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Performance improvement of a quiet zone using multichannel real-time active noise control system

다채널 실시간 능동 소음제어 시스템을 이용한 정숙공간 성능개선

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ABSTRACT: Generation of a quiet zone in noisy environment is undoubtedly of considerable realistic significance. This paper describes development and implementation of a multichannel real-time active noise control (ANC) system for 3 dimensional noisy environment to enhance the quiet zone performance in terms of size and noise cancellation gain. The proposed ANC system employs a multichannel delay-compensated filtered-X least mean square (FXLMS) algorithm; its real-time implementation is designed in TMS320C6713 digital signal processor (DSP) board. The system is evaluated for cancelling various tonal frequency noises in the range from 100 to 500 Hz, and the performance is then illustrated by measuring the quiet zone in terms of sound pressure level (SPL) attenuation. Experiment results show that a quiet zone of quiet with satisfactory size and maximum 24 dB noise attenuation is successfully generated.
Keywords: Multichannel real-time ANC, Delay-compensated FXLMS, DSP board, Quiet zone

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초 록: 3차원 잡음환경에서 정숙공간은 현실적으로 매우 중요한 문제이다. 본 논문은 3차원 잡음 환경에서 정숙공간의 성능을 크기와 소음제거 면에서 향상시키는 다채널 실시간 능동소음 제어시스템의 개발과 구현을 다루고 있다. 제안된 능동소음제어 시스템은 delay-compensated Filtered-X Least Mean Square (FXLMS) 알고리즘을 적용한다. 이와 같은 시스템의 실시간 적용을 위해서 TMS320C6713 DSP 프로세서 기반으로 설계되었다. 제안된 실시간 다채널 능동소음제어기의 성능평가는 100 ~ 500 Hz 범위의 다양한 잡음 환경에서 잡음제거를 수행하고, 정숙공간에서 음압레벨(Sound Pressure Level, SPL)측정하여 평가하였다. 실험결과는 정숙공간의 크기는 만족스러우며 최대 24 dB의 소음 감쇄가 성공적으로 생성된 것을 보여준다.
핵심용어: 실시간 다채널 능동 소음 제어, Delay-compensated FXLMS, DSP 프로세서, 정숙공간

I. Introduction

ANC (Active Noise Control) cancels the primary noise by generating and combining an anti-noise (with equal amplitude but opposite phase) at the location of the error microphone, and it efficiently attenuates low frequency noise with benefits in size and cost.[1,2] Over the past few decades a great progress has been made in ANC, of which main purpose is generating a quiet zone, where the noise is cancelled. Generally, the quiet zone is generated around the position of error microphone, however, the size of this quiet zone is usually limited especially in a single channel (which means one reference microphone, one cancelling

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speaker and one error microphone) ANC system.[3] Thus generating a quiet zone with sufficient size and good performance is of considerable significance.

This paper applies a multichannel ANC structure to create a quiet zone with satisfactory size. The multichannel FXLMS algorithm[2] has been widely used for ANC application.[4-6] To improve the performance of quiet zone, a multichannel delay-compensated FXLMS (Filtered-X Least Mean Square) algorithm[7,8] with faster convergence rate and better performance is selected for our multichannel ANC. The proposed multichannel ANC employes one reference microphone, two cancelling speakers and four error microphones, details on it are discussed in later section. The real-time system is designed and implemented in TMS320C6713 floating-point DSP board, other devices, including microphones and speakers, are connected to DSP board by the self-designed circuit.

The system is tested to cancel 100 to 500 Hz tonal frequency noises generated by a function generator and primary loudspeaker. Experiment results prove that our system generates a quiet zone with sufficient size and maximum 24 dB noise attenuation.

The organization of the paper is as follows: Section II presents the multichannel delay-compensated FXLMS algorithm. Section III describes the real-time system setup and implementation. Section IV illustrates the results the quiet zone. The conclusions are summarized in Section V.

II. Multichannel delay-compensated FXLMS algorithm

FXLMS algorithm has been widely used in practical ANC due to its robustness and simplicity, numerous algorithms are researched to improve its performance, but most of them achieve better performance with a significant computational complexity increase, which cannot be applied for real-time implementation. In the proposed system, a multichannel delay-compensated FXLMS algorithm is applied. It achieves faster convergence rate and better performance with only double computation load, and

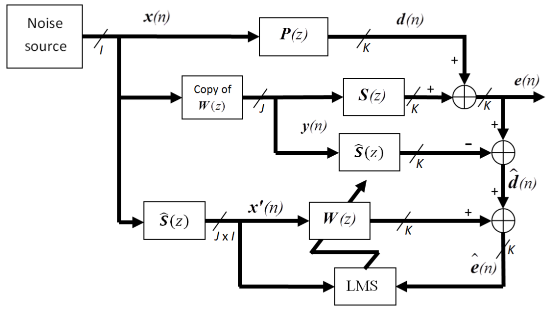

Fig. 1. Block diagram of multichannel delay-compensated FXLMS algorithm.

guarantees the possibility of real-time implementation. The multichannel delay- compensated FXLMS is illustrated in Fig. 1. It employs I reference sensors to form the reference signal vector $\boldsymbol{x}(n)$, generates J cancelling signals $\boldsymbol{y}(n)$ to drive the corresponding secondary sources, and distributes K error sensors over desired locations to measure the residual noise components $\boldsymbol{e}(n)$.

The wide arrows represent an array of signals that are symbolically expressed as vectors. The matrix $\boldsymbol{P}(z)$ represents $K \times I$ primary path transfer functions, $P_{ki}(z)$, from the reference signal $x_i(n)$ to each error sensor output $e_k(n)$. The matrix $\boldsymbol{S}(z)$ represents $K \times J$ secondary-path transfer functions, $s_{kj}(z)$, from J secondary sources to K error sensors. There are $J \times I$ possible feedforward channels, each demanding a separate adaptive filter, and these adaptive filters are represented by the matrix $\boldsymbol{W}(z)$. The delay-compensated FXLMS algorithm uses the estimated plant models $\hat{s}_{kj}(z)$ to subtract the contribution of the secondary path from the error signals, so that the estimated primary field signals $\hat{d}_k(n)$ are obtained, indeed, the adaptive filters $\boldsymbol{W}(z)$ then try to predict the estimated signals $\hat{d}_k(n)$ instead of the original $d_k(n)$ signals.

Using the following additional notations, we then obtain the multichannel delay-compensated FXLMS algorithm described by Eqs.(1) ~ (5), where L is the length of finite impulse response (FIR) adaptive filters in $\boldsymbol{W}(z)$, M is the length of FIR adaptive filters in $\hat{\boldsymbol{S}}(z)$, $y_j(n)$ is the value at time n of the jth cancelling noise signal, $w_{j,i,l}(n)$ is the value at time n of the lth coefficient in $w_{j,i}$, $\hat{s}_{k,j,m}$ is the value of the mth coefficient of $\hat{s}_{kj}(z)$, $x'_{i,j,k}(n)$ is the value at time n of the filtered reference signal, $\hat{d}_k(n)$ is the estimate of the value at time n of the primary sound field $d_k(n)$ at the kth error sensor, and $\hat{e}_k(n)$ is the alternative error signal.

$$y_j(n) = \sum_{i=1}^{I} w_{j,i}^T(n) x_i(n),$$
$$\text{where } x_i(n) = [x_i(n), x_i(n-1), \ldots, x_i(n-L+1)]^T. \tag{1}$$

$$x'_{i,j,k}(n) = s_{k,j}^T x_i(n),$$
$$\text{where } x_i(n) = [x_i(n), x_i(n-1), \ldots x_i(n-M+1)]^T. \tag{2}$$

$$\hat{d}_k(n) = e_k(n) - \sum_{j=1}^{J} s_{k,j}^T y_j(n),$$
$$\text{where } y_i(n) = [y_i(n), y_i(n-1), \ldots y_i(n-M+1)]^T. \tag{3}$$

$$\hat{e}_k(n) = \hat{d}_k(n) + \sum_{i=1}^{I}\sum_{j=1}^{J} w_{j,i}^T(n) x'_{i,j,k}(n),$$
$$\text{where } x'_{i,j,k}(n) = [x'_{i,j,k}(n), x'_{i,j,k}(n-1), \ldots, x'_{i,j,k}(n-L+1)]^T. \tag{4}$$

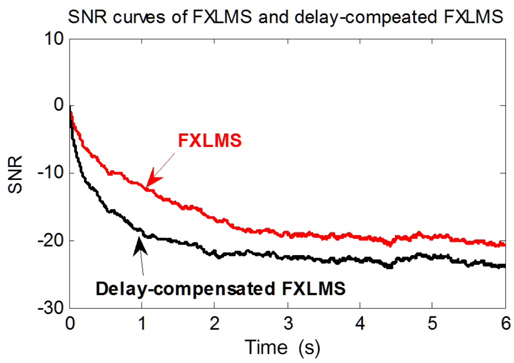

Fig. 2. SNR curves of FXLMS and delay-compensated FXLMS.

$$w_{j,i}(n+1) = w_{j,i}(n) - \mu \sum_{k=1}^{K} x'_{i,j,k}(n) \hat{e}_k(n). \tag{5}$$

Table 1 presents the number of multiples required for an iteration of both algorithms. Comparing with multichannel FXLMS algorithm, the delay-compensated FXLMS algorithm has almost twice computational load, but much faster convergence speed and better performance (see Fig. 2). 10 fan noises, 10 factory noises and 10 car noises are tested to the performances of two algorithms. The SNR is defined as the ratio between the power of residual noise and power of primary noise. The average SNR curves of 30 simulations are presented in Fig. 2.

III. Real-time ANC implementation

A single channel real-time ANC system[3] is presented in our previous research. It applies one reference microphone, one cancelling speaker, one error microphone and FXLMS algorithm. Its structure is shown in Fig. 3.

The experiment result in the next section proves a quiet zone is generated around the location of error microphone. But it must be noticed that the area of quiet zone is limited, and it is inefficient and meaningless for practical usage. To increase the size of quiet zone, multichannel ANC system with one reference microphone, two cancelling speakers and four error microphones is designed.

The proposed multichannel system with multichannel delay-compensated FXLMS algorithm is shown in Fig. 4.

The primary noise is generated by function generator and output from a powered studio monitor speaker-YAMAHA HS50, which has a frequency response from 100 to 20 kHz with a maximum power output of 70 W.

The reference microphone and error microphones are

Table 1. Computational complexity (number of multiplies) of multichannel algorithms.

	FXLMS	Delay-compensated FXLMS
Analytical computational load	$L(IJ+IJK)+M(IJK)+K$	$L(IJ+2IJK)+M(IJK+JK)+K$
Computational load of $I=1, J=1, K=1, L=50, M=40$	141	231
Computational load of $I=1, J=2, K=4, L=50, M=40$	774	1544

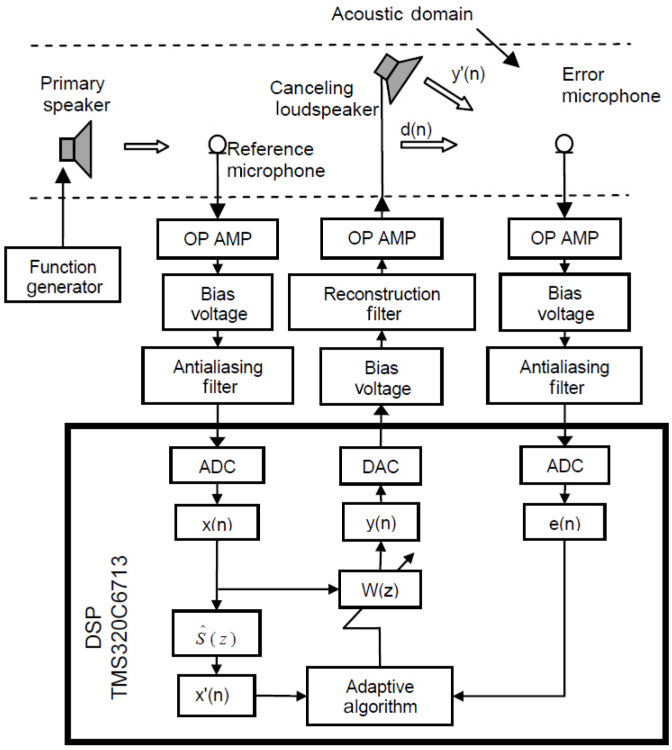

Fig. 3. The structure of single channel real-time ANC system.

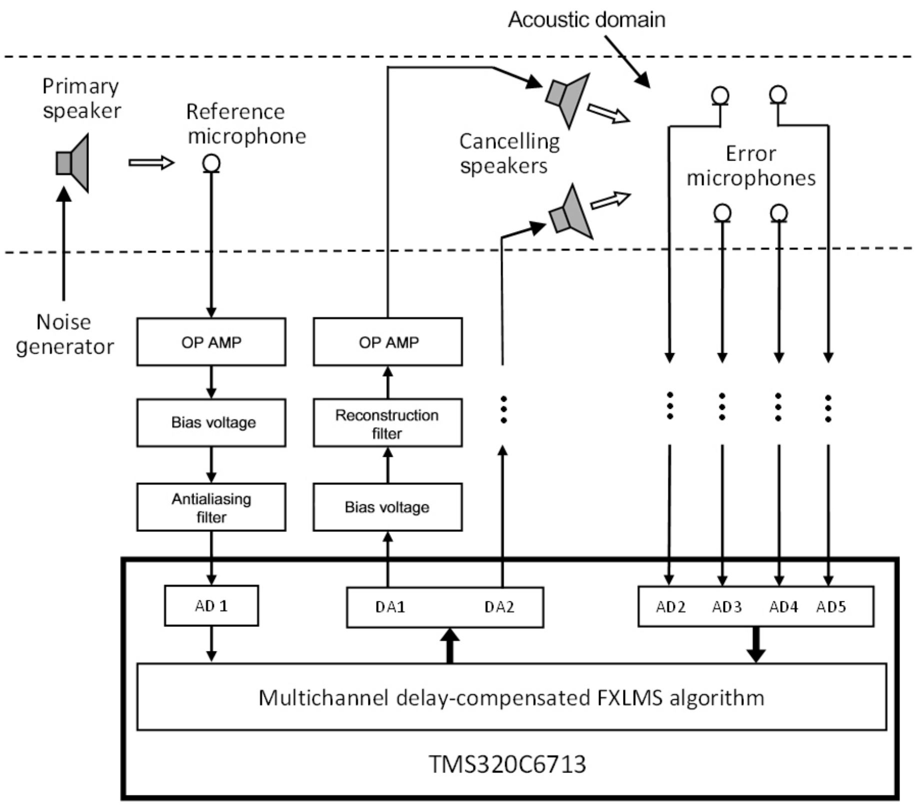

Fig. 4. The structure of proposed multichannel real-time ANC system.

unidirectional condenser microphones with 2.2 kΩ impedance, -46 ± 4 dB sensitivity and 70 to 20 kHz frequency response range. The cancelling loudspeakers have the same specifications as the primary loudspeaker.

The system is implemented in TMS320C6713 DSP board, the board employs 12-bit 8-channel A/D convertor AD7888 and 12-bit 4-channel D/A convertor AD7564, both analog input voltage of A/D convertor and analog output voltage of D/A convertor are 0 to 2.5 V. That is the reason we designed +1.25 V bias voltage before the ADC signal input and -1.25 V bias voltage after the DAC signal output. The anti-aliasing filter and reconstruction filter are 4-order low-pass Butterworth filters; their passband frequency is 750 Hz and stopband frequency is set to 1000 Hz with -9 dB stopband attenuation. For the operational amplifier, a balance input microphone amplifier circuit with adjustable gain is designed using high fidelity audio operational amplifier IC LME 49740.

To better understand our system, the equipment setup of multichannel ANC is illustrated in Fig. 5. To simulate the 3 dimensional environment, our system is implemented in a 15 × 10 × 3 m room, the heights of microphones and speakers are 1.5 m with their stands.

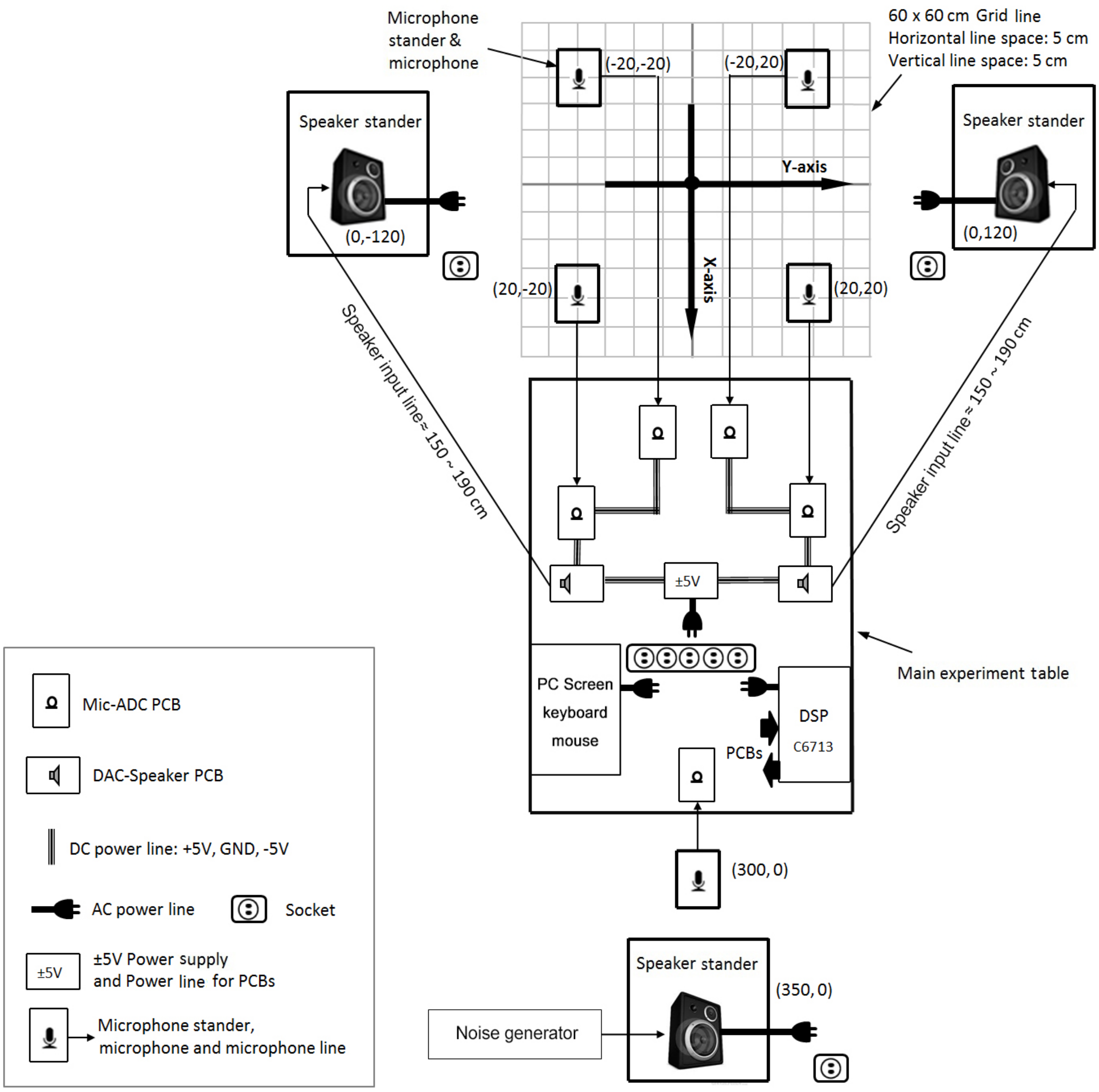

Fig. 5. System setup of multichannel real-time ANC.

IV. Experimental Results

The real-time ANC system is tested to cancel tonal noises, with frequency range from 100 to 500 Hz, generated by function generator. The system uses 8000 Hz sampling rate, the length of adaptive filter $S(z)$ and $W(z)$ are selected as 64 and 128 respectively. A SPL meter is used to measure the quiet zone in terms of noise attenuation (dB). The results are shown in Figs. 6 ~ 8, which present the quiet zone in a 60×60 cm area generated by single channel ANC, multichannel ANC with FXLMS and proposed multichannel ANC with delay- compensated FXLMS.

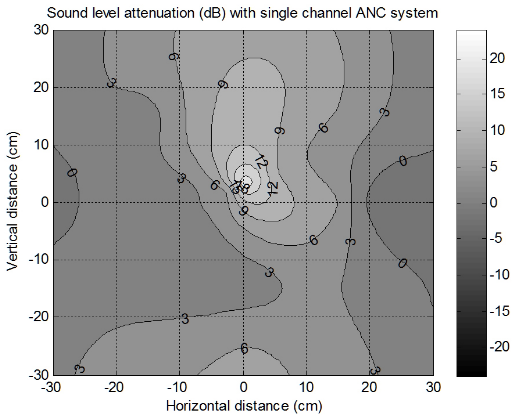

Fig. 6. The quiet zone generated by single channel ANC system.

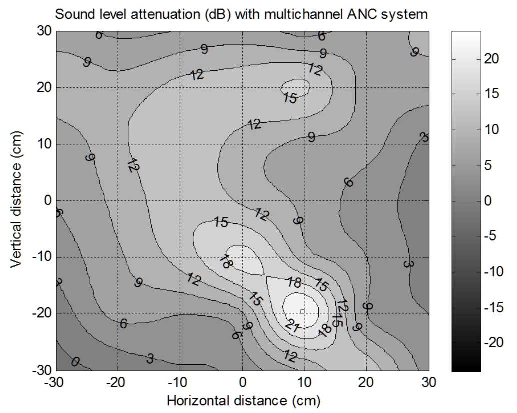

Fig. 7. The quiet zone generated by multichannel ANC system with FXLMS.

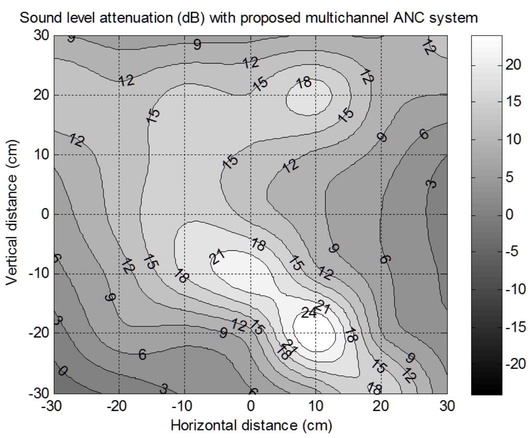

Fig. 8. The quiet zone generated by proposed multi-channel ANC system.

A coordinate system, as shown in Fig. 5, is to better present the location of equipment. In single channel ANC implementation. The primary noise speaker is placed in (200, 0) (in centimeter), the reference microphone is placed in (150, 0), the cancelling speaker is placed in (40, 40) and error microphone is set at (0, 0). From Fig. 6, a quiet zone with maximum 20 dB attenuation is generated around the location of error microphone. But the area of quiet zone is limited, the quiet zone with 12dB noise attenuation is smaller than a 10×10 cm area, that is inefficient and meaningless for practical usage.

For multichannel ANC systems, the primary noise speaker is placed in (350, 0), the reference microphone is placed in (300, 0), the cancelling speakers are symmetrically placed in (0, -120) and (0, 120), the error microphone are distributed in (20, 20), (-20, 20), (-20, -20) and (20, -20). Comparing the results in Figs. 6 and 7, a multichannel structure successfully enlarges the quiet zone to a satisfactory size. From Figs. 7 and 8, the noise attenuation is evidently improved with multichannel delay-compensated FXLMS algorithm. From Fig. 8, the quiet zone with 12 dB noise attenuation is up to a 40×40 cm area. This proves the quiet zone performance of proposed ANC system is successfully enhanced in both in terms of zone size and noise attenuation.

V. Conclusions

Generating a quiet zone, in which the primary noise is cancelled, is the main objective of ANC. For practical application, this quiet zone must be large enough to satisfy the needs. To enhance the noise cancellation in quiet zone, including enlargement of zone size and improvement of noise cancellation gain, the proposed multichannel real-time ANC system is designed. The proposed system employs multichannel structure, which includes one reference microphone, two cancelling speakers and four error microphones, to enlarge the quiet zone; a multichannel delay-compensated FXLMS is then applied to improve the noise attenuation.

The quiet zones generated by ANC system are measured by SPL meter in terms of noise attenuation in dB. The results measured in a 60×60 cm area prove that, comparing with the single channel ANC system and multichannel ANC with FXLMS algorithm, the proposed ANC system successfully enhances the quiet zone performance in terms of zone size and noise attenuation.

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Profile

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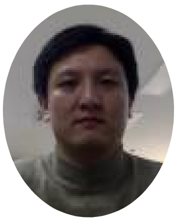

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