

PAPER

Individual equalization in binaural reproduction of a reverberant sound field

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Abstract: Currently, there is a constraint in the individual equalization of the differences in head-related transfer functions (HRTFs) between a dummy head and a listener for a binaural system. Therefore, the equalization function of a listener must be measured in the original sound field. To remove this constraint, we have developed a method that replaces the function measured in the original sound field with that measured in another sound field (an equalization sound field). The two functions' profiles are similar. Therefore, the original function is approximated by adding reflected sound components to the transfer function measured in an equalization sound field. In the proposed method, the reflected sound components are extracted by applying the auto regressive moving-average (ARMA) model to the transfer function measured in the original sound field. An ear canal transfer function is introduced into the proposed method to calculate the transfer function measured at the entrance of the ear canal from that at the eardrum. A psychoacoustical experiment is carried out to evaluate the equalized sound with respect to four criteria and localization. The experimental results confirm that the proposed method is effective for all listeners.

Keywords: Binaural technique, Individual equalization, Head-related transfer function, Ear canal transfer function, Equalization sound field

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1. INTRODUCTION

When sound signals recorded in an original sound field (OSF) such as a concert hall are precisely reproduced, a listener in another sound field will have the impression that he or she exists in the OSF. The binaural technique [1,2] is known as one of the methods for realizing high-definition audio reproduction. The basis of this technique is that an auditory experience is assumed to be reproduced by recording sound pressures at the eardrums of a listener and reproducing them accurately.

In the binaural technique, a dummy head with microphones positioned at the eardrums is usually used for recording in an OSF; subsequently, the recorded sounds are presented to a listener through headphones. This system requires equalization to cancel the duplicate frequency characteristics of the external ears during recording and reproduction [3,4]. In addition, an equalization method to

compensate for individual differences in the head-related transfer functions (HRTFs) caused by the transformation of the spectrum by reflection and diffraction around the external ear, head and torso has been proposed [5–7]. However, there is a constraint that measurement of the HRTF of the listener in the OSF is necessary. To overcome this constraint, Ozawa *et al.* [8] proposed an equalization method that replaced the equalization function measured in the OSF with that measured in another sound field (an equalization sound field, ESF) when the OSF was a dead room. In this article, we propose an equalization method for a reverberant sound field.

2. RESOLUTION OF THE CONSTRAINT ON SOUND FIELD

2.1. Formulation of Compensation

Figure 1 shows a block diagram of the binaural recording and reproduction system. In this system, in accordance with the literature [4–7], the frequency spectra observed in an OSF and a reproduced sound field are defined as follows.

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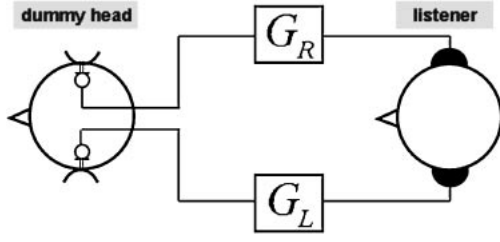


Fig. 1 Block diagram of the binaural recording and reproduction system.

H_3 : transfer function from a sound source to the entrance of the open ear canal in the OSF

H_4 : transfer function from a sound source to the eardrum in the OSF

M_1 : transfer function of Microphone 1 placed at the eardrum

M_2 : transfer function of Microphone 2 placed at the entrance of the ear canal

E_{hp} : voltage of the input terminal of a headphone for reproduction

E_{mic} : voltage of the output terminal of Microphone 2

Note that a transfer function includes sensitivity as well as frequency response. In [4], H_1 and H_2 are also defined: H_1 is a transfer function from a sound source to the center of the listener's head when the listener is not present; H_2 is a transfer function from a sound source to the entrance of a blocked ear canal. In this study, it is assumed that the transfer functions are measured at an open ear canal. Thus, it is not necessary to refer to transfer functions H_1 and H_2 . A prime symbol attached to a transfer function implies that it is measured with a dummy head. A suffix []^{OSF} denotes that the transfer function is measured in the OSF. When binaural signals are recorded at the eardrums of a dummy head and reproduced to a listener through headphones, the equalization function G is given as Eq. (1) [5,7]:

$$G = \frac{M_2 \cdot [H_3]^{OSF}}{M_1 \cdot [H_4]^{OSF}} \cdot \frac{E_{hp}}{E_{mic}}. \quad (1)$$

The first term in Eq. (1) is the individual equalization function, whereas the second term compensates for the characteristics of the listener's external ear that overlap because listening is carried out with headphones. The latter is called the headphone transfer function (HpTF) [3,4]. The equalization function G is calculated separately for the right and left ears to compensate for sound signals recorded with a dummy head as shown in Fig. 1.

According to Eq. (1), it is necessary to measure a transfer function in the OSF for each listener, $[H_3]^{OSF}$, which is a constraint. In this study, we propose to replace the transfer function measured in the OSF with that measured in another sound field (an equalization sound field, ESF) to approximate the first term in Eq. (1).

2.2. Comparison between Transfer Functions Measured in the OSF and the ESF

In order to compare $[H_3]^{OSF}$ and $[H_3]^{ESF}$, they were measured in two sound fields: one was a meeting room ($W: 3.8 \times D: 6.3 \times H: 2.8$ m) as the OSF; the other was a soundproof chamber ($W: 2.4 \times D: 3.3 \times H: 2.0$ m, Music Cabin, SC-6) as the ESF. The reverberation times were 295 ms and 43 ms, respectively. In each room, a loudspeaker (Yamaha, NS-3MX) was placed to the front right (at 45°), 1.5 m from the center of the subject's head. The height of the loudspeaker was the same as the subject's ears. In the experiment conducted in this study, stimuli were presented to a subject in the OSF using a loudspeaker or headphones (Sennheiser, HD600). The transfer functions of each subject were measured while the headphones were worn. A time-stretched pulse (TSP) signal [9] (sample length: 16,384 points, sampling frequency: 44.1 kHz) was used to measure the impulse response by a probe microphone (Brüel & Kjær, 4182) at the entrance of the ear canal. Then, the transfer function was obtained by Fourier transformation of the impulse response. Figure 2 shows the impulse responses measured for a subject. In Fig. 2(a), the response measured in the OSF as a reverberant sound field exhibits a direct sound component and reflected sound components. In Fig. 2(b), however, the response measured in the ESF as a near-dead sound field shows that the direct sound component is dominant. Figure 3 shows the frequency characteristics corresponding to the impulse responses in Fig. 2. The characteristics measured in the OSF have numerous peaks and dips, which are derived from reflected sounds; those in the characteristics measured in the ESF are few. However, the profiles of the two spectra are similar. For example, dips around 1.6 kHz and 4 kHz and peaks around 1.2 kHz and 4.5 kHz, indicated by arrows in the figure, are observed in both transfer functions. Therefore, it is assumed that the transfer function in the OSF can be approximated to the transfer function in the ESF by adding peaks and dips due to reflected sounds.

2.3. Proposed Method to Approximate the Transfer Function in the OSF

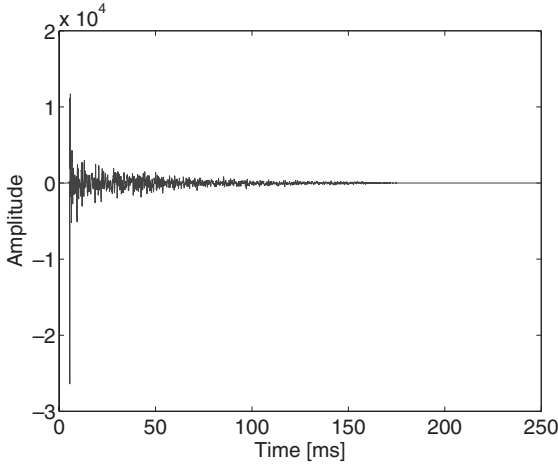
2.3.1. Derivation of the proposed method

Because the transfer function measured in the OSF consists of a direct sound component and reflected sound components that arrive after a delay, it is given as

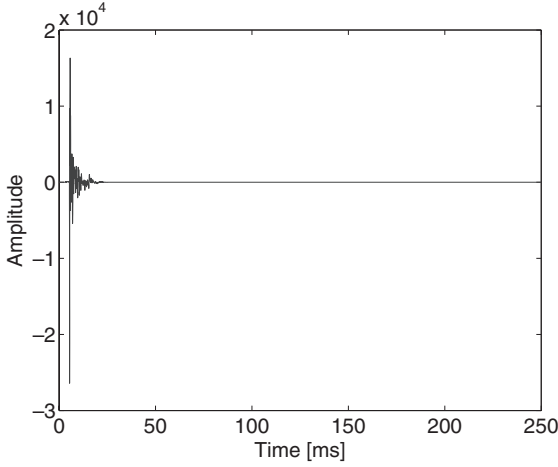
$$[H_3]^{OSF} = [H_3]_{direct}^{OSF} + [H_3]_{reflect}^{OSF}, \quad (2)$$

where the suffixes "direct" and "reflect" respectively denote the direct sound component and reflected sound components. The transfer function is described in the time domain as follows:

$$[h_3]^{OSF} = [h_3]_{direct}^{OSF} + [h_3]_{reflect}^{OSF}, \quad (3)$$



(a) Original sound field (meeting room)



(b) Equalization sound field (soundproof chamber)

Fig. 2 Impulse responses measured in the OSF and ESF (subject A, the right ear).

where $[h_3]^{\text{OSF}}$ is the impulse response measured in the OSF, $[h_3]_{\text{direct}}^{\text{OSF}}$ is the direct sound component, and $[h_3]_{\text{reflect}}^{\text{OSF}}$ are the reflected sound components.

Because the direct sound is dominant in the ESF, it is assumed that the direct sound component in the OSF, $[h_3]_{\text{direct}}^{\text{OSF}}$, can be estimated from the impulse response in the ESF, $[h_3]^{\text{ESF}}$:

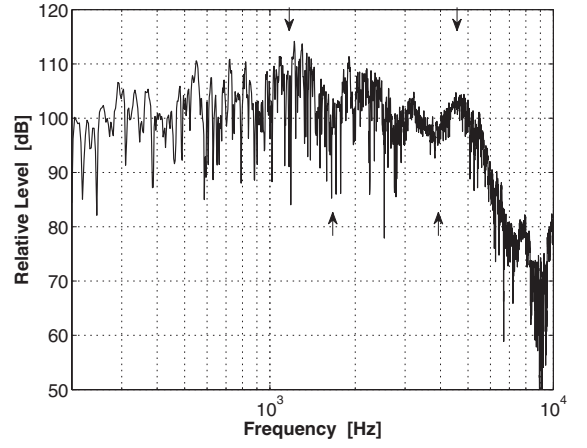
$$[\hat{h}_3]_{\text{direct}}^{\text{OSF}} \simeq [h_3]^{\text{ESF}}, \quad (4)$$

where $\hat{\cdot}$ denotes an estimated or approximated transfer function. On the other hand, because the reflected sound components for a listener, $[h_3]_{\text{reflect}}^{\text{OSF}}$, are difficult to estimate, we substitute the components by those obtained using a dummy head, $[h'_3]_{\text{reflect}}^{\text{OSF}}$ as follows:

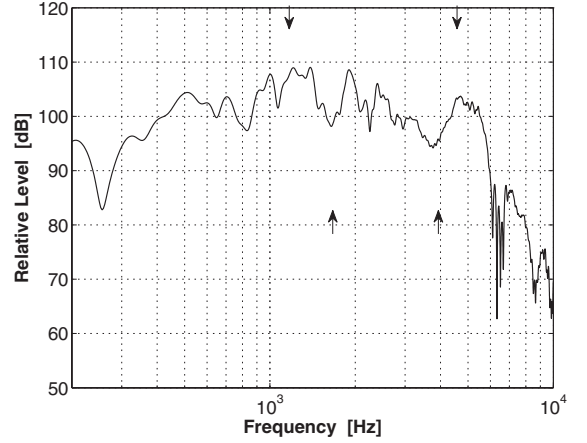
$$[\hat{h}_3]_{\text{reflect}}^{\text{OSF}} \simeq [h'_3]_{\text{reflect}}^{\text{OSF}}. \quad (5)$$

Because Eq. (3) is also applicable for the impulse response for a dummy head, $[h'_3]_{\text{reflect}}^{\text{OSF}}$ is given as

$$[h'_3]_{\text{reflect}}^{\text{OSF}} = [h'_3]^{\text{OSF}} - [h'_3]_{\text{direct}}^{\text{OSF}}. \quad (6)$$



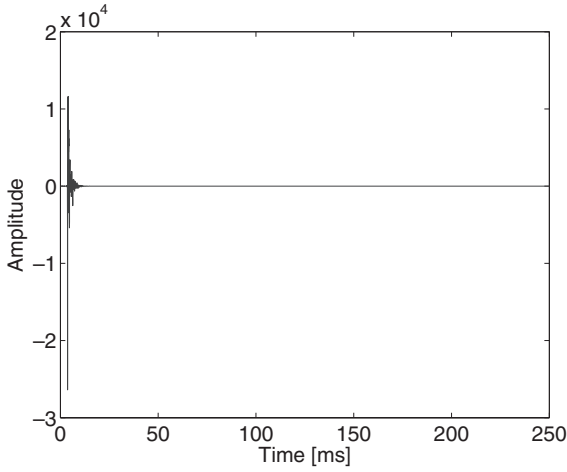
(a) Original sound field (meeting room)



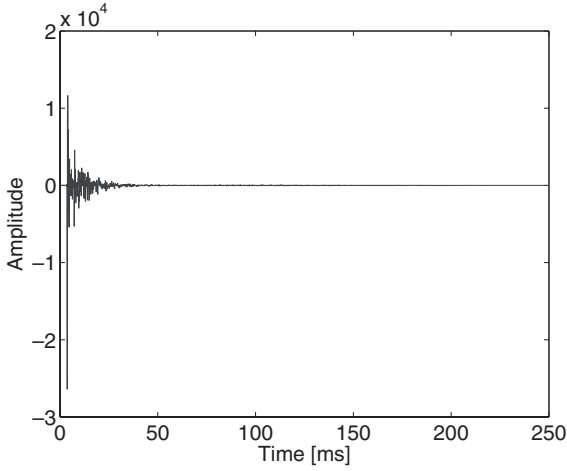
(b) Equalization sound field (soundproof chamber)

Fig. 3 Frequency characteristics of transfer functions measured in the OSF and ESF (subject A, the right ear).

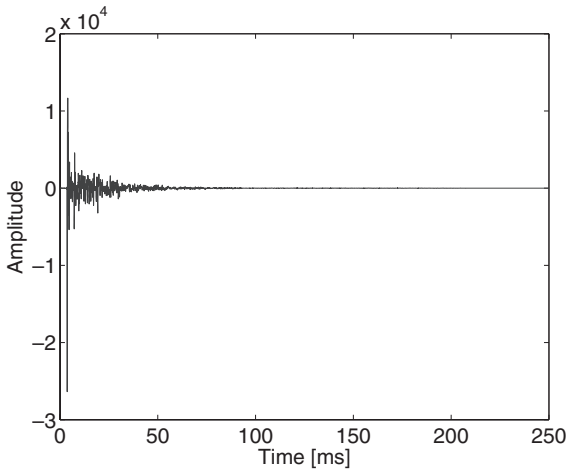
In this study, $[h'_3]_{\text{direct}}^{\text{OSF}}$ is estimated by applying the autoregressive moving-average (ARMA) model [10] to the transfer function, $[h'_3]^{\text{OSF}}$, to remove the reflected sound components by reducing the numbers of poles and zeros. The ARMA model is implemented by using the function “prony” in MATLAB 7.1 R14 (MathWorks). Prony’s method [11] is a method for designing IIR filter coefficients in the time domain impulse response. This method uses AR modeling to determine the AR coefficients. Then, it determines the MA coefficients for which the impulse response of the output filter exactly matches the first $n + 1$ samples corresponding to the order of the model. Therefore, this process corresponds to extracting the beginning of the impulse response $[h'_3]^{\text{OSF}}$, i.e., the direct sound component $[h'_3]_{\text{direct}}^{\text{OSF}}$. By subtracting the estimated component from $[h'_3]^{\text{OSF}}$, $[h'_3]_{\text{reflect}}^{\text{OSF}}$ can be calculated. Figure 4 shows the estimated impulse responses when the orders of poles/zeros of the model are 100/100, 500/500, and 1,000/1,000; these conditions were considered to be effective on the basis of a preliminary experiment [12]. The response



(a) Orders of the ARMA model: 100/100



(b) Orders of the ARMA model: 500/500



(c) Orders of the ARMA model: 1,000/1,000

Fig. 4 Impulse responses estimated by ARMA model for each order (dummy head, the right ear).

shown in Fig. 4(c) includes part of the reflected sound components. Therefore, the orders of the model can control the length of the reflected sound components.

Then, the impulse response $[\hat{h}_3]^{\text{OSF}}$ is estimated as

$$[\hat{h}_3]^{\text{OSF}} = [h_3]^{\text{ESF}} + [h'_3]^{\text{OSF}}_{\text{reflect}}. \quad (7)$$

In the frequency domain, the above equation is expressed as

$$[\hat{H}_3]^{\text{OSF}} = [H_3]^{\text{ESF}} + [H'_3]^{\text{OSF}}_{\text{reflect}}. \quad (8)$$

The estimation of $[H'_3]^{\text{OSF}}_{\text{reflect}}$ requires the transfer function measured at the entrance of the ear canal of the dummy head, $[H'_3]^{\text{OSF}}$. In this study, however, the measurement of the transfer function $[H'_4]^{\text{OSF}}$ for the dummy head is carried out at the eardrum. Thus, the authors calculate $[H'_3]^{\text{OSF}}$ from the measured $[H'_4]^{\text{OSF}}$. The transfer function measured at the eardrum is represented by the product of the transfer function measured at the entrance of the ear canal by the ear canal transfer function H'_e (the transfer function from the entrance of the ear canal to the eardrum of the dummy head):

$$[H'_4]^{\text{OSF}} = H'_e [H'_3]^{\text{OSF}}. \quad (9)$$

Therefore, $[H'_3]^{\text{OSF}}$ is obtained using $[H'_4]^{\text{OSF}}$ if the ear canal transfer function is measured in advance. The ear canal transfer function is independent of the source direction [4]. Thus, the function measured for a sound source placed in a certain direction is valid for calculating $[H'_3]^{\text{OSF}}$ for any source location.

On the basis of this idea and Eqs. (4) to (9), $[H_3]^{\text{OSF}}$ in Eq. (2) is approximated as follows:

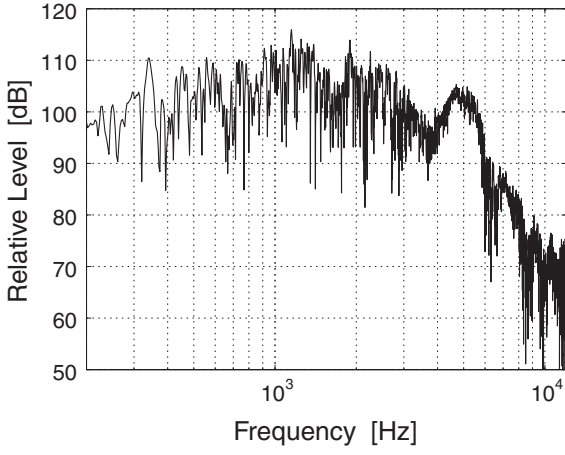
$$\begin{aligned} [\hat{H}_3]^{\text{OSF}} &= [H_3]^{\text{ESF}} + [\hat{H}_3]^{\text{OSF}}_{\text{reflect}} \\ &= [H_3]^{\text{ESF}} + [H'_3]^{\text{OSF}} - [H'_3]^{\text{OSF}}_{\text{direct}} \\ &= [H_3]^{\text{ESF}} + \frac{[H'_4]^{\text{OSF}} - [H'_4]^{\text{OSF}}_{\text{direct}}}{H'_e}. \end{aligned} \quad (10)$$

The constraint that a listener must visit the OSF once to measure the transfer function is removed if this approximation is sufficiently accurate. In this study, we propose the replacement of the transfer function $[H_3]^{\text{OSF}}$ in Eq. (1) by $[\hat{H}_3]^{\text{OSF}}$ in Eq. (10) so that the equalization function becomes the following:

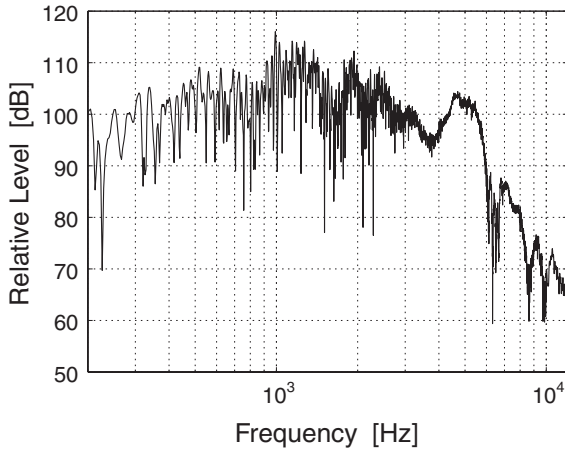
$$G = \frac{M_2 \cdot [\hat{H}_3]^{\text{OSF}}}{M_1 \cdot [H'_4]^{\text{OSF}}} \cdot \frac{E_{\text{hp}}}{E_{\text{mic}}}. \quad (11)$$

2.3.2. Results of approximation of $[\hat{H}_3]^{\text{OSF}}$

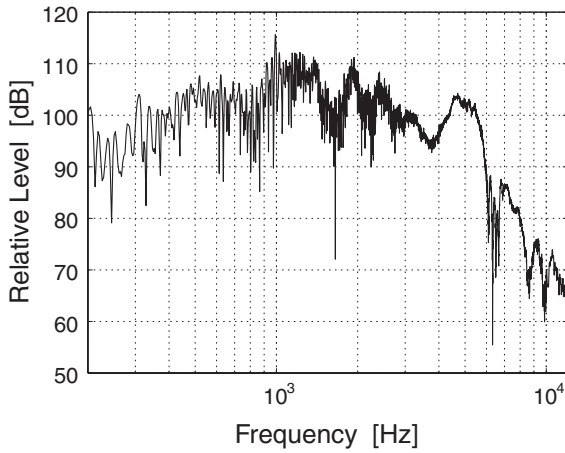
Figure 5 shows an example of a frequency characteristic of $[\hat{H}_3]^{\text{OSF}}$ approximated by adding the reflected sound components to the impulse response in Fig. 2(b). The ear transfer function H_e was measured for a sound source placed at 0° in the same soundproof chamber as the ESF (reverberation time: 43 ms). Panels (a) to (c) in the figure respectively show the results for the orders of poles/zeros of the ARMA model of 100/100, 500/100, and 1,000/1,000 used to extract the direct sound component in Eq. (6). When the orders are higher, the extracted sound component



(a) Orders of the ARMA model: 100/100



(b) Orders of the ARMA model: 500/500



(c) Orders of the ARMA model: 1,000/1,000

Fig. 5 Frequency characteristics of $[\hat{H}_3]^{\text{OSF}}$ for each order of ARMA model (subject A, the right ear).

corresponding to the beginning of the impulse response becomes longer (as shown in Fig. 4) and the numbers of peaks and dips corresponding to the reflected sound components decrease (as shown in Fig. 5). Therefore, it is necessary to set the orders appropriately as the spectrum becomes closer to that shown in Fig. 3(a).

3. EVALUATION EXPERIMENT

3.1. Outline of the Experiment

In order to evaluate the validity of the proposed method, an experiment using the binaural technique was conducted to simulate the situation of listening to a sound radiated from a loudspeaker.

Each subject was asked to compare a standard sound that was radiated from a loudspeaker in the OSF with a test sound presented via headphones; then the subject evaluated the test sound with respect to four criteria: “naturalness (1 to 5),” “blurring of sound image (−5 to 5),” “sound quality (−5 to 5),” and “reverberation (−5 to 5).” The numbers in parentheses represent the scores that could be assigned by the subject.

The instructions to the subjects were as follows.

- Two stimuli will be presented in the order of a standard sound followed by a test sound.
- Evaluate the “naturalness” of the test sound compared with the standard sound (whose value is 5).
- Evaluate the “blurring of sound image,” “sound quality,” and “reverberation” of the test sound. A negative value means that the test sound is worse and a positive value means that it is better.

After evaluation, the subject reported the perceived position of the test sound. A diagram of a circle, whose center was the head of the subject and whose radius indicated the distance between the subject and the loudspeaker, was displayed on the screen of a computer. The subjects were asked to point and click the position on the diagram corresponding to the sound image with a mouse. The sound source was a tune that was exempt from copyright, “Aura Lee” (RWC-MDB-J-2001 No. 37, Aoi Kato), with a duration of 15 s. Six subjects (three males and three females) with normal hearing acuity participated in the experiment. They were asked to wear open-type headphones during the experiment, even when listening to the standard sound. According to our measurement, the sound leakage from the open-type headphones attenuated at least 35 dB at a distance of 1 m when the sound was presented from open-type headphones. Therefore, there is little effect of sound leakage because the amplitude of the sound leakage is very small compared with the test sounds. The inverse HpTF [3,4], which is the second term in Eq. (11), was measured for each subject to compensate for individual differences.

3.2. Test Sounds

Ten test sounds were prepared for each subject as shown in Table 1. Test sound 1 was the ideal condition; the sound was recorded at the entrance of the subject’s ear canal in the OSF. In this case, no individual equalization was required because the numerator and denominator of

Table 1 Stimuli used in the experiment.

| No. | recorded with | the 1st term (Eq. (1) or Eq. (11)) | |
|-----|---------------|---|--------------------|
| | | $[\hat{H}_3]^{OSF}$ | orders of ARMA* |
| 1 | subject | not required | |
| 2 | DH** | $[H_3]^{OSF}$ | without estimation |
| 3 | DH** | $[H_3]^{ESF}$ | without estimation |
| 4 | DH** | $[H_3]^{ESF} + [H'_3]^{OSF}_{reflect}$ | 100/100 |
| 5 | DH** | $[H_3]^{ESF} + [H'_3]^{OSF}_{reflect}$ | 500/500 |
| 6 | DH** | $[H_3]^{ESF} + [H'_3]^{OSF}_{reflect}$ | 1,000/1,000 |
| 7 | DH** | $[H_3]^{ESF} + [H'_4]^{OSF}_{reflect}/H'_e$ | 100/100 |
| 8 | DH** | $[H_3]^{ESF} + [H'_4]^{OSF}_{reflect}/H'_e$ | 500/500 |
| 9 | DH** | $[H_3]^{ESF} + [H'_4]^{OSF}_{reflect}/H'_e$ | 1,000/1,000 |
| 10 | DH** | without compensation | |

*: Orders of the ARMA model in Eq. (6), **: dummy head.

the first term of the equalization function canceled out. Other test sounds were prepared to evaluate the effects of equalization on the sounds recorded at a dummy head's eardrums in the OSF. Test sound 2 was the high-fidelity (hi-fi) condition, i.e., the sound was equalized with the transfer function measured in the OSF for the subject in accordance with to Eq. (1). Test sound 3 was equalized with the function measured in the ESF, that is, only individual differences were equalized without respect to the characteristics of the sound field. Test sounds 4 to 6 were equalized on the basis of Eq. (11) but with the measured $[H'_3]^{OSF}$ to evaluate the effect of the ear canal transfer function used in the proposed method. Test sounds 7 to 9 were equalized using the proposed method with the measured $[H'_4]^{OSF}$ and H'_e . The differences among test sounds 4 to 6 and 7 to 9 were the orders of poles and zeros in the ARMA model used to estimate the reflected sound components $[H'_3]^{OSF}_{reflect}$ in Eq. (8). Test sound 10 was produced without any individual equalization. Each test sound was evaluated five times in random order.

3.3. Results and Discussion

3.3.1. Results of “naturalness”

The results averaged for all subjects with respect to “naturalness” are shown in Fig. 6. The ordinate is the mean and the error bar denotes the standard deviation of the 30 (6 subjects \times 5 times) evaluations. The thick horizontal lines show the conditions under which the obtained evaluation for the test sound is not significantly different ($p < 0.05$, Wilcoxon rank-sum test) from the evaluation of the ideal condition, test sound 1. For example, in Fig. 6, sounds 2 and 4 to 8 are not significantly different from sound 1.

The difference between sounds 1 and 2 is not statistically significant. Therefore, the equalization with the subject's own transfer function is effective [5–7]. As shown in Fig. 6, the score for test sound 3 is extremely low,

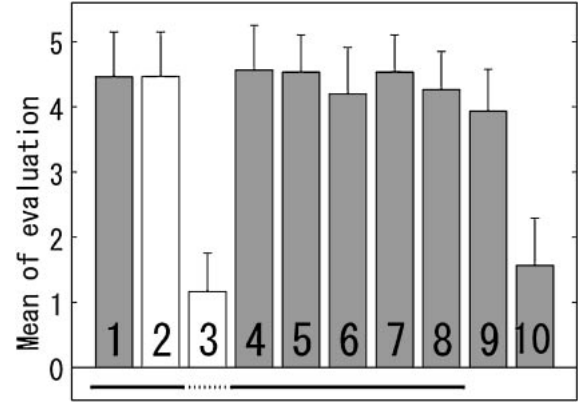


Fig. 6 Experimental results of all subjects for “naturalness”; the thick horizontal lines show the conditions under which the obtained evaluation for the test sound is not significantly different ($p < 0.05$, Wilcoxon rank-sum test) from the evaluation of the ideal condition, test sound 1.

which indicates that individual equalization using the transfer function measured in a sound field other than the OSF is not effective. The score for test sound 10 is also low. Therefore, the results confirm that the first term of the equalization function is important.

The tendency of the results for test sounds 7 to 9 using the proposed method is almost the same as that for test sounds 4 to 6, which indicates that using the ear canal transfer function is effective for estimating $[H'_3]^{OSF}$ from $[H'_4]^{OSF}$. In addition, test sound 7 exhibits the maximum score. This suggests that, when the orders of poles/zeros were set at 100/100 to estimate the reflected sound, a satisfactory effect resulting from the proposed equalization is expected.

3.3.2. Results for criteria other than “naturalness”

The results averaged for all subjects with respect to other criteria are shown in Fig. 7. The thick horizontal lines have the same meaning as those in Fig. 6. Test sounds 2, 4 to 6, and 8 are not significantly different from sound 1. Although there is a significant difference for test sound 7 in the result for “blurring,” this is not problematic because the score is almost zero, that is, the sound is evaluated to be equivalent to the standard sound radiated from the loud-speaker. The results show that the evaluations for test sounds 7 to 9 using the proposed method are almost the same as those for test sounds 4 to 6.

From the results, it is concluded that the sounds are accurately reproduced with respect to sound quality, blurring and reverberation as well as naturalness using the proposed method.

3.3.3. Results for localization

Figure 8 shows the results of localization for each test sound. In this figure, the origin indicated by a square is the center of the subject's head, and the cross is the position of

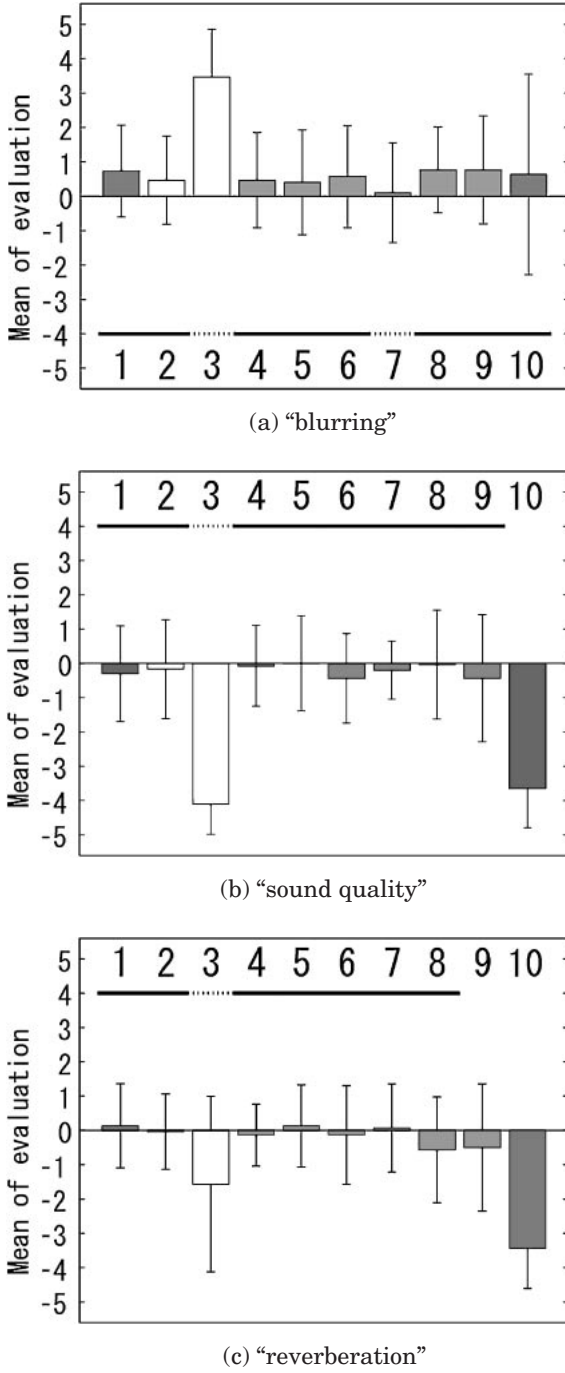


Fig. 7 Experimental results of all subjects for “blurring,” “sound quality,” and “reverberation”; the thick horizontal lines show the conditions under which the obtained evaluation for the test sound is not significantly different ($p < 0.05$, Wilcoxon rank-sum test) from the evaluation of the ideal condition, test sound 1.

a loudspeaker at $(1.5/\sqrt{2}, 1.5/\sqrt{2})$. Each circle in this figure is plotted so that its area is proportional to the number of subject responses. For test sounds 1 and 2, almost all responses are localized around the loudspeaker. For test sound 3, some of the localized points are inside or near the heads. From the results, it is indicated that suitable equalization is important for localization. For test sound 7,

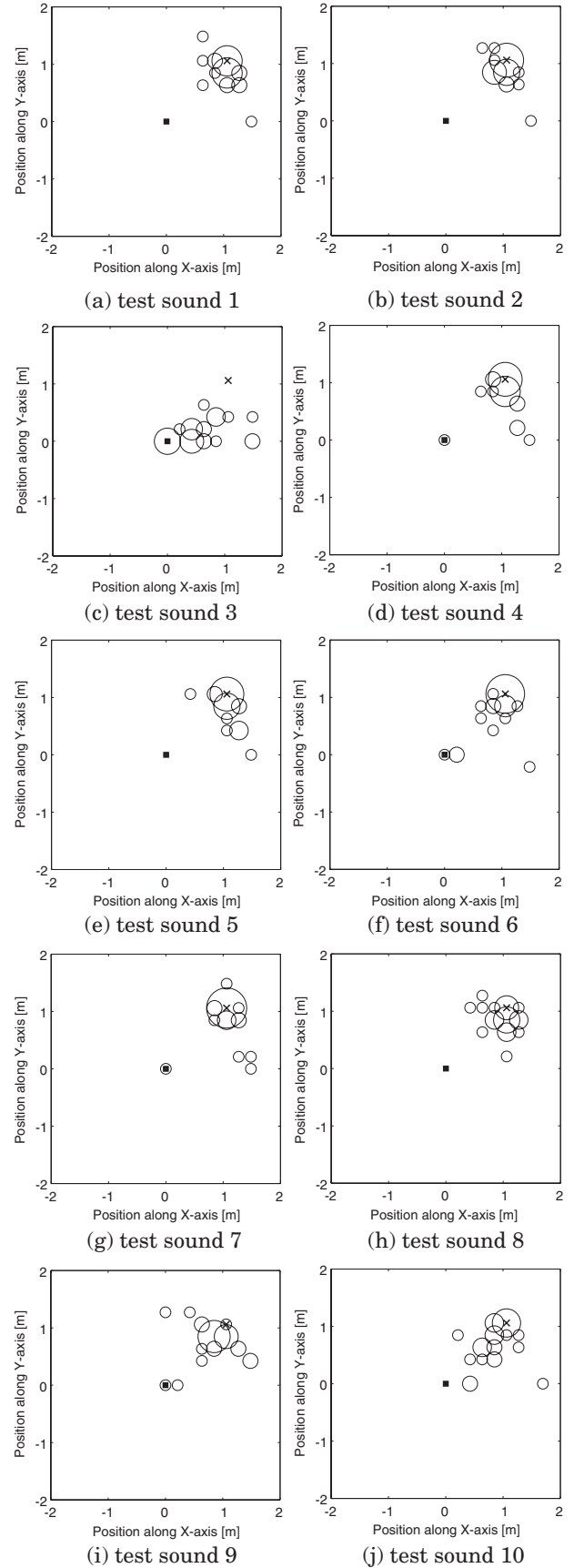


Fig. 8 Positions of sound localization.

which was related highly for all criteria, most of the subject responses are around the loudspeaker. This indicates that the proposed method reproduces the OSF in terms of localization.

3.3.4. General discussion

The results are summarized as follows: the approximation of the transfer function in the OSF by adding the reflected sound components in the OSF to the transfer function measured in the ESF is effective for individual equalization. Therefore, the constraint that the transfer function in the OSF must be measured for each listener is overcome by the proposed method.

In this paper, the experiment was carried out under limited conditions. Although the validity of the proposed method should be evaluated for all directions, the experiment was carried out for only one source direction (45°) as a typical direction including both interaural and spectral cues as a basic evaluation. The OSF addressed in this study was restricted to a meeting room. Thus, it is indicated that the proposed method is effective for a speech or a one-to-one meeting in a general room.

Investigations of equalization for the cases of an OSF with strong reverberation, such as in a concert hall, and an OSF with multiple sound sources are future works.

4. CONCLUSION

In this study, an individual equalization method for the binaural technique is proposed with respect to a reverberant sound field. In the method, the reflected sound components in an OSF are added to the transfer function measured in an ESF to approximate the transfer function in the OSF. The results of an experiment show that the equalization function using the proposed method is effective. The combination of this method with that proposed by Ozawa *et al.* [8], equalization for a dead sound field, overcomes the constraint that the transfer function of the listener must be measured in the OSF.

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