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DESIGN OF A BROADBAND ACTIVE SOUND CONTROL SYSTEM

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Summary - Hitherto the filtering techniques used in active sound control systems have been analogue: these are rapidly becoming obsolete because of their lack of stability and versatility. The microprocessor gives digital techniques an edge both on grounds of convenience and economics: moreover the performance of digital filtering is far superior to that of analogue methods.

An algorithm is presented which can be used to assess the optimum filter characteristics required for active sound control systems with a single degree of freedom. In the process the system is subjected to three random noise tests which directly yield the characteristics of the filter. The algorithm has been tried out in practical applications and shown to be both quick and convenient to use.

Introduction - This paper describes how a controller for plane-wave sound can be constructed using digital filtering techniques. Plane sound waves travelling away from a noise source, N, are detected at a detector, D. The signal from the detector is used by a controller with transfer function T, to drive a second source, S, so that beyond it, at a point P, say, the sound wave is not heard. Depending upon the complexity of the second source its effect will be to, either reflect the sound back to the noise source, or absorb the radiated sound. The transfer function, T, is specified by the requirement for the sound at P to be zero. A_{dn} is the transfer function between the points specified by the subscripts (ie. detector output and noise input respectively).

We derive the noise reduction for any transfer function (T_a say) and hence find its value for the maximum reduction.

$$S = T_a D \tag{1}$$

The sound at D with both the source and the noise operating is

$$D = A_{ds} S + A_{dn} N \tag{2}$$

and at P it is

$$P = A_{ps} S + A_{pn} N \tag{3}$$

From equations (1), (2) and (3)

$$P = \frac{A_{ps} T_a A_{dn} N}{1 - T_a A_{ds}} + A_{pn} N$$

Note that without the cancelling signal we have $P = A_{pn}N$ so the ratio of the sound with the control system to that without it is

$$\frac{P_{\text{with controller}}}{P_{\text{without}}} = \frac{1 - T_a(A_{ds}A_{pn} - A_{ps}A_{dn})/A_{pn}}{1 - T_a A_{ds}} \qquad (4)$$

Writing $$T_d = \frac{A_{pn}}{A_{ds}A_{pn} - A_{ps}A_{dn}} \qquad (5)$$

and substituting it into (4) the reduction in sound level is

$$\text{Noise reduction} = \frac{1 - T_a/T_d}{1 - T_a A_{ds}} \qquad (6)$$

So, if $T_a = T_d$, then the observer will be in silence. We may also estimate the expected result for an imperfect controller using equation (6).

Conventional methods of estimating the desired transfer function, T_d, would measure the four transfer functions, A_{ds}, A_{dn}, A_{ps} and A_{pn}, and perform the manipulation of (5). The manipulation is most conveniently accomplished in the frequency domain by simple arithmetic but this necessitates an initial transformation from the time domain. In the time domain the calculation is complex as it involves convolution. This paper presents a convenient method of obtaining the characteristic in the discrete time domain without resorting to convolution; it also leads directly to a specification of a digital filter which will realise the characteristic. This is achieved by manipulating the signals before the transfer functions are measured. Two time series are produced, by cascading the four transfer functions, in such a way that they are related by the desired controller characteristic. A model (and thus a digital filter) for this controller is then estimated from cross- and auto-correlations of the two time series using the techniques of System Identification. We now describe the particular way that this manipulation can be done for a system with a single degree of freedom.

The Algorithm - Either the existing noise source can be used to provide the initial excitation of the system, or, and preferably, a loudspeaker driven by filtered white noise can be positioned close to the noise source. As the noise, N, is being generated, it is simultaneously recorded at the detector, D, and the observer, P, microphones (these signals are denoted by $A_{dn}N$ and $A_{pn}N$ respectively). The noise source is then stopped and the signal recorded at the detector, $A_{dn}N$, is fed to the source and recorded again at the observer, changing it to $A_{ps}A_{dn}N$. Next, the other signal, $A_{pn}N$, is also replayed through the source and recorded at the detector changing it to $A_{ds}A_{pn}N$. The difference between these two signals, $A_{ds}A_{pn}N - A_{ps}A_{dn}N$, and the original signal $A_{pn}N$ are equivalent to the input and output of the desired controller. The ratio of the Fourier transform of the time series gives the frequency domain specification of the desired transfer function T_d.

In order to maintain our signal to noise ratio throughout the test and thus ensure coherence, it is desirable to shape the spectral content of the signals so that they are as near to white as possible. This can be done in two places; firstly, if we have control of the noise, N, then we can boost its contents where A_{pn} and A_{dn} have weak responses. Secondly, once the two signals have been recorded we can filter them both again to compensate for weak response in A_{ds} and A_{ps} and ensure that the signal to noise ratio is uniform at all frequencies. We are able to filter these two signals at this stage provided we do so with identical filters because we will only need their ratio in the end.

Once the two signals have been recorded they are used to specify the digital filter controller. A digital filter can be considered, most conveniently, as a program running in a computer; the computer is made to sample an input at regularly spaced intervals in time and produce an output. Generally the output depends upon a weighted average of the input and its past values and past values of the output. System Identification can been used to estimate these weighting coefficients from the recorded input and output time series.

A Practical Demonstration - This theoretical method of designing a controller for systems with a single degree of freedom was tested in an experiment conducted on a rig constructed in the Department of Engineering, Cambridge for Dr. Swinbanks. The rig is a model of an air conditioning duct ten metres long with a fifty centimetre square cross section. For sound waves of wavelength larger than the first transverse mode the only method of propagating down the duct is as a plane wave: it has, therefore, only one degree of freedom for long wavelength sound.

The noise source is a loudspeaker which forms one end of the duct. The detector was a three microphone array positioned on the axis of the duct. The cancelling source was built into the wall further down the duct and the performance was measured by another microphone at the duct opening.

The noise source loudspeaker was driven by either pre-recorded random noise or computer generated signals. The three microphones of the phased detector array were electronically connected according to Swinbanks' formulae with a time delay arranged to give no output for sound travelling towards the noise source loudspeaker (backward travelling waves). Their spacing determines their effective frequency bandwidth and their output is compensated so as to remove any distorting effect of the time delays for sound travelling in the forward direction (away from the noise source). The resultant signal is fed through filters (in order to reduce frequency aliasing; a problem introduced by signal sampling) into the digital controller. The output from the controller is passed through a low-pass filter to smooth the stepped sample and hold output and thus reduce the power in the harmonics of the signal. The filter output is connected to the cancelling source.

The testing method, as it has just been described, was used in a computer-controlled series of random noise tests on the duct to determine the form of the controller required. The computer was then programmed to come as close as possible to this desired controller; rather remarkably it worked first time. The system gave some 15 to 20 decibels of suppression over more than 3 octaves of random sound. The attenuation falls off for frequencies above 350Hz since a 6^{th} order Butterworth filter with this turnover frequency was put in line to reduce frequency aliasing. It is envisaged that this frequency could be increased up to the limit of the first transverse mode; above this frequency more than one cancelling loudspeaker would be needed to excite the higher modes. The poor results below 50Hz correspond to quite a large discrepancy between the actual and desired transfer functions. This is caused by low coherence between the two test signals $A_{ds}A_{pn}N - A_{ps}A_{dn}N$ and $A_{pn}N$.

Improvements on the basic system - To improve the performance below 50 Hz the spectral content of the signal $A_{pn}N$ was made nearly white by prefiltering N to compensate for the characteristic, A_{pn}. A suitable filter to do this can be obtained by identifying the signal $A_{pn}N$ as an autoregressive process. The filter, a moving average one, is then the inverse of this process. When the subsequent identification procedure was carried out using this improved signal, the match (measured using equation (6)) between the desired transfer function and the model was much improved.

Conclusions - When the resulting filters produced by all the separate methods were actually tried on the duct they performed closely to the expected results calculated from equation (6). Between 15 and 20 dB of reduction was achieved from 25 to 350 Hz. The agreement demonstrates that the method of determining the expected results is sufficiently reliable that we can be confident of the likely results of any active control system before it is built. For the larger applications it is necessary to have concrete predictions of the performance so that they can be weighed against the predicted financial outlay. One application, which has recently been successful, involved the control of the noise level inside the acoustic test chamber at R.S.R.E. Malvern which was affected by low-frequency sources outside. Some 10 decibels of the low-frequency sound entering the anechoic chamber was blocked using a single degree of freedom system; it was set-up, tested and demonstrated in a day.