AN ITERATIVE INVERSE FILTER DESIGN FOR THE MULTI-CHANNEL SOUND FIELD REPRODUCTION SYSTEM

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ABSTRACT

In order to realize the multi-channel sound field reproduction system, it is indispensable to design inverse filters that remove the effect of room transfer functions. The design method in the frequency domain based on the least-norm-solution (LNS) is less memory and less calculation than the design method in the time domain. However, the LNS method cannot guarantee the causality and stability of the filters. In this paper, a design method of a time domain inverse filter using iterative processing in the frequency domain for multi-channel sound field reproduction is proposed, and the numerical analysis result is described. The proposed method can decrease the squared error for every sensor by 3–12 dB. Furthermore, the reproduced sound by this method attains over 13 dB improvement in the segmental SNR compared with one designed by the LNS method for real environment impulse responses.

KEYWORDS: multi-channel sound field reproduction, inverse filter, least-mean-square, least-norm-solution, iterative processing

INTRODUCTION

In order to realize good acoustic localization in the large sound field, we have to control many points monitored via error sensors by many secondary acoustic sources. Multi-channel transaural-system[1] is one of the practical approach based on above-mentioned principle. In this system, we have to design inverse filters to cancel the room transfer function. Because of less memory and calculation, the design method in the frequency domain based on the Least-Norm-Solution (LNS) is more efficient approach than that in the time domain[2]. These inverse filters must be finite length, however the LNS method cannot guarantee the causality and stability of the filters.

Nakajima et al. has proposed an iterative design method of the time domain inverse filter for the improvement of reproduction accuracy of inverse filter designed in the frequency domain[3]. This paper describes a new design method of a time domain inverse filter using iterative processing in
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Design Method Based on LNS. Let's consider the reproduction system that the number of loudspeakers (as secondary sources) is $M$, and the number of microphones (as control points) is $N$. We obtain the following equation as

$$GH = I_N \quad (1)$$

where $G = [G_{NM}(\omega)]$ is the $N \times M$ transfer function matrix, $H = [H_{MN}(\omega)]$ is the $M \times N$ inverse filter matrix, $I_N$ is $N \times N$ identity matrix, and $\omega$ denotes a frequency. Here, we can find the optimal $H$ to solve $H = G^+ I_N$. If $M > N$, we find $H_{LNS}$ as LNS; this can be given by

$$H_{LNS} = G' (GG')^{-1} \quad (2)$$

where $G'$ is the Hermitian transposed matrix of $G$.

Design Method Based on Iterative Processing. Consider to find the $(k+1)$ th approximate solution $h^{(k+1)}$ of inverse filter by using the $k$ th approximate solution $h^{(k)}$. In the case of $k = 0$, we adopt $h_{LNS}$ as $h^{(0)}$. In the case of $k > 0$, $h^{(k+1)}$ is given by

$$e^{(k)} = d - F^{-1}\{GH^{(k)}\}, \quad (3)$$

$$H_{\text{diff}}^{(k)} = H_{LNS} E^{(k)}, \quad (4)$$

$$h^{(k+1)} = h^{(k)} + h_{\text{diff}}^{(k)} \quad (5)$$

where $F^{-1}\{}$ is the inverse Fourier transform, $d = F^{-1}\{I_N\}$ is the desired signal of every control point. $e^{(k)}$, $h^{(k)}$ and $h_{\text{diff}}^{(k)}$ are the time-domain representation of $E^{(k)}$, $H^{(k)}$ and $H_{\text{diff}}^{(k)}$, respectively.

EXPERIMENTAL CONDITIONS

Measurement of Impulse Response. The impulse responses used in this study were measured in our acoustic experiment room, where reverberation time was about 0.2 seconds. The arrangement of the apparatus is shown in Fig.1. Every loudspeaker was used to radiate the sound source signal, and the received signals were recorded through six microphones fixed on the head of Head And Torso Simulator (HATS) as shown in Fig.2. All sound data prepared in this experiment were sampled at 48 kHz with 16-bit resolution. To obtain the impulse response, Time Stretched Pulse (TSP)[4] of 131072-point length was used as the sound source signal.

Design of Inverse Filter. First, $H_{LNS}$ was calculated by Eqn.(2), in which the impulse response of 9600-point length was used and transformed into the frequency domain. These initial inverse filters were filtered by a bandpass filter with the pass band ranging from 150 to 4000 Hz. Finally, these were transformed into the time domain by Inverse FFT, and were cut as filter length to be 16384-points.

Next, we found an improved inverse filter by using the proposed method. The inverse filters were convolved with impulse responses, and the error from the desired signal, $e^{(k)}$, of every control point in the time domain was measured. The $e^{(k)}$ was transformed into the frequency domain by 65536-point FFT, and $H_{\text{diff}}^{(k)}$ was calculated by Eqn.(4). Finally, we obtained $h^{(k+1)}$ by Eqn.(5). This iterative processing was carried out to 50 times.
EVALUATION BY NUMERICAL ANALYSIS

Evaluation on System Transfer Function. Inverse filters for \( M \cdot N \) transaural-system (\( M \cdot N \) system) shown in Fig.1 were designed, where \( M \) is the number of loudspeakers (\( M = 4, 8, 16 \)), and \( N \) is the number of microphones (\( N = 2, 6 \)). Table.1 shows the arrangement of used loudspeakers. In the case of \( N = 2 \), microphones were fixed on \( C_2 \) and \( C_3 \) shown in Fig.2. Inverse filters were convolved with impulse responses, and the squared error from the desired signal was calculated at every control point. The relation between the average of squared error of each control point and iteration times at every system is shown in Fig.3. The squared error decreases by the iterative processing even in the case of all systems, at about 3–12 dB. This phenomenon indicates that the proposed method is effective for the inverse filter design. The effectiveness is more enhanced, when the number of control points is both 2 and 6, as the number of loudspeakers increases. From these results, it is expected that the more number of loudspeakers are required as we obtain more reduction of the squared error.

Evaluation on Reproduced Sound Quality. On the assumption of 16-6 system under the condition of Fig.1, both inverse filters by the LNS method and the proposed method (50-time iteration) were designed, respectively. We can obtain the original sound source by convolving English voice by male speaker (reproduction time was 6 seconds, and frequency band was 150–4000 Hz) with the impulse response between the primary source and each microphone shown in Fig.1. Then, we also reproduce the same sound to be evaluated by using the proposed 16-6 system. In each control point, Signal-to-Noise Ratio (SNR) of each reproduction sound for the original sound source was calculated every 30 ms. Table.2 shows the result of the averaged SNR in the whole voice interval. In all control points, the SNRs of the reproduction sound by proposed method were improved by about 14 dB in comparison with those of reproduction sounds by LNS method. Fig.4 shows a speech pause in the original sound source and in the reproduction sounds observed at the control point \( C_2 \). Although the noise, which sounds like pre-echo, arises in 0.93–1.02 second.
Table 2 Reproduction accuracy of every control point in 16-6 system

<table>
<thead>
<tr>
<th>Design method</th>
<th>SNR [dB]</th>
</tr>
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<tbody>
<tr>
<td></td>
<td>C_1</td>
</tr>
<tr>
<td>LNS</td>
<td>14.0</td>
</tr>
<tr>
<td>proposed</td>
<td>28.5</td>
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</tbody>
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Fig. 3 The relation between iteration times and the average of the squared error from the desired signal at each control point in M-N system.

Fig. 4 The waveform of reproduction sound recorded at near left ear. Top figure is the original sound source, middle figure is by LNS method, and bottom figure is by proposed method.

(surrounded area by the dotted line in Fig. 4) of the reproduction sound by LNS method, the proposed method can reduce this kind of noise in the reproduction sound.

SUMMARY

This paper describes a new inverse-filter-design method that uses iterative processing in the frequency domain for the multi-channel sound field reproduction system. As a result of numerical simulation, the squared error from the desired signal decreased by iterative processing in all control points. Also the noise sounds like a pre-echo was reduced in the reproduction sound by the proposed method. The effectiveness was remarkably enhanced as the scale of the system got larger. From these results, it was evident that the proposed method can be applicable to the large reproduction system.

REFERENCES