The development of the filtered-U algorithm for active noise control

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Citation: The Journal of the Acoustical Society of America **87**, S82 (1990); doi: 10.1121/1.2028384
View online: https://doi.org/10.1121/1.2028384
View Table of Contents: https://asa.scitation.org/toc/jas/87/S1
Published by the Acoustical Society of America

be achieved with a low number of control transducers. However, little is known about the characteristics of the nearfield intensity and pressure distribution over the panel surface when control is applied. In this paper experimental work is reported in which these variables are measured using a microphone probe traversed near the surface of a vibrating baffled rectangular panel located in an anechoic chamber. Two different thicknesses of panel were considered in order to give a range of test frequencies relative to plate size. Control was applied by a single piezoceramic actuator bonded to the panel surface while error information was taken from a microphone located in the farfield. Results are presented for various modes of vibration, and are related to the distribution of the modal amplitudes of the panel. The influence of the location of the noise input and error sensor on the intensity and pressure distributions is studied. The work has lead to an increased understanding of the mechanisms of control inherent in the technique. [Work supported by NASA Langley Research Center and the Australian Electrical Board.]

1:50

HH3. The development of the filtered-U algorithm for active noise control. L. J. Eriksson (Corporate Research Dept., Nelson Industries, Inc., Stoughton, WI 53589-0600)

The filtered-X algorithm discussed by Widrow and Burgess is a modified form of the LMS algorithm for use when there are transfer functions in the auxiliary path following the adaptive filter. To ensure convergence of the algorithm, the input to the error correlators is filtered by a copy of these auxiliary path transfer functions. More recently, Eriksson has presented a new approach to active noise control in the presence of acoustic feedback that uses an infinite impulse response (IIR) filter structure with a modified form of the RLMS algorithm. This algorithm may be described as a filtered-U algorithm since it uses a copy of the auxiliary path transfer functions to filter the generalized input vector to the error correlators of both the direct and recursive elements of the filter to ensure convergence. The relationship of the filtered-U to the filtered-X algorithm and other earlier concepts will be discussed.

Contributed Papers

2:15

HH4. Adaptive on-line modeling technique for active noise control systems. Sen M. Kuo and Javier Tapia (Dept. of Elec. Eng., Northern Illinois Univ., DeKalb, IL 60115)

Active noise control (ANC) systems are required to model direct and acoustic feedback paths as well as the error path on an on-line basis. In this paper a new adaptive on-line error-path modeling technique is presented. An adaptive IIR filter using SHARF algorithm is used to generate the canceling signal. The error path is modeled by a LMS adaptive filter. This new modeling technique uses the canceling signal to train an adaptive model that is placed in parallel with the error path. This process is performed only when certain conditions are satisfied in order to avoid undesired undulations. The modeling process technique differs from two other methods. Ericksson *et al.* [U.S. Patent number 4,677,676, 30 June 1987] used an additional auxiliary low level, uncorrelated noise for training processes. Warnaka *et al.* [U.S. Patent number 4,473,906, 25 September 1984] used the canceling signal for modeling, but the plant being continuously modeled is not really the error path. Comparative performance and real-time testing results show that this new technique is suitable for active noise control systems.

2:30

HH5. Synchronous active noise cancellation with LPC noise synthesizer. Sen M. Kuo, Liang Chen, and Bob Lee (Dept. of Elec. Eng., Northern Illinois Univ., DeKalb, IL 60115)

A new active noise cancellation scheme has been developed for the purpose of engine noise cancellation. The major difference between the new approach and the conventional two-microphone adaptive noise cancellation methods is that the new approach does not require the reference signal. Therefore, the problem of acoustical feedback from the canceling speaker to the reference channel is eliminated. In this new scheme, a linear predictive coding (LPC) technique is employed to synthesize an "engine noise," which is used as a reference signal, and this synthesized noise is processed by an LMS adaptive filter. Simulation results using a set of digitized engine noise data show that the adaptive filter has a faster convergence rate and more than 10-dB mean-squared error improvement by using an LPC noise synthesizer in comparison with using an impulse train as input only. In the case that engine speed is varying or the engine rpm is not available, a multipulse excitation synthesizer has been used to generate the synthesized engine noise. The multipulse excitation method has also provided an improvement of 10 dB in mean-squared error.

2:45

HH6. Active control of harmonic noise in large reverberant spaces. M. Mitchell-Dignan, G. A. Abe, and J. Clift (BBN Systems and Technologies Inc., New London, CT 06320)

A study was conducted on the control of a distributed harmonic source in a reverberant space of moderate to high modal density. Many large reverberant spaces exhibit high modal density at the operating frequencies of the sources they enclose. Often, the sources in such a space must be treated as distributed sources and cannot be practically controlled by placing secondary sources near their surface. Therefore, the noise must be controlled with secondary sources placed in the reverberant field. Both analytical and experimental models were used to develop a control system for a large reverberant laboratory in which the noise, generated by a distributed source, was controlled using multiple secondary sources. The secondary source locations were optimized based on a modal model of the space. Significant reduction of the total acoustic energy was attained near a peak composed of several resonances. A modest reduction in acoustic energy was seen at a frequency near an antiresonance. Less impressive reductions were observed at frequencies of higher modal densities.