The active control of sound

Active control of sound results from destructive interference between the sound field of an original acoustic source and that from a controllable array of 'secondary' acoustic sources. For this destructive interference to occur over an appreciable region of space the sound field of the secondary sources must match that from the primary source in both time and space. The spatial matching requirement leads to an upper frequency of applicability of active control. Active control complements conventional passive methods of sound control, which do not work well at low frequencies. Practical feedforward controllers, using a multichannel generalisation of the well known LMS adaptive algorithm, have been developed, using as many as 16 loudspeakers and 32 microphones, and applied with considerable success in the control of low-frequency propeller noise inside aircraft and low-frequency engine noise inside cars.

by S. J. Elliott and P. A. Nelson

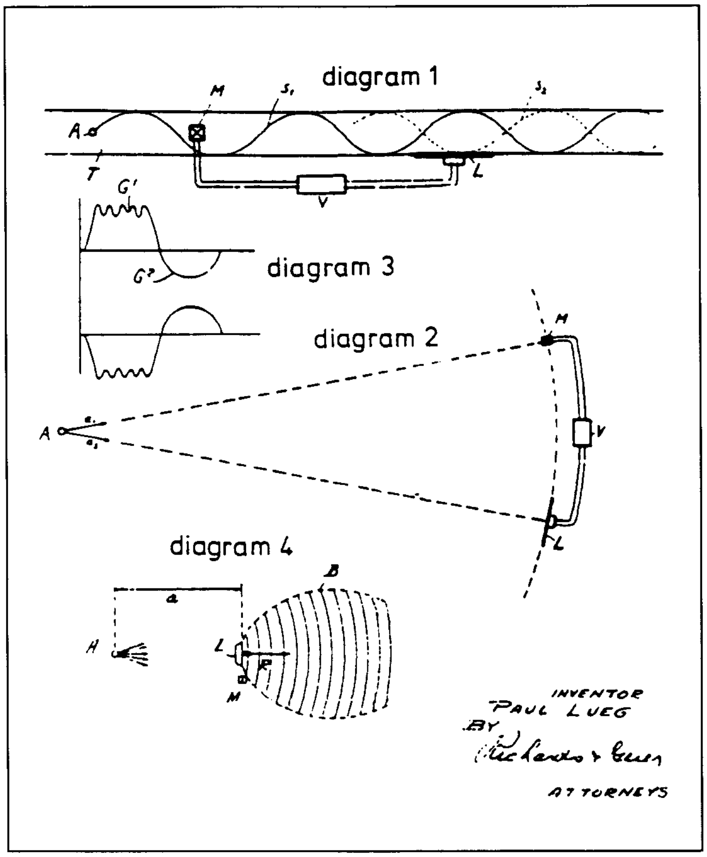
1 Introduction

The pressure fluctuations that we perceive as sound are generally very small modulations of a much larger steady, ambient pressure. For example, in air at normal atmospheric pressure, a very loud sound, with a sound pressure level of 100 dB (measured with respect to a reference level of 20 μPa) is a modulation of only about 2 Pa (2 N/m^2) on top of an ambient pressure of about 10^5 Pa. Sound propagates as a longitudinal wave motion, which involves an interaction between the compressibility of the air and its inertia, at a wave velocity in air of about 340 ms^{-1}, which is a factor of 10^6 or so slower than the speed of propagation of electromagnetic radiation. This means that the wavelength of a sound wave is very much shorter than that of an electromagnetic wave of a comparable frequency. The frequency range of audible sound is from about 20 Hz to 20 kHz, which gives a wavelength in air of between 17 m and 17 mm. This range of wavelengths corresponds to the VHF to microwave region of electromagnetic waves, which obviously have very much higher frequencies.

An important property of sound waves at normal amplitudes is that they propagate linearly, so that the net effect at any one point of two separate sound waves will be the superposition of the effects of the sound waves acting individually. This linear property of sound means that, if an artificially generated sound wave can be engineered to be exactly out of phase with that from some annoying acoustic source, the two waves will destructively interfere and the result will be silence. This is the basis of active sound control. To turn this laboratory curiosity into a useful noise control strategy requires an understanding of both the acoustics and electrical control technology appropriate to the particular problem under consideration. In this brief review we will consider these two aspects of the problem separately, and then look at some successful practical applications of the technique. But first we outline the historical development of active sound control and the reasons why it has rapidly come of age over the past few years. The interested reader is also referred to some of the other review articles which have been published over the last few years,[1-4] and to the recent textbook on the subject written by the present authors.[5]

In a pioneering piece of work, first published in 1934, the German physicist Paul Lueg[6] outlined the basic philosophy of active sound control in ducts, and in free space. Fig. 1 is reproduced from this original patent and clearly shows, in diagram 1, the detection of an offending sinusoidal sound wave S_1 propagating in a duct using microphone M, whose electrical output is passed through some electronic controller V to drive an acoustic 'secondary' source L. This source generates the antiphase acoustic wave S_2 that cancels the initial wave. Diagram 3 illustrates the cancellation of nonsinusoidal acoustic waveforms, and diagrams 2 and 4 illustrate, in a rather idealised way, the generation of an actively generated 'acoustic shadow' behind a secondary source L when excited by a three-dimensional, freely propagating sound field generated by source A. It should be remembered that this patent was published nearly 50 years ago, only a few years after devices for the transduction of sound into electrical signals and vice versa had first become widely available (Guicking[7]).

Notable developments were made in the United States in the 1950s by Harry Olson[8] (1953), who identified the possible application of the technology to controlling the noise in cars and aircraft, and William Conover[9] (1956), who was working on controlling the sound radiated by large transformers. Fig. 2 is reproduced from Conover's 1956 paper and shows a feedforward control strategy in which a sinusoidal reference signal, at a harmonic of the line



1 Diagrams from Paul Lueg's 1934 patent[6]

frequency, is manually adjusted in amplitude and phase before being fed to the secondary loudspeaker. The amplitude and phase are varied to minimise the pressure at some distant microphone location and the hope was that in cancelling the sound at the microphone position a null would be generated in the sound radiation pattern of the transformer. Conover discussed the possibility of having an automatic control system to perform the amplitude and phase adjustment, but felt that, since the level could change by 6 dB in an hour, such a control system was beyond the state of the art at that time.

In the same paper, Conover also talks about using multiple secondary loudspeakers to obtain reductions in radiated sound over a larger angle in the directivity pattern. Such multichannel feedforward active sound control systems are now starting to find their way into production, not so much for the control of radiated sound, but more for the active control of enclosed sound fields in applications where the alternating pressure waveforms are almost periodic. Examples are the low-frequency boom inside cars due to the engine firing frequency, and the low-frequency drone in the passenger cabin of aircraft due to the propellers.

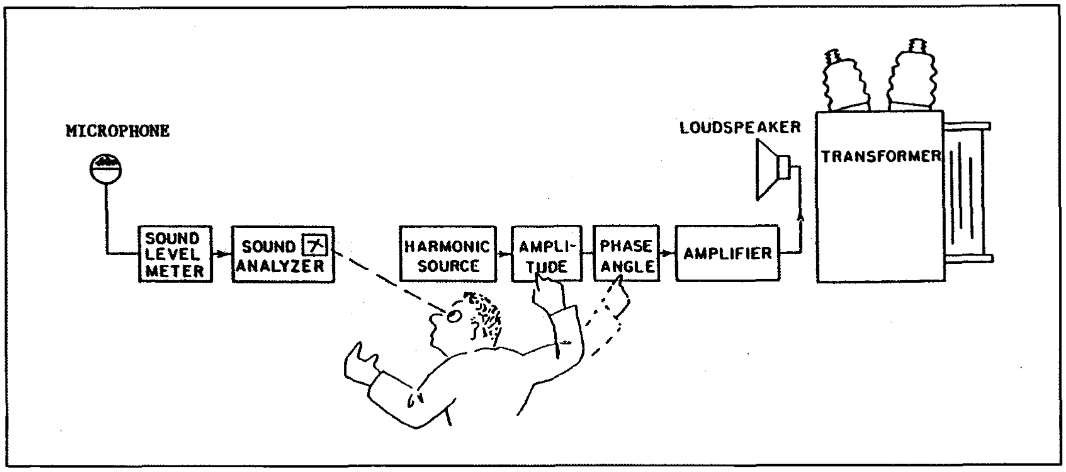
The physical reasons why these applications are confined to relatively low audio frequencies will be discussed in the next section, and after that the advances in control strategy which allow rapid adaptation of the controller (compared to that envisaged by Conover) will be described. This rapid adaptation is needed for the automotive application in particular since the frequency of excitation is constantly varying with engine speed and the level of excitation is very dependent on the engine load. Adaptation times of about 0·1 s have been achieved for a multichannel active control system in such an application.

2 Acoustic principles of active sound control

Destructive interference at a point between two wavefields that have the same frequency is familiar in a number of different types of wave phenomena. The classic demonstrations of interference were performed in optics by Thomas Young at the beginning of the nineteenth century. Fig. 3, for example, illustates Young's famous experiment in which two radially expanding waveforms were generated by illuminating a piece of paper with two closely spaced thin slits cut into it. Although the two waves interfere destructively at certain points in such demonstrations, there are other points at which the two waves interfere constructively and so increase the amplitude of the wave. It was Young who first clearly understood how the principle of superposition applies to interference between waves.

When sound is controlled by destructive interference between the original sound wave and that generated by a controllable secondary source, we are generally not just seeking to cancel at a point but to extend the area over which this destructive cancellation occurs over as wide a region of space as possible. The main acoustic objective of designing an active sound control system is thus to match, as well as we can, the spatial variations of the sound fields generated by the original (primary) acoustic source and the controllable (secondary) acoustic source. This geometric requirement is in addition to the temporal requirement: that the acoustic waveform produced by the secondary source must exactly mirror that of the primary source. The spatial matching of sound fields is only possible in a rather small number of physical situations, and these define the geometries in which active sound control will work best.

Perhaps the simplest situation in which the spatial variation of the primary and secondary sound fields can be visualised is the one-dimensional case of plane sound waves propagating in a duct, as discussed by Lueg.[6] The spatial distribution of a sinusoidal sound wave propagating from left to right in a duct is illustrated at one instant in time in Fig. 4*a* The distribution of a sound wave generated by the secondary source is also shown in this figure in which the waves on either side of



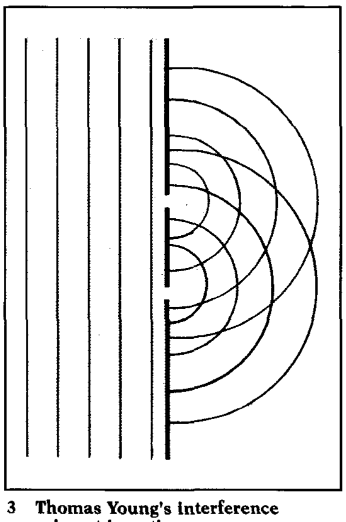
2 Manually adaptive, feedforward system for the active control of transformer noise, proposed in 1956 by William Conover[9]

the secondary source are propagating away from it. If the two waves are added together at every point in space the net pressure distribution is as illustrated in Fig. 4*b*. Clearly the two sound fields destructively interfere to the right of the secondary source, so the original sound wave has been silenced in this region. The two sound fields constructively interfere to the left of the secondary source, however, and a standing wave is formed.

It should be remembered that this net pressure distribution only exists at one instant in time; at some later time, when the wave has progressed, the pressure waveform due to the primary source will be at a maximum next to the secondary source. The waveform generated by the secondary source will then be at a minimum and at this instant the two waves not only cancel each other out to the right of the secondary source, but also the whole of the standing wave to the left (which is all in-phase) is also zero. The diagram shown by Lueg and reproduced in Fig. 1 is thus physically correct at one very particular instant in time, but does not really illustrate the overall behaviour of the active control system.

The physical effect of the secondary source is to drive the pressure immediately in front of it to zero at all times. The original sound wave propagating in the duct thus sees a large impedance discontinuity and is reflected back along the duct, exactly as would happen in an electrical transmission line by an open-circuit termination. Such an impedance discontinuity could also be engineered, for example, by using a passive tuned side branch in the duct. This arrangement would only work at certain discrete frequencies however, and the advantage of an active system is that the sound waves can be reflected back along the duct over a broad range of frequencies. Active sound control does not introduce any physical obstruction into the duct, which may impede the air flow, and this is in contrast with conventional noise control methods, which generally require splitters or baffled expansion chambers.

The second example of matching the spatial distributions of secondary and primary sound fields that we will consider is that of two sinusoidal monopole

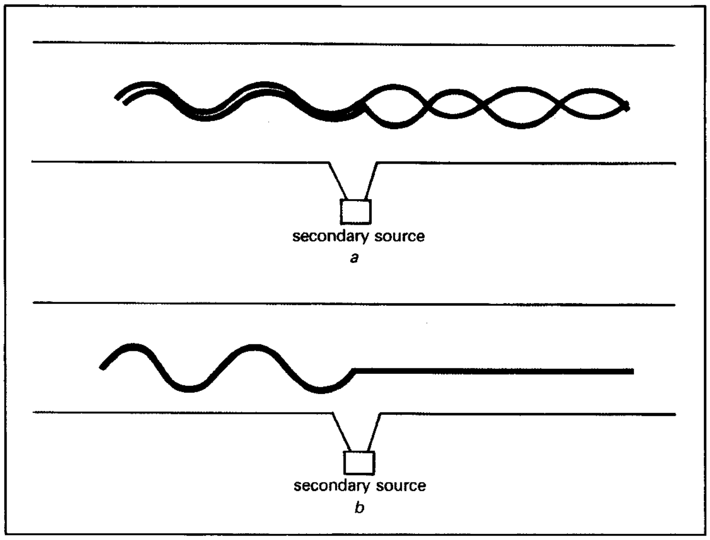


3 Thomas Young's interference experiment in optics

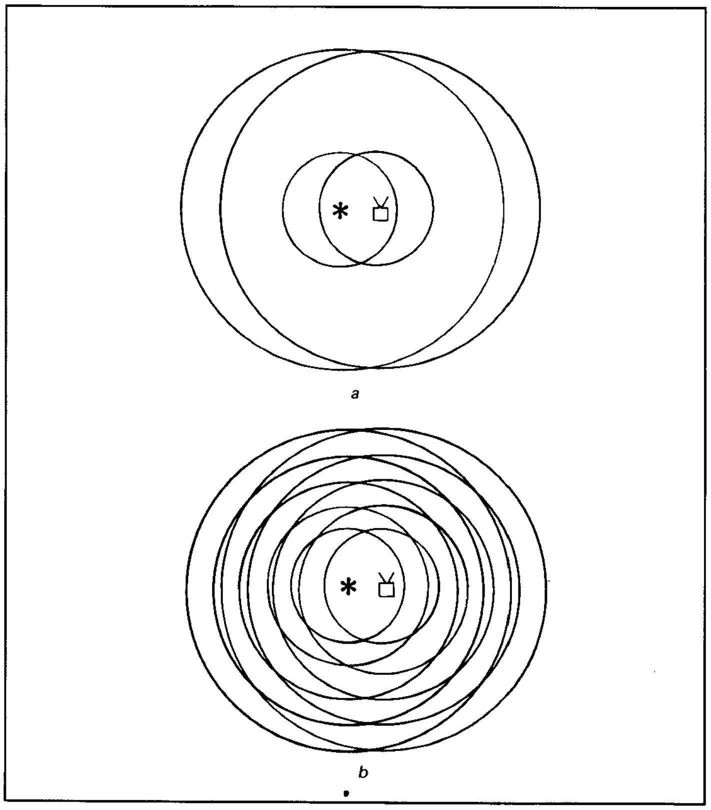
acoustic sources radiating into free space. Such an arrangement is illustrated in Fig. 5, in which the positions of the wavefronts due to the two sources are drawn in different colours.

In Fig. 5*a* the frequency of the two sources is low, so that the acoustic wavelength is large and the wavefronts are well separated compared with the distance between the sources. In this case the sound waves generated by the two sources are reasonably well matched some distance from the sources; that is, the distance between the two sets of wavefronts (which corresponds to the phase difference between the waveforms) is small in comparison with the distance between successive wave fronts from either source (which corresponds to a complete cycle of the waveform). If one source is arranged to be out of phase with the other, then clearly the two wavefields will destructively interfere, to a large extent, in this far-field region. Close to the sources the pressure levels will still be significant and the spatial distribution of the sound field will be rather complicated, but the cancellation of the far-field pressure will still result in a greatly diminished net acoustic *power output* from the two sources. The active control of radiated sound can clearly be very successful when the original source is compact and the secondary source can be positioned a small distance away, compared with the acoustic wavelength.

In Fig. 5*a*, a monopole type acoustic source has been converted into a dipole type source with a subsequent decrease in



4 Active control of a plane sinusoidal sound wave propagating from right to left in a duct: (a) spatial distributions of pressure at one instant due to primary wave (blue line) and secondary source (red line); (b) nett pressure field showing destructive interference to the right of the secondary source, and a standing wave to the left



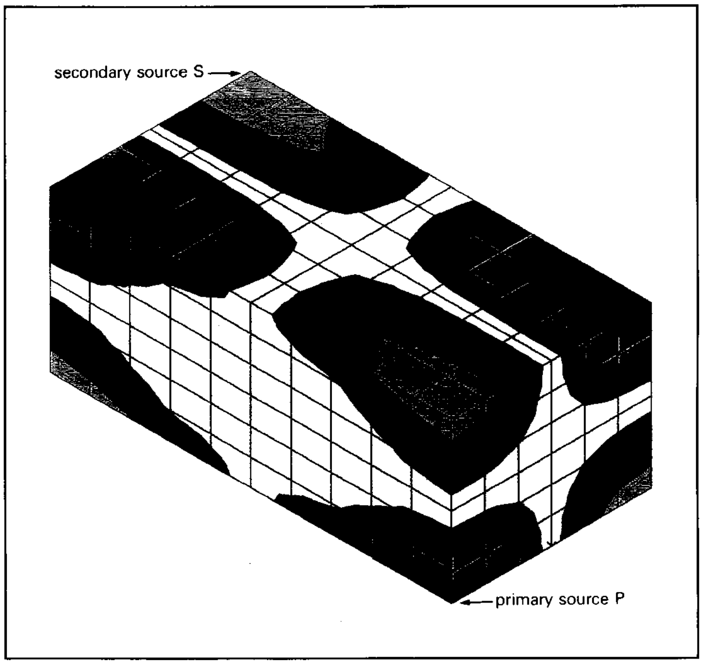
5 Wavefronts from a primary source (*) and a secondary source (☐) propagating into free space: (a) at a frequency at which the wavelength is large compared to the spacing of the sources; (b) at a frequency at which the wavelength is small compared to the spacing of the sources

radiation efficiency. In Fig. 5*b*, the wavefronts are much closer together, indicating that the frequency of the acoustic sources is higher, and the wavelength in this case is not large compared to the separation of the sources. This situation is more analogous to the Young's slit experiment illustrated in Fig. 3, in which the interference between the wavefields is at some points destructive, but at other points constructive, even far from the sources. In this case the total acoustic power radiated by the two sources will be greater than that generated by the primary source on its own. This undersirable feature can be suppressed by gradually reducing the strength of the secondary source as the frequency increases,[10] but although enhancement of the net power output can be prevented, it is still not possible to achieve significant reductions in the power output once the separation between the sources becomes comparable with half the acoustic wavelength.

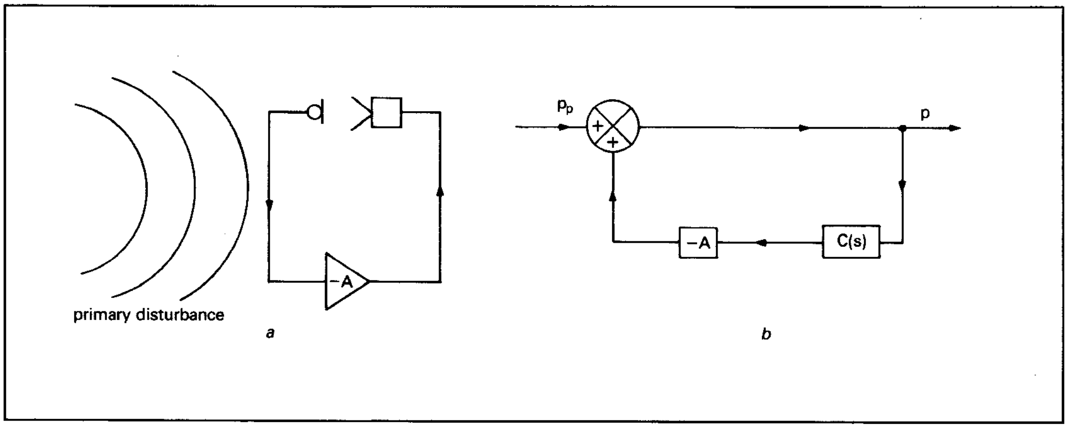
Fig. 5 illustrates a very important feature of active sound control systems: generally they will only work well if the acoustic wavelength is long compared with the separation between the primary and secondary source. It is unusual in practice to be able to position a secondary source much closer than a metre or so from the primary source, even assuming that the primary source is relatively compact compared with this distance and does behave as a monopole acoustic source. This geometric constraint imposes an upper frequency limitation on the effective operation of such active sound control systems of a few hundred hertz. It is very significant, however, that active sound control will generally work progressively better the lower the frequency becomes (i.e. the longer the wavelength), whereas conventional, passive methods of noise control generally get progressively worse at lower frequencies. This observation leads one to the conclusion that the active control of sound will not replace conventional, passive, noise control solutions in the majority of applications where high-frequency noise is important. There are some areas, however, in which low-frequency sound predominates and conventional solutions would be too bulky or weighty to be practical, in the interior of cars and aircraft for example. It is here that active sound control comes into its own.

An enclosed sound field differs from a freely propagating one in that, at certain frequencies, resonances can be set up within the enclosure that will cause an increase in the acoustic response of the enclosure at these frequencies. The characteristic pressure distribution in a rectangular enclosure for one of these acoustic resonances (or room modes) is illustrated in Fig. 6, in which the sinusoidal pressure waveform varies in amplitude (being zero along the 'nodal planes') and is either in-phase (blue) or out-of-phase (red) with the acoustic source driving the enclosure (which is assumed to be that in the bottom left-hand corner marked *P*). A secondary acoustic source in the opposite corner, marked *S*, will couple into this room mode in the opposite phase to the original source, and so Fig. 6 could equally well illustrate the pressure distribution in the enclosure when driven by source *S* with the regions coloured in red being in-phase with source *S* and those in blue being out-of-phase with *S*. Since the spatial distributions of the pressure in the enclosure due to the two sources are identical, but their waveforms are exactly out-of-phase, then complete cancellation of this room mode can be achieved by driving the secondary source to precisely the same extent as the primary source.

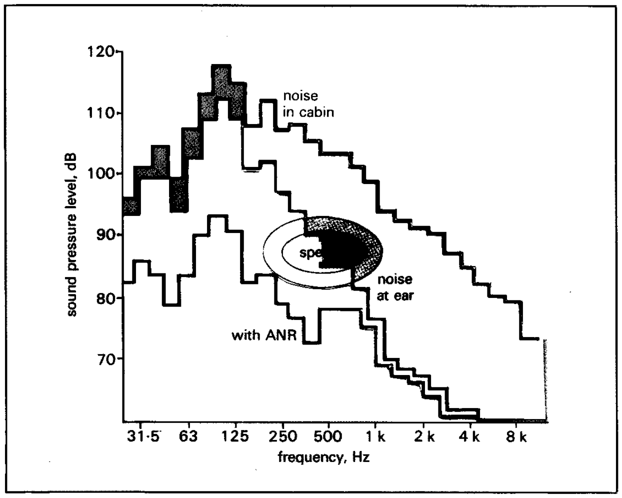
In practice, however, these room modes are fairly heavily damped (with a typical Q factor of about five) and even if an enclosure is driven by a single sinusoidal source, then many room modes will be excited to some degree. Although cancellation under such circumstances will not be perfect, the suppression of a dominant acoustic resonance is clearly possible, and by using multiple secondary sources some control over a number of room modes can be achieved, giving reductions in overall pressure level over a wider frequency range. It is an unfortunate fact that the number of acoustic modes in a room with natural frequency below the excitation frequency, rises in proportion to the cube of the excitation frequency. This means that, even if a large number of secondary sources are used, there rapidly comes a point, as the excitation frequency is increased, when the number of room modes significantly excited in the enclosure is very much larger than the number of secondary sources, and overall, global control of the sound field is not possible using an active control system. This very definite upper frequency limit in enclosures is due to much the same reasons as the upper frequency limit in the free field, namely the requirement for good spatial matching of the sound fields from primary and secondary sources breaking down as the wavelength gets smaller.



6 Pressure distribution in a rectangular enclosure due to one acoustic resonance (room mode)



7 (*a*) Physical block diagram and (*b*) equivalent electrical block diagram of a feedback active sound control system



8 1/3 octave sound pressure level spectrum in the cockpit of a jet aircraft (upper curve) and at the pilot's ear when wearing a conventional headset (middle curve). The lowest curve is the spectrum at the pilot's ear using a headset with active noise reduction (ANR) developed by Racal Acoustics

3 Electrical controller strategies in active sound control

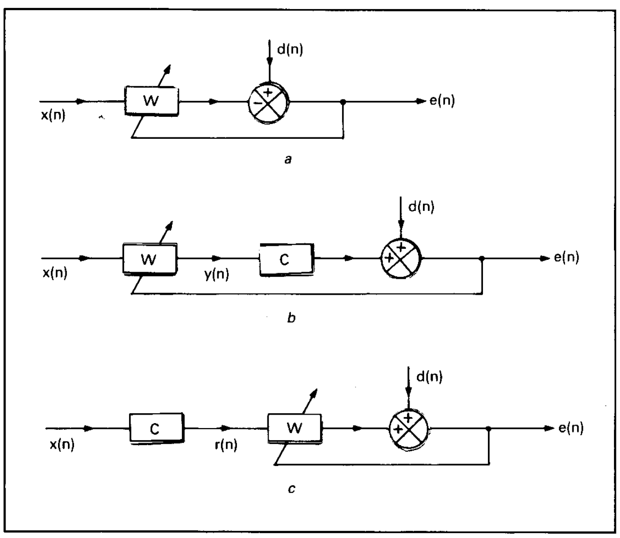
The design of electrical controllers in active sound control systems can be broadly divided into those operating using a feedforward principle (such as that of Conover[9]) and those operating using a feedback principle (such as that of Olson[8]). The feedback control approach requires no knowledge of the waveform of the primary source, and is most often implemented as the simple negative feedback loop illustrated in Fig. 7*a*, in which a loudspeaker is driven by the signal from a closely-spaced microphone after passing through a high-gain inverting amplifier. The equivalent block diagram for this arrangement is illustrated in Fig. 7*b*, in which $-A$ is the gain of the inverting amplifier and $C(s)$ is the purely electrical transfer function between the loudspeaker input and microphone output, which contains the electroacoustic response of both these transducers, together with the response of the acoustic coupling between them. p_p is the acoustic pressure at the microphone due to the primary source alone and p is the acoustic pressure at the microphone with the active control system operating. The transfer function of the complete feedback control system clearly has the form

$$\frac{p(s)}{p_p(s)} = \frac{1}{1+AC(s)} \qquad (1)$$

Provided the phase shift round the electroacoustic loop $C(s)$ is not too great, the amplifier gain A can be made large, and significant reductions in the pressure at the microphone can be achieved.

The problem obviously comes when the phase shift round the external loop $C(s)$ approaches 180° (as it will do at higher frequencies, since $C(s)$ contains some element of delay), in which case the system can become unstable if the amplifier gain is too high. Although compensators can be used in series with the amplifier to extend the usable frequency range, this tendency to instability at higher frequencies determines the practical limits of operation of such a feedback system. The stability problem is further complicated by the changes in the response of the electroacoustic loop $C(s)$ because of the variability in the response of the transducers and the acoustic environment in which they operate.

Despite these practical difficulties, a number of companies have developed production versions of feedback active control systems for controlling the noise inside the earmuffs of headsets. Fig. 8, for example, shows a 1/3 octave sound pressure spectrum representative of that in the cabin of a military jet aircraft (upper curve) and that at the ear of the pilot wearing a conventional headset (middle curve). The noise level at the ear is still clearly excessive at low frequencies and can interfere with the perception of speech, whose range of frequencies and levels are also



9 Block diagrams of an adaptive digital filter *W* used for (*a*) the adaptive cancellation of electrical noise, and (*b*) the active control of sound. (*c*) Another version of (*b*) assuming the filter is slowly time-varying

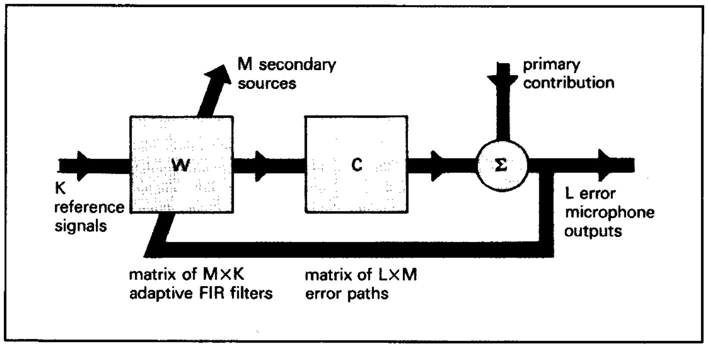
shown as the grey area in the centre of the figure. The lower curve represents the noise at the pilot's ear when wearing a headset fitted with an active noise reduction (ANR) system developed at Racal Acoustics.[11]

The other control strategy which can be adopted (feedforward contol) is applicable when a reference signal can be generated which is related to the sound radiated by the primary source. An example of this approach has already been illustrated in Fig. 2, as used by Conover in the active control of transformer noise. Because the sound field generated by the primary source is generally nonstationary, such a feedforward controller must be adaptive to track these changes in the primary field. Conover illustrates manual adaptation to compensate for changes that occur in the sound propagating from his transformer, due partly to atmospheric changes over periods of several hours. In controlling the engine noise inside cars, however, changes in the amplitude, phase and frequency of the primary pressure field occur much more quickly than this, on a timescale of a second, and a way of rapidly adapting such a feedforward controller must be found.

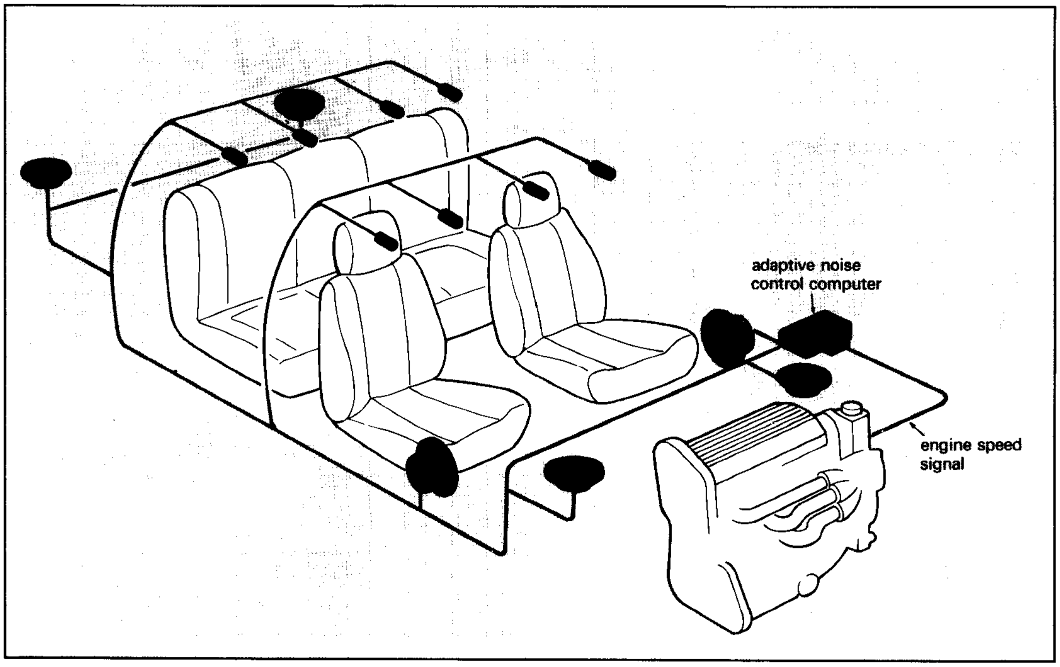
One very successful area of signal processing that has developed rapidly over the past few decades is that of adaptive digital filtering. In Fig. 9 we compare block diagrams in which an adaptive digital filter is used in an *electrical* noise cancellation problem (Fig. 9*a*) and in which an adaptive digital filter is used as the feedforward controller in an active *sound* control system (Fig. 9*b*), such as that illustrated in Fig. 2. In Figs. 9*a* and 9*b*, $x(n)$ is the sampled reference signal. $d(n)$ is the 'desired signal' in the electrical cancellation problem of Fig. 9*a* and the signal due to the primary field in the active sound control problem of Fig. 9*b*. $e(n)$ is the residual electrical error signal in Fig. 9*a* and the residual acoustical signal, due to both primary and secondary sources, in Fig. 9*b*. The major difference between these two diagrams is the presence of the electroacoustic path C between the input to the loudspeaker and output from the microphone in Fig. 9*b*. The presence of this 'error path' means that the usual algorithms used to update such adaptive digital filters, such as the LMS algorithm, will not generally converge in such applications. The LMS algorithm for the electrical noise cancellation problem (Fig. 9*a*) may be written as

$$\mathbf{w}(n+1) = \mathbf{w}(n)+\alpha e(n)\mathbf{x}(n) \qquad (2)$$

where $\mathbf{w}(n)$ is the vector of FIR filter coefficients at the nth sample



10 Block diagram of a multichannel feedforward active sound control system



11 Schematic diagram of a six-loudspeaker, eight-microphone active sound control system for reducing the engine boom inside a car

time, $\mathbf{x}(n)$ is the vector of previous reference signals, $e(n)$ is the instantaneous error signal and α is a convergence coefficient.

The failure of this algorithm in Fig. 9*b* is due to the presence of the error path operating on $e(n)$ but not on $x(n)$, which biases the estimate of the crosscorrelation between these signals, which is the basis for the update quantity $e(n)\mathbf{x}(n)$ used in eqn. 2. A solution to this problem, put forward independently by Widrow *et al.*[12] and Burgess[13] in 1981, was that the order of the transfer functions of the filter and error path can be notionally reversed (Fig. 9*c*), in which case the adaptive filter is operating directly on the error signal, as in Fig. 9*a*, but is now being driven by $x(n)$ after having been filtered by a representation of the error path C to produce the filtered reference signal $r(n)$. This commuting of the elements of the block diagram is not strictly valid if the adaptive filter is time varying, but it does suggest a modification to the LMS algorithm which has been very successful used in practical applications.

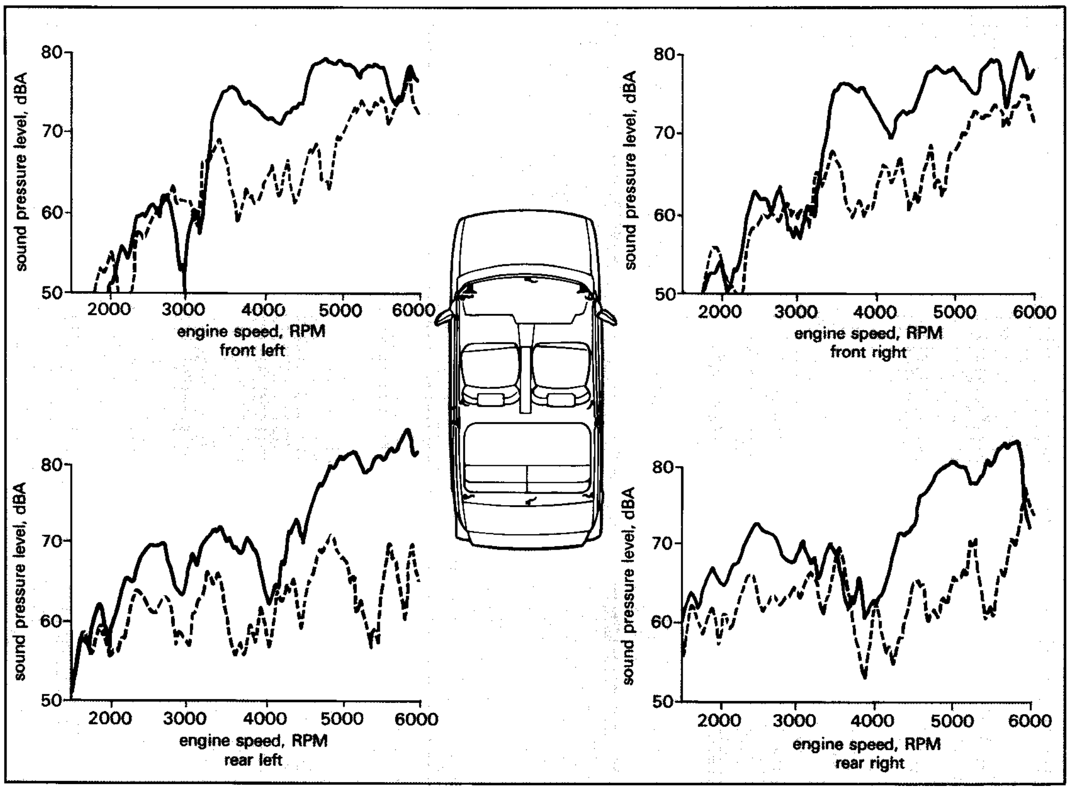
$$\mathbf{w}(n+1) = \mathbf{w}(n) - \alpha e(n)\mathbf{r}(n) \qquad (3)$$

where $\mathbf{r}(n)$ is now the vector of previous reference signals filtered by a representation of the error path C. Eqn. 3 is known as the 'filtered x' LMS algorithm. Although in practice the filtered reference signal now has to be generated (by passing $x(n)$ through some electrical model of the error path) to perform the update on the adaptive filter, so that some knowledge of the system under control is required, the convergence of the algorithm has been found to be very robust to errors in this error path model.

To actively control the sound throughout an enclosure it is generally necessary to use a number of secondary acoustic sources to minimise the sum of the squared pressures at a number of error microphones. The general block diagram of such a multichannel adaptive active control system is shown in Fig. 10, in which it has been additionally assumed that multiple reference signals are being used to drive a matrix of adaptive control filters. The multichannel generalisation of the filtered x LMS algorithm, which minimises the sum of the squared error signals by adjusting each of the coefficients of an array of digital filters, can be written as[14]

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \alpha \mathbf{R}(n)\mathbf{e}(n) \qquad (4)$$

where $\mathbf{w}(n)$ is a vector containing the coefficients of all the adaptive filters, $\mathbf{e}(n)$ is a vector containing all the error signals and $\mathbf{R}(n)$ is a matrix containing each of the delayed reference signals passed through every error path from each loudspeaker to each microphone. The calculation of the exact least squares solution for $\mathbf{w}(n)$ involves the inverse of the matrix $\mathbf{R}^T(n)\mathbf{R}(n)$, and by carefully ordering the various filter coefficients in the vector $\mathbf{w}(n)$ this matrix can be arranged to be block Toeplitz,[15] which can result in more efficient numerical solution. The multichannel LMS algorithm described by eqn. 4 has been implemented, for example, in a practical system for the control of the sound at the first three harmonics of the blade-passing frequency inside a propeller



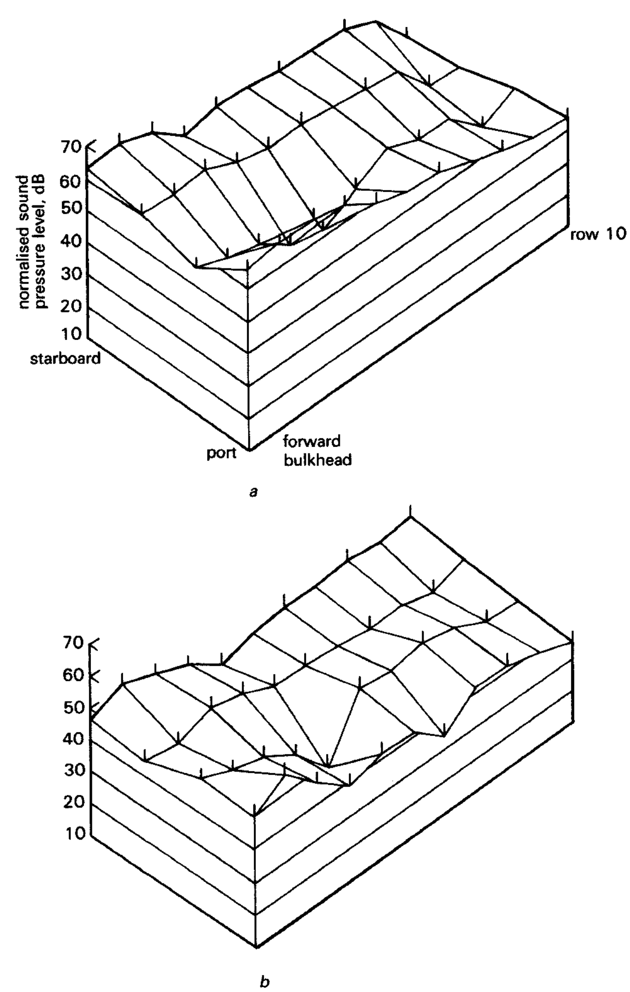
12 A-weighted sound pressure level due to the engine firing frequency at head height in the four seat positions of the small hatchback car illustrated when accelerated hard in second gear
—— standard car; ----- with active sound control system

aircraft using 16 loudspeakers and 32 microphones. It is acoustically desirable to minimise the sum of the squares of the pressures at a greater number of microphones than there are secondary sources, and recent work has suggested that under such conditions the stability of the multichannel LMS algorithm is even more robust to errors in the estimates of the error path than the single-channel filtered x algorithm.

In practical situations, then, where some prior knowledge of the primary sound field is available, feedforward control methods using adaptive digital filters can provide stable control that can still track nonstationarities. The algorithms generally used to adapt such digital filters can also be generalised to cope with the case of multiple reference signals, secondary sources and error microphones.

4 Some examples of the practical application of active sound control inside cars and aircraft

As remarked earlier, the sound inside cars and propeller aircraft tends to be dominated by the low-frequency periodic noise from the car engine, or from the propeller sweeping past the aircraft fuselage. In both these cases, reference signals of the appropriate frequency are readily generated from once-per-revolution tachometer signals obtained from the car or aircraft engines. Adaptive feedforward control systems have been built and used in practice for both these applications and it is informative to briefly discuss the common features and the differences between the two control systems.



13 Spatial distribution of the normalised sound pressure level in the passenger cabin of a British Aerospace 748 propeller aircraft at the blade passing frequency: (*a*) in the standard aircraft; (*b*) with active noise control

Although the size of passenger cabin in a 50-seat propeller aircraft, as used for the flight tests of a practical control system, is obviously somewhat larger than the size of a typical car interior, the upper frequency of operation (about 200–300 Hz) of both systems is quite similar. This is because the acoustic limit on global control is due to the rise in the acoustic modal density with frequency, which is similar in the two cases. Because of the difference in size, however, the number of secondary loudspeakers (16) and error microphones (32) required to maintain reasonable active control in the aircraft cabin is considerably higher than that necessary to control the noise at any one speed in the car, which is typically two loudspeakers and four microphones. The frequency of excitation in a car can vary by a factor of up to ten to one as the engine is taken from idle to maximum speed, whereas the rotational speed of the propellers is normally kept within a much narrower range. This means that a two-loudspeaker, four-microphone system is not usually sufficient to couple into the variety of acoustic modes that are excited over the speed range experienced in a car, and a four- or six-loudspeaker, eight-microphone system is more typically used. The controller for the car, then, has to manage a smaller number of channels than that for the aircraft, but must be able to adapt more quickly to cope with the rapid changes in engine speed and engine load.[16] A schematic diagram of a practical control system for a car is shown in Fig. 11. The positioning of the loudspeakers and microphones, together with the fine tuning of the adaptation coefficient and various other algorithm parameters must be optimised from one model of vehicle to another for the best results. Fig. 12 shows the sound pressure level at the engine firing frequency only, passed through an 'A-weighting' frequency-selective filter, at various positions in a small hatchback car as it is being accelerated hard along a test track in second gear. The solid lines are the levels in the normal production car and the dashed lines are with the benefit of a four-loudspeaker, eight-microphone active control system. Large reductions of 10–15 dB are achieved in the front seats by the active control system above an engine speed of about 3000 RPM (which corresponds to an engine firing frequency of 100 Hz), and at somewhat lower engine speeds in the rear seats. The engine firing frequency is not the only component of the sound in a car but, as cars become more powerful and lighter, this low-frequency 'boom' increasingly dominates the overall sound level, and is very difficult to control using conventional, passive techniques to the extent shown in Fig. 12 without significantly increasing the overall weight of the vehicle.

A similar dilemma faces the designer of a propeller aircraft: reductions in the dominant low-frequency sound in the passenger compartment can only be achieved, using conventional methods, with a significant weight penalty. Flight trials of an active sound control system in a BAe 748, however, have shown that up to 14 dB can be taken off the sum of the squared outputs of the 32 control microphones at the blade passing frequency of 88 Hz using 16 internal loudspeakers as secondary sources.[17] Fig. 13 is an isometric plot of the magnitude of the pressure at 88 Hz measured at the 32 error microphones uniformly distributed at seated head height in the passenger cabin of the aircraft used for the in-flight experiments. The normalised sound pressure levels in decibels are plotted as dashed points at the four microphone positions across the cabin (from port to starboard) and at ten seat row positions (from forward bulkhead to row 10) going along the cabin. Fig. 13*a* shows the sound field without active control and Fig. 13*b* shows the pressure field, measured in the same positions, after the active control system was switched on. The overall reduction in level and a general flattening of the spatial variation in the sound field can be seen clearly.

5 Conclusions

Although the basic principles of active sound control have been known for over 50 years, it is only recently that advances in digital signal-processing technology have enabled practical multichannel active control systems to be realised. The possibilities opened up by these advances have also stimulated a re-examination of the basic physical principles of active sound control.

In general, it is only possible for the sound field from an array of controlled secondary sources to destructively interfere with that from some original primary source of sound over a useful volume of space if

(*a*) the waveform of the secondary source is the mirror image of the primary source (a temporal constraint) and

(*b*) the soundfield distribution from the secondary source nearly matches that from the primary source (a spatial constraint).

In practice, the latter condition can often only be met by positioning the secondary source within a fraction of an acoustic wavelength of the primary source. This leads to a fundamental upper frequency range of operation for global active sound control of a few hundred hertz. Active noise control systems working in a more restricted volume, such as an earmuff, can work to somewhat higher frequencies, but physical limitations always limit the upper frequency range of operation.

The fact that active control works better at lower frequencies complements more conventional noise control methods, using absorptive materials for example, which tend to work better at higher audio frequencies. In applications where strong low-frequency components are a problem, and in which the additional weight associated with passive shielding or absorption cannot be tolerated, active sound control offers an attractive alternative. Current applications being developed include controlling the low-frequency engine noise in cars and the low-frequency propeller noise in the passenger cabins of aircraft. In both these applications very significant reductions in sound pressure level have been experimentally demonstrated.

Acknowledgments

Fundamental work on active control at the Institute of Sound and Vibration Research at Southampton University has been supported by the DTI and UK SERC. The practical applications in aircraft and cars have been the result of co-operation with British Aerospace and Lotus Engineering, and we are grateful to Lotus for supplying Figs. 11 and 12 in this paper and to Racal Acoustics for providing Fig. 8.

References

1 WARNAKA, G. E.: 'Active attenuation of noise—the state of the art', *Noise Control Eng.*, May/June 1982, pp. 100–110

2 CHAPLIN, G. B. B.: 'Anti-noise—the Essex breakthrough', *Chart. Mech. Eng.*, 1983, pp. 41–47

3 FFOWCS-WILLIAMS, J. E.: 'Anti-sound', *Proc. R. Soc. London*, A395, 1984, pp. 63–88

4 LEITCH, R. R., and TOKHI, M. O.: 'Active noise control systems', *IEE Proc. A*, 1987, **134,** (6), pp. 525–546

5 NELSON, P. A., and ELLIOTT, S. J.: 'Active control of sound' (Academic Press, 1990) (in press)

6 LUEG, P.: 'Process of silencing sound oscillations' US Patent 2 043 416, 1934

7 GUICKING, D.: 'From Paul Lueg to digital signal processing—55 years of active noise control'. IEEE ASSP Workshop on Applications of signal processing to audio and acoustics, 1989

8 OLSON, H. F., and MAY, E. G.: 'Electronic sound absorber', *J. Acoust. Soc. Amer.*, November 1953, **25,** 1130–1136

9 CONOVER, W. B.: 'Fighting noise with noise', *Noise Control*, 1956, pp. 78–92

10 NELSON, P. A., CURTIS, A. R. D., ELLIOTT, S. J., and BULLMORE, A. J.: 'The minimum power output of free field point sources and the active control of sound', *J. Sound & Vib.*, 1987, **116,** (3), pp. 397–414

11 WHEELER, P. D.: 'The role of noise cancellation techniques in aircrew voice communications systems'. Proc. Royal Aeronautical Soc. Symp. on Helmets and helmet mounted devices, 1987

12 WIDROW, B., SHUR, D., and SHAFFER: 'On adaptive inverse control'. Proc. 15th ASILOMAR Conf. on Circuits, systems and computers, 1981

13 BURGESS, J. C.: 'Active adaptive sound control in a duct: a computer simulation', *J. Acoust. Soc. Amer.*, 1981, **70,** (3), pp. 715–726

14 ELLIOTT, S. J., STOTHERS, I. M., and NELSON, P. A.: 'A multiple error LMS algorithm and its application to the active control of sound and vibrations', *IEEE Trans.*, 1987, **ASSP-35,** (1), pp. 1423–1434

15 NELSON, P. A., and ELLIOTT, S. J.: 'Multi-channel adaptive filters for the reproduction and active control of sound fields'. Proc. IEEE ASSP Workshop on Applications of signal processing to audio and acoustics, New York, 1989

16 ELLIOTT, S. J., STOTHERS, I. M., NELSON, P. A., MCDONALD, A. M., QUINN, D. C., and SAUNDERS, T.: 'The active control of engine noise inside cars'. Proc. Inter-Noise 88, 1988, pp. 987–990

17 ELLIOTT, S. J., NELSON, P. A., STOTHERS, I. M., and BOUCHER, C. C.: 'In-flight experiments on the active control of propeller-induced cabin noise', *J. Sound & Vib.*, 1990 (in press)

First received 21st May and in revised form 22nd June 1990

The authors are with the Institute of Sound and Vibration Research, The University, Southampton SO9 5NH, UK.