Hypotheses for constructing a precise, straightforward, robust and versatile sound field reproduction system

Omoto, Akira
Faculty of Design, Kyushu University

Kashiwazaki, Hiroshi
Graduate School of Design, Kyushu University

INVITED REVIEW

Hypotheses for constructing a precise, straightforward, robust and versatile sound field reproduction system

Akira Omoto[1,*] and Hiroshi Kashiwazaki[2,†]

[1]*Faculty of Design, Kyushu University,*
4–9–1 Shiobaru, Minami-ku, Fukuoka, 815–8540 Japan
[2]*Graduate School of Design, Kyushu University,*
4–9–1 Shiobaru, Minami-ku, Fukuoka, 815–8540 Japan

Abstract: The three dimensional sound field reproduction systems can be categorized mainly into two types, physical reproduction, and artistic reproduction. The former is sometimes referred to as scientific or engineering, and the latter is sometimes recognized as psychological reproduction using phantom images produced by, for example, amplitude panning and the other effects. The purpose of the reproduction system is widely spread. The system can be a design tool of enclosed space, such as a concert hall, before practical construction by reproducing physical characteristics accurately. Also, the system can be a pure entertainment tool, mostly with visual images. Of course, the scale and necessary conditions vary with their purpose and objectives; however, it might be interesting to investigate what are the essential factors for the higher total performance of reproduction systems. We currently hypothesize that the following four conditions might be necessary for the total performance of the versatile sound field reproduction system. A) physical accuracy, B) robustness against disturbance, C) flexibility for additional direction, D) capability of integration with visual stimuli. As a platform of examination, 24-channel narrow directional microphone array and 24-channel loudspeaker array are used. The boundary surface control principle and its modified version are adopted for the physical background. As examples, several practical efforts are attempted to assure the total performance of the system effectively.

Keywords: Sound field reproduction, Boundary surface control principle, Inverse filter matrix, Narrow directivity microphone

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1. INTRODUCTION

The sound field reproduction systems seem to have roughly two principles; reproducing characteristics of certain target sound field physically or creating a desired sound field by mostly artistic technique. These aspects can also be called as engineering and art, respectively. Figure 1 shows typical methods, in which the top row includes several physical methods and left column includes aesthetic methods. In the former category, the typical methods using plural loudspeakers are the Ambisonics [1–4] including higher-order version [5–11], Wave Field Synthesis [12–16], and the Boundary Surface Control [17–21].

Also, as shown in the bottom left of the figure, there are several attempts to integrate both methods. For example, the series of research for SIRR (Spatial Impulse Response

Rendering) [22,23], DirAC (Directional Audio Coding) [24,25], and SDM (Spatial Decomposition Method) [26] measure the impulse responses and analyze the crucial parameters such as location and the timing of dominant reflections and the amount of reverberation. These methods synthesize the responses to reproduce the perceptual cues of the primary sound field. The focus of the reproduction is therefore not on the accurate reproduction of the wave quantity but on the perceptually or psychologically essential items such as inter-aural time and level differences, inter-aural coherence, monaural localization cue, and the timbre. These are powerful tools, especially for the reproduction of concert halls.

The authors had attempted to improve the performance of immersive sound field reproduction system based on the physical principle. Examples were the Sound Cask [20,21] which used fullerene shaped 80-channel microphone array and 96-channel loudspeakers arranged in a nonagonal shaped small enclosure. The Boundary Surface Control

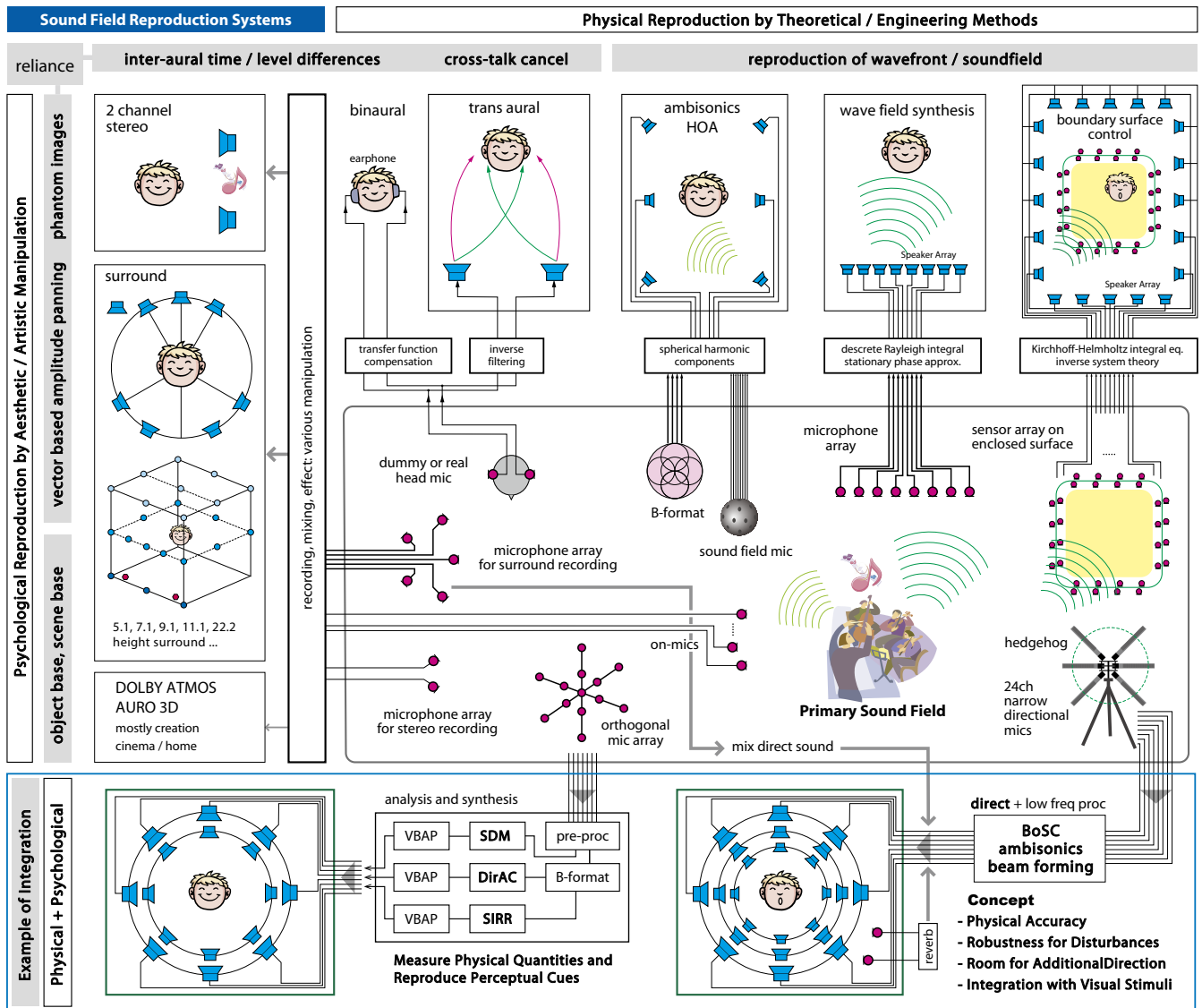*e-mail: omoto@design.kyushu-u.ac.jp
†e-mail: hrs@penr.in

Fig. 1 Schematic diagram of the various sound field reproduction system, in which the top row indicates the physical method and the left column indicates the psychological methods. The bottom row indicates the attempt to integrate both methods. The system introduced in this manuscript is shown in the bottom right. Adopted and modified from A. Omoto, "A note on sound field reproduction," *Journal of the Acoustical Society of Japan*, **67**, 520–525 (2011).

principle (BoSC, hereafter) developed by Ise [17], was used in these systems as the physical backbone.

During the series of examination, as well as precise subjective performance evaluation of the systems, many demonstrations were conducted to many ordinary peoples. The number of participants was more than few hundreds per day in most cases. Due to limited time and space, several peoples sometimes heard the demonstration simultaneously. This situation was unexpected for the system, and inevitable and additive disturbances would occur in the reproduction. However, the reactions from ordinary peoples were mostly positive. We think the reasons as follows;

– the sounds were coming from roughly correct directions, or the movements of the sound source can be recognized,

– reproduced sounds were high fidelity,
– hearing environment, surrounded by a large number of loudspeakers, was unusual and amazing.

Besides, the professionals such as recording or mixing engineer and acoustic designer, who usually create sound field mostly by aesthetic sense, also experienced such demonstration. In most cases, while recognizing its uniqueness and potentials, they complained about the insufficient low-frequency sound, the lack of force of sound, inaccurate phantom images, and the other causes.

The original development purpose of the sound field reproducing system is to reproduce any sound field correctly in physical meanings. This achievement could be measured by the reproduction degree of several physical parameters or wavefront itself.

However, considering various suggestions from ordinary peoples and professionals, to improve the total performance of the system, there might be the other factors in addition to the physical indexes that would be necessary when sound reproduction is carried out.

Another exciting proposal provided through a demonstration is a demand to use a sound field reproducing system for the extensive range application, such as noise control or purely entertainment, and the others.

Currently, we assume that the following four items are valid to realize the versatile and high-performance reproduction system. These hypotheses aim the integration of physical and psychological reproduction strategies. The conceptual diagram of the system introduced in this manuscript is shown in the bottom row, right, of Fig. 1.

A) **Physical accuracy of reproduction**

As an essential backbone of reproduction, some physical principle should be used to obtain directional information of sound. The boundary surface control principle is adopted in our examination.

B) **Robustness against disturbance**

Robustness against unexpected disturbance would be necessary. As mentioned above, the existence of plural listeners and their movement are the typical causes.

C) **Flexibility for additional direction**

With a broad interpretation of the purpose of the sound field reproducing system, flexibility for additional direction might be necessary. Examples of the additional direction are the mixing with the sound recorded by other technique such as conventional stereo recording and additional reverberation at reproducing sound field. These examples are shown in the diagram.

D) **Capability of integration with visual stimuli**

Balancing the above conditions A) and B) is generally challenging, due to inevitable disturbances such as the existence of listeners. For example, many systems are often designed to reproduce the wave information at the point or region without considering the movement of the listeners. The other modality, visual stimuli might be effective to compensate inevitable degradation of the physical performance. The 360-degree cylindrical moving image is currently used with the reproduction system, and the effect is examined.

As mentioned above, the four hypotheses include the content that is contradictory or complementary with each other. Therefore, it is thought that the sophisticated and bold action from various points of view is necessary. These conditions are hypotheses, and this manuscript introduces the example which we are inspecting currently. Scientific conclusive evidence has not been necessarily provided.

Fig. 2 Systems used in the examination, including the hedgehog shaped microphone array and the loudspeaker array called G-VRAWS.

2. PRACTICAL STRATEGIES

2.1. Platform of Examination

As a platform of examination, the system shown in Fig. 2 is used. In this system, a 24-channel hedgehog-shaped narrow directional microphone array is used as a sound acquiring device. As a sound-emitting device, the 24-channel loudspeakers called G-VRAWS are arranged at every $45°$ in azimuth and three layers in height. The term VRAWS is an abbreviation of Variable Reflection Acoustic Wall System [27,28], in which the additional reverberation can be added in the reproducing (secondary) sound field. This function is often used as an example of additional direction, which emphasizes the reverberation.

In the system, the BoSC principle is often used as a physical reproduction method. In this case, the total 576 $(= 24 \times 24)$ inverse filters are necessary to compensate for the amplitude and phase distortions in the reproducing field. The minimum distance between adjacent microphones are around $0.2 \, \mathrm{m}$, and therefore, the critical frequency for spatial aliasing is around $850 \, \mathrm{Hz}$. The precise reproduction at higher than this frequency is hardly expected.

Also, the other simple methods such as direct-reproduction in which the recorded signal at one narrow directional microphone is emitted from the loudspeaker at the almost corresponding direction are used. This direct reproduction seems to be a natural extension of six-channel reproduction method [29].

In G-VRAWS with BoSC principle, physically precise reproduction is hardly expected at high frequency, due to insufficient numbers of microphones and loudspeakers, and therefore the large distances between adjacent sensors. However, the system, using the proven equipment in music recording and monitoring in the studio, might have potentials with various bold efforts for higher total performance. Example in this manuscript was performed in G-VRAWS.

2.2. Strategies Assumed in Our Examination

Several strategies used in our current examination are shown in this section. These are again suggestions and not necessarily confirmed in a scientific manner. We hope these would be the chance for future discussions.

a) Efficient Inverse Filter

In the system using the BoSC principle, the inverse filter matrix is usually necessary to compensate phase and amplitude distortion in the reproducing field. Several methods, such as using singular value decomposition and its variation, introducing and some optimization of Tikhonov's regularization parameter [30], etc., are known for the calculation. The characteristics of the obtained filter directly affect the quality of reproducing sound and therefore, the physical accuracy of the reproduction, A) above.

Currently, the highest-rated sound by subjective evaluation is obtained with the filter that was calculated with a heuristically determined regularization parameter in our reproducing system. More systematic and efficient calculation method, i.e., using error criteria due to the existence of the listener [31], introducing mixed norm [32], using genetic algorithm [33] are the current research targets.

One of the recent research trends is a method of introducing sparsity into a transfer matrix to realize a sharp sound image localization; see, for example, [34,35]. Too sparse structure, however, results in abrupt changes of the spatial sound image. Therefore, several efforts, such as mixed norm penalties, are examined. We expect that our microphone array having narrow directional characteristic provide moderate sparsity in the transfer matrix [36].

b) Reduced Scale Inverse Filter

Reducing the scale of the inverse filter is attempted, although this item might be a disagreeing with a) mentioned above. Introducing narrow directional microphone array limits the number of loudspeakers that affect one microphone dominantly due to its directivity. The scale of the inverse filter matrix can thus be reduced reasonably.

We have tried to reduce the number of the inverse filter from 576 ($= 24 \times 24$, full size) to 104, and confirmed the performance degradation in reproducing wavefront was not fatal [36].

We now assume the advantages of reduced scale filter as follows;

– The fewer calculation cost enables us to perform the inverse filtering in real-time, for example, by using plug-in software of sampling reverberator.
– Reduced scale filter reproduce the minimum directional information in the sound field. Although the reproduced field might not be so much accurate, we expect the robustness against additional disturbance such as the existence of listener.

c) Simple Compensation Filter

A more simplified filter is attempted. In this case, the combination of one microphone and one loudspeaker located at almost corresponding direction is determined, and the amplitude and the phase alignment is carried out by 24 independent filters. As mentioned above, no sweet spot is expected by introducing such a simple filter. However, moderate directional information might be reproduced.

d) Using Directional Impulse Responses

If the reproduction of reverberation at the particular sound field is desired, the convolutions of directional impulse responses, that are measured for example by hedgehog microphone, and the dry source, might be useful. These are sometimes referred to as spatial impulse responses [26]. The combination, dry sources recorded at several directions which can reproduce the directional characteristics of musical instruments and the directional impulse responses which reflects the directional characteristics of the sound field, are effective [37,38].

e) Mixing with Conventionally Recorded Sound

In the case of music contents recorded in a concert hall, for example, mixing with separately recorded sound in front direction is often desired in addition to the normal reproduction. This method is no longer the physically assured sound field reproduction, but an additional direction for further pleasure in the listening experience. Also, this additional direction might be useful to improve the robustness or to compensate for the insufficient performance of the system.

f) Additional Reverberation in Reproducing Field

As an application of VRAWS for sound field reproduction system, artificially produced reverberation can be added in the reproducing environment. The VRAWS provide listeners experiences as if they are in a reverberant environment such as a concert hall. This function is also categorized as an additional direction to reproduce the atmosphere of the reverberant environment effectively. In our case, several microphones are arranged in the reproducing field, and loudspeakers of the system reproduce the picked-up signal with additional reverberation generated by reverb effectors.

g) Integration with Visual Information

The effect of visual information, which is recorded simultaneously with the audio signal, is currently examined. Cylindrical movie recorded by 360° panorama camera or by several cameras and stitched is projected on a cylindrical screen that surrounds listeners. The reproducing loudspeakers are located behind the screen. Again, compensation of insufficient performance of sound reproduction is expected. An example is shown in Fig. 3.

The relationship between four hypothesis and practical efforts is shown in Fig. 4.

Fig. 3 Trial of integration with visual stimuli, in which the cylindrical screen was used for 360° movies. The 24-channel G-VRAWS was set behind the screen.
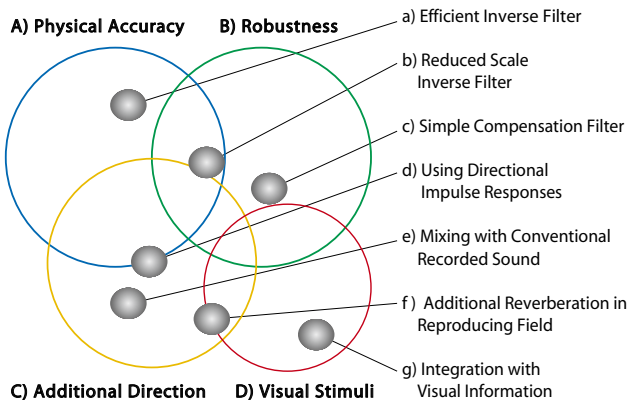

Fig. 4 Relationship between four hypothesis and the practical efforts.

Fig. 5 Block diagram of experiment, in which 24 simple compensation filters were obtained in the process of system identification. The optimum Wiener filters were calculated in this case.

3. EXAMPLE OF PRACTICAL STRATEGIES

3.1. Conditions

Using G-VRAWS and hedgehog microphone, several examinations were carried out. As examples of strategies mentioned above, the simple compensation filter (1to1 Filter, hereafter), the direct reproduction (Direct), and full-size inverse filtering (inv576) have been attempted. The block diagram of the experiment is shown in Fig. 5. The cylindrical screen of common white cloth was located just in front of G-VRAWS loudspeakers in the reproducing field. The level reduction, especially at high frequency, is expected.

The 1to1 Filter is calculated by system identification process shown in the top diagram in the secondary sound field in Fig. 5. The test signal $x(n)$ was inserted to i-th loudspeaker and the impulse response was estimated with an output signal of i-th microphone that was at the same azimuth angle and same layer (top, middle, or bottom) as the loudspeaker.
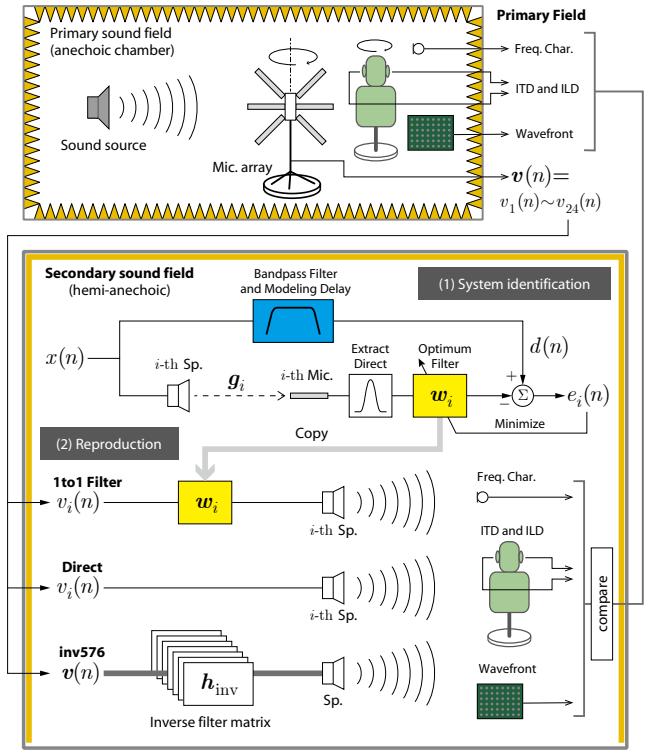
The direct sound was then extracted from the impulse response and passed through the compensation filter. This signal was compared with a reference signal which was generated by passing through the smooth bandpass filter such as 50 Hz to 20,000 Hz. The optimum filter was then obtained by the conventional Wiener optimal solution.

The Direct reproduction emit the recorded sound from the loudspeaker located at the almost corresponding direction, i.e., the same azimuth angle and the same layer.

The full size inverse filter, inv576, uses 576 (= 24 × 24) filters calculated with heuristically optimized regularization parameters.

The frequency characteristics were measured by a one-point omnidirectional microphone, the inter-aural time and level differences (ITD, ILD) were measured by dummy head, and the wavefront was measured by using MEMS microphone array [36], all in the anechoic primary field and hemi-anechoic secondary field, respectively.

3.2. Results

Frequency Characteristics Figure 6 shows the results of the frequency characteristics. Almost flat characteristics are obtained in the range of 60 Hz to 10,000 Hz in the primary field. The inv576 showed closest characteristics to the primary field and the 1to1 Filter gave the better results compared with Direct method. The steeper decreasing
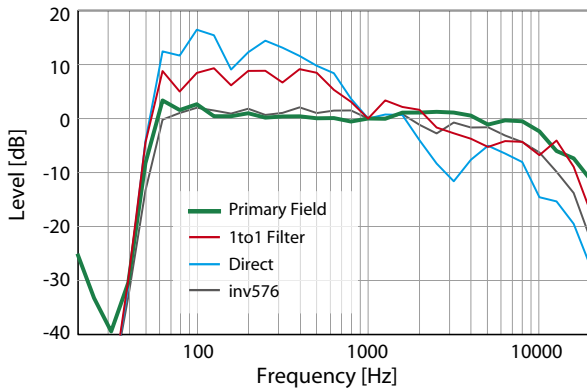
Fig. 6 Frequency characteristics of primary sound and reproduced sound, in which all levels at 1 kHz were normalized to 0 dB.
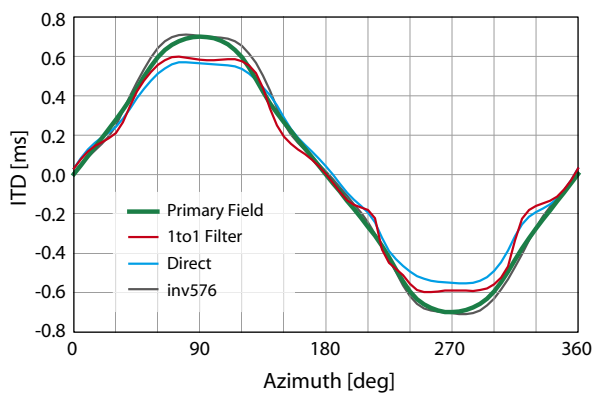

Fig. 7 Interaural time difference in primary and reproduced field, in which the cross correlation was calculated for impulse responses at both ear positions after low-pass filtered at 1.6 kHz and up-sampled.

characteristics in Direct reproduction compared with the primary field was due to the existence of the screen immediately in front of the loudspeakers.

Interaural Time and Level Differences Figures 7 and 8 show the results of the ITD and ILD, respectively. These results were obtained by the measurement with dummy head put on the turn-table which was rotated 5° step in the primary and secondary fields. In the calculation of ITD, the cross-correlation was calculated with impulse responses measured at both ear positions after low-pass filtered at 1.6 kHz and up-sampled. Also, in the calculation of ILD, window functions of frequency-dependent length were applied to the responses to remove the inevitable reflections from the floor and screens.

All the methods reproduced the characteristics of ITD in the primary field quite well. Notably, the inv576 showed the closest results. In the results of ILD at 250 Hz, all methods showed similar results. However, at 500 Hz and 1,000 Hz, the inv576 traced the shape of the results of the primary field well. At higher frequencies such as 4 kHz and
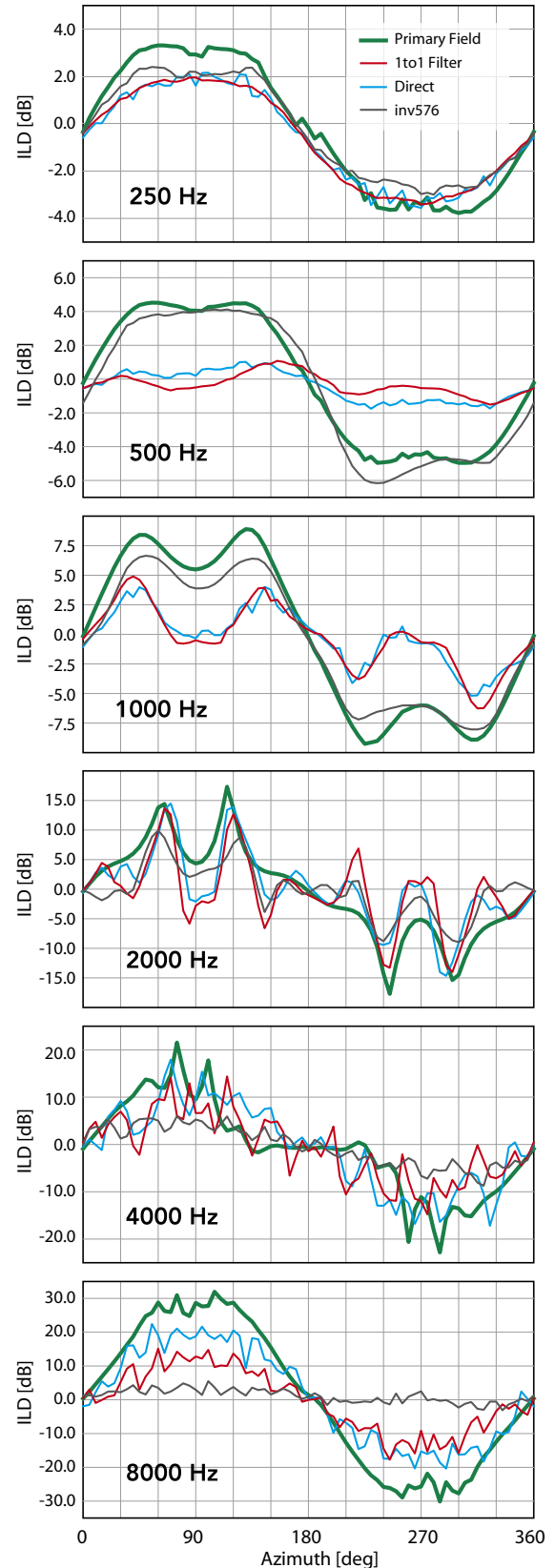

Fig. 8 Interaural level difference in primary and reproduced field calculated from 250 Hz to 8,000 Hz 1/1 octave bands.
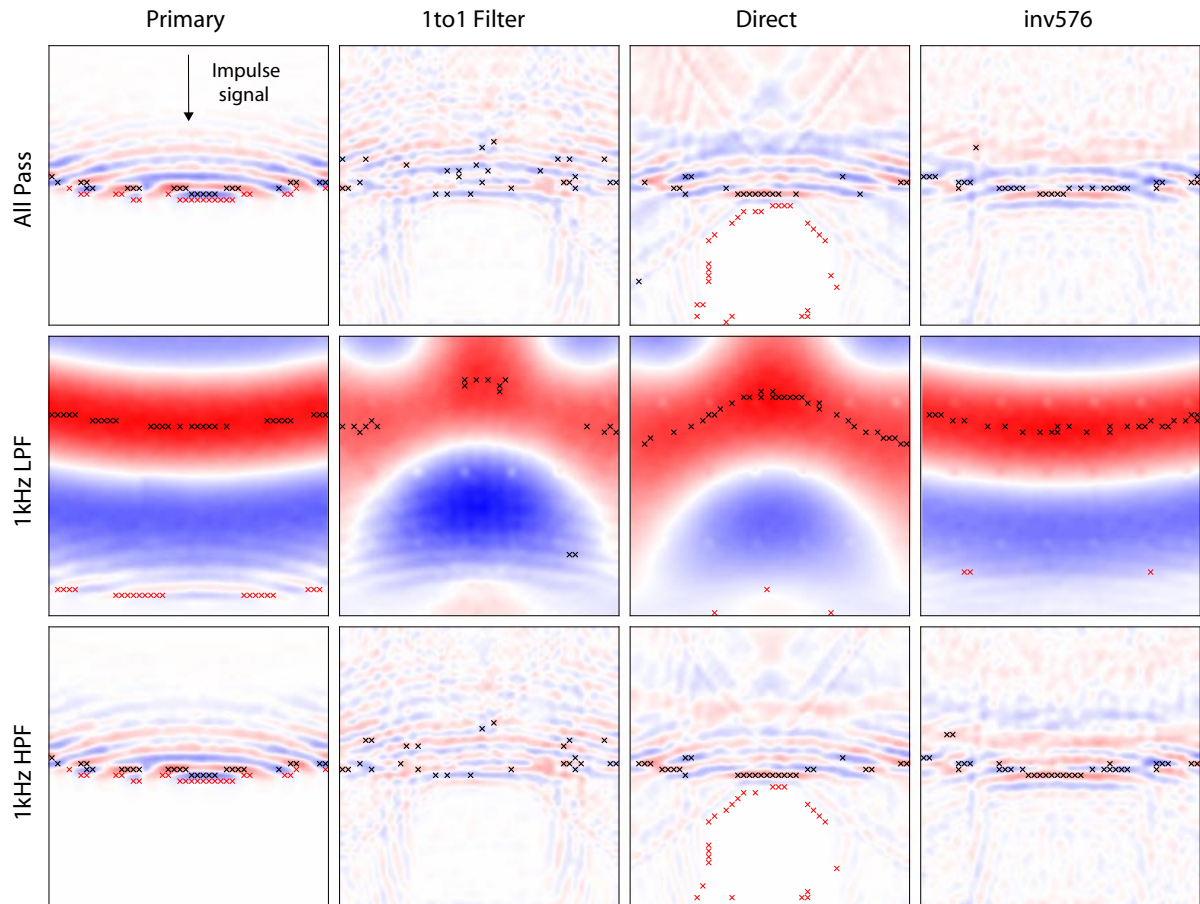
Fig. 9 Snapshots of wavefronts measured in the primary field and the reproduced field with 1to1 Filter, Direct and inv576. The sound source was located at 2 m away from the center of the observation area. The upper row is the case of all passed frequency range, the middle and the lower are the cases of low-passed and the high-passed, respectively, and the cut-off frequency was set to 1 kHz in this case. The red x-marks in the figures corresponds to the first peaks in the waveform, and the black x-mark indicates the peak positions with maximum amplitude in the waveforms.

8 kHz, the methods of 1 to 1 Filter and Direct showed better results than inv576, even though the shapes were somewhat rough.

Wavefront By using the microphone array of square grid shape, the propagation of wavefront from the loudspeaker located 2 m away was measured at 0.72 m × 0.72 m area. The snapshot, when the impulse wave was at almost center of the measured area, was captured. Figure 9 indicted for the primary field, and in the secondary field with each reproduction method.

Several disturbances can be observed before the main wavefront, in all the reproduction methods and the all-passed frequency range. They are due to the filters applied to the reproducing signals in the cases of 1to1 Filter and inv576. The characteristic octagon shape of red x-mark, which corresponds to the loudspeaker location at every 45°, can be seen in the case of Direct. Unexpectedly measured impulse signals generated this first wavefront at every microphone, i.e., if the directional characteristics were narrow enough, no such wavefront could be generated.

In the case of low-passed (below 1 kHz) frequency range, inv576, which was based on the BoSC principle reproduced the wavefront. In contrast, the cases 1to1 Filter and Direct showed relatively large distortions in the shape of the wavefront.

The octagon-shaped first wavefront was again observed in the case of Direct method in the high-frequency range. However, the total shape of the wavefront was similar to the primary field.

Further examination such as robustness against obstacle would be necessary for further discussion.

4. CONCLUDING REMARKS

The basic idea to improve the total performance of the versatile sound field reproduction system is proposed. Here we assume that the key points are A) physical accuracy, B) robustness, C) flexibility for additional direction and D) capability of integration with visual stimuli. Several practical strategies have also been proposed to achieve these conditions.

As an example of our recent trials, the application of boundary surface control principle to the system, which consists of the narrow directional microphone array and the same numbers of loudspeakers, have been attempted. Also, simple direct reproduction method and characteristics compensation filter has been tried.

The results indicated the superiority of physically assured method, BoSC in this case, in the low-frequency range. Almost comparable performance of simple methods in higher frequency range has also been confirmed. As attempted in our previous work [36], the hybrid scheme that uses wave-based reproduction in the low-frequency range, and uses narrow directional characteristics of the microphone in the high-frequency range, might be useful.

In any case, we believe that the integration of physical reproduction technique and aesthetic operation for psychological reproduction and effective direction is indispensable to realize four hypotheses for improving the total performance of the system thoroughly. Further examinations are continuously carried out for the realization of a system that can provide correct and pleasant sound.

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Akira Omoto   graduated from the Department of Acoustic Design, Kyushu Institute of Design, in 1987 and received his Ph.D. degree from the University of Tokyo in 1995. From 1987 to 1991, he worked as a research and development engineer at Nittobo Acoustic Engineering Co., Ltd., Tokyo. In 1991, he was appointed as a Research Assistant at Kyushu Institute of Design and was made an Associate Professor in 1997. After integration of universities in 2003, he was made an Associate Professor of Kyushu University and was made full professor in 2014. His research interest includes measurement, evaluation, and control of sound field.

Hiroshi Kashiwazaki   was born in 1994 and graduated from the Department of Acoustic Design, Kyushu University in 2017. He is currently a student of Ph.D. degree program, in Communication Design Science Course, Graduate School of Design, Kyushu University.