

microphones 30-1 and 30-2 is supplied as first inputs to adder 44 and ^{time delay} ~~delay time~~ 45, respectively. Forward delay 46 provides a time delay, τ , where $\tau = \frac{L}{c}$ and L is the separation distance of microphones 30-1 and 30-2 and c is the speed of sound in duct 32, as a second input to time delay 45. Delay 45 provides a second input to adder 44. The output of adder 44 is supplied as a first input to adder 48. The output of adder 48 represents the forward pressure wave, P_f , and is supplied via switch S2 to filters 14-1 and 22 and is supplied as a first input to time delay 50 in the feedback loop. Feedback delay 52, having a time delay of 2τ , provides a second input to delay 50. A loss term of 0.95 appears in the feedback loop as block 54 which receives an input from delay 50 and supplies a second input to adder 48. This small leakage in the feedback loop controls the stability of the filter 40 and keeps the filter gain at its poles within reasonable limits. This value was arbitrarily set to 0.95, however, any value between 0.9 and 0.99 could be chosen without appreciable loss in accuracy.

In Figure 2, filters 22 and 14-1 receive the output of PWF 40, which represents P_f at microphone 30-1, as an input. Filter 14-1 which is copied from filter 14, as described above with respect to Figure 1, provides a first input to multiplier 26 and an output signal from error microphone 18 is provided as a second input to multiplier 26. The output of multiplier 26 is provided as a second input to filter 22. Filter 22 has an output representing the corrected forward pressure wave which is supplied via adder 38 to adder 41 as a first input. The output of filter 22 accounts for the time delay from microphone 30-1 to the canceling loudspeaker 13 and any anomaly associated with the frequency response of loudspeaker 13. The output of adder 38 which represents the driving force for speaker 13 and any gain corrections that may be required due to system effects is supplied to speaker 13 via adder 41. Referring again to Figure 3, power supplied to coil 13-6 via adder 41 causes its movement and the movement of integral cone 13-3 which produces sound. Coil 13-7 moves therewith and the movement of coil 13-7 in the air gap between pole 13-1 and pole 13-2 induces a voltage in coil 13-7 which is related to the movement/velocity of coil 13-7. Since coil 13-7 is moving as a unit with cone 13-3 and coil 13-6, the voltage induced by movement of coil 13-7 is a direct indication of the velocity of movement of cone 13-3

and therefore the sound being produced by speaker 13 since the velocity of cone 13-3 is directly proportional to the forward pressure wave of the speaker (P_{fs}) caused by its movement. The voltage induced in coil 13-7 is sensed, passed through feedback gain step 42 as gain K and supplied as second input to adder 41 thereby correcting the driving signal for speaker 13 responsive to the actual operation of speaker 13.

In comparing Figures 1 and 2, it will be noted that the Figure 2 device eliminates filters 14-2 and 24 and multiplier 28 and adder 36.

Turning now to Figure 5, the progressive wave filter 40, PWF, of Figure 4 can be replaced with forward pressure wave approximation filter 100. The noise sensed by sensing microphone 30-1 is supplied as a first input to adder 101 and as an input to divider 102. The noise sensed by sensing microphone 30-2 is supplied as a second input to adder 101. The output of adder 101 is supplied to integrator 103 which supplies an input to divider 104. The outputs of dividers 104 and 102 are supplied as first and second inputs, respectively, to adder 105 which has an output P_f . The embodiment of the PWF 40 described in Figure 4 reduces to that in Figure 5 when $kL < \lambda/8$ where k is the acoustic wave number, L is the separation distance between microphones 30-1 and 30-2, and λ = the acoustic wavelength.

Before proceeding with the description of the embodiments of Figures 6 and 7, note their common servo (feedback) mechanism at the loudspeaker 112 and power amplifier 113. The servo mechanism provides a feedback signal that is proportional to the loudspeaker's cone velocity through feedback gain stage 114, having a gain K . The velocity feedback signal could be achieved via a variety of mechanisms such as providing a coil on the cone, as in Figure 3, which moves with respect to the magnet of loudspeaker 112 so as to produce a signal indicative of movement of the cone and thereby of the sound being generated by loudspeaker 112. The feedback gain, K is not known and would have to be predetermined before control starts. This gain would be loudspeaker dependent and in general would be on the order of $1/100$. In addition, the power amplifier 113 is assumed to have a unity power transfer function. The power

amplifier 113 is essentially a current amplifier that supplies drive current to the loudspeaker 112 for required actuation.

A major difference between the embodiments of Figures 6 and 7 from those of Figures 2-4 is that microphone 130-2 is used in both the PW filter 132 and as the error sensor, being placed directly over the control loudspeaker 112. Alternatively, if desired, it could also be located downstream of control loudspeaker 112. The indicated noise source 134 can be either on, or off during Steps 1 and 2. It is assumed that the noise source 134 is on during Step 3. If it is off, the ANC system will be essentially inoperative.

To initiate calibration of loudspeaker 112, switch S3 is closed, switches S4 (Figure 6 only) and S5 are open and noise source 134 is on. The white noise source 110 (constant amplitude, broadband frequency distribution) supplies a signal to the Loudspeaker Correction Filter (adaptive Finite Impulse Response (FIR) structure) 116, H_C , multiplier 129, and Desired Loudspeaker Velocity Response Filter, 117, H_D , in loudspeaker adaptive correction block or circuit 190. Circuit 190 has the function of computing the required correction filter 116, H_C , for the loudspeaker's velocity ~~response~~ ^{response} based upon a desired response, H_D of filter 117. The output of H_C correction filter 116 is supplied as an input to the servo-loudspeaker via closed switch S3 and adder 138. The servo-output (i.e. velocity of the loudspeaker's cone before gain, K) is fed back to adder 128 negated and summed with the output of the velocity response filter 117. This signal represents an error signal and is the deviation of the actual loudspeaker's cone velocity from the desired loudspeaker's cone velocity. The error signal is combined in a least mean square, LMS, fashion with the input signal from the noise generator 110. Specifically the signal from adder 128 is multiplied with an input signal from noise generator 110 in multiplier 129 and a small constant (not shown), generally referred to as a convergence parameter, which is typically 0.1% of the input power. The process continues until the error signal is reduced to a predetermined small value. After convergence, the H_C -filter 116 is copied to filter 116-1 of the FIR controller 192 or 192', as indicated, and filter 116-2 of C-plant

CLAIMS

What is Claimed is:

1. A noise canceling system for an air distribution structure comprising:

duct means (32) for delivering air;

a source of noise (34) located with respect to said duct means so as to transmit noise into said duct means as a forward component which is subject to reflection due to coaction with said duct means to produce a reflected component whereby noise from said source of noise can be present in said duct means as both a forward component and a reflected component;

*insert ** → sensing means (30-1; 30-2; 130-1; 130-2) located in said duct means;
noise canceling means (13; 112) located with respect to said duct means so as to transmit a canceling noise into said duct means;

circuitry connected to said sensing means and said noise canceling means and including means for distinguishing between said forward component and said reflected component and for producing an output representative of ^{only} said forward component (40; 100; 132) and means for driving said noise canceling means so as to produce noise corresponding ^{only} to said forward component.

2. The system of claim 1 wherein said means for distinguishing between said forward component and said reflected component is a progressive wave filter (40; 132).

3. The system of claim 1 wherein said means for distinguishing between said forward component and said reflected component is a forward pressure wave approximation filter (100).

* said sensing means includes a pair of sensors in a spaced relationship along said duct means relative to said source of noise;

4. The system of claim 1 wherein said circuitry includes means (13-7) for sensing a parameter corresponding to actual output of said noise canceling means and means (42; 114) for feeding back said sensed output to said means for driving said noise canceling means so as to adjust said means for driving said noise canceling means.

5. The system of claim 1 wherein said circuitry provides a time delay (22) corresponding to the time necessary for noise from said source of noise to travel between said sensing means and said noise canceling means.

~~6. The system of claim 1 wherein said sensing means (30-1, 30-2; 130-1; 130-2) is a pair of sensors in a spaced relationship relative to said source of noise.~~

~~6~~ 6. The system of claim ~~1~~ further including:
a second sensing means (18) located in said duct means at a location such that said noise canceling means is located intermediate said second sensing means and said source of noise;

said second sensing means being connected to said circuitry means so as to provide a signal representative of the result of interaction between noise from said source of noise and said noise canceling means;

said circuitry including means (41) responsive to said signal representative of the result of interaction between noise from said source of noise and said noise canceling means to adjust said means for driving said noise canceling means.

~~7~~ 7. The system of claim ~~1~~ wherein one of said pair of sensors (130-2) is located opposite said noise canceling means, and said one of said pair of sensors (130-1) is connected to error sensing means (132) which forms a part of said circuitry.

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The system of claim 8 wherein:

said circuitry includes a progressive wave filter (40; 132) connected to said pair of sensors and a finite impulse response controller (192; 192') connected to one of said pair of sensors and said progressive wave filter and providing an output to said noise canceling means.

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10. The system of claim 9 further including:

a loudspeaker calibration circuit including a white noise source (10; 110), an adaptive transverse filter (140), and a desired loudspeaker velocity response transfer function; and

means (S3) for selectively connecting said calibration circuit to said noise canceling means and disabling said output of said controller.

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11. The system of claim 10 further including:

a an adaptive error path identification circuit including an adaptive transverse filter (14; 116) and connected to said white noise source.

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12. The system of claim 11 wherein said controller includes a filter (140-1) copied from said adaptive transverse filter of said identification circuit.

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13. The system of claim 12 wherein said controller includes a filter (116-1) copied from said adaptive transverse filter of said speaker calibration circuit.

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14. The system of claim 13 wherein said controller includes a filter (116-1) copied from said adaptive transverse filter of said speaker calibration circuit.

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15. The system of claim 1 wherein said sensing means includes a pair of sensors in a spaced relationship relative to said source of noise located upstream of said noise canceling means.