

A FEEDFORWARD CONTROL SYSTEM FOR THE ACTIVE NOISE CONTROL OF A TRIPLE-PANE AIRCRAFT-TYPE WINDOW

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Aircraft windows are a significant path for structure-borne and air-borne noise transmission in aircraft. Turbulent Boundary Layer noise is mainly transmitted into the cabin by airborne paths, but structure-borne noise, associated with engine vibration and the interaction between aerodynamic wakes and aircraft structure, makes a significant contribution to interior noise levels, especially at certain discrete frequencies. Active systems can be efficiently designed in order to reduce the amount of noise transmitted through the aircraft window, contributing to improve passenger's comfort perception of the interior noise. In this paper, an active noise control system applied to a triple-pane aircraft window prototype is presented. Piezoelectric actuators are integrated on a window prototype in order to adaptively enhance its vibro-acoustic absorption capabilities and to control the acoustic radiation from the vibrating panes. The active noise control performances of the developed Filtered-x LMS controller, driving the piezo stacks mounted on the window prototype, are numerically evaluated in a reference acoustic enclosure subjected to a primary tonal noise source.

1. Introduction

Airplane fuselage sidewalls in general, and passenger windows in particular, are the dominant path of external noise entry causing high interior noise levels in a treated aircraft, [1]. Fuselage sidewalls radiate sound because of external forcing, causing the surface to vibrate, and absorbs energy from the incident noise field because of the finite surface acoustic impedance. The distinction between

the incident and the radiated field is not obvious, even in the near field, and recently novel methods have been proposed to experimentally distinguish the different sound field components via the spatial information captured by beam-forming techniques, [2]. Transmission Loss tests on window panels installed between a reverberant and an anechoic room, [3], have demonstrated that the total sound intensity is mainly radiated by the window area and partially absorbed by the trim panel.

In order to counteract the structure-borne noise and the air-borne noise transmitted into the aircraft interior, recent works have analyzed the acoustic benefits provided by the use of damped plexiglass windows with transparent viscoelastic damping material sandwiched between the layers, [3]. On the other hand, the introduction of active systems on window panels can be crucial to enhance its vibro-acoustic absorption capabilities by efficiently controlling the structure-borne noise radiation. Active noise control systems applied to conventional homogeneous windows have been proposed in [4]. The main target noise components, impinging the aircraft cabin, and the transmission mechanism through the aircraft window, have a great influence on the overall acoustic field of a cabin. In this paper, the noise generated by turboprop aircraft engines, flying in stationary conditions and having the fundamental tonal frequency of 124 Hz (3-bladed propeller), is considered. The broadband case is analyzed as well by assuming a chirp signal varying between 120-128 Hz. A feed-forward control algorithm is proposed to control the anti-noise signals. The reference signal is derived from the Blade Passage Frequency (BPF) sensed by a tachometer. The error signal is measured by a microphone located at the head of the aircraft seat. Fig. 1 schematically describes the system architecture. The canceling sound is emitted by a window prototype, Fig. 2a, properly designed to generate secondary noise signals from the vibration induced by piezo stack actuators exciting the structure. The assembled prototype and the actuation mechanism are detailed in Fig. 2b. Numerical simulations of active noise control are performed to demonstrate the active system capabilities in suppressing the experienced acoustic field in a reference acoustic enclosure, represented by a steel box mounting the window prototype. The experimental set-up is shown in Fig. 4. The unknown transfer functions between the control microphone and the primary and the secondary noise sources are experimentally measured and numerically fitted.

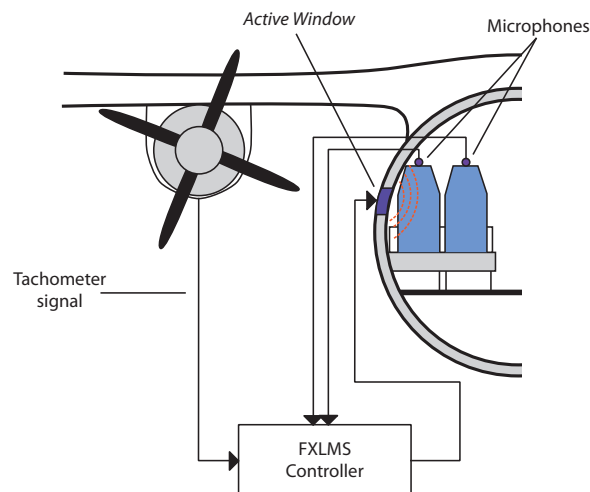


Figure 1. The FXLMS controller driving the *Active Window* on a turboprop aircraft.

2. The Filtered-x LMS algorithm

The principle of the *filtered-reference LMS* algorithm is to find coefficients of a FIR filter which minimize a quadratic cost function given by the mean square of the error signal, as described in Fig. 3.

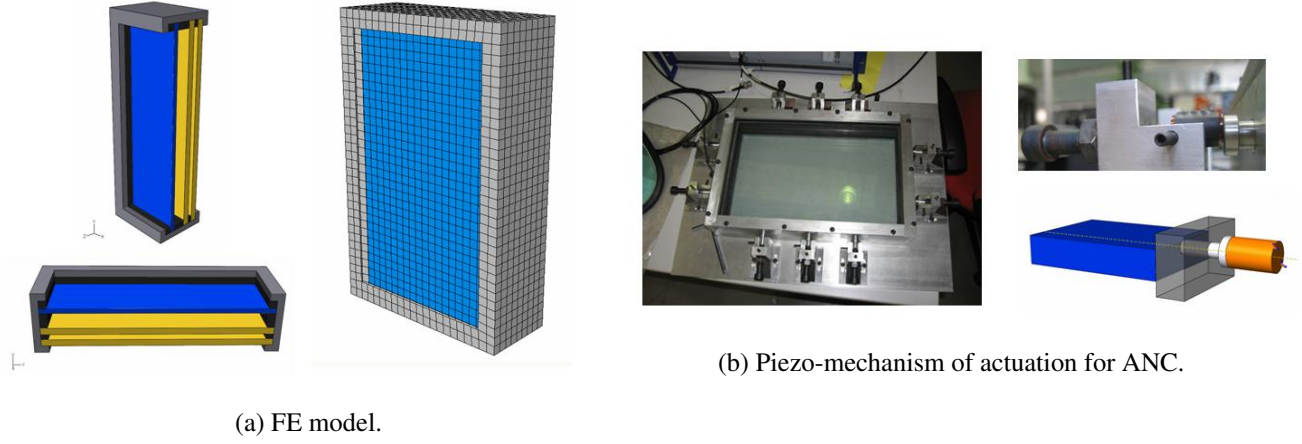


Figure 2. Active window prototype.

It is an adaptive control of feedforward type, because it needs a reference signal correlated with the disturbance. The FIR filter output is given by:

$$y(n) = \sum_{i=0}^{I-1} w_i x(n-i) \quad (1)$$

where w_i are the *coefficients*, or *weights*, of the digital filter of I dimension. The quadratic cost function to minimize is defined as:

$$J = E[e^2(n)] \quad (2)$$

where E denotes the *expectation operator*.

The *error signal*, $e(n)$, is given by the sum of the *disturbance signal*, $d(n)$, and the *filtered reference signal*, $r(n)$, filtered by the feedforward controller of FIR filter type, with coefficients w_i , so that

$$e(n) = d(n) + \sum_{i=0}^{I-1} w_i r(n-i) = d(n) + \mathbf{w}^T \mathbf{r}(n) \quad (3)$$

where

$$\mathbf{w} = [w_0 \ \cdots \ w_{I-1}]^T \quad (4)$$

is the vector of controller coefficients and

$$\mathbf{r}(n) = [r(n) \ \cdots \ r(n-I+1)]^T \quad (5)$$

is the vector of past values of the filtered reference signal. The superscript T denotes the transpose of the vectors, which are assumed to be column vectors.

The value of the coefficients of the FIR filter that reduces the mean-square error to a minimum can be found by differentiating the cost function with respect to each coefficient and setting all of the resulting derivatives to zero. This optimal filter coefficients vector can be expressed as

$$\mathbf{w}_{opt} = -\mathbf{R}_{rr}^{-1} \mathbf{r}_{rd} \quad (6)$$

where $\mathbf{R}_{rr}$ is the autocorrelation matrix of the filtered reference signal, and $\mathbf{r}_{rd}$ is the cross-correlation vector between the filtered reference signal and the disturbance signal. The filter that has these optimal coefficients is often called the *Wiener filter*. By making the coefficients of such a filter adaptive, i.e. sequentially adjusting the filter coefficients so that they evolve in a direction which minimizes the mean-square error, the adaptation algorithm is defined as:

$$\mathbf{w}(new) = \mathbf{w}(old) - \mu \frac{\partial J}{\partial \mathbf{w}}(old) \quad (7)$$

which is called the *steepest-descent* algorithm, where μ is a convergence factor. Using an instantaneous estimate of the gradient, which is sometimes called the *stochastic gradient*, the adaptation algorithm becomes:

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \alpha \mathbf{r}(n)e(n) \quad (8)$$

where $\alpha = 2\mu$ is the convergence coefficient. This is known as the *filtered-reference LMS*, or the *filtered-x LMS* (FXLMS), since the reference signal is generally denoted as x . In practice, the filtered reference signal is generated using an *estimated* version of the true plant response represented by a *plant model*. This can be implemented as a separate real-time filter, $\hat{G}(z)$, which is used to generate the filtered reference signal, $\hat{\mathbf{r}}(n)$, as illustrated in Fig. 3. The practical version of the filtered-reference LMS algorithm can thus be written as:

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \alpha \hat{\mathbf{r}}(n)e(n) \quad (9)$$

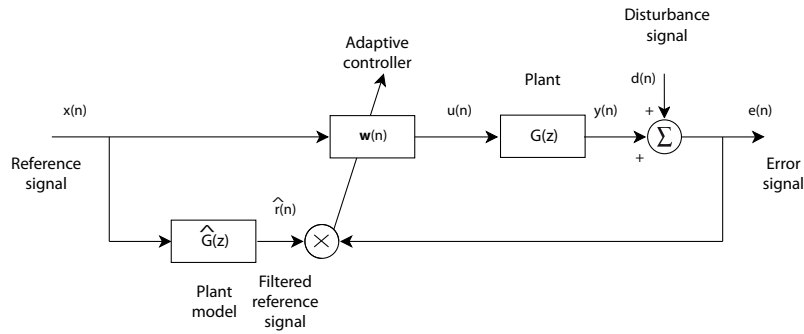


Figure 3. Block diagram of the filtered-reference LMS algorithm

3. Experimental System Identification

A pure feed-forward control approach requires a very precise model of the system to perform well. An accurate knowledge of the system under control can be obtained by numerical simulations and/or deduced from experimental measurements based on input-output approach. The model can be analytical, and should account for all the features of the real system, such as three dimensional effects, time varying behavior, temperature, etc. But the model can also be experimental, previously-recorded, and stored in the control algorithm. This second solution has the advantage of being more precise for complicated systems because it includes their uniqueness and it can be revaluated during the lifetime if any change occurs in the environment or the system itself.

Considering the hardware used for implementing the control, an experimental model-based approach has been chosen. The experimental models of the unknown transfer functions have been recorded in advance, before running the control algorithm. Plant responses have been measured for both the primary source and the active window prototype. They have been numerically fitted in order to derive the transfer functions needed to design the noise control algorithm and to perform the simulations. The mathematical model of the primary and secondary noise sources have been computed using the System Identification approach. Both the internal loudspeaker and the piezo stacks of the active window prototype have been driven by a broadband signal (chirp signal) in the 50-300 Hz frequency range. The same signal has been employed to simultaneously excite the piezo actuators. No. 4 microphones have been placed in the acoustic enclosure in order to measure the respective transfer functions.

The fitting of each experimental curve has been realized using the MATLAB[®] `invfreqs` function from the Signal Processing Toolbox, which tries to find the best coefficients b_i and a_i of the

numerator and the denominator of the polynomial function. An appropriate number of states for each transfer function has been identified through the cost function defined in Eq. 10 ranking the results of the MATLAB[®] `invfreqs` function, and coupled with a routine computing the results for a specified range of states, as follows:

$$f(n_p, n_k) = ||H_{exp} - H_{fit}||^2 \quad (10)$$

where $f(n_p, n_k)$ is the cost function to minimize; n_p and n_k are respectively the number of states of the numerator and the denominator of the transfer function; $||H_{fit} - H_{exp}||$ is the Euclidean norm of the difference between the experimental Frequency Response Function (FRF) and the fitted transfer function. The coefficients of these transfer functions have been derived from the measured data, and converted then in the canonical form of state-space systems for more reliable computations:

$$\begin{aligned} \dot{x} &= Ax + Bu \\ y &= Cx + Du \end{aligned} \quad (11)$$

The acoustic transfer functions between the input stimulus (voltage) and the acoustic pressures measured by the internal microphones have been experimentally characterized inside the acoustic enclosure, as shown in Fig. 4. Microphone no.4 has been chosen as control microphone in a SISO architecture. The anti-noise source has been obtained by controlling the vibrations induced by the piezo-stack actuators on the window prototype. An actuation mechanism forcing the window edges allowed exciting the flexural modes of the window pane causing an anti-noise field useful for ANC based on destructive interferences.

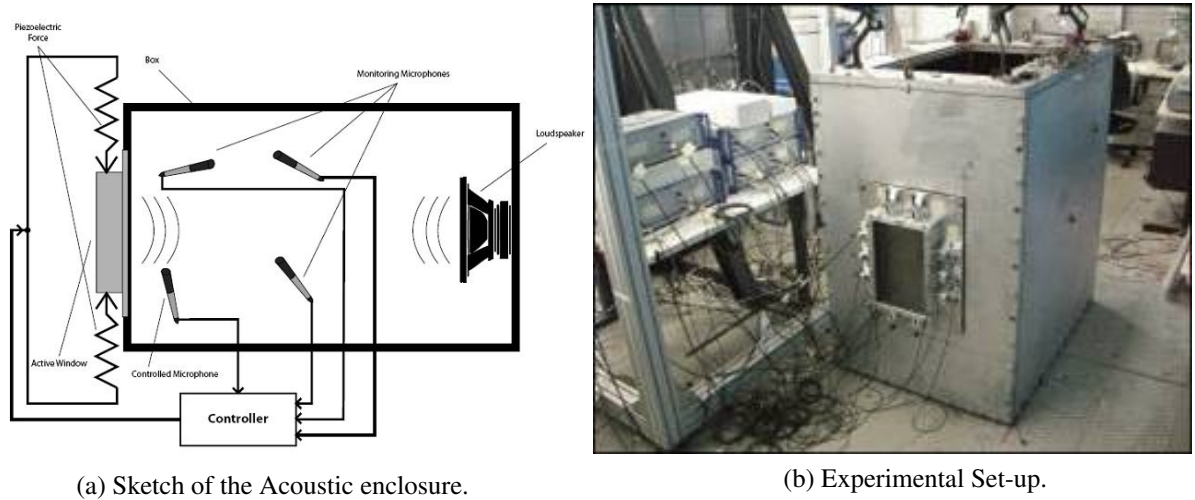


Figure 4. Active interior noise control of the box structure.

Fig. 5 shows the experimental and the fitted transfer functions obtained for the primary noise source and the *Active Window* prototype.

4. Numerical Simulation

4.1 Simulation model

A single-channel feedforward ANC system with one error microphone and one reference signal, driving no.8 piezo actuators, has been implemented in Simulink[®]. In addition, no.3 microphones have been simulated in order to evaluate the size of the controlled area. The effectiveness of the developed algorithm in controlling an acoustic field has been tested on a box structure. The transfer functions of the internal loudspeaker and the *Active Window*, the primary and the secondary noise sources respectively, have been modeled and then used to numerically calculate the disturbance and

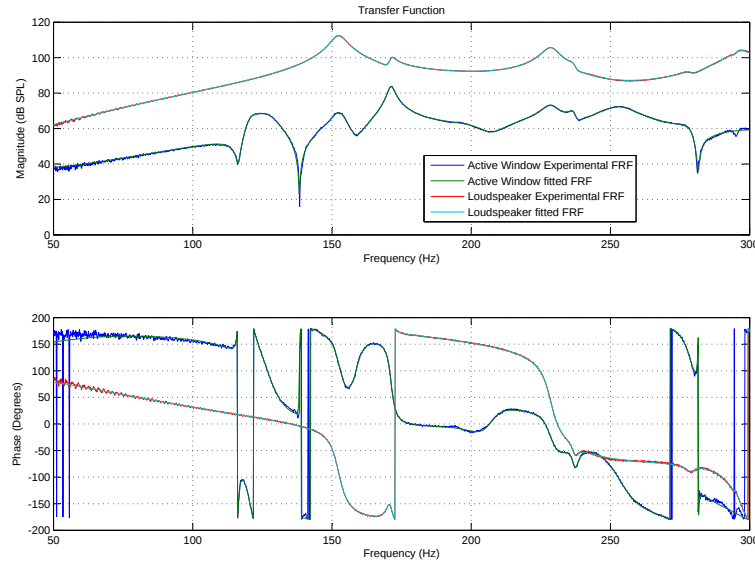


Figure 5. Experimental and fitted transfer functions of the primary source and the Active Window prototype.

the anti-noise signals. The same plant, fitted by the *Active Window* transfer function, has been used to simulate both the *plant* $G(z)$ and the *plant model* $\hat{G}(z)$ of Fig. 3, by assuming a perfect correspondence between the modeled and the experimental TFs. The *reference signal* $x(n)$, driving the internal loudspeaker, has been imposed either by a sine wave, at a frequency of 124 Hz correspondent to the typical fundamental frequency of the structure-borne noise emitted by a turboprop aircraft in horizontal flight, either by a chirp for slightly varying BPF. The *disturbance signal* $d(n)$ has been derived from the *reference signal* by the experimental plant of the loudspeaker. A FIR filter of 512 coefficients has been applied throughout the simulation.

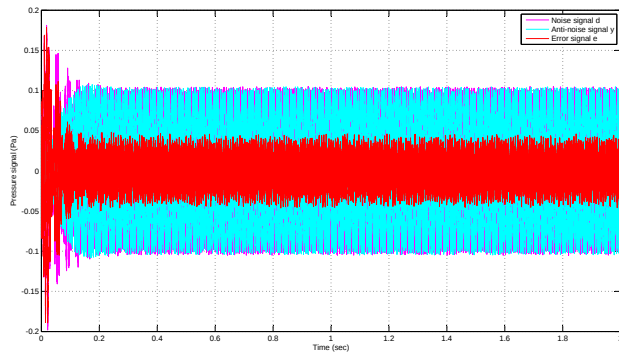
4.2 Tonal disturbance

4.2.1 Optimal filter coefficients

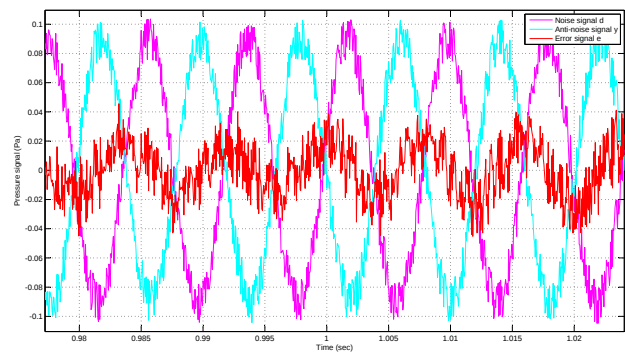
Firstly, the theoretical optimal FIR filter coefficients have been computed, as defined in Eq. (6). A random signal has been added to the tonal disturbance at 124 Hz to simulate a realistic signal-to-noise ratio measured in a real experiment. By imposing a convergence coefficient α of 0,003, the coefficients have been computed so that to generate an anti-phase *noise signal* y counteracting the *disturbance signal* d . Results are illustrated in Fig. 6. By comparing the root mean square (RMS) pressure of the *noise signal* d and of the *error signal* e , a reduction in the noise signal of 66,7 % has been obtained for the disturbance at 124 Hz. By converting the RMS pressure signals in decibels of sound pressure level, the noise abatement is estimated equal to 9,6 dB SPL.

4.2.2 Adaptive filter coefficients

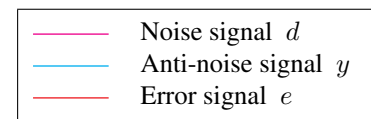
The adaptivity and the robustness of the self-tuning noise controller has been evaluated by computing the adaptive filter coefficients, needed to suppress the same noise disturbance, starting from different initial conditions. The convergence coefficient α has been imposed equal to 0,003. Fig. 7 shows the results obtained by imposing zero initial conditions to the filter coefficients. For each case, the *anti-noise signal* y has been observed still in anti-phase of the *noise signal* d , and capable to adapt its amplitude to completely cancel the noise wave. Once again, the controlled noise condition has been compared to the control-off condition, by calculating the RMS pressure of both the error and the disturbance signals. A reduction of the noise signal of 80,9 %, corresponding to an



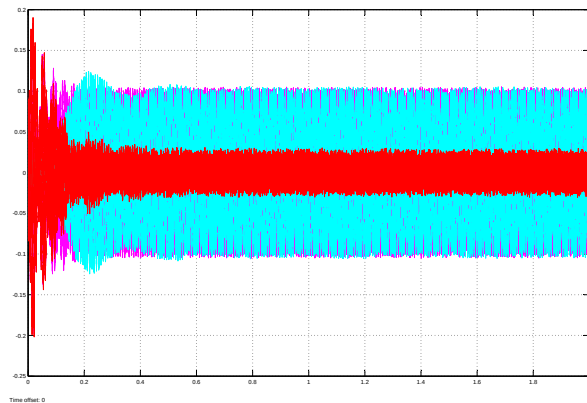
(a) Tonal disturbance at 124 Hz.



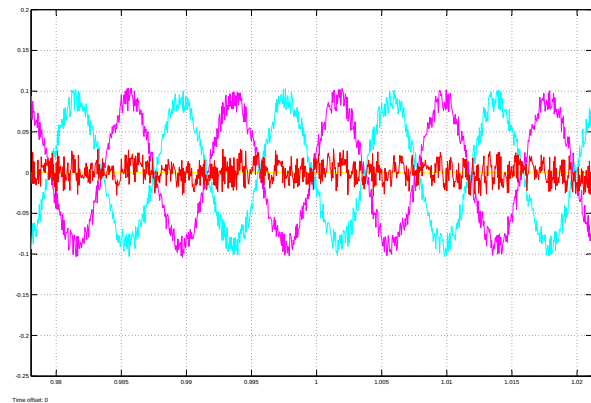
(b) Tonal disturbance at 124 Hz - zoom.

**Figure 6.** Results for tonal disturbances with optimal filter coefficients.

abatement of 14, 4 dB SPL, has been obtained for the disturbance at 124 Hz. Table 1 summarizes the results obtained for the optimal and the adaptive filter coefficients.



(a) Tonal disturbance at 124 Hz.



(b) Tonal disturbance at 124 Hz - zoom.

Figure 7. Results for tonal disturbances with adaptive filter coefficients.**Table 1.** Comparison between the results of the optimal filter coefficients and the adaptive filter coefficients.

Tonal disturbance at 124 Hz	Noise reduction (dB SPL)
Optimal filter coefficients	9, 6
Adaptive filter coefficients	14, 4

4.3 Narrow-band disturbance

Finally the control capabilities have been tested for a narrow-band disturbance modeled by a bidirectional chirp in a range of 120-128 Hz with a “one-way” sweep duration of 10 seconds. As it is shown in Fig. 8, by slightly varying both the amplitude and the frequency of the disturbance noise,

the controller has shown to converge rapidly to a self-tuned anti-noise signal y , causing a reduction of 72,9 % of the disturbance, corresponding to an abatement of 11,3 dB SPL.

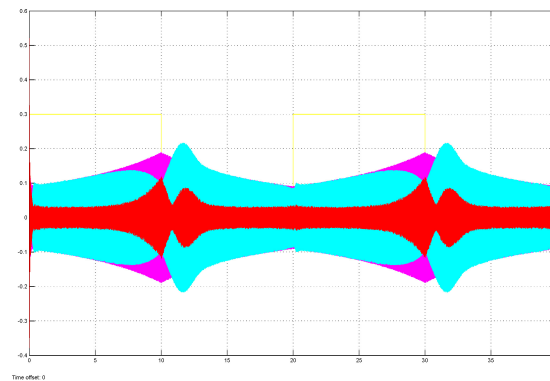


Figure 8. Disturbance from 120 to 128 Hz.

5. Conclusions

An active window prototype has been developed from a conventional, triple pane window configuration, by adding a number of piezoelectric stacks ensuring actuating and sensing capabilities. The actuator mechanism is based on no. 10 piezo stacks distributed on the window frame in order to generate in-plane and eccentric forces controlling the inner glass window pane. A reference enclosure, simulating an aircraft cabin environment and including the developed window panel, has been numerically analyzed. An accurate knowledge of the noise field inside the enclosure under control has been deduced from experimental measurements taken directly from the system. The plant models of the primary and secondary noise sources have been characterized using an input-output approach, rather than using a state variable model predicted by numerical simulations. The identification of the plant responses has been performed prior to the control system being switched on (off-line system identification) while the effects of plant uncertainty have not been considered on the robustness of the developed adaptive control law. The system demonstrated promising capabilities in suppression both tonal and broadband signals. The anti-noise field needed to counteract the imposed primary noise field has been simulated by computing the optimal and the adaptive filter coefficients. Such a suitable control strategy will be implemented in a DSP control board in order to evaluate the controller performance in real-time experiments.

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