Active noise control using adaptive digital signal processing

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ACTIVE NOISE CONTROL USING ADAPTIVE DIGITAL SIGNAL PROCESSING

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ABSTRACT

Active noise control utilizes microphones, electronics, and loudspeakers to generate an acoustical wave to cancel undesired sound. The basic approaches to active noise control will be briefly reviewed. Primary emphasis will be on a system identification approach to the problem using adaptive digital signal processing. A unique solution to problems caused by the acoustical nature of the application will be presented. This includes a discussion of the use of an IIR adaptive filter to model the primary plant and compensate for acoustic feedback, the use of an independent random noise generator to continuously model the characteristics of the source and error plant, and the effect of uncorrelated noise on both processes.

I. INTRODUCTION

Active noise control uses a secondary acoustic sound source to generate a sound wave that will cancel an undesired noise from a primary noise source. The basic concept is quite old, originating in the 1930s. However, only with the recent advent of adaptive digital signal processing and relatively inexpensive digital signal processing hardware has the process become practical [1].

This paper will focus on the particular problem of the active control of sound waves in one-dimensional waveguides. Although many of the comments apply to sound waves in extended solid structures such as beams or rods as well as sound waves in liquid-filled conduits such as noise from pumps, primary attention will be on airborne sound waves in ducts. Typical examples include discharge or intake noise from industrial or commercial fans, blowers, or compressors.

A wide variety of approaches to generating the sound cancelling wave have been used in the past [1]. However, a system identification approach will be used in this paper. In this approach, the sound cancelling wave is generated from a signal obtained by passing an input signal through a computer-based model that accurately represents the response of the acoustic plant.

As shown in Fig. 1, an input microphone is used to measure the undesired noise. The acoustic system between the input microphone and cancelling source, known as the plant, is modelled on a computer such that the model output can be used to generate the cancelling signal. A key advantage of this approach is that with the proper model of the plant, the system can respond instantaneously to changes in the input signal caused by changes in fan speed or loading.

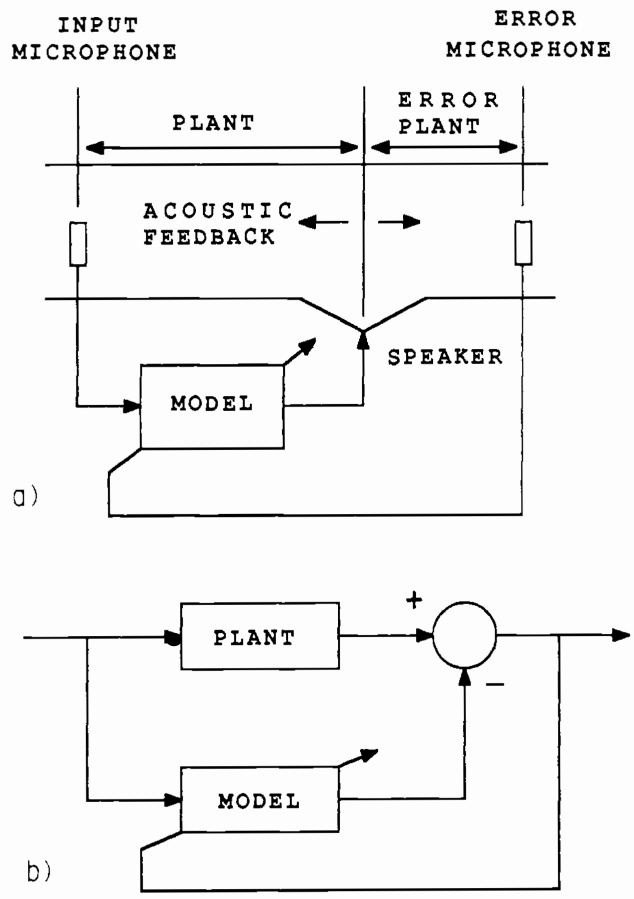

Fig. 1 - a) Schematic diagram of active attenuation problem.
b) System identification approach to problem.

II. PRIMARY MODEL AND ACOUSTIC FEEDBACK

Due to changes in the characteristics of the acoustic plant, it is not practical to determine the model prior to operation of the system. For example, temperature and flow variations cause changes in the sound velocity in the duct, and transducer response characteristics change over time. Thus, it is necessary for the model to be determined adaptively as the system operates.

Initial efforts in this direction were limited by the effects of acoustic feedback from the cancelling source to the input microphone. This potentially destabilizing factor is difficult to avoid due to the omnidirectional nature of acoustic transducers at the low frequencies where active noise control is of primary interest.

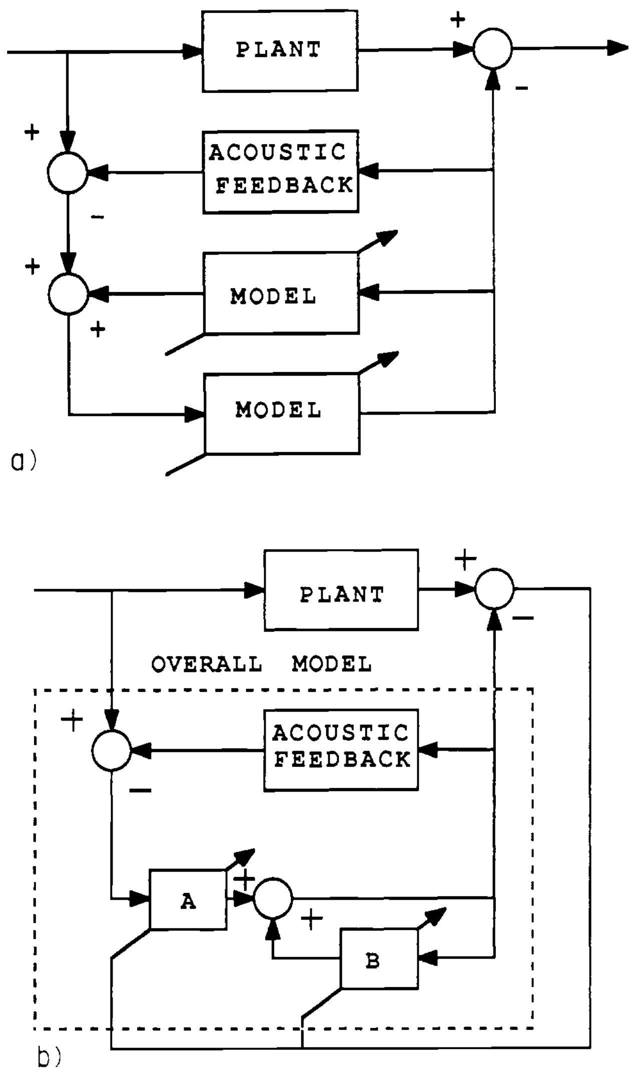

Fig. 2 - a) Two model approach to feedback compensation
b) New single model approach using RLMS algorithm (the elements "A" and "B" are adaptive LMS algorithms) for IIR modelling [2-4].

Previous methods have been restricted due to the use of a two model approach to the acoustic feedback problem. One model was used to compensate for the acoustic feedback while a second model was used to provide the cancelling signal, as shown in Fig. 2. A more powerful approach is to recognize that the acoustic feedback is part of a single overall model and results in uncontrolled poles in the transfer function of this model that may be cancelled using a pole-zero or infinite impulse response (IIR) adaptive filter [2,3].

Although there are a number of pole-zero, IIR adaptive filter algorithms that would be suitable for this problem, the recursive least mean squares (RLMS) algorithm of Feintuch [4] was selected for use in the system. One of the problems with IIR adaptive filters is the potential for instability. In particular, the solution for a high order filter used with a spectrum containing few frequencies is non-unique. Pole-zero cancellation must occur and some of these unconstrained poles can become unstable. However, in real applications the input is very rich with many frequencies and a unique, stable solution can be obtained.

III. AUXILIARY MODEL AND SPEAKER/ERROR PATH MODELLING

Proper convergence of the RLMS algorithm requires knowledge of the transfer functions present in the so-called auxiliary path following the model as well as the error path. This problem has been previously discussed by Widrow [5], Morgan [6], and Burgess [7] for the LMS algorithm. For the RLMS algorithm, a similar situation occurs as shown in Fig. 3. In this case, either the transfer functions for the speaker (S) and error path (E) must be determined for use in the weight update calculation or their effects must be compensated for through the use of inverse transfer functions in the error path. The latter approach requires use of delay in the weight update calculation since only delayed inverses are possible in a causal system.

To ensure proper operation with changes in S and E as well as to avoid the need for complicated start-up procedures, it is desirable that S and E be adaptively determined as the system is operating. An early attempt at solving this problem involved the use of three microphones [8]. However, a more powerful approach is to use an independent, random noise generator to continuously model S and E using an independent adaptive filter [9,10]. Although the adaptive filter C could use a variety of adaptive algorithms, one simple choice is to use the least mean squares (LMS) algorithm of Widrow [5]. The random noise generator is referred to as a Galois noise source after the Galois sequence that can be used for its generation [11].

IV. EFFECT OF PRIMARY SIGNALS ON AUXILIARY MODEL

Although the Galois modelling process functions independently of the primary RLMS model, the primary model output as well as the primary plant

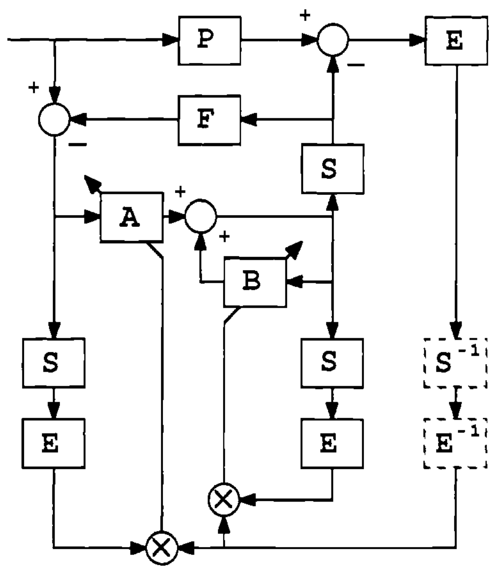

Fig. 3 - Block diagram of active attenuation problem ("P" is the direct plant and "F" is the feedback plant) showing general solution to compensation for transfer functions due to loudspeaker "S" and error path "E" through the addition of duplicate transfer functions or inverse transfer functions.

output constitute noise for this process, as illustrated in Fig. 4. A parallel model approach is preferable due to the fact that although the error signal contains noise from the primary modelling system, the final adapted weights for the filter C are unaffected by this noise since it does not appear on the input to the adaptive filter. In a series approach, the error signal as well as the final converged values of the weights are affected. Thus, a less accurate estimate of the transfer functions S and E is obtained. The final system used the parallel modelling approach which does not have this problem, as shown in Fig. 5 [9,10].

V. EFFECT OF GALOIS NOISE ON PRIMARY MODEL

In addition to allowing the independent determination of S and E through the adaptive filter C, the Galois noise source introduces noise into the primary modelling process through the acoustic feedback path. Since this noise is present on both the error signal and input signal for the primary RLMS adaptive filter, it does affect the final converged value of the weights of the filter. Using the approach of Widrow and Stearns [12], the optimal model transfer function, $W^*(z)$, becomes:

$$\frac{A(z)}{1-B(z)} = \frac{P(z)}{1-P(z)F(z)+(\Phi_{gg}(z)/\Phi_{uu}(z))|F(z)|^2}$$

where P(z) and F(z) are as shown in Fig. 3 with $\Phi_{gg}(z)$ and $\Phi_{uu}(z)$ representing the auto power

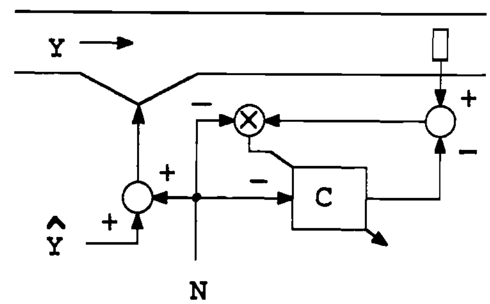

Fig. 4 - Direct modelling approach for the determination of the transfer function of the speaker and error path, SE, using an independent random noise source "N" with adaptive filter "C" ("$\hat{Y}$" is the primary plant output and "Y" is the primary model output).

spectra of the Galois noise source and system input respectively. Thus, the Galois noise source results in a change in the converged weight values. However, it can be shown that the combined system output still converges to zero as the amplitude of the Galois noise source is made vanishingly small. In practice, a low amplitude Galois noise source does not product any significant adverse effects while still producing a good, real time model of S and E. The above equation also explicitly demonstrates the role of the acoustic feedback in introducing poles in the optimal adaptive filter response.

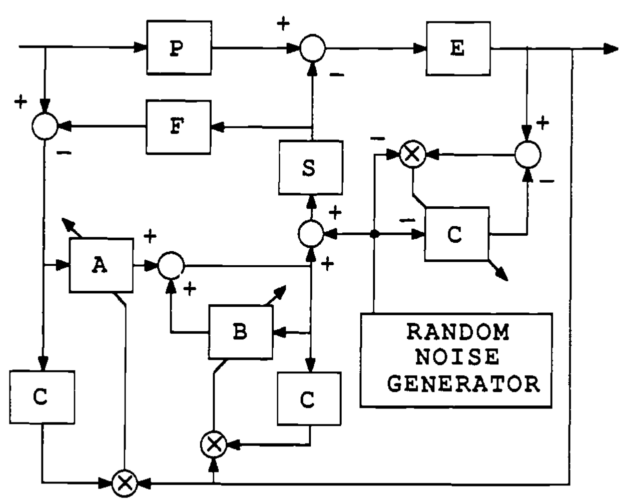

Fig. 5 - New approach to on-line modelling of speaker "S" and error path "E" with adaptive filter "C" and using results in RLMS model with acoustic feedback to form fully adaptive active attenuation system [9-10].

VI. IMPLEMENTATION OF SYSTEM

The system shown in Fig. 5 has been implemented using Texas Instruments TMS32010 and TMS32020 digital signal processing microprocessors in the Nelson/DIGISONIX™ dX-30 and dX-40 Digital Sound Cancellation Systems, respectively [13]. A discussion of the hardware is available in a companion paper [14], and performance results on a variety of installations have been reported [1,2,9,15-17]. In actual field use on large fans, noise reductions of about 30 dB on narrowband, tonal sounds and about 15 dB on broadband, continuous spectra associated with random signals have been obtained.

VII. SUMMARY

This paper has described a complete, fully adaptive, active noise control system for use in ducts, pipes, conduits, beams, or other types of acoustical waveguides containing propagating plane waves [15-16]. Additional discussion of the acoustical effects on the performance of this system will be described in an upcoming paper [17]. Current work is concerned with applying these same principles to closed resonant systems as well as systems containing higher order mode propagation.

ACKNOWLEDGMENTS

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