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ACTIVE CONTROL OF MULTI-DIMENSIONAL RANDOM SOUND IN DUCTS

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INTRODUCTION

Previous work has demonstrated how active control may be applied to the control of random noise in ducts.[1,2] These implementations however have been restricted to frequencies where only plane waves are propagating in the duct. In spite of this, this technology has progressed to the point where commercial products that apply these concepts are currently available.[3] Extending the frequency range of this technology for a fixed duct geometry requires the development of multi-variate rather than single channel control systems and solutions to the problems inherent in controlling higher order modes. Eriksson et al[4] used two independently operating controllers for higher order mode control and obtained reasonable results for singular modes. Tichy[5] suggests independent control of each mode in a multi-variate system.

This paper examines the application of active control to the multi-dimensional propagation of random noise in waveguides. The problem of designing a realizable controller using a finite impulse response filter and controlling the acoustic feedback path of the control source-detector microphone path are addressed using an acoustically lined section. An adaptive system is implemented using measured system frequency response functions. Experimental results are presented illustrating suppressions of 15 to 30 dB for random noise propagating in multiple modes.

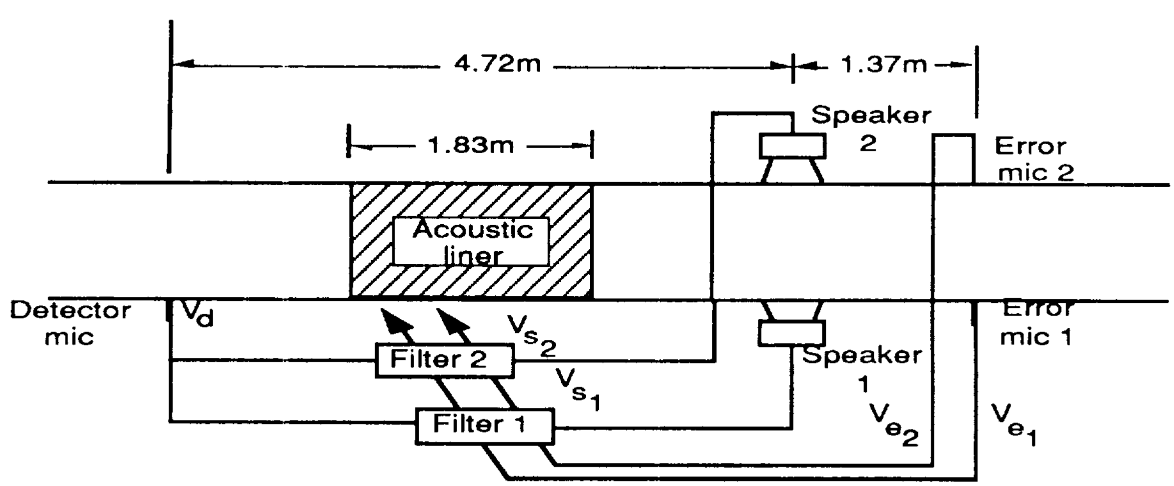

Figure 1. Schematic of duct and control system.

CONTROLLER DESIGN

The design of the control system parallels that of reference 1. The combination of acoustical and control systems, shown in figure 1, is defined as a single input, multiple output system. A block diagram incorporating the various elements of the duct and control system is shown in figure 2. Some excitation voltage, V_p, is assumed to cause an undesired acoustic response in the duct such that only the plane wave (mode 0) and the first cross mode (mode 1) are excited. The output of a single detector microphone, V_d, is used as an input to a feed forward control system. Two control voltages, V_{si}, excite two loudspeakers on opposite sides of the duct. Finally, the acoustic response downstream of the loudspeakers is represented as the error microphone outputs, V_{ej}.

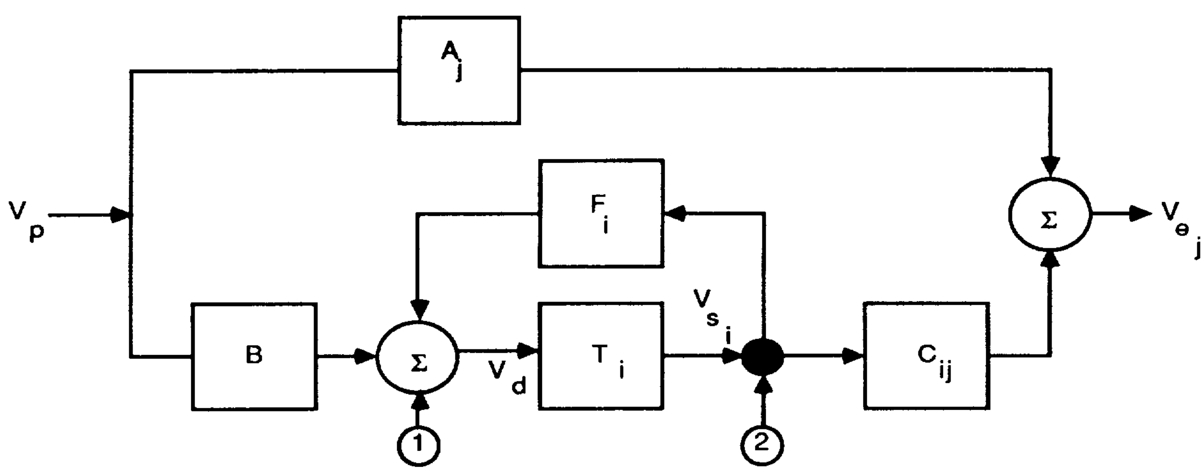

Figure 2. Block diagram of control system.

Transfer functions between the applicable excitation voltages and the sensor output voltages are defined as the following:

$$\left.\frac{V_{e_j}}{V_p}\right|_{V_{s_1}=V_{s_2}=0} = A_j(\omega) \qquad \left.\frac{V_{e_j}}{V_{s_i}}\right|_{V_p=0} = C_{ji}(\omega)$$

$$\left.\frac{V_d}{V_p}\right|_{V_{s_1}=V_{s_2}=0} = B(\omega) \qquad \left.\frac{V_d}{V_{s_i}}\right|_{V_p=0} = F_i(\omega)$$

Designating the transfer functions of filter 1 and filter 2 of figure 1 as the vector T_i, the sensor output vector relative to the excitation voltage, V_p, is given as

$$\frac{V_{e_j}}{V_p} = A_j + \frac{C_{ji} T_i B}{1 - F_k T_k}$$

Setting this to zero and substituting E_j for the ratio A_j/B yields the expression for the controller functions T_i that force the transmitted wave field to zero.

$$T_i = [E_j F_i - C_{ji}]^{-1} E_j$$

The function E_j is the transfer function measured between the error sensors and the detector sensor. This makes the design of the controller independent of V_p. For previous work involving only plane waves, this ideal controller was correct for any changing source condition so long as the controller parameters remained constant. Since only plane wave propagation was permitted, this arrangement could predict everything about the source in terms of it structure and content. However, in this case, not only the frequency content but the source structure or modal content is subject to change. The single detector microphone cannot adequately define the magitude and phase of the spatial modes excited by the source. However, this information is contained in the transfer function measurements between the two error microphones and the detector microphone. Since this information is integrated into the controller only when it is initialized, any changes in this structure will result in degraded performance of the controller even under ideal conditions. To overcome this problem, additional detector microphones would be necessary in order to define the source structure in the feedforward section rather than the feedback section. This approach however imposes additional problems beyond the scope of this paper as well as additional hardware complications.

Adaptive techniques may be applied to overcome this limitation for a slowly varying source structure. Consider an additional voltage vector, ΔV_{si}, is input at point 2 of figure 2, with the controller functions having been previously set to nonzero values, T_i. The additional voltage appearing at the inputs to the loudspeakers are

$$\Delta V_{s_i} + V_{s_i}\Big|_{V_p=0} = \Delta V_{s_i} + \left(\frac{\Delta V_{s_k} F_k}{1 - F_k T_k}\right) T_i$$

The single detector microphone output, V_d, will be modified by the above voltages multiplied by F_i and summed. Adding an additional controller function, ΔT_i in parallel with T_i (with inputs V_d and outputs summed to the outputs of T_i) a new expression may be written for the error sensor outputs.

$$\frac{V_{e_j}}{V_d} = E_j^1 + \frac{[(1 - F_k T_k) C_{ji} \Delta T_i + C_{jk} T_k F_i \Delta_i]}{1 - F_k T_k - F_k \Delta T_k}$$

In this equation, E_j^1 is the transfer function between the j[th] error sensor and V_d after the controller function T_i is implemented. Setting V_{ej} again to zero, ΔT_i may be determined and a general update equation determined.

$$T_i^{m+1} = T_i^m + (1 - F_k T_k^m)[E_j^{m+1} F_i - (1 - F_k T_k^m) C_{ji} - C_{jk} T_k^m F_i]^{-1} E_j^{m+1}$$

Here, E_j^{m+1} refers to the transfer function measured after the implementation of controller T_i^m. Note that for $T_i^m = 0$, the update algorithm reduces to the original controller definition.

One final comment note on the controller design. For low frequencies where only the plane wave propagates, the controller equation was found to be nearly singular. For true plane waves with no measurement error, this would be the case. This is due to the fact that the 2 channel controller is redundant since only one channel is necessary. This may be used to advantage in an arrangement where a single woofer is used to control low frequency plane waves but a pair of midrange speakers is used to control higher frequency multimodal propagation. For the implementation here, the controller for each source/error microphone pair was treated as a plane wave system and the resulting scalar controller equation was solved. This applied for frequencies up to 250 Hz, beyond which the above matrix equation was applied.

EXPERIMENTAL CONFIGURATION

In order to provide a reasonable amount of damping, a section of the duct between the detector sensor and controller sources was lined with acoustic treatment. This provided attenuation of the waves propagating in both directions. Thus the reflected wave field at the detector sensor is attenuated relative to the incident acoustic wave field at the detector sensor. This has the effect of eliminating the singularity in the controller implementation, shortening the reverberation time of the feedback path and consequently the required controller filter length. When the treated section is coupled to the hardwall sections at either end, any residual cut-off modes scattered by the junction of the lined and unlined sections are attenuated so that the necessary controller inputs are not affected.

The schematic of the duct and associated sources and sensors in shown is figure 1. The duct cross section is 0.535m by 0.229m with the acoustic liner spanning the larger dimension. The loudspeakers were oval and their flat cone made up the narrower wall dimension at their mounting location. This design yielded a first cross mode cut-on frequency of 325 Hz and a second cross mode frequency of 650 Hz. Both of these modes have their spatial variation in the wider dimension (0.535m) and the focus of this work is to control the plane wave and the first cross mode in a frequency range from 50 Hz to 600 Hz. The source excitation consisted of two speakers mounted in a similar arrangement to the controller sources. This provided control over the input source structure. These were driven using a common random excitation but with the speakers out of phase and at different levels in order to drive a combination of modes.

The above control algorithm was implemented using standard digital signal processing (DSP) techniques to determine the necessary frequency response functions. The two controller functions were implemented as finite impulse response digital filters on a TMS320C25 DSP chip using 256 coefficients each. All data was sampled and output at 2048 samples/second. The frequency response function of each controller was Fourier transformed to an impulse response for implementation in the time domain.

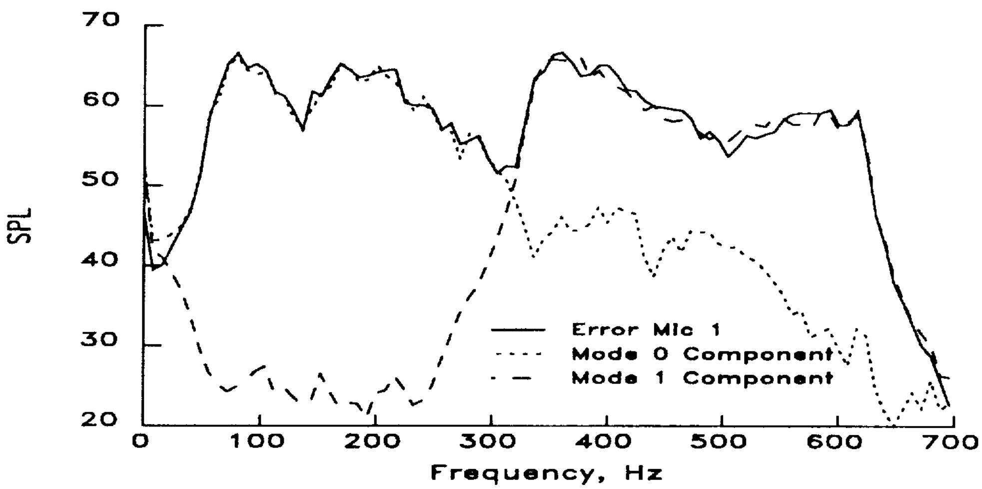

Figure 3. SPL and source modal distribution at error microphone location.

EXPERIMENTAL RESULTS

Figure 3 shows the spectra of the sound pressure level at error microphone 1 and the contributions due to both the plane wave and mode 1. These were obtained by summing and differencing the two error microphones and are valid only below the 2nd mode cut-on frequency of 650 Hz. The overall SPLs at error microphones 1 and 2 were 102.6 and 102.7 dB. The plane wave dominates the pressure spectrum by more than 30 dB below the cut-on of the first mode at 325 Hz. Above mode cut-on, the higher order mode dominates the response by 10 to 20 dB.

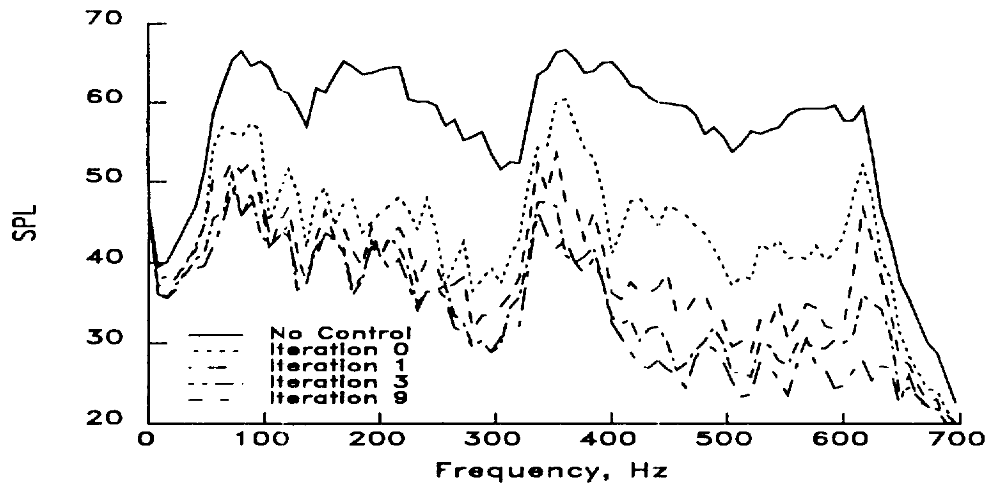

Figure 4. Effect of controller operation on SPL at error microphone 1.

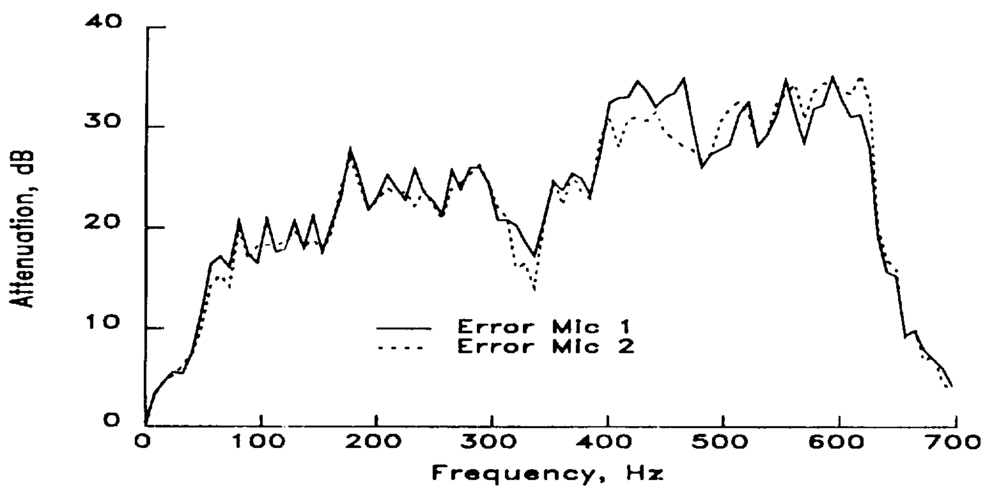

Figure 5. Attenuation at error microphones.

The effect of optimizing the controller on the SPL at error microphone 1 is shown in figure 4. The controller is seen to provide significant attenuation across the spectrum, converging to a minimum level in 3 iterations and remaining stable through nine iterations. The levels at iteration 0, 1, 3, and 9 are shown with most of the suppression attained in the first two iterations. The overall reductions in SPL obtained at these iteration steps were 10.3 dB, 16.4 dB, 19.9 dB and 20.7 dB respectively. The attenuation spectra at both error microphones are shown in figure 5.

This shows the frequency dependence of the attenuation. In the frequency range where the plane wave dominates (<325 Hz), the attenuation ranges from 15 to 25 dB. At higher frequencies, the attenuation is 10 dB greater. At the mode cut-on, the resonant behavior of the duct and the limited frequency resolution of the controller implementation limits the attenuation to about 15 dB.

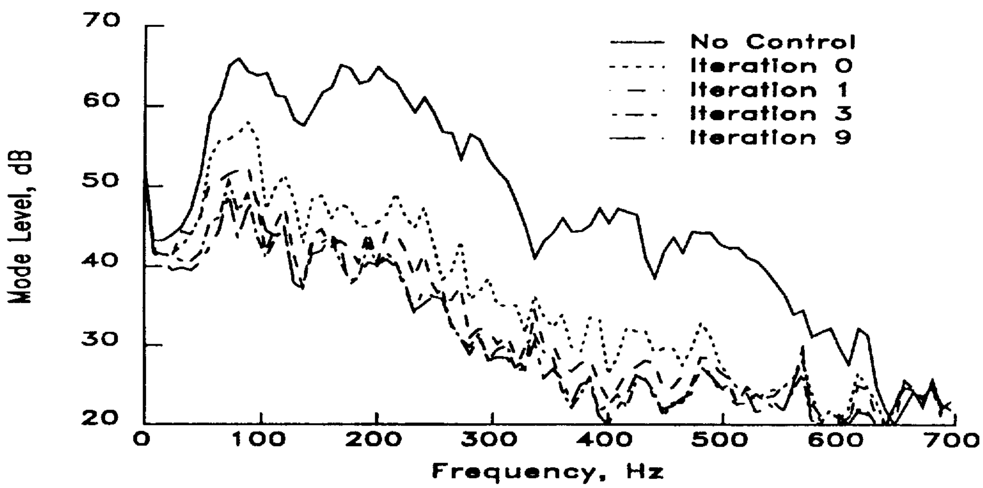

Figure 6. Effect of controller operation on plane wave.

Figure 6 shows the level of the plane wave at similar iterations of the controller. An overall attenuation of 19.1 dB has been obtained across the spectra from 50 to 600 Hz. It is expected that noise floor limits were obtained over the higher frequency range. Figure 7 shows a similar result for mode 1 where an overall reduction of 22.6 dB was attained. Both modes are quite controllable using this configuration with better results obtained in the frequency region where both modes are freely propagating. Above the resonance of the (1,0) mode, both modes are controlled to a common noise floor level.

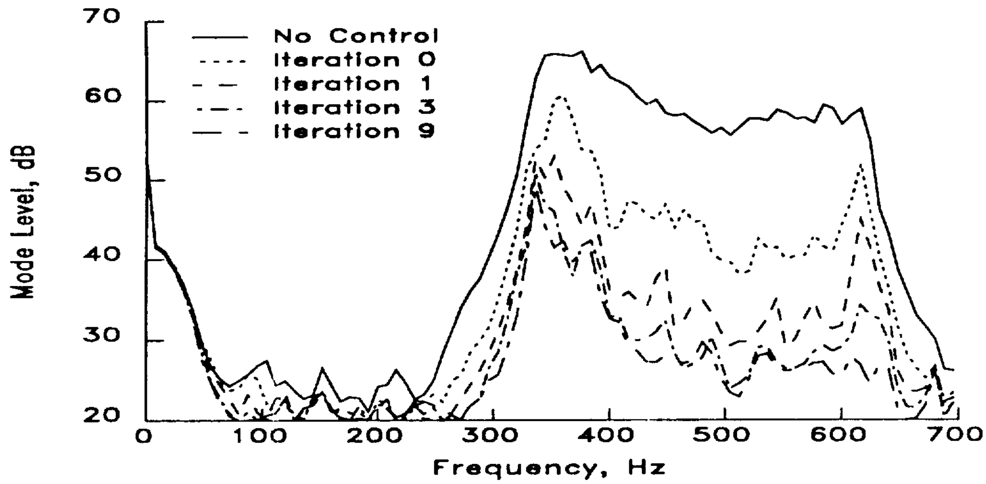

Figure 7. Effect of controller operation on mode 1.

CONCLUDING REMARKS

These results indicate that multi-mode propagation may be controlled effectively using active control techniques. Overall attenuations of 20 dB in SPL were obtained over a frequency range where two duct modes were shown to be propagating. An adaptive technique was used to mimimize the sound pressure at two error sensors with no discernible instabilities. This controller used two independent control sources and three sensor elements to provide effective control. This arrangement is believed to require the minimum transducers for this source problem and is adaptable to other control methodologies. It is noted that this technique is sensitive to changes in the source modal content and must be adaptive in order to be able to handle any changes in the source structure.

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16. Abstract

Previous work has demonstrated how active control may be applied to the control of random noise in ducts. These implementations, however, have been restricted to frequencies where only plane waves are propagating in the duct. In spite of this, the need for this technology at low frequencies has progressed to the point where commercial products that apply these concepts are currently available. Extending the frequency range of this technology requires the extension of current single channel controllers to multi-variate control systems as well as addressing the problems inherent in controlling higher order modes.

This paper examines the application of active control in the multi-dimensional propagation of random noise in waveguides. An adaptive system is implemented using measured system frequency response functions. Experimental results are presented illustrating attained suppressions of 15 to 30 dB for random noise propagating in multiple modes.

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