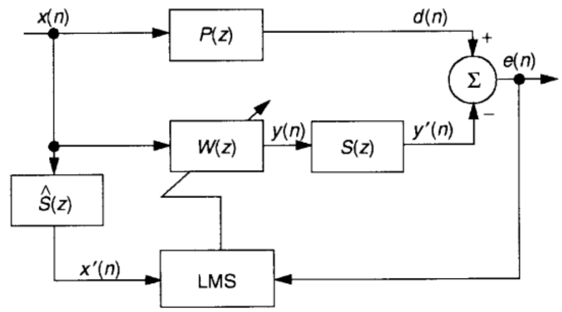
2) Accounting for Secondary Path Effect



Here the adaptive filter output is filtered by a filter *S(z),* called the secondary path, which produces a delay of the filter output with respect to its input. The secondary path represents the effect of the filters, A/D and D/A converters, microphones, loudspeakers and the acoustic path between the canceling loudspeaker and the microphone. From the above figure, the z-transform of the error signal is *E(z)* = [*P(z)–S(z)W(z)*] *X(z).* This way, the system responds instantaneously to changes in the input signal. Now For *E(z)* = 0, *W(z) = P(z) / S(z).*

3) **Filtered-X LMS Algorithm**

The introduction of the secondary path transfer function causes instability as the error signal is not aligned correctly in time with the reference signal as *S(z)* is introduced. An effective solution to this problem is introduction of an identical filter in the reference signal path, which is more efficient than placing an inverse filter 1/*S(z)* in series with *S(z)* to remove its effect.



4) **Leaky FXLMS Algorithm**

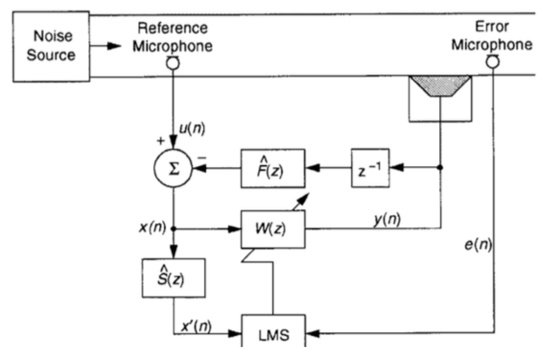
The direct application of the Filtered X LMS (FXLMS) algorithm leads to increased noise levels due to low frequency resonances, which may cause overloading of the secondary source (stalling effect). This can be solved by introduction of output power constraints or adaptive filter weight constraints. The introduction of the leaky FXMLS algorithm (the leakage term) has a stabilizing effect on the adaptive algorithm. It adds to the complexity of the system, but more robustness of the adaptive filter is ultimately achieved.

5) **Feedback effect**

As discussed earlier the ANC uses a reference sensor to read the noise signals. When the anti-noise is produced by the cancelling speakers it may travel up the primary path and get picked up by the reference microphone leading to a corrupted x(n) signal which will inhibit the performance of the system. This scenario is known as acoustic feedback effect and it can be eliminated in 2 ways:

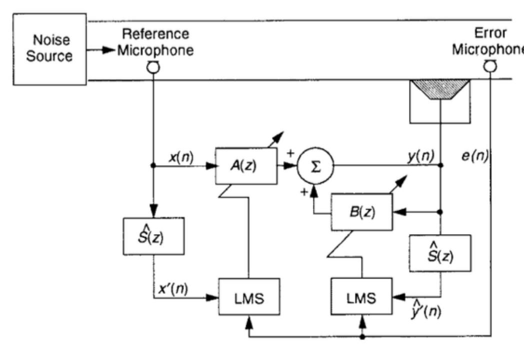
A) Feedback neutralization

This is the simplest approach and it makes use of a separate filter within the controller to cancel the feedback. The output of the adaptive filter is canceled from the reference signal which eliminates the feedback present in reference. The adaptive feedback cancellation filter uses offline adaptive methods for determining the transfer function of feedback path.



B) Active IIR filter

This is the optimal solution and it makes use of an IIR filter to compensate for the feedback. The filtered-U recursive LMS algorithm is used in this model as it is best able to eliminate the effect of the feedback even though it is relatively less stable, may converge to local minima and has relatively slow convergence rate.



Conclusion

Through this project we intend to examine and develop a deeper understanding of the working and effectiveness of different ANC models. All the ANC systems we examined cancel unwanted noise by generating antinoises of equal amplitude and opposite phase through a secondary source using the input received from the reference sensor/microphone. The performance of the system is improved using adaptive filters that change filter weights to minimize the error reading at the error sensor/microphone.

Progress Report:

Done:

1. Gained good understanding of working and purpose of these models
2. Implemented basic adaptive LMS ANC system

3. Implemented FxLMS ANC system

To-Do:

1. Implement feedback neutralizing ANC system
2. Implement filtered-U recursive LMS ANC system
3. Analyze the working of these systems mathematically

References

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- Adaptive Filter Theory (3rd edition) by Simon Haykin