A STUDY OF THE PERFORMANCE OF
AN OLSON-TYPE ACTIVE NOISE CONTROLLER
AND THE POSSIBILITY OF THE REDUCTION OF CABIN NOISE

by

S. E. Keith and H. S. B. Scholaert

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Abstract

In contrast to orthodox sound insulating techniques, the active noise controller is a device designed to reduce sound levels by means of an electronic transducing system. The device is a basic feedback control system composed of a speaker, microphone, amplifier and control unit. The scheme can be effective in reducing low frequency noise; thus, it is of particular interest to the transportation industry, in particular to aircraft manufacturers: as attenuation of low frequency noise, to increase passenger comfort, can be at once costly and cumbersome, when conventional sound absorption methods are employed.

The idea of active noise control was pioneered in the early fifties by H. F. Olson and E. G. May. They produced an electronic sound absorber which appeared to be successful over small volumes, in a unidirectional sound field. This work has re-examined these accomplishments and more recent developments to test their suitability to the aircraft industry. The results suggest only limited possible use for all systems studied.
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<tr>
<td>A</td>
<td>speaker radius (cm)</td>
</tr>
<tr>
<td>B</td>
<td>large dimensionless number corresponding to feedback loop gain</td>
</tr>
<tr>
<td>c</td>
<td>speed of sound (m/sec)</td>
</tr>
<tr>
<td>C</td>
<td>capacitance (Farads)</td>
</tr>
<tr>
<td>d</td>
<td>effective distance (m)</td>
</tr>
<tr>
<td>D</td>
<td>speaker separation (m)</td>
</tr>
<tr>
<td>f</td>
<td>frequency (Hz)</td>
</tr>
<tr>
<td>G</td>
<td>gain (dB)</td>
</tr>
<tr>
<td>H</td>
<td>transfer function of the controller</td>
</tr>
<tr>
<td>I</td>
<td>sound intensity (watt/m²)</td>
</tr>
<tr>
<td>k</td>
<td>wavelength constant (m⁻¹)</td>
</tr>
<tr>
<td>K</td>
<td>arbitrary constant</td>
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<tr>
<td>L</td>
<td>inductance (Henry)</td>
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<tr>
<td>P</td>
<td>sound pressure (Newton/m²)</td>
</tr>
<tr>
<td>r</td>
<td>distance to speaker (m)</td>
</tr>
<tr>
<td>R</td>
<td>resistance (Ω)</td>
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<tr>
<td>U₀</td>
<td>piston velocity (m/sec)</td>
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<tr>
<td>V</td>
<td>voltage (volts)</td>
</tr>
<tr>
<td>Zₘ</td>
<td>mechanical impedance</td>
</tr>
<tr>
<td>Zₐ</td>
<td>radiation impedance</td>
</tr>
<tr>
<td>φ</td>
<td>phase (deg)</td>
</tr>
<tr>
<td>ρ₀</td>
<td>density of air (kg/m³)</td>
</tr>
<tr>
<td>θ</td>
<td>angle (radian, deg)</td>
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<tr>
<td>ω</td>
<td>angular frequency (radians)</td>
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1. **INTRODUCTION**

To increase passenger comfort, modern aircraft have been designed to minimize the noise levels heard within their cabins. The structure of the fuselage attenuates the high frequency noise quite well; however, at low and sometimes moderate frequencies, a considerable amount of noise energy is transmitted into the cabin. It is in this frequency range that active noise control could possibly reduce perceived sound pressure levels by as much as 10 to 20 dB, without significant increases in structural weight. To obtain similar results using orthodox sound insulating techniques, the weight of the structure would have to be increased by a factor of two to three.

The essence of active noise control is based on the cancellation of sound by a second sound out of phase with the first. The development of a successful active noise control system, however, hinges on stable operation. The design of such a suitable controller is not altogether trivial as attested by the limited success to date.

The basic scheme had already been proposed in the early fifties by Olson and May [1]. At present, several researchers have succeeded in applying active noise control to the suppression of noise in ducts [2], in the reduction of transformer noise [3], and even in the attenuation of low frequency noise in ear defenders [4].

It is the pioneering work of Olson and May which is re-examined herein, along with some more recent developments. The results of the study are applied to the design of a suitable prototype noise controller in order to reduce the perceived sound pressure levels in the vicinity of the head of a passenger.

2. **OVERVIEW OF THE CONTROL PROBLEM**

2.1 **The Device**

The active noise controller is a feedback control system. It consists of a microphone, an amplifier, a speaker, and a control unit. The microphone is used to sense the magnitude and frequency of the signal to be attenuated. The control unit is designed to process the signal such that it becomes approximately out of phase with the original input. Finally, the amplifier-speaker combination provides the acoustical output required. A block diagram of the system is shown in Fig. 1.

It would seem that the design of the control segment is relatively simple. It must be indicated, however, that this unit need not only produce a 180 degree phase shift but must also compensate for the detrimental phase shifts incurred by the speaker and the time delay between reception of the signal to be controlled and the control signal emitted by the speaker. This delay is inherent in the system.

2.2 **Phase and Magnitude Response of Individual Elements**

The microphone unit, for all useful purposes, can be considered to possess constant gain and constant phase shift over the frequency range of interest, namely from 20 to 500 Hz. On the other hand, the amplifier-speaker combination introduces a new independent transfer function.
Using a digital signal processor, the amplifier-speaker system was studied in order to obtain the magnitude and phase response. As can be seen in Fig. 2 the amplitude response remains relatively constant. The phase, however, as a function of frequency, does not behave in a manner consistent with classical high or low frequency roll-offs.

In any case, the phase plot can be approximated by a series of straight lines in the range of 50 to 500 Hz and it is theoretically possible for the control unit to compensate for such arbitrary behaviour of the phase. The controller would require a circuit with an increasing phase angle and be based on a series of phase shifters.

2.3 A Simulated Controller

In the initial trials, a basic active noise controller was set up, whereby a summing amplifier simulated the algebraic properties of incident sound waves. Hence, a series of "black box" control units could be tested.

The attenuation was measured in terms of the input voltage of the summing amplifier. The inputs to the latter being a signal produced by a random noise generator and this same signal after having been processed by the control unit and the speaker-microphone system (see Fig. 3).

A computer program was written, and then used in order to best fit the data obtained for the phase response of the speaker-microphone combination, with the general equations of two phase shifters placed in series. This led to the calculation of the appropriate R-C parameters involved for optimum attenuation.

Satisfactory results were obtained when employing the network illustrated in Fig. 4. Fine tuning enabled an attenuation of up to 28 dB to be achieved. Furthermore, the system provided attenuation over an extremely wide range of frequencies, that is, from 50 to at least 800 Hz.

This system, although representing a theoretically suitable control unit, does however lack a critical component: the feedback loop. Once the feedback loop was integrated into the actual control system, certain difficulties arose.

For any realizable active noise control system, the gain of the control unit plays a major role in the feasibility. Extremely high gains are required of this segment, without inducing system instabilities. The destabilizing effect of time delay, discussed below, make it difficult to incorporate the prerequisite high gains.

Control theory states that for any electronic control inserted in the feedback path, there can be no poles with positive real parts in the corresponding transfer function, if the system under consideration is to be stable. Furthermore, for transfer functions which can be represented as ratios of polynomials, an increase in the slope of the phase response must be accompanied by an increase in the slope of the magnitude response. Hence, the use of the control units designed above, in a configuration analogous to Olson's, inevitably led to instabilities.

2.4 Time Delay

The problem due to time delay arises from the finite spacing between the control microphone and the speaker that emits the controlling sound waves. In
any feedback control system, such as the present one, a time delay will always create the possibility of instability. Nyquist analysis shows that for some sufficiently large gain, the system response curve is bound to encircle the critical -1 point in a clockwise manner. On the other hand, we have already noted that high gain is a prerequisite for a successful feedback control system of this type.

A second problem arises in connection with the coherence time of random noise. The larger the signal bandwidth, the shorter the coherence time. If the time delay is of the order of the coherence time, then it is impossible to realize any significant degree of control.

3. CONTROLLER NETWORKS

3.1 Delay Line System

At first, an attempt was made to eliminate the adverse effects of the speaker at the microphone, the idea of the scheme being to render the microphone insensitive to the output of the speaker. This in turn would make it possible to obtain a closed loop gain that is approximately unity. The only modifications to the input signal would then be solely due to the equipment's phase relationships. Such a scheme can be realized by subtracting a time delayed replicate of the control signal from the instantaneous microphone signal. A test delay line system was constructed using two identical, but separate, and out of phase speaker-microphone combinations feeding into a common summing circuit. High and low pass filters were then used to reduce spurious noise in both circuits. Each configuration enhanced the stability of the other, allowing greater than unity gain through the entire system. Such a gain provides good control beyond the microphone.

Certain problems were however inherent to the electronic arrangement discussed above. The system introduced further instabilities. It was effective only in a restricted bandwidth, having a rather complicated filter characteristic. Furthermore, the delay line, based on a "bucket-brigade" type of system, produced an additional unwanted, and yet irrepresible noise due to the large gains involved in the feedback path. Hence, although the idea of eliminating the control signal from the microphone signal ought to work in principle, its feasibility is somewhat impaired due to practical limitations of a real system.

3.2 Other Systems

Another possible approach requires the use of a unidirectional microphone. The directional response of such a microphone, resembling a cardidoid pattern, would provide in effect a system analogous to the electronic delay previously discussed.

The output of the cardidoid microphones available can be modelled as the sum of the responses of a velocity microphone and a pressure microphone. When operating at distances of less than three meters from a speaker, a pressure microphone will display a uniform response, however a velocity microphone will exhibit an increase in sensitivity as the signal frequency decreases. This is particularly harmful to any system modelled along Olson's proposal, since low frequencies and small microphone-to-speaker distances are involved.
A velocity microphone by itself might, however, be suitable in a controller aimed at attenuation of noise from specified sources. Since the microphone response pattern is that of a figure eight, one only needs to point a lobe at the incoming signal and keep the null point towards the speaker. An attempt was then made to discover the effects of simple high and low pass filters. This resulted in minimal attenuation, over a wide range, but with unavoidable high and low frequency enhancement (see Fig. 5). The idea, in itself, was therefore found unsuitable.

3.3 Duplication of Speaker Characteristics

In order to find a suitable approach to the design of a control unit, the electromechanical properties of the speaker system were studied. Following along the lines of Olson, the speaker was modelled by certain equivalent electrical networks (see Fig. 6).

The principal parameters governing the sound radiated by a speaker are the radiation impedance \((Z_R)\) and the mechanical impedance \((Z_M)\) of the speaker. The radiation impedance is modelled as a resistance \(R_1\), which accounts for the energy lost by radiation, and an inductance \(L_1\), due to the apparent or virtual mass associated with the incompressible motion of the air in front of the speaker. The speaker mass \((\omega_0^2)\), the stiffness of the suspension \(C_1\), and damping \((\alpha R_2)\) govern the mechanical impedance. The cabinet enclosure provides extra damping and stiffness \((R_3, C_2)\).

The sound pressure \(p_2\), at or near the speaker, is the sum of the incident pressure \(p\), and the pressure generated by the speaker \((\omega p_3)\). Thus, one may write:

\[
\frac{p_1 - p_2}{Z_R} = \frac{p_2 - p_3}{Z_M}
\]

\(p_3\) is governed by the controller and \(p_2\) is to be minimized for an arbitrary \(p_1\). Furthermore, \(p_2\) can be considered to be the signal received by the control microphone. If one postulates \(p_3 = H p_2\), one finds that \(p_2\) is minimized if

\[
H = \left(1 - \frac{B Z_M}{Z_R}\right)
\]

where \(B\) is a large number (i.e., high gain). The transfer function \(H\) can be realized if all the system properties \((R_1, L, R_2, \text{etc.})\) are known.

Using standard transducer theory, simple experimental procedures and calculations provided data for each parameter involved. An electronic circuit duplicating the transfer function of the acoustical network was then designed. This was done in the hope of counteracting the effects of the loudspeaker. The output of the circuit would simply be inverted.

The finalized system, although stable, consisted of some rather incompatible parameters. The electronic circuit was basically composed of three branches, one of which was relatively ineffective in terms of processing the signal (see Fig. 7). This branch, in fact, corresponded to the effects of the parameters \(C_2, R_3\) and \(C_1\). Due to its lack of influence, the branch was eliminated altogether, resulting in a simpler and compact system.
Once activated, the controller produced rather interesting results. An attenuation of up to 10 dB was obtained at the control microphone over a large frequency range (see Fig. 8). Although this was not yet quite suitable for our purpose, the unit proved to be rather useful in the implementation of a final design.

3.4 Narrow-Band Type Controller

The use of a narrow band filter resulted in a successful way to control the system. The filter increased the feedback loop gain to a very high value at specific frequencies. This was then followed by a rapid drop-off just before the network reaches the point of instability. A narrow band is sought so that the phase will not vary greatly. Otherwise, the decreasing phase angle caused by the speaker and time delay will lead to instabilities. High gain is required at the centre frequency, in order to make the feedback control system most effective.

A Bruel and Kjaer type 2107 frequency analyzer was found to provide the characteristics required. It is a constant percentage bandwidth analyzer composed of 3 tunable stages. Not only did the filter possess an acceptable phase and gain response for the system (see Fig. 9), but it was also tunable over the entire audio spectrum. The latter attribute made the filter an ideal test model.

It was found that more than one such unit could be used. The individual response of various filters could be summed and hence a piecewise fit to the required curve would be obtained. For the case discussed herein, three analyzers were used and the output added by a summing amplifier. The results were most promising (see Fig. 10). An attenuation of up to 25 dB at the control microphone could be obtained at a variable centre frequency. A lower limiting frequency however occurred where the response became somewhat unstable. With a chosen centre frequency of about 150 Hz reasonable attenuation over a bandwidth of 150 Hz was found. Beyond this range two peaks appeared. The enhancement is produced at the lower end of the spectrum, near 60 Hz and at the upper end at about 380 Hz.

To compensate for these undesirable effects, the results of our studies in the electromechanical properties of the speaker system were used. Using the response of the electronic circuit modelling the speaker's acoustical network and that of the three analyzers, a new system was constructed (see Fig. 11). The final results were, as expected, a combination of the best controlling achievements to date (see Fig. 12). By lowering the centre frequency, an attenuation of up to 30 dB was maintained at the control microphone for a bandwidth of 80 Hz. The lower frequency peak was greatly reduced and forced to the lower end of the spectrum, near 30 Hz. Furthermore, for the frequencies under consideration, the enhancement at the upper end of the spectrum was totally eliminated.

4. LIMITATIONS OF THE NOISE CONTROLLER

4.1 Localized Performance of the Controller

Although the attenuation obtained at the microphone was quite high, the performance was found to be very localized indeed. A second microphone was used to sense sound pressure levels at a distance from the controller. The measurements show a rapid decrease in attenuation as distance from the controlling microphone is increased (see Fig. 13). These observations are in fact close to, if not worse, than predictions of simple theory.
Consider that the sound source behaves as a radiating piston; one finds a rapid drop in sound intensity as distance to the source increases. In the near field, the intensity (a mean square pressure) is given by:

\[
I = 2\rho_o U_o^2 c \sin \left( \frac{\pi f}{c} \left( \sqrt{r^2 + A^2} - r \right) \right)
\]

At low frequencies and for small speaker diameters \((a = 12.5 \text{ cm})\) this reduces to:

\[
I = 2\rho_o U_o^2 f(\sqrt{r^2 + A^2} - r)
\]

Similarly the far field intensity

\[
I = \rho_o c k^2 U_o^2 \left( \frac{mA^2}{2\pi} \right)^2 \left[ \frac{2J_1(kA \sin \theta)}{kA \sin \theta} \right]^2
\]

simplifies to

\[
I = \rho_o c U_o^2 \frac{A^2}{8r^2}
\]

Now since

\[
\frac{-2}{2\rho_o c} = I
\]

one is able to use these relations to predict a loss in pressure amplitude as a function of distance and hence the resulting loss in attenuation. For example, in the far field, the pressure drop is proportional to \(1/r\). Assuming perfect cancellation at the control microphone an effective attenuation of 10 dB is restricted to a volume bounded by a radius of half the microphone to speaker distance.

Further calculations produced a plot of useful controller range versus microphone to speaker distance, all in terms of the speaker radius \(a\) (see Fig. 13). From this graph it can be seen that the most effective results are in the near field of large speakers. However feasibility must be considered; such a large speaker becomes rather impractical for the suggested scheme. Another extreme alternative is the placement of the microphone at large distances from a speaker of reasonable size. Therefore, for an arrangement such as in our original setup, an optimum in terms of size and microphone to speaker distance must be chosen. To obtain reasonable attenuation of at least 10 dB for a range of about 10 cm, a speaker of at least a 20 cm radius is required with a minimum microphone to speaker distance of 40 cm. This is clearly not suitable; the unit would no longer be compact and integrated as originally intended.

Another problem limiting the system to very localized performance is the change in the phase angle of the sound source. For many noise sources at a distance, we can assume that there is some set of parallel planes in which the phase is at least locally constant. As one moves from one plane to another, the phase will change with respect to frequency and distance:
Locally this is true for the anti-sound emanating from the control speaker. To study this effect of phase change on attenuation a graph can be constructed (see Fig. 15), whereby attenuation is plotted as a function of phase and effective pressure ratio for the far field.

Successful control depends on the existence of a point or area where phase differs by 180 degrees and magnitude is approximately equal to that of the noise. Hence, depending on the way the system is optimized, reasonable attenuation may never occur. When noise and control anti-sound are propagating in the same direction, the phase between the two will remain constant. Hence, if the anti-noise is exactly out of phase with the noise at the microphone, it will always be so, and controlling will simply be limited by the distance for which the anti-noise maintains a comparable magnitude to the noise (see Fig. 16). On the other hand, if the anti-sound is not out of phase at the microphone, the noise may never be controlled at all. If the noise and anti-noise are propagating towards each other, a new situation occurs. The rate of change of the phase difference between the two sources, with respect to distance, is twice the rate of change of phase for a single source. Therefore, in this case, the volume in which attenuation may be controlled is effectively reduced (see Fig. 16). Finally, for a control source perpendicular to the control speaker axis, the phase difference between the two sounds will vary with change in distance. The worst case would have the phase change with distance. A better situation is also possible where no phase change will occur. The final result is some asymmetric controllable volume in which attenuation occurs (see Fig. 16).

In the general case, noise arrives from different directions, and the point for optimum phase difference of 180 degrees will vary. The greater this change, the greater the distance will be from the controlled volume to the microphone. However the point of optimum magnitude is always the same. Therefore, since the phase must be greater than 162 degrees and no less than 198 degrees for a 10 dB cancellation, it can be seen that the appropriate magnitude of the control noise will vary in terms of position to such an extent that no control is possible.

For Olson's system, control of the noise was attempted at a distance in front of the control microphone. The magnitude of the anti-noise required was greater than that of the noise at the microphone, and the phase relationships were optimized so that attenuation occurred in front of the control microphone. The magnitude of the anti-noise required was greater than that of the noise at the microphone, and the phase relationships were optimized so that attenuation occurs for noise propagating perpendicularly to the anti-noise. The only way a specific volume can be controlled in a random noise field is to have the control optimized for best attenuation at the microphone. In doing so, the optimum controlling point never changes while there is always a definite volume where attenuation is possible regardless of the origin of the noise.

In any case, considering changes in sound pressure ratios and phase angle with distance (see Fig. 15), alternatives in terms of speaker-microphone arrangements were sought and evaluated.
4.2 Viable Systems

The scheme, somewhat resembling Olson's suggestion, is altogether inapplicable to an aircraft cabin. In reference to our previous discussion, we can point out that this system is ultimately influenced by the direction in which the noise is propagating. In other words, in an unspecified sound field, as that of an aircraft cabin, the scheme appears to be unreliable.

To avoid rapid changes in anti-sound intensity as one approaches the control speaker, a new arrangement is proposed. The system would utilize a total of four control speakers. The implementation of new sound sources produces a much more uniform sound field (see Fig. 17).

The control microphone is placed in the vicinity of the passenger's head. The speakers are then arranged in an appropriate manner, at the ceiling, so as to produce a suitable sound field (see Fig. 18). Such a field is relatively constant in the horizontal plane; the most rapid changes in anti-sound intensity occur along the vertical axis. The change in phase angle with distance is now much less of a potential threat to suitable attenuation. The phase can be shown to be approximately constant over the equipotential lines, as those of Fig. 17.

For this scheme, the best possible circumstances occur for the case of noise propagation downwards from the ceiling (see Fig. 19). Successful attenuation occurs over the complete horizontal and vertical planes because of the constant noise to anti-noise phase relationships over these areas. If the noise is incident normal to the speaker axis, then the areas in which attenuation occurs are limited by the change in anti-noise phase with distance (see Fig. 19). For example, an attenuation of at least 10 dB, at a frequency of up to 100 Hz, is indicated in Fig. 20 for a maximum range of 17 cm to either side of the microphone. This is due to the much more rapid change in phase angles between the noise and anti-noise.

In the worst possible case, we can conclude that the control volume is disk shaped having a minimum height of 17 cm and a comparable radius (see Fig. 19). This can be relatively suitable for the applications suggested. The passenger's freedom of movement is, however, somewhat restricted.

To increase the minimum controllable volume, the implementation of more speakers is suggested. For example, eight control speakers in a cubic formation can produce an anti-sound field of very constant intensity (see Fig. 21). The only attenuation restrictions inside the cube are caused once again by the familiar phase relationships. The worst case of column control would result in attenuation inside a sphere of 17 cm in radius. This is in fact the best possible controlling situation for any system employing a single microphone.

If one is to carry this process to extremes, a multitude of individually controlled speakers would be ideal in reproducing a general sound field, as the Huygen's principle suggests [5]. A similar inference can be drawn from the analysis of Kempton [6].

5. SUMMARY

An active noise controller as suggested by Olson is only viable if the noise is characterized by rather restrictive features, as in the case of noise propagation
in ducts. Otherwise, in a general sound field, the scheme can only be successful over very small volumes. This makes the idea suitable to the attenuation of noise in ear defenders.

The active noise controller in this form is not appropriate to the cancellation of a random noise field, as encountered in an aircraft cabin. For any such device the effective distance over which attenuation will occur is limited by the decrease in sound intensity as the distance from the controller increases. The change in sound pressure amplitude is inversely proportional to the radial distance to the sound source. A second limitation occurs due to the change in the effective phase difference between the noise and anti-noise. If one required at least 10 dB attenuation, the relationship between effective distance and frequency becomes:

\[ d = \frac{32c}{2\pi f} \]

Therefore, with a uniform anti-noise sound field, for a frequency of 100 Hz, the system will provide attenuation over a range of at most 17 cm to either side of the control point.

The overall performance of the scheme can be somewhat improved by employing more than one speaker. In fact, the ideal configuration would require an infinity of separately controlled sound sources to duplicate the noise field, as suggested by Huygen's principle. Any idea is however limited in this application to providing a simple system which could easily be integrated into the design of an aircraft's interior. Hence, a system of four control speakers and one microphone, as in Fig. 18, is proposed as the most suitable design.

Although the control volume is still rather limited by the effective distance relationship previously described, this configuration makes it possible to obtain a much more uniform anti-noise sound field. It has been shown that this provides a workable scheme regardless of the direction of the incident noise.

The concept of active noise control inside aircraft cabins, or for that matter any other vehicle, is still thought to be very limited. Unless very low frequencies are a problem, any system such as the ones described above cannot produce a very large control volume. The suggested design will surely provide considerable noise attenuation in the vicinity of a passenger's head. However, the passenger's movement would be very limited, and it would not be difficult for one to place his head beyond the effects of the controller.

It is only by the use of much more extensive speaker-microphone combinations that active noise control can be successful at attenuating random sound fields. However, such a system at the moment is not foreseeably applicable to commercial uses in the aircraft industry.
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FIG. 1 BLOCK DIAGRAM OF SIMPLE NOISE CONTROLLER.

FIG. 2 MAGNITUDE AND PHASE RESPONSE OF SPEAKER-AMPLIFIER COMBINATION.
FIG. 3 SIMULATION OF AN ACTIVE NOISE CONTROLLER; ACOUSTIC SIGNAL MODELS
SPEAKER RESPONSE AND TIME DELAY, $V_{out}$ = HYPOTHETICAL SOUND PRESSURE
NEAR CONTROL MICROPHONE.

FIG. 4 (a) CONTROL NETWORK, (b) TYPICAL FREQUENCY RESPONSE CURVES.
FIG. 5 ATTENUATION OF CONTROL NETWORK (FIG. 4) WITH LOW PASS FILTER SET AT 660 Hz. REGIONS OF ENHANCEMENT ARE SHADED.

FIG. 6 EQUIVALENT NETWORK OF THE ACOUSTICAL PORTION OF THE ACTIVE NOISE CONTROLLER.

FIG. 7 ELECTRONIC CONTROL CIRCUITRY DRIVING P₂ IN FIG. 6 (UPPER BRANCH WAS INEFFECTIVE).
FIG. 8 PERFORMANCE CURVES OF THE ACTIVE NOISE CONTROLLER.

FIG. 9 BODE PLOT FOR B&K 2107 ANALYZER.

FIG. 10 PERFORMANCE CURVES OF THE ACTIVE NOISE CONTROLLER UTILIZING A B&K 2107 FILTER.
FIG. 11  ACTIVE NOISE CONTROL NETWORK UTILIZING THREE B&K 2107 FILTERS. 
= HiFi PREAMPLIFIER WITH BASS AND TREBLE CONTROL.

FIG. 12  PERFORMANCE CURVE OF THE ACTIVE NOISE CONTROLLER IN FIG. 11.
FIG. 13 INFLUENCE OF OBSERVER-MICROPHONE SEPARATION.

FIG. 14 NOISE REDUCTION (dB) AS A FUNCTION OF DISTANCE FROM THE CONTROL MICROPHONE AND SPEAKER-CONTROL MICROPHONE SEPARATION. DISTANCES ARE NORMALIZED BY LOUDSPEAKER RADIUS.
FIG. 15 NOISE REDUCTION (dB) AS A FUNCTION OF RELATIVE CONTROL SIGNAL
LEVEL AND PHASE.

FIG. 16 SCHEMATIC OF CONTROLLABLE VOLUME (SHADEd) WITH OPTIMUM ATTENUATION
AT THE MICROPHONE (a) AND AT AN ARBITRARY FIELD POINT (b).
© LOCATION OF MAXIMUM NOISE REDUCTION.
FIG. 17 EQUIPOTENTIAL LINES OF SOUND INTENSITY IN A PLANE BISECTING "FOUR SPEAKER SQUARE CONFIGURATION" OF DIMENSION D.

FIG. 18 SUGGESTED SPEAKER-MICROPHONE PLACEMENT.
FIG. 19 SCHEMATIC OF CONTROLLABLE VOLUMES FOR SEVERAL FOUR-SPEAKER CONFIGURATIONS; DIMENSIONS APPLY AT $f = 100$ Hz.
FIG. 20 EFFECTIVE RANGE OF CONTROL VS. FREQUENCY.

FIG. 21 MINIMUM CONTROL VOLUME FOR AN EIGHT-SPEAKER CONFIGURATION.
A STUDY OF THE PERFORMANCE OF AN OLSON-TYPE ACTIVE NOISE CONTROLLER AND THE POSSIBILITY OF THE REDUCTION OF CABIN NOISE

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In contrast to orthodox sound insulating techniques, the active noise controller is a device designed to reduce sound levels by means of an electronic transducing system. The device is a basic feedback control system composed of a speaker, microphone, amplifier and control unit. The scheme can be effective in reducing low frequency noise; thus, it is of particular interest to the transportation industry, in particular to aircraft manufacturers: as attenuation of low frequency noise, to increase passenger comfort, can be at once costly and cumbersome, when conventional sound absorption methods are employed.

The idea of active noise control was pioneered in the early fifties by H. F. Olson and E. O. May. They produced an electronic sound absorber which appeared to be successful over small volumes, in a unidirectional sound field. This work has re-examined these accomplishments and more recent developments to test their suitability to the aircraft industry. The results suggest only limited possible use for all systems studied.

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