

LOCAL ACTIVE SOUND CONTROL USING THE REMOTE MICROPHONE TECHNIQUE AND HEAD-TRACKING FOR TONAL AND BROADBAND NOISE SOURCES

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Local active sound control can be combined with the remote microphone technique and head-tracking for the case when direct measurement of the error signals at a listener's ears is not possible and the position of the listener's head is continuously changing. This paper presents the performance of the combined active control system when it is applied to reduce either tonal or broadband random noise. First of all, effects of the combined system were experimentally investigated when periodic noise from a single primary source was adaptively controlled in real time. By directly applying observation filters and acoustical responses in the frequency domain, it was shown that active control of tonal noise can be effectively improved by the remote microphone technique and head-tracking. In addition, the ability of the combined active control system to control stationary random broadband noise was investigated. However, in this case, the causality of the observation filter must be considered due to the unpredictability of random noise. Therefore, the causally constrained remote microphone technique and the corresponding broadband active noise control system were studied.

Keywords: remote microphone technique, head-tracking

1. Introduction

The active headrest system, in which secondary loudspeakers and error microphones are installed around a headrest of a vehicle seat, has been suggested as a practical local active control system to reduce the interior noise around ears of the driver and passengers [1] [2]. However, when direct measurement of the error signals at a listener's ears is not possible, this can be a constraint on the application of the active headrest system. Especially, because the small zone of quiet is generated at higher frequencies, the separation between the ears and the error microphones can degrade the attenuation performance around the ears. To overcome this limitation, a shift of the zone of quiet has been suggested by a number of virtual sensing algorithms [3] [4]. Among these algorithms, the remote microphone technique has been investigated for the nearfield estimation of signals at the virtual error microphones from measured signals at remotely installed monitoring microphones through simulations and off-line analysis from measured data [5]. The present paper investigates the accuracy of the remote microphone techniques through both real-time experiments for tonal sounds and off-line analysis for stationary random broadband sounds.

As another practical issue, head movements of the listener can produce changes in sound fields and acoustic responses. To prevent these changes from degrading the convergence stability and attenuation performance of the adaptive active headrest system, the application of head-tracking has been suggested [5] [6]. In previous research, off-line test results in the frequency domain have shown

that the head-tracking can improve the stability and attenuation performance of the active headrest system when the listener's head is moving [5]. In this paper, a complete active headrest system combined with both the remote microphone technique and a commercial head-tracking device is used to adaptively control tonal sounds.

In Section 2, a theoretical analysis for the integrated active headrest system is presented in frequency domain for controlling tonal disturbances but an analysis in time domain is also included for a causal controller for broadband random disturbances. It is shown in Section 3 that real-time experiments about the nearfield estimation and integrated active headrest system are implemented to reduce tonal noise. Section 3 also describes results of off-line analysis with measured acoustic responses when the causal filters for the remote microphone technique and active control are applied to reduce broadband random sounds. Finally, in Section 4, conclusions are drawn.

2. The nearfield estimation and active control of the integrated active control system

2.1 Frequency domain analysis

When the remote microphone technique is combined with a feedforward active control system, the block diagram of the combined algorithm is shown in Fig. 1. This combined active control system is initially investigated in frequency domain in this section. Results in frequency domain allow a straightforward assessment of both the nearfield estimation and attenuation performance when it is aimed to reduce tonal noise regardless of the causality constraint.

In Fig. 1, the primary sources, which have a vector of source strengths $\mathbf{v}^T = [v_1, v_2 \dots v_{N_v}]$, produce a vector of disturbance signals $\mathbf{d}_e^T = [d_{e1}, d_{e2} \dots d_{eN_e}]$ at the virtual error microphones and a vector of reference signals $\mathbf{x}^T = [x_1, x_2 \dots x_{N_x}]$ at the reference microphones via the matrices of transfer responses \mathbf{P}_e and \mathbf{R} , respectively. As the reference sensor signals drive the matrix of the active control filter, \mathbf{W} , to produce a vector of control signals $\mathbf{u}^T = [u_1, u_2 \dots u_{N_u}]$, the control signals are transmitted to the virtual error microphones, via the matrix of plant responses \mathbf{G}_e . A vector of error signals, \mathbf{e} , at the virtual error microphones after control can be thus defined as

$$\mathbf{e} = \mathbf{d}_e + \mathbf{G}_e \mathbf{u} = \mathbf{d}_e + \mathbf{G}_e \mathbf{W} \mathbf{x} = \mathbf{P}_e \mathbf{v} + \mathbf{G}_e \mathbf{W} \mathbf{R} \mathbf{v}. \quad (1)$$

When it is necessary to estimate the virtual error signals, \mathbf{e} , from the monitoring microphone signals $\mathbf{m}^T = [m_1, m_2 \dots m_{N_m}]$, which are defined as

$$\mathbf{m} = \mathbf{d}_m + \mathbf{G}_m \mathbf{u} = \mathbf{d}_m + \mathbf{G}_m \mathbf{W} \mathbf{x} = \mathbf{P}_m \mathbf{v} + \mathbf{G}_m \mathbf{W} \mathbf{R} \mathbf{v}, \quad (2)$$

where $\mathbf{d}_m^T = [d_{m1}, d_{m2} \dots d_{mN_m}]$ is a vector of disturbance signals at the monitoring microphones and \mathbf{G}_m is the matrix of plant responses between the secondary sources and the monitoring microphones, the estimated error signals, $\hat{\mathbf{e}}$, at the virtual error microphones can be written as

$$\hat{\mathbf{e}} = \hat{\mathbf{d}}_e + \hat{\mathbf{G}}_e \mathbf{u} = \hat{\mathbf{O}} \hat{\mathbf{d}}_m + \hat{\mathbf{G}}_e \mathbf{u} = \hat{\mathbf{O}} (\mathbf{m} - \hat{\mathbf{G}}_m \mathbf{u}) + \hat{\mathbf{G}}_e \mathbf{u}, \quad (3)$$

where the superscript $\hat{}$ represents an estimate of an actual value and $\hat{\mathbf{O}}$ is the linear observation filter used to estimate $\hat{\mathbf{d}}_e$ from $\hat{\mathbf{d}}_m$.

Since the active control system attempts to minimise the estimated error signals, $\hat{\mathbf{e}}$, the accuracy of the nearfield estimation with the observation filter, $\hat{\mathbf{O}}$, can be an important factor for better attenuation at the error microphones. Previous research has shown that when the number of monitoring microphones (N_m) is larger than the number of error microphones (N_e), the problem is mathematically overdetermined, the optimal observation filter, \mathbf{O}_{opt} , is given as

$$\mathbf{O}_{\text{opt}} = \mathbf{S}_{d_m d_e} (\mathbf{S}_{d_m d_m} + \beta \mathbf{I})^{-1} = \mathbf{P}_e \mathbf{S}_{vv} \mathbf{P}_m^H (\mathbf{P}_m \mathbf{S}_{vv} \mathbf{P}_m^H + \beta \mathbf{I})^{-1}, \quad (4)$$

where $\mathbf{S}_{d_m d_e} = E[\mathbf{d}_e \mathbf{d}_m^H]$ is the cross spectral density matrix between \mathbf{d}_m and \mathbf{d}_e , $\mathbf{S}_{d_m d_m} = E[\mathbf{d}_m \mathbf{d}_m^H]$ is the power spectral density matrix for \mathbf{d}_m when $E[\cdot]$ is the expectation operator and H is the Hermitian, complex conjugate transpose, and β is a positive real effort-weighting parameter to improve the robustness of the optimal observation filter and \mathbf{I} is the identity matrix having the same dimensions as $\mathbf{S}_{d_m d_m}$ [4] [5]. If \mathbf{S}_{vv} is the power spectral density matrix of primary source strengths, \mathbf{O}_{opt} , can be written in the alternative form on the right-hand side of Eq. (4). From Eq. (4), the optimal observation filter, \mathbf{O}_{opt} , can be calculated from pre-measured data of \mathbf{d}_e and \mathbf{d}_m before active control.

To calculate the optimal active controller, \mathbf{W}_{opt} , by substituting Eq. (2) into Eq. (3), Eq. (3) can be expressed as

$$\hat{\mathbf{e}} = \hat{\mathbf{O}} \mathbf{d}_m + \left[\hat{\mathbf{G}}_e + \hat{\mathbf{O}}(\mathbf{G}_m - \hat{\mathbf{G}}_m) \right] \mathbf{W} \mathbf{x}. \quad (5)$$

In Eq. (5), the term, $\hat{\mathbf{G}}_e + \hat{\mathbf{O}}(\mathbf{G}_m - \hat{\mathbf{G}}_m)$ can be defined as the effective plant response, \mathbf{G} , between \mathbf{u} and $\hat{\mathbf{e}}$. By minimising the cost function, which is defined as $J_1 = \text{trace} \left\{ E[\hat{\mathbf{e}} \hat{\mathbf{e}}^H] \right\}$, the optimal active controller, \mathbf{W}_{opt} can be obtained using the derivation detailed in Ref. [2] [4] as

$$\mathbf{W}_{\text{opt}} = -(\mathbf{G}^H \mathbf{G})^{-1} \mathbf{G}^H \mathbf{O}_{\text{opt}} \mathbf{S}_{x d_m} \mathbf{S}_{x x}^{-1}, \quad (6)$$

where the optimal observation filter, \mathbf{O}_{opt} , is applied and $\mathbf{S}_{x d_m} = E[\mathbf{d}_m \mathbf{x}^H]$ is the cross spectral density matrix between \mathbf{d}_m and \mathbf{x} , $\mathbf{S}_{x x} = E[\mathbf{x} \mathbf{x}^H]$ is the power spectral density matrix for \mathbf{x} .

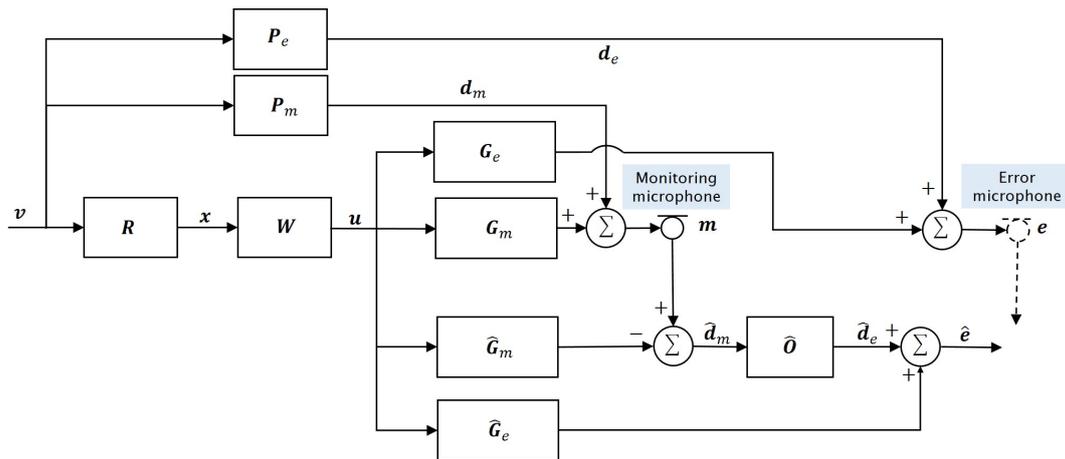


Figure 1: Block diagram of the feedforward active control algorithm, combined with the remote microphone technique.

2.2 Time domain analysis

For stationary random broadband noise, it is required that the observation filter and the active controller are causally constrained due to the unpredictability of random noise. Therefore, we will investigate the nearfield estimation and attenuation performance in time domain.

In Fig. 1, a vector of N_x reference sensor signals in time domain, at the n th sample time is expressed as $\mathbf{x}(n)^T = [x_1(n), x_2(n) \dots x_{N_x}(n)]$ and a vector of control signals for the N_u secondary sources is described as $\mathbf{u}(n)^T = [u_1(n), u_2(n) \dots u_{N_u}(n)]$ from the controller which is the I th order FIR filter with coefficients, $w_{n_u n_x i}$. In addition, the matrix of plant responses, \mathbf{G}_e is described by the J th order FIR filters with arbitrary accuracy, which have the impulse response from the n_u th secondary source to the n_e th error microphone as $g_{e, n_e n_u j}$. The vectors of error signals and disturbance signals for N_e error microphones are $\mathbf{e}(n)^T = [e_1(n), e_2(n) \dots e_{N_e}(n)]$ and

$\mathbf{d}_e(n)^T = [d_{e1}(n), d_{e2}(n) \dots d_{eN_e}(n)]$, respectively and thus the n_e th error microphone signal can be written as [2]

$$e_{n_e}(n) = d_{en_e} + \sum_{n_u=1}^{N_u} \sum_{j=0}^{J-1} \sum_{n_x=1}^{N_x} \sum_{i=0}^{I-1} g_{e,n_en_uj} w_{n_un_xi} x_{n_x}(n-i-j). \quad (7)$$

When the $N_e N_u N_x$ filtered-reference signals for the error microphones are written as

$$r_{e,n_en_un_x}(n) = \sum_{j=0}^{J-1} g_{e,n_en_uj} x_{n_x}(n-j), \quad (8)$$

Eq. (7) can be expressed substituting Eq. (8) to Eq. (7) as

$$e_{n_e}(n) = d_{en_e} + \sum_{i=0}^{I-1} \mathbf{w}_i^T \mathbf{r}_{en_e}(n-i), \quad (9)$$

where $\mathbf{w}_i^T = [w_{11i} \ w_{12i} \ \dots \ w_{1N_x i} \ w_{21i} \ w_{22i} \ \dots \ w_{N_u N_x i}]$ and $\mathbf{r}_{en_e}^T = [r_{e,n_e11} \ r_{e,n_e12} \ \dots \ r_{e,n_e1N_x} \ r_{e,n_e21} \ r_{e,n_e22} \ \dots \ r_{e,n_eN_u N_x}]$. Therefore, the vector of signals at the error microphones in time domain can be expressed as

$$\mathbf{e}(n) = \mathbf{d}_e(n) + \mathbf{R}_e(n)\mathbf{w}(n), \quad (10)$$

where

$$\mathbf{R}_e(n) = \begin{pmatrix} \mathbf{r}_{e1}^T(n) & \mathbf{r}_{e1}^T(n-1) & \dots & \mathbf{r}_{e1}^T(n-I+1) \\ \mathbf{r}_{e2}^T(n) & \mathbf{r}_{e2}^T(n-1) & \dots & \mathbf{r}_{e2}^T(n-I+1) \\ \vdots & \vdots & \ddots & \vdots \\ \mathbf{r}_{en_e}^T(n) & \mathbf{r}_{en_e}^T(n-1) & \dots & \mathbf{r}_{en_e}^T(n-I+1) \end{pmatrix}, \quad (11)$$

and $\mathbf{w}(n) = [\mathbf{w}_0^T \ \mathbf{w}_1^T \ \dots \ \mathbf{w}_{I-1}^T]^T$. Similarly, the vector of signals at N_m monitoring microphones, $\mathbf{m}(n)^T = [m_1(n), m_2(n) \dots m_{N_m}(n)]$ is defined as

$$\mathbf{m}(n) = \mathbf{d}_m(n) + \mathbf{R}_m(n)\mathbf{w}(n), \quad (12)$$

where $\mathbf{d}_m(n)^T = [d_{m1}(n), d_{m2}(n) \dots d_{mN_m}(n)]$ and $\mathbf{R}_m(n)$ has a similar form with $\mathbf{R}_e(n)$ in Eq. (11), except that the impulse response from the n_u th secondary source to the n_m th monitoring microphone, $g_{m,n_m n_u j}$ is used in Eq. (8), instead of $g_{e,n_e n_u j}$.

The vector of estimated error signals in time domain, $\hat{\mathbf{e}}(n)$ from a similar process with Eq. (3) is written as

$$\hat{\mathbf{e}}(n) = \hat{\mathbf{d}}_e(n) + \hat{\mathbf{R}}_e(n)\mathbf{w}(n) = \hat{\mathbf{O}}(n)\{\mathbf{m}(n) - \hat{\mathbf{R}}_m(n)\mathbf{w}(n)\}' + \hat{\mathbf{R}}_e(n)\mathbf{w}(n), \quad (13)$$

where $\hat{\mathbf{O}}(n)$ is the matrix of observation filter coefficients in time domain, which is defined as $\hat{\mathbf{O}}(n) = [\mathbf{O}_1^T \ \mathbf{O}_2^T \ \dots \ \mathbf{O}_{N_e}^T]^T$ and $\mathbf{O}_{n_e} = [\mathbf{O}_{n_e1}^T \ \mathbf{O}_{n_e2}^T \ \dots \ \mathbf{O}_{n_e N_m}^T]^T$ and $\{\}'$ represents the $JN_m \times 1$ vector of values between braces when the impulse response from the n_m th monitoring microphone to the n_e th error microphone, $\mathbf{O}_{n_e n_m}$ is modelled by a FIR filter with J filter coefficients as

$$\mathbf{O}_{n_e n_m} = [\mathbf{O}_{n_e n_m 0} \ \mathbf{O}_{n_e n_m 1} \ \dots \ \mathbf{O}_{n_e n_m (J-1)}]^T. \quad (14)$$

The matrix of the optimal observation filter coefficients in time domain, \mathbf{O}_{opt} can be obtained using the derivation detailed in Ref.[4] as

$$\mathbf{O}_{opt} = \left(\{E[\mathbf{d}'_m(n)\mathbf{d}'_m(n)] + \beta\mathbf{I}\}'^{-1} \{E[\mathbf{d}'_m(n)\mathbf{d}'_e(n)]\}' \right)^T, \quad (15)$$

where $\mathbf{d}'_m(n)$ also represents the $JN_m \times 1$ vector, which is defined as  $\mathbf{d}'_m(n) = [\mathbf{d}'_{m1}(n) \ \mathbf{d}'_{m2}(n) \ \dots \ \mathbf{d}'_{mN_m}(n)]^T$ and $\mathbf{d}'_{mnm}(n) = [d_{mnm}(n) \ d_{mnm}(n-1) \ \dots \ d_{mnm}(n-J+1)]^T$, and \mathbf{I} is the identity matrix having the same dimensions as $E[\mathbf{d}'_m(n)\mathbf{d}'_m(n)]$.

To calculate the causal constrained optimal controller, \mathbf{w}_{opt} , in time domain to minimise the cost function, $J_2 = \text{trace} \left\{ E [\hat{\mathbf{e}}(n) \hat{\mathbf{e}}^T(n)] \right\}$, it is required that Eq. (13) is expressed in terms of $\mathbf{w}(n)$ by substituting Eq. (12) to Eq. (13) as

$$\hat{\mathbf{e}}(n) = \hat{\mathbf{O}}(n)\mathbf{d}'_m(n) + [\hat{\mathbf{O}}(n)\Delta\mathbf{R}'_m(n) + \hat{\mathbf{R}}_e(n)]\mathbf{w}(n), \quad (16)$$

where $\Delta\mathbf{R}'_m(n)\mathbf{w}(n) = \{[\mathbf{R}_m(n) - \hat{\mathbf{R}}_m(n)]\mathbf{w}(n)\}'$. Therefore, the causal constrained optimal controller, \mathbf{w}_{opt} , with the optimal observation filter, \mathbf{O}_{opt} , can be given by [2] [4]

$$\mathbf{w}_{opt} = -\{E[\mathbf{R}^T(n)\mathbf{R}(n)]\}^{-1}E[\mathbf{R}^T(n)\mathbf{O}_{opt}\mathbf{d}'_m(n)], \quad (17)$$

where $\mathbf{R} = [\mathbf{O}_{opt}\Delta\mathbf{R}'_m(n) + \hat{\mathbf{R}}_e(n)]$.

2.3 Head-tracking for local active sound control

When the position of a listener's head is continuously changing, the head-tracking technology can allow the local active control system combined with the remote microphone technique to improve the attenuation performance and convergence stability, as suggested by previous research [5] [6].

If $\hat{\mathbf{G}}_e$, $\hat{\mathbf{G}}_m$ and \mathbf{O}_{opt} in Eq. (6) and $\hat{\mathbf{R}}_e(n)$, $\hat{\mathbf{R}}_m(n)$ and \mathbf{O}_{opt} in Eq. (17) are preliminary decided at a single fixed position, the head movements can make differences between actual and estimated error signals and thus degrade the attenuation performance at the error microphones. To overcome this problem, the active headrest system can be designed to use the head position information, which is given by a head-tracking device, to update $\hat{\mathbf{G}}_e$, $\hat{\mathbf{G}}_m$ and \mathbf{O}_{opt} for tonal control and $\hat{\mathbf{R}}_e(n)$, $\hat{\mathbf{R}}_m(n)$ and \mathbf{O}_{opt} for broadband random control from a pre-calculated lookup table for different head positions.

3. Experimental investigation

3.1 Experimental arrangement

Fig. 2 shows the experimental installation used for measuring acoustic responses and the real-time implementation in an anechoic chamber. Four monitoring microphones were installed, with two on the back of the headrest and two on the supporting structure, as shown in Fig. 2. The positions of these monitoring microphones were carefully selected through a series of preliminary experiments to obtain an accurate nearfield estimation of the disturbance signals at the virtual error sensors. To compare the estimated disturbance signals at the virtual error microphones with the actual disturbance signals, two microphones were also installed in the ears of a dummy head. The dummy head and physical error microphones can be removed after the preliminary measurements and the system can be used by a real listener. A single loudspeaker was installed behind the active headrest system to act as the primary source and driven to produce either a sinusoidal sound or a stationary random sound.

For real-time tonal control, the optimal observation filters in frequency domain were calculated from Eq. (4) with the pre-measured acoustic responses and the improved filtered-reference LMS algorithm suggested by [5] [6] was applied for the integrated local active control. In addition, the performance for broadband control in-off-line was estimated from measured data using Eq. (15) and Eq. (17). The full set of transfer responses between the primary and secondary sources and the monitoring and physical error microphones was measured when the dummy head was located at 20 different positions on a 5 x 4 grid of points spaced 5 cm apart. Finally, a commercial headtracking device, the Microsoft Kinect, was installed in front of the dummy head. The information from the Kinect was decoded in real time using a plug-in software implemented in MaxMSP and this was passed to

a dSPACE-based adaptive controller, which already had stored in it the various plant responses and pre-calculated observation filters for 20 head positions.

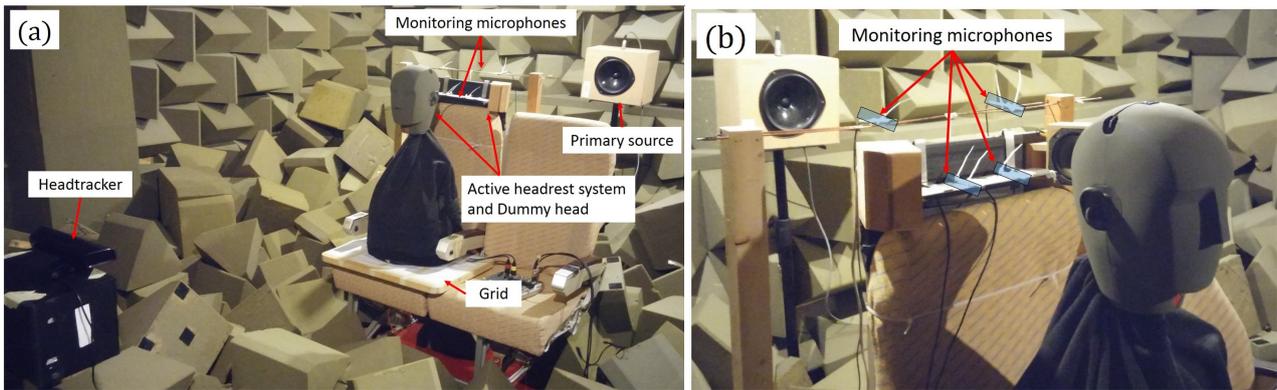


Figure 2: (a) The overall installation for both the nearfield estimation and the integrated active headrest system for controlling tonal disturbances in real-time and stationary random broadband disturbances in off-line (b) 4 monitoring microphones on both a mounting structure and a headrest

3.2 Real-time implementation for controlling tonal disturbances

When the regularization factor, β , in Eq. (4) was selected from the trade-off between the estimation accuracy and robustness to uncertainties as $\beta = 10^{-3}$, tonal disturbance signals at the error microphones of the dummy head at different frequencies were estimated from the monitoring microphone signals with the pre-calculated observation filter. The estimation error was defined as a normalised difference between actual and estimated disturbance signals at the error microphones, as presented by [5]. Therefore, when the dummy head was located in the nominal position, the measured estimation errors in Table 1 show that tonal disturbance signals at the error microphones can be estimated with considerable accuracy. With this nearfield estimation accuracy, the achieved attenuation performance of the integrated active headrest system is also shown in Table 1. It is shown that attenuation performance of greater than 15 dB is achieved at different frequencies.

The effect of the head-tracking device on the integrated active control system was then investigated by moving the dummy head from the nominal position to different positions on the grid. When the dummy was moved both 10 cm forward and 10 cm right, time history of the signal from the right microphone in the dummy during control is shown in Fig. 3. It can be seen in Fig. 3 that when the active headrest system is implemented without the head tracker, the disturbance signal at 600 Hz is slightly reduced but the disturbance signal at 700 Hz is enhanced due to the significant difference between the virtual and actual error signals. However, when the head tracker is in operation, the attenuation performance is significantly improved at both frequencies.

Table 1: The real-time nearfield estimation error and attenuation performance of the integrated active headrest system when tonal disturbance signals at the error microphones of the dummy head in the nominal position are estimated by the 4 monitoring microphones around the headrest with the pre-calculated observation filter and are reduced by local active control in the anechoic chamber .

Frequencies	Real-time estimation error right / left ear, (dB)	Achieved attenuation performance right / left ear, (dB)
500 Hz	-19.3 / -15.1	-23.9 / -20.4
600 Hz	-21.1 / -17.1	-15.9 / -28.6
700 Hz	-17.0 / -16.0	-15.0 / -26.0

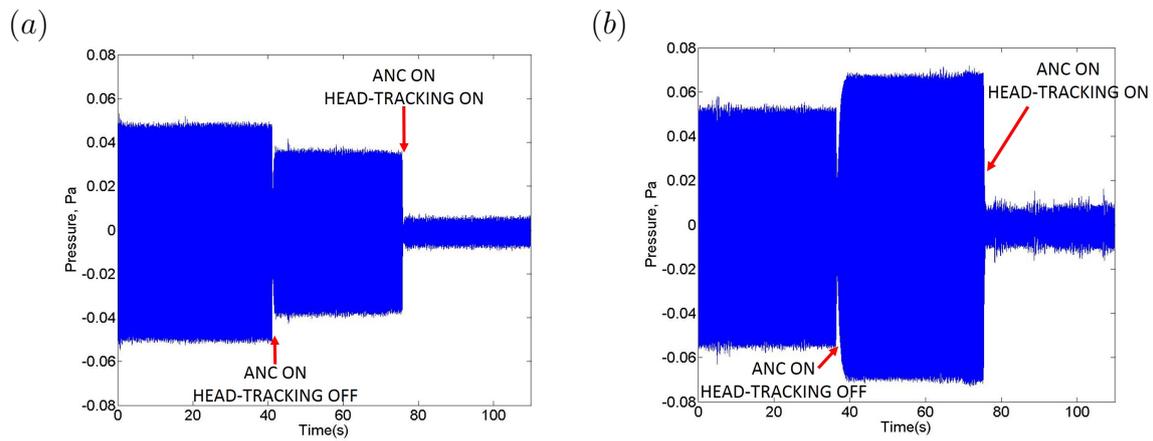


Figure 3: Measured signals at the right error microphone of the dummy head at a head position moved both 10cm forward and 10cm right from the nominal position when the integrated active headrest system with the remote microphone technique and the head tracker reduces the tonal disturbance signals from a single primary source in the anechoic chamber: (a) 600 Hz, (b) 700 Hz

3.3 Off-line analysis for controlling broadband random disturbances

When the primary source was driven to produce stationary random broadband sound, measured signals from the monitoring and error microphones in Fig. 2 were used to investigate the attenuation performance of the active headrest system in off-line. First of all, when the optimal observation filter with the causality constraint was obtained from Eq.(15), the impulse response of O_{12} among the FIR filters of O_{opt} is shown in Fig. 4 (a). When O_{12} of O_{opt} in Eq.(4) is transformed by inverse Fourier transform and compared with the impulse response of O_{12} of O_{opt} in Eq.(15), the impulse response of O_{12} is similar with the causal response of O_{12} but the noncausal response of O_{12} is excluded from the the impulse response of O_{12} due to the causality constraint. Fig. 4 (b) shows that the causal constrained filter can estimate stationary random disturbances at the error microphones with less than -10 dB estimation error except degradation at around 250 Hz due to the corresponding anti-resonance.

The attenuation performance of the causal constrained controller is compared when the error signals are directly measured by the physical error microphones and are estimated by the causal observation filter. Although the attenuation performance with the remote microphone technique is generally lower than that of the physical error microphones, more than 10 dB is still achievable with the remote microphone technique. In this study, the causal nearfield estimation and active control produce the remarkable performance because the primary source was behind the microphones and the noncausal response of the observation filter was thus negligible. However, if the primary source is in front of the microphones and the noncausal response becomes more important, the improved remote microphone technique should be considered to overcome this problem.

4. Conclusions

The attenuation performance of an active headrest system integrated with the remote microphone technique and head-tracking has been investigated. The optimal observation filter and active controller have been initially derived in frequency domain for controlling tonal disturbances. In addition, the optimal observation filter and active controller with the causal constraint have been discussed in time domain for broadband random disturbances. To improve the performance of these controllers, the head-tracking has also been suggested, when the position of a listener's head is changing.

The nearfield estimation and noise reduction of the integrated system have been verified through real-time experiments for tonal control and off-line analysis for random broadband control. The real-

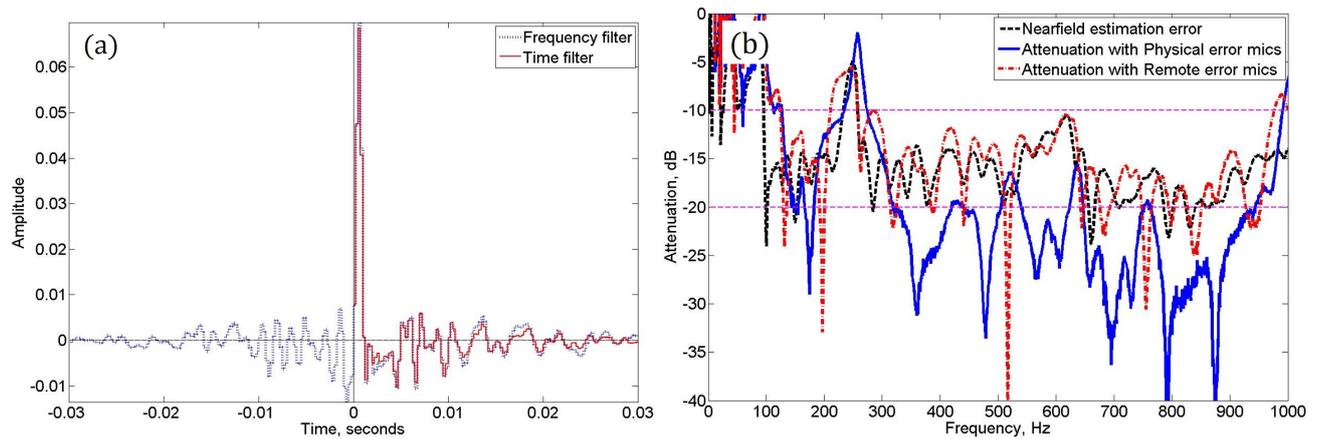


Figure 4: (a) Comparison of the impulse response of O_{12} among the FIR filters of the causal constrained O_{opt} with the inverse Fourier transformed O_{12} of O_{opt} without the causal constraint, (b) the nearfield estimation error (dash line) when a stationary random broadband sound is estimated by the monitoring microphones in Fig. 2 and the attenuation performance of the integrated active headrest system in Fig. 2 when the error signals are provided by either the physical error microphones (solid line) or the causal remote microphone technique (dash-dot line).

time experimental results have demonstrated that disturbance signals at the error microphones were estimated by the monitoring microphones with less than -15 dB error and the adaptive active control achieved more than 15 dB attenuation. The application of a head-tracking device has improved the attenuation performance when the dummy was moved to different positions from the nominal position. The results from the off-line analysis have shown that the causal filters for the nearfield estimation and active control allow to reduce stationary random disturbances when noncausal responses are less important.

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