Simplified active noise control system with decorrelation reference signals

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ABSTRACT

Remote microphone technology (RMT) aims to address the issue of the impracticality of installing physical microphones in specific locations for reducing noise. To guarantee the noise reduction performance, the number of observation microphones in a typical RMT system is generally not less than the number of noise sources, which will lead to a high device cost and a heavy computational burden. Our goal is to design a simplified RMT system that cuts down on the quantity of reference and observation microphones required in settings with multiple noise sources. The first step involves examining the relationship of the sound field to find the best quantity of microphones. Next, the observation filter is designed using the best number of observation microphones. Furthermore, the blind separation method reconstructs reference signals while the observation filter estimates virtual signals for the RMT system. Experimental evidence confirms that the proposed RMT system can lower the number of microphones required while still achieving comparable noise reduction performance to traditional systems in scenarios with multiple noise sources.

Keywords: Active noise control, remote microphone technique, blind separation, independent component analysis

1. INTRODUCTION

Active noise control (ANC) involves using secondary speakers to produce anti-noise and reduce unwanted sounds effectively[1]. For the past few years, the application of ANC in cars[2], in flight[3], in high-speed trains[4], and through an open window[5], has been greatly developed.

In general, it is inconvenient, especially for a driver or a passenger, to place the physical error microphones in the desired ear position for noise reduction. To overcome this problem, a number of complicated virtual sensing algorithms have been developed[6,7], such as virtual microphone arrangement (VMA)[8], virtual microphone control (VMC)[9], remote microphone technique (RMT)[10], and so on. Buck and Das et al. conducted the research to study the performance of the above virtual sensing method[11,12]. The RMT system was proved to be effective in the environment of fixed primary noise. In the subsequent research on virtual sensing techniques, RMT gradually became one of the research focuses of scholars.

In order to study the quiet zone which directly affects the effectiveness of noise reduction, Wrona et al. focused on finding the best arrangement for the transducers (microphones and loudspeakers) in terms of spatial layout[13] as it can define the layout of the quiet zones that are obtained. Cheer and Lam et al. studied the effect of the error microphone position on the quiet zone[14,15]. To improve the practical noise reduction capabilities, previous researches focus[16-19] more attention on discussing the transducers position, size and even the microphone arrays. Larger control loudspeakers can achieve wider noise reduction frequency bands. Using more microphones or other sensors can achieve better quiet zones. However, excessive sensors will increase the channels of RMT, bringing computational burden to the control system. A higher power transducer will increase the cost of the system. For practical application, it is important to avoid unnecessary expenses and complications in RMT systems. The main goal of the suggested RMT system is to maintain noise reduction effectiveness by cutting down on sensor usage, ultimately leading to a decrease in both cost and complexity of the system.

2. THE MULTICHANNEL RMT METHOD

The RMT system utilizes the signals captured by observation microphones to predict signals at a virtual location through the use of an observation filter connecting the observation microphones and virtual microphones, as depicted in Figure 1(a). The virtual microphones are usually placed at the location to be noise-reduced such as the ear of the head, while

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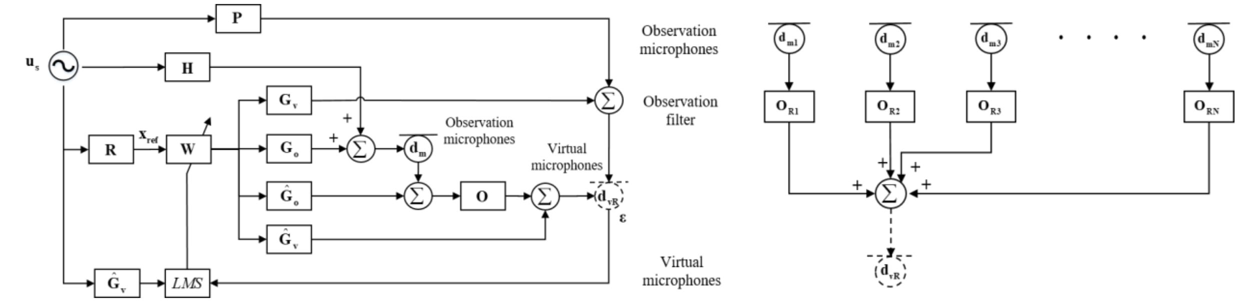
observation microphones are placed near the location to be noise-reduced. For simplicity, Figure 1(b) shows the schematic diagram of observation filter estimation.



(a) (b)

Figure 1. The schematic diagram of the RMT method.

The primary noise $\mathbf{u}_s$ picked by N observation microphones is the concatenation of multiple observation microphone signals and can be defined as $\mathbf{d_m} = [\mathbf{d_{m1}}, \mathbf{d_{m2}}, \cdots, \mathbf{d_{mN}}]$. The primary noise detected by right virtual microphone is defined as $\mathbf{d_{vR}}$. $\hat{\mathbf{d}}_{\mathbf{vR}}$ is the estimate of $\mathbf{d_{vR}}$, which can be expressed as:

$$\hat{\mathbf{d}}_{\mathbf{vR}} = \hat{\mathbf{O}}_{\mathbf{R}} \mathbf{d_m} \tag{1}$$

where $\hat{\mathbf{O}}_{\mathbf{R}}$ is the observation filter $\hat{\mathbf{O}}_{\mathbf{R}} = [\mathbf{O_{R1}}, \mathbf{O_{R2}}, \cdots, \mathbf{O_{RN}}]$ between the observation signals and the virtual signal. The coefficients of the observation filter are calculated by minimizing the mean squared error between the estimated virtual signal vector $\hat{\mathbf{d}}_{\mathbf{vR}}$ and the true virtual signal $\mathbf{d_{vR}}$, as shown in the cost function:

$$\mathrm{J}_{\hat{\mathbf{O}}} = \mathrm{Tr}\left\{ E\left[\left(\mathbf{d_{vR}} - \hat{\mathbf{O}}_{\mathbf{R}}\mathbf{d_m}\right)\left(\mathbf{d_{vR}} - \hat{\mathbf{O}}_{\mathbf{R}}\mathbf{d_m}\right)^{\mathbf{T}} + \beta \hat{\mathbf{O}}_{\mathbf{R}} \hat{\mathbf{O}}_{\mathbf{R}}^{\mathbf{T}} \right] \right\} \tag{2}$$

where $\mathrm{Tr}\{\ \}$ denotes the trace of the matrix, $E[\ ]$ denotes the expectation operation, $(\ )^{\mathbf{T}}$ denotes the transpose. A small positive real number β is typically chosen as the regularization parameter. By performing trace derivation calculations on the matrix, the optimal observation filter can be obtained as:

$$\hat{\mathbf{O}}_{\mathbf{opt}} = E\left[\mathbf{d_{vR}}\mathbf{d_m^T}\right] E\left[\mathbf{d_m}\mathbf{d_m^T} + \beta \mathbf{I}\right]^{-1} \tag{3}$$

where $E\left[\mathbf{d_{vR}}\mathbf{d_m^T}\right]$ is the cross spectral density matrix of the $\mathbf{d_{vR}}$ and $\mathbf{d_m}$, and $E\left[\mathbf{d_m}\mathbf{d_m^T}\right]$ is the power spectral density matrix of $\mathbf{d_m}$.

3. PROPOSED METHOD

When the reference signals show correlation, this correlation causes a slow convergence rate and a high susceptibility to measurement errors. In addition, if two reference signals are linearly dependent, the RMT system is underdetermined, meaning there is no unique solution. An adaptive method is introduced to address the issue of ill-conditioning by decorrelating the reference signals. To enhance the efficiency of a multiple reference control system, it is important to manipulate the reference signals to ensure they are not related to each other. FastICA, as a type of blind source separation algorithm, can separate independent signals from mixed signals.

Based on the above theory, to further reduce the use of the microphone in the RMT system, the proposed method operates in three stages. (1) Pre-training Stage: obtaining the optimal number of observation microphone and reference microphone. (2) Identification Stage: modeling the observation filter and secondary path. (3) Control stage: using the estimated virtual error signals and reconstructed reference signals for the RMT system. The flow diagram of the system is illustrated in Figure 2.

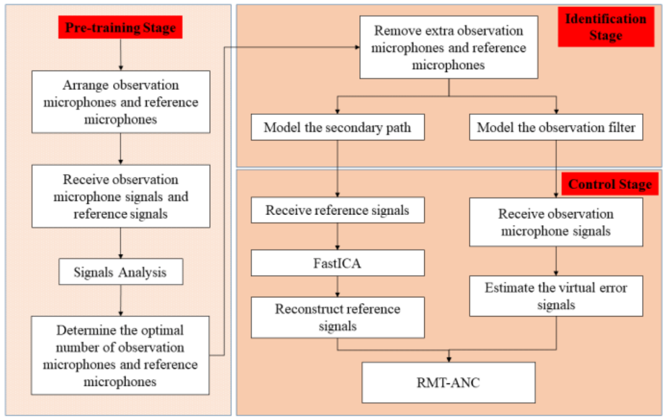


Figure 2. Flow diagram of the proposed method.

3.1 Pre-training stage

Suppose that the number of primary noise sources in the sound field is N, in order to balance estimation performance and computational burden, N observation microphones are placed near the virtual error microphone. To determine the optimal observation microphone number, signal analysis is performed to the observation microphone signals $\mathbf{d_m}$, the covariance matrix $\mathbf{C_{d_m d_m}}$ needs to be calculated firstly as follows:

$$\mathbf{C_{d_m d_m}} = \frac{1}{N}\mathbf{d_m d_m^T} \tag{4}$$

Since the covariance matrix $\mathbf{C_{d_m d_m}}$ is a symmetric matrix, there is a eigenvalue decomposition in the covariance matrix. It can be expressed as:

$$\mathbf{U \Lambda U^T} = \mathbf{C_{d_m d_m}} \tag{5}$$

where $\mathbf{U}$ is the eigenvector matrix $\mathbf{\Lambda} = \mathrm{diag}\left(\lambda_1, \lambda_2, \cdots, \lambda_N\right)$ is a diagonal matrix, and its diagonal elements are the eigenvalues in descending order. If λ_N is much smaller than other eigenvalues, it means that one of the observation microphone signals can be linearly represented by other signals. In this multiple noise source environment, only $(N-1)$ observation microphones are needed to accurately estimate the virtual signal. Therefore, in practical systems, reducing the number of observation microphones to $(N-1)$ can still accurately estimate virtual signals in the sound field environment. Under normal usage[20], to yield the best noise reduction performance, reference microphones are placed next to each primary sound source to collect reference signals for the RMT system. We have taken this view as well. There is a reference microphone next to each primary noise, with a total of N reference microphones, used to collect reference signals $\mathbf{x_{ref}} = [\boldsymbol{x}_{ref\,1}(n), \boldsymbol{x}_{ref\,2}(n), \cdots, \boldsymbol{x}_{ref\,\mathrm{N}}(n)]$. Similar to signal analysis on the observation microphone signals, perform the same operation on the reference signals as well and then we can obtain the optimal reference microphone number.

3.2 Identification stage

In the identification stage, if the optimal number of reference microphone and observation microphone are P and Q respectively, extra microphones are removed from the RMT system according to the optimal number of observation microphone and reference microphone. Then, observation microphone signals and virtual error signals are used to model the observation filter by the cost function in Section 2. In addition, white noise was used to drive the secondary loudspeaker to obtain the secondary paths.

3.3 Control stage

When the N primary loudspeakers are playing noise, the signal received by the P reference microphone at nth sample time form the new L length reference signals matrix $\mathbf{x_{ref}} = [\boldsymbol{x}_{ref1}(n), \boldsymbol{x}_{ref2}(n), \cdots, \boldsymbol{x}_{ref\,\mathrm{P}}(n)]$. The reference signals are centralized by:

$$\mathbf{x_{refc}} = \mathbf{x_{ref}} - E\left[\mathbf{x_{ref}}\right] \tag{6}$$

where $E\left[\mathbf{x_{ref}}\right]$ is the mean of each reference signal vector. the covariance matrix of the $\mathbf{x_{ref}}$ is calculated as follows

$$\mathbf{C_{refc}} = \frac{1}{P}\mathbf{x_{refc}}\mathbf{x_{refc}^{T}} \tag{7}$$

The feature decomposition of covariance matrix $\mathbf{C_{ref}}$ is carried out by

$$\mathbf{C_{refc}} = \mathbf{EDE^{T}} \tag{8}$$

where $\mathbf{E}$ is the eigenvector matrix and $\mathbf{D}$ denotes the eigenvalue matrix. The whitening transformation is as follows

$$\mathbf{x_{refW}} = \mathbf{D}^{-1/2}\mathbf{E^{T}}\mathbf{x_{refc}} \tag{9}$$

The FastICA algorithm usually uses a nonlinear function g to approximate the maximum negentropy $\boldsymbol{w}^{\mathbf{T}}\mathbf{x_{refW}}$, where $\boldsymbol{w}$ is an initialized random vector. And its iterative update rules are as follows:

$$\boldsymbol{w}^{+} = E\left[\mathbf{x_{refW}}g\left(\mathbf{w^{T}}\mathbf{x_{refW}}\right)\right] - E\left[g^{'}\left(\mathbf{w^{T}}\mathbf{x_{refW}}\right)\right]\boldsymbol{w} \tag{10}$$

Above rules are repeated until convergence, where the nonlinear function g can be expressed as $g\left(u\right) = \tanh\left(au\right)$, and a is typically a fixed value, commonly assumed to be $a = 1$. It's important to note that convergence implies that the previous and current values of $\boldsymbol{w}$ are aligned, with their dot product nearly equal to 1.

Once all the weight vectors $\mathbf{W} = \left[\boldsymbol{w}_1, \boldsymbol{w}_2, \cdots, \boldsymbol{w}_n\right]$ are found, the reference signals can be reconstructed by the following formula.

$$\mathbf{x_{refR}} = \mathbf{W^{T}}\mathbf{x_{ref}} \tag{11}$$

Subsequently, the signals received by the Q observation microphones are used to estimate the virtual error signals with the observation filter. The reconstructed reference signals and the estimated virtual error signals mentioned above are used in the RMT system to realize the effect of the noise reduction while reducing the use of microphones.

4. EXPERIMENTS

Experiments were conducted in a multimedia space to confirm the effectiveness of the new technique, as depicted in Figure 3. The primary noise is generated by six loudspeakers placed in a horizontal circle. The dummy head is in the center of the primary loudspeaker. The dummy head had virtual microphones placed at its ears and six observation microphones evenly distributed around it. Reference microphones are placed at each primary loudspeakers. The count of observation microphones and reference microphone corresponds to the count of primary noise emitters. The diagram of the experimental setup is shown in Figure 3.

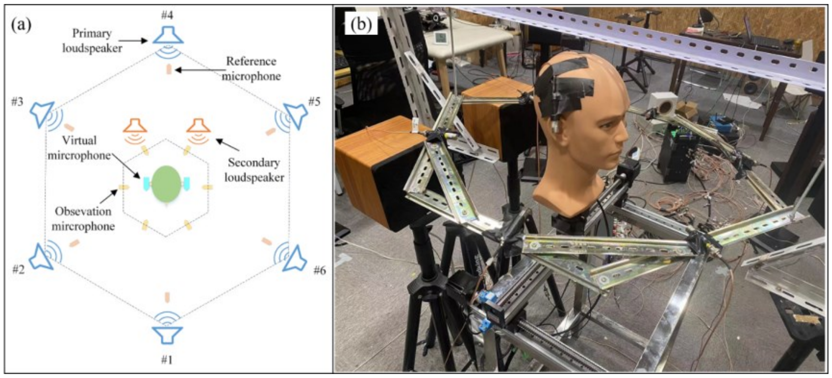


Figure 3. Experimental setup.

To compare the noise reduction performance, the noise attenuation capability of the proposed simplified RMT system and traditional RMT system were presented as shown in Table 1 and Figure 4. In simplified RMT system, the noise reduction levels below 300 Hz are slightly higher compared to traditional RMT system with all reference signals. The new simplified RMT system can match the noise control performance of the traditional RMT system across the entire 0-1000 Hz frequency range. It means simplifying the RMT system based on the correlation of the primary sound field in a multi primary noise environment does not deteriorate the control effect.

Table 1. Noise reduction in the 0-1000Hz frequency band (Left ear/Right ear).

Method	**Traditional RMT**	**Proposed RMT**
NR (dBA)	16.0/12.0	16.4/13.0

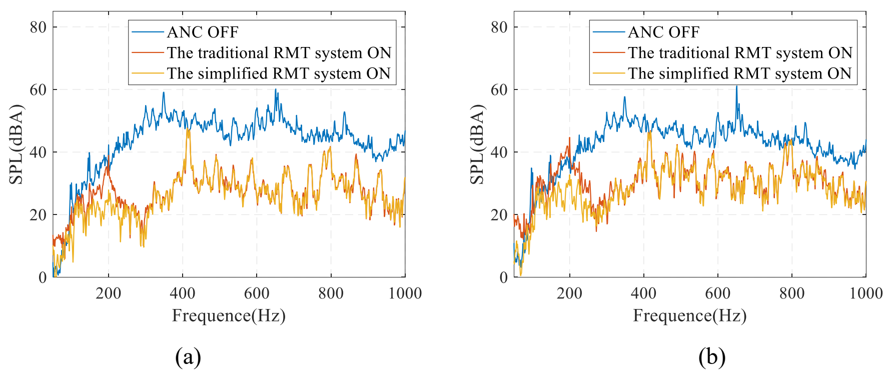


Figure 4. The noise reduction performance. (a): Left ear; (b): Right ear.

5. CONCLUSIONS

In this paper, a multichannel simplified active noise control system for remote microphone technique combined with FastICA technique in multiple noise sources environment was proposed. In the pre-training stage, the optimal number of microphone is obtained using the correlation analysis performed on primary sound field. During the identification stage, the extra microphones are removed from RMT system according to the pre-training result and then the secondary path and observation filter are modeled. Finally, in the control stage, estimated virtual signals and reconstructed reference signals are used for simplified RMT system to attenuate the noise. The experiments show that the proposed simplified The RMT system helps in determining the best combination of observation and reference microphones. Compared to the traditional RMT system, our simplified RMT system has the similar noise reduction performance which can offer a resolution for safeguarding the microphone in practical settings.

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