

Research on Active Noise Reduction Systems at Building Openings

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Abstract: *This paper proposes an active noise reduction system design based on a digital signal processing (DSP) platform to address the impact of external noise on the indoor environment. The system is specially designed to be installed in the openings of a building and adopts the FxLMS (Filtered-x Least Mean Squares) algorithm to control the intrusion of noise effectively. By comprehensively considering the design of the hardware system and software systems' design, this paper verifies the feasibility of the proposed active noise reduction system and discusses its noise reduction performance in depth. Through experiments, this paper analyses the effect of secondary acoustic feedback on the system performance and the coherence between the reference signal and the error signal on the impact of the multi-channel active noise cancellation system. This paper provides a practical basis for further developing active noise reduction technology in practical applications.*

Keywords: Multi-channel FxLMS algorithm, Building openings, Active Noise Cancellation System Design.

1. Introduction

With the rapid improvement of the modern industrial level, people's lives are more and more convenient. Still, at the same time, the living and working environment has increased a variety of noise, noise spread through the building openings to the indoors, people for a long time in the noise environment will bring serious physical and mental harm. To create a good living and working environment and reduce the harm of noise to people, corresponding noise control measures must be taken. Noise propagated to the room through the opening is often low-frequency noise, traditional passive noise control is not ideal for low-frequency noise, German physicist Paul Leug based on the principle of acoustic phase cancellation proposed the concept of active noise control (Active Noise Control, ANC), and used to eliminate the noise [1]. Since then, the active noise control technology about openings has developed rapidly.

Long Shuai developed a semi-physical active noise cancellation system based on the LabVIEW platform and conducted experimental studies in a daily life environment [2]. A multi-channel simplified hybrid active noise control system was developed at Zhejiang University and applied to the left rear seat of a large sport utility vehicle to reduce in-vehicle road noise, which was shown through simulation results to produce significant suppression of narrow-band noise peaks between 75 and 80 Hz using either vibration or acoustic reference signals [3]. Professor Chen Ke'an led a team to research active noise control technology for aircraft cabins and completed ground tests, a very big breakthrough in the field of domestic aircraft applications [4]. Tang and Lin proposed a narrowband concomitant control algorithm and corresponding multi-channel ANC optimization structure, which reduces the amount of computation and improves the system convergence speed and stability [5]. Yuan Jun proposed an active noise control system based on adaptive IIR, using Xilinx FPGA as the core control module of the system for the hardware design of the system [6]. The measured

results show that the method has a noise reduction effect of up to 20 dBA in the low and medium frequency bands.

Oldham combined active and passive control to reduce the noise into the vents and experimentally proved that the attenuation of traffic noise in the octave bands of 63 Hz and 250 Hz can reach more than 7.5 to 8.5 dB [7]. Buttarazzi designed an active noise cancellation system for the noise generated by ambulance sirens, and the experimental results showed that the system could achieve almost complete noise cancellation in the specified frequency range [8]. Kuo analog/digital hybrid feedback active noise cancellation system was applied to headphones to improve the robustness and noise reduction of the system [9, 10].

Jphann used the surface impedance method to optimize the configuration of an ANC system applied on a window and the optimized ANC configuration was able to achieve an average attenuation of 9.2 dB [11]. Carme divided the vents into small zones, each including an ANC system with noise reduction of up to 30 dB [12]. Wongeun investigated the effect of ANC windows when the noise source is the human voice and showed that the energy of the average speech spectrum can be reduced [13].

In this paper, we design an active noise cancellation system based on the multi-channel FxLMS algorithm, which will be installed in the openings to reduce the impact of external noise on the room.

2. System Overall Design

2.1 System Hardware Selection

A complete active noise cancellation system mainly includes two parts: the upper active control software system and the hardware system, which is divided into four main parts: the signal acquisition module, the signal processing module, the loudspeaker array, and the main frame. When building the

system, the upper secondary sound source and the lower secondary sound source are 40 cm away from each other, the left secondary sound source and the right secondary sound source are 100 cm away from each other, the lower secondary sound source is 55 cm away from the ground, the error microphone is in the center of the secondary sound source array, the error microphone is arranged in a triangle, the lower error microphone and the error microphone in the center are 30 cm away from each other, and the plane where the error microphone is located is 25 cm away from the secondary sound source. 25 cm, the hardware system is shown in Figure 1.

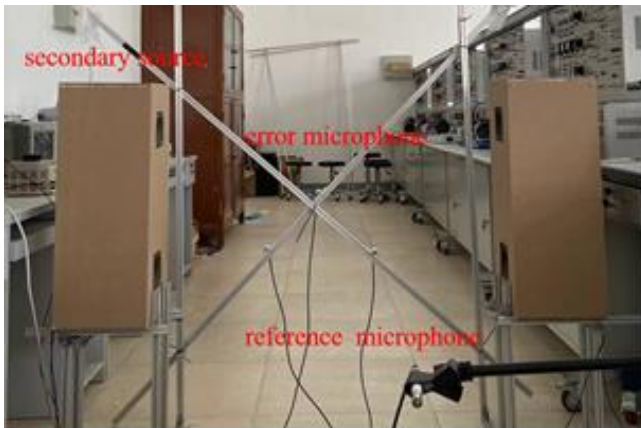


Figure 1: Hardware system diagram

The active noise cancellation system built in this paper is a $1 \times 4 \times 3$ system, one reference microphone, four secondary sound sources, and three error microphones, which is a large amount of computation, and the DSP has a powerful processing capability for digital signals, so the signal processing module in this paper selects the TMS320C6748 development board as the controller, with a main frequency of 456 MHz, and processes the signals acquired by the microphone to compute the inverse noise signal.

In this study, the MPA 201 condenser microphone was selected, which is known for its excellent immunity to interference and high-precision detection performance. Its sensitivity is as high as 50 mV/Pa and its frequency response ranges from 20 to 20000 Hz, which makes it ideal for acquiring reference and error signals.

In the conversion process of analog to digital signals, the system uses a 16-bit 8-channel parallel sampling successive approximation type A/D converter module AD7606. 8 channels of the module support up to 200 KHz sampling frequency, to ensure that the signal acquisition and conversion of high precision, the microphone will capture the analog acoustic signals accurately converted to digital signals.

For the D/A converter, the TL5724 module was chosen for this system, which is known for its high resolution of 12 bits and four-channel output. The module has a flexible power supply and a build-up time of up to 10 microseconds, which ensures the accuracy and responsiveness of the output signals.

Speakers, this system uses the WF100-9B30 model speakers.

Considering the acoustic characteristics of wood audio, such as good sound wave absorption and the ability to reduce resonance, especially in the low-frequency performance, as well as its density is greater than the characteristics of plastic, this system selected a 6 mm thick wood panel as the speaker enclosure, and added acoustic cotton in the interior, to optimize the sound quality transfer and sound absorption performance. The speakers are responsible for sending out the calculated inverse noise signal to effectively control the noise.

For the power amplifier, this system chooses the SA-36A pro power amplifier with the TDA7492PE module as the core. This Class D power amplifier is favored for its high efficiency, thermal overload protection, and short-circuit protection features, as well as having a smaller heat dissipation load, which can more accurately restore audio signals and output drive signals, providing the system with powerful audio output capability.

2.2 Software System Design

The hardware part of the system realizes the physical basis of data acquisition, processing, and storage through interconnection and collaboration, and the software design of the active noise reduction system is mainly to complete the control of each piece of hardware, transforming the demand into the instruction that the hardware system can understand and execute, and completing the real-time acquisition of data, signal processing, and signal output through the instruction to finally control the noise. Hardware systems and software systems are closely linked, and they depend on and interact with each other. The hardware system provides the basis for the operation of the software system, while the software system controls the work of the hardware system so that it can fulfill specific tasks. Only when the hardware system and the software system are coordinated, the system can operate normally and achieve various functions. The software system mainly includes: a system initialization procedure, secondary pathway offline modeling procedure, and multi-channel active noise control algorithm procedure.

In the design of the ANC software system, it is mainly for the programming of algorithmic procedures. As in this experiment using the TI chip as the core of the TMS320C6748 as the controller, the use of TI's program debugging software provided by Code Composer Studio5.5 (CCS5.5), the software is based on the C programming language, with Starterware function library, you can quickly achieve the design of applications based on the DSP from the concept to the writing, debugging, through the analysis of the program correction, testing.

The core of the software system is the multi-channel FxLMS algorithm [14], assuming that there are I reference microphones, J secondary sound sources, and K error microphones in the multi-channel feed-forward control system. $W(z)$ is the control filter coefficients, and the length is L . $H_p(z)$ is the transfer function of the primary path, and the transfer function of the secondary path, and the length is M . The block diagram of the algorithm is shown below.

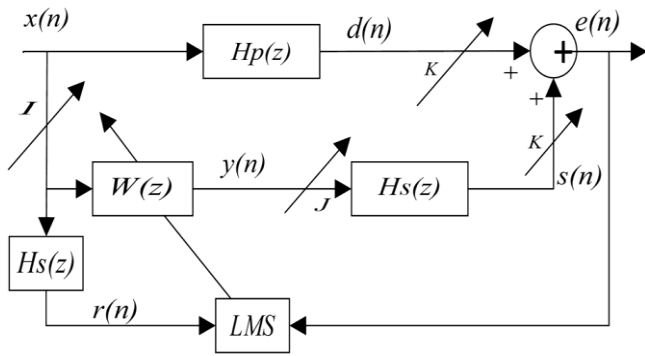


Figure 2: Block diagram of the multichannel FxLMS algorithm

The updated formula for the control filter coefficients:

$$W(n+1) = W(n) - 2\mu r(n)e(n) \quad (1)$$

Where μ is the step size and takes the range of values:

$$\mu_{max} \approx \frac{2}{p_x \times L} \quad (2)$$

The overall flowchart of the software system is shown in Figure 3.

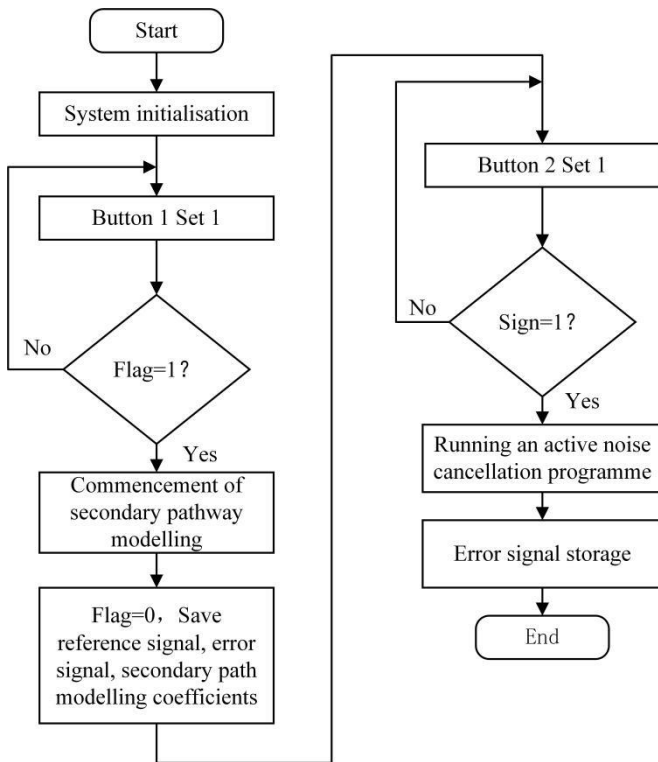


Figure 3: Overall flow chart of the software system

3. Active Noise Reduction Experiment and Result Analysis

3.1 Secondary Pathway Modeling

Due to the presence of secondary paths in the ANC system, accurate modeling of the secondary paths is required before active noise control to ensure that the filter is correctly adapted to the characteristics of the system. The accuracy of the secondary path modeling plays a key role in active noise control, which determines the accuracy and stability of the filter weight coefficients when iterating. Therefore, we need to adopt appropriate methods and techniques to optimize

secondary pathway modeling to improve the effectiveness of active noise control. Since the location of the secondary sound source and the error microphone are relatively fixed, this paper adopts an offline modeling approach using the LMS (Least Mean Square) algorithm [15] to model the secondary pathway. The modeling process is to drive the secondary source sound and error microphone acquisition. After the secondary source sounds for 4-10 seconds, the program is stopped and the error microphone acquisition signal is saved to the SD card and imported into MATLAB for secondary pathway modeling. The modeling error converges as shown in Figure 4. The error converges rapidly to a small range within 1 s. The modeling error is shown in Figure 5, which shows the convergence of the modeling error.

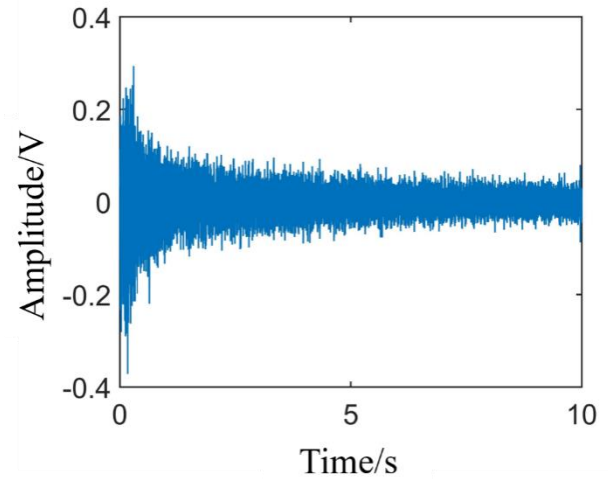


Figure 4: Error convergence diagram

3.2 Active Noise Reduction Experiment

After the secondary path modeling is completed the secondary path transfer function matrix is imported into the noise reduction algorithm for active noise control experiments. In this paper, the noise source is 3 m away from the noise cancellation system, the white noise signal with a frequency of 630 Hz or less is played, the sampling frequency is 6000 Hz, and the experimental flow is shown in Figure 6.

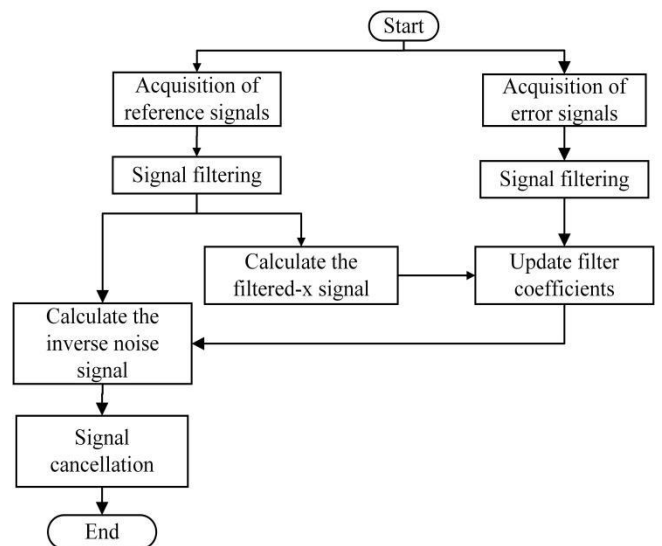


Figure 5: Experimental flow chart

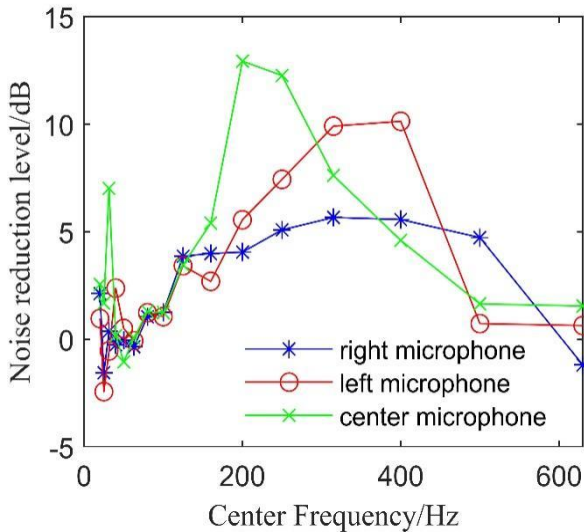


Figure 6: Plot of relative sound pressure levels of error microphones

At 0-200 Hz, the noise reduction of all three microphones is relatively low and does not vary much. As the frequency increases, the center error microphone reaches the highest noise reduction between 200 Hz and 300 Hz, with a maximum noise reduction of 13 dB, the left error microphone has a noise reduction effect of about 10 dB at the center frequencies of 315 Hz and 400 Hz, and the right error microphone is relatively ineffective, with a noise reduction of about 5 dB in the frequency range of 125 -500 Hz. At 400-600 Hz, the noise reduction of all three microphones decreases and tends to be close.

3.3 Effects of Secondary Acoustic Feedback

The back-noise emitted by the secondary sound source will be received by the reference microphone thus affecting the effect of noise control, so it is necessary to add an acoustic feedback simulation path to the system to eliminate the effect of secondary acoustic feedback, the schematic diagram is shown in Figure 7. In the experimental setup, the reference microphone is 2 m, 2.5 m and 2.6 m away from the error microphone, with a height of 0.6 m. The experimental setup is shown in Figure 8, and the experimental result is the one-third octave plot of the error microphone on the right side, as shown in Figure 9.

It can be seen from Figure 9 that when the reference microphone is 2 m away from the error microphone, the noise reduction performance is greatly reduced due to the influence of secondary acoustic feedback, which needs to be eliminated. When the reference microphone is more than 2.5 m away from the error microphone, a better control effect can be obtained regardless of the presence of acoustic feedback cancellation, and a noise reduction effect of 10 dB is achieved at the center frequency of 200 Hz. It can therefore be surmised that the system is not affected by secondary acoustic feedback

when the reference microphone is more than 2.5 m away from the error microphone.

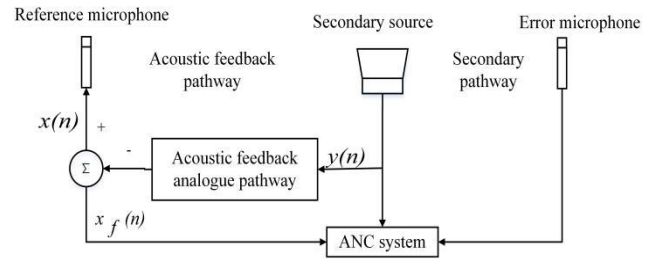


Figure 7: Secondary Acoustic Feedback Cancellation Schematic

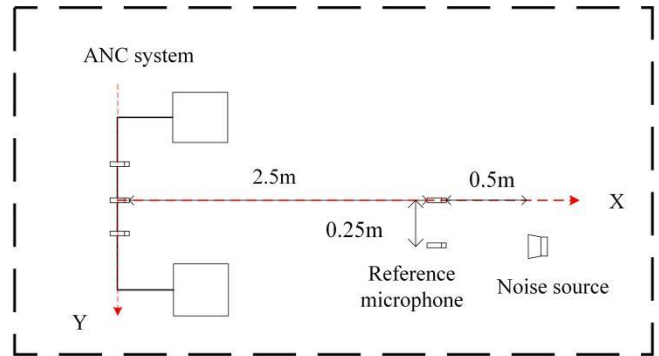


Figure 8: Experimental setup diagram

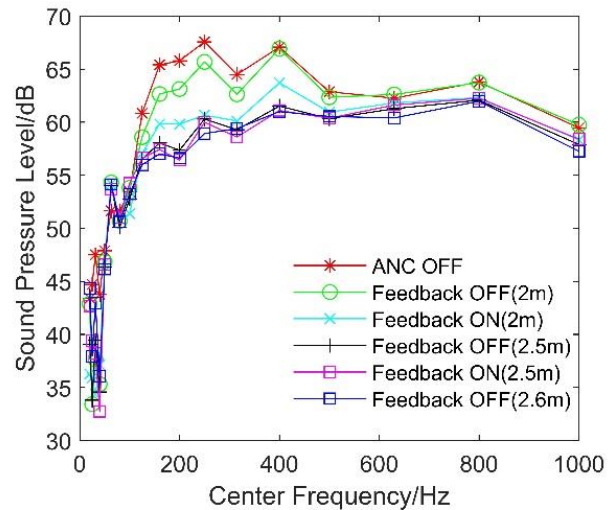


Figure 9: Error microphone relative sound pressure level graph

3.4 Effect of Coherence

In the active noise cancellation experiment, the level of coherence between the reference signal and the error signal will have a large impact on the control effect of the noise. Firstly, the reference microphone is set at a position 2.5 m away from the error microphone with a lateral offset of 0 m and 0.25 m. The noise source is 0.5 m away from the reference microphone, and the experimental setup is shown in Figure 10.

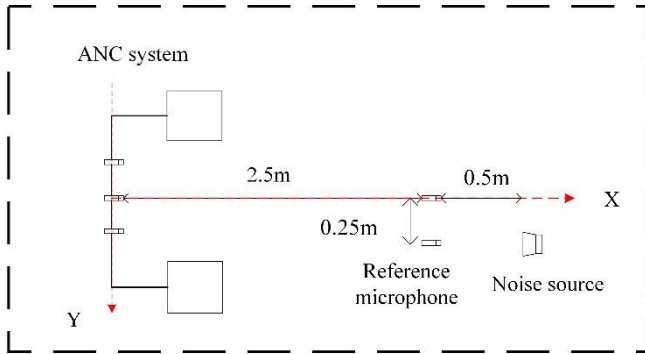
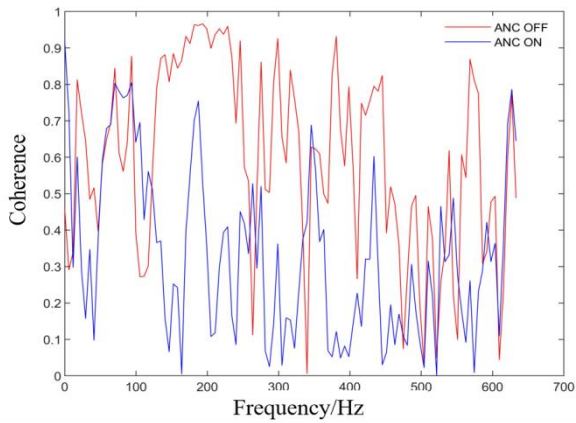
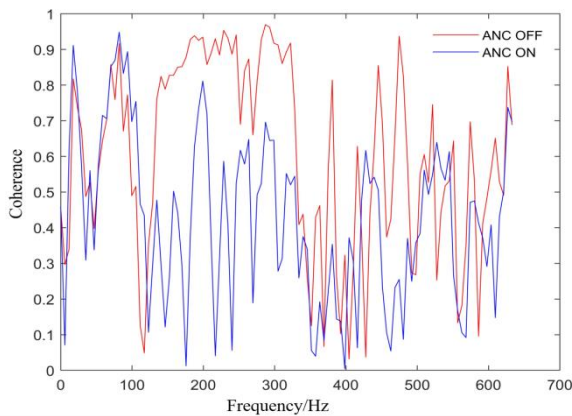


Figure 10: Experimental setup diagram



(a) $x=2.5$ m, $y=0.25$ m



(b) $x=2.5$ m, $y=0$ m

Figure 11: Coherence plot of reference and error signals

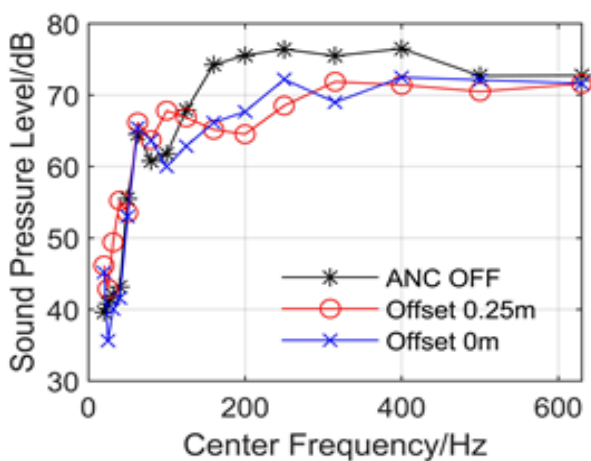


Figure 12: Error microphone relative sound pressure level graph

As shown in Figure 11 and Figure 12, the coherence between the reference signal and the error signal is greatest in the frequency range of 100 to 250 Hz, so the control of noise is greatest in this range. Above the frequency of 300 Hz, the coherence decreases and the amount of noise reduction decreases with it. Between 300 and 450 Hz, the coherence increases dramatically when the reference microphone is offset by 0.25 m, and the control effect increases. Therefore, it can be studied that the coherence of the reference and error signals before the active noise reduction system is switched on predicts the control effect of the noise to a certain extent.

4. Conclusion

In this paper, for the influence of external low-frequency noise on the room, a multi-channel active noise reduction system is introduced and constructed in terms of both hardware system and software system, which is used to realize the control of noise at the opening. The experimental results show that a maximum noise reduction of 13 dB can be achieved at 200 Hz. In addition, this paper investigates the influence of secondary acoustic feedback and coherence on the noise control effect. Without the influence of secondary acoustic feedback, the higher the coherence of the reference signal and the error signal, the better the noise control effect is, therefore, choosing the position where the reference signal achieves a higher coherence can achieve a more excellent noise reduction effect.

References

- [1] Paul L. Process of silencing sound oscillations. US Patent: 2043416,1936-09-06.
- [2] Long S, Tan P, Yang L B, et al. Design of An Active Noise Reduction System Based on Labview. *Scientific and Technological Innovation*, 2023, (15): 83-86.
- [3] Ma X, Chen Z. Research on Frequency-Selective Output Constraint Algorithm for Active Vibration Control. *Applied Sciences*, 2021, 11(1): 201-214.
- [4] Chen K A. Active noise control. Beijing: National Defense Industry Press, 2014.
- [5] Yuan J, Li J, Meng X S, et al. Research on active noise control system of compressor noise. *Electronic Measurement Technology*, 2022, 45(4): 33-38.
- [6] Tang R, Lin B He H, et al. Design and Research of Active Noise Control Structure Based on NALMS Algorithm. *Journal of Vibration Engineering & Technologies*, 2023, 11(5): 2181-2192.
- [7] Oldham D J, De Salis M H, Sharples S. Reducing the ingress of urban noise through natural ventilation openings. *Indoor Air*, 2004, 14(8): 118-126.
- [8] Buttarazzi M G, Borchi F, Mambelli A, et al. An Active Noise Control System for Reducing Siren Noise Inside the Ambulance. *International Conference of the Italian Association of Design Methods and Tools for Industrial Engineering*. Cham: Springer Nature Switzerland, 2023, 273-282.
- [9] Song Y, Gong Y, Kuo S M. A robust hybrid feedback active noise cancellation headset. *IEEE transactions on speech and audio processing*, 2005, 13(4): 607-617.
- [10] Kuo S M, Mitra S, Gan W. Active noise control system for headphone applications. *IEEE transactions on control systems technology*, 2006, 14(2): 331-335.

- [11] Tan J K A, Du L, Lau S K. Optimization of single-channel active noise control performance in a plenum window using the surface impedance approach. *The Journal of the Acoustical Society of America*, 2024, 155(2): 1570-1582.
- [12] Carne C, Schevin O, Romerowski C, et al. Active opening windows. *Proceedings of the ICSV*, 2016.
- [13] Wongeun O. Open-Window Active Noise Control for Environmental Noise. *International Conference on Research & Innovation in Environment, Civil and Architecture Engineering*, 2017.
- [14] Burgess J C. Active adaptive sound control in a duct: A computer simulation. *The Journal of the Acoustical Society of America*, 1981, 70(3): 715-726.
- [15] Widrow B, Glover J R, McCool J M, et al. Adaptive noise cancelling: Principles and applications. *Proceedings of the IEEE*, 1975, 63(12): 1692-1716.

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