Multichannel Compatible Two-Layer Speaker Array System

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Abstract**—**In this paper, we introduce a multichannel compatible two-layer speaker array system that delivers immersive sound. Owing to the appearance of UHD-TV and 3DTV, 3D rendering of sound is now required these days to keep pace with the immersive visual experience of such systems. However, while traditional multichannel loudspeaker formats such as 5.1-ch or 7.1-ch provide reasonable surround sound quality, such systems require a secluded space and are a challenge to install for laypersons. Furthermore, if each speaker is not properly placed, the original sound image will be distorted. Therefore, in this paper, we introduce a two-layer array speaker system that renders an immersive audio effect and is easy to install by placing it around the user's TV set. We introduce the system layout of the two-layer array speaker system and the simulation results of the sound reflection. In addition, the signal flow of our proposed system is described. Unlike conventional speaker configurations, which are only available for their own formats, our proposed two-layer speaker system is compatible with all formats used up to 22.2-ch, as well as all stereo types, including 5.1-ch, 7.1-ch, 9.1-ch, 10.1-ch.

Keywords-*3D sound; array loudspeaker; WFS*

I. INTRODUCTION

In a home theater system, easy use and compatibility are two of the most crucial issues for users. However, a discrete traditional multichannel loudspeaker is difficult to install and requires meticulous fine-tuning to create the proper surround sound image. To solve this problem, we developed a two-layer loudspeaker array system that is able to reproduce a multichannel sound format and virtual sound source. If this system is developed in conjunction with an automatic setup procedure, lay persons will be far more comfortable and less challenged in setting up and enjoying this surround sound system.

The proposed system is technically based on a Wave Field Synthesis (WFS) technique. The original technique was proposed during the late 1980s [1]. The basic physical concept of WFS is based on Huygens' principle. This basic physical concept can be analyzed using a Kirchhoff-Helmholtz integral [2]. Although it is impossible to realize a continuous acoustic source in real life, this theoretical background is still partially valid for an array of discrete loudspeaker system.

As the biggest merit of the proposed system, which is equipped with a loudspeaker array and uses the WFS technique, we can place an unlimited number of virtual sound sources on a horizontal plane. Therefore, we can place a sound object source or multichannel surround source signal. Thus, regardless of the multichannel input format, the proposed system can reproduce an adequate surround sound image for the listener. In addition, the proposed double-layered loudspeaker provides a sound image in the middle of the screen.

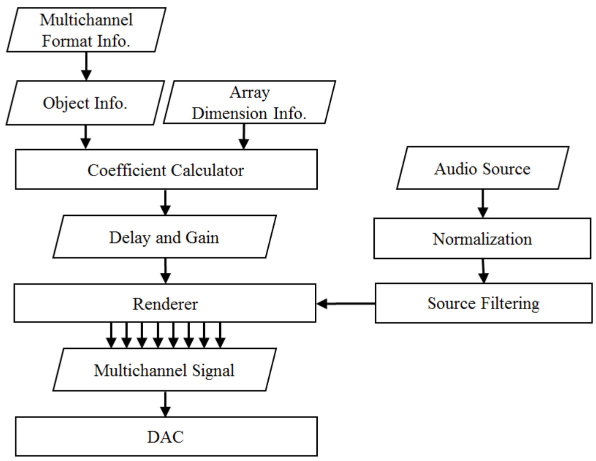

Figure 1. Dataflow of the proposed system

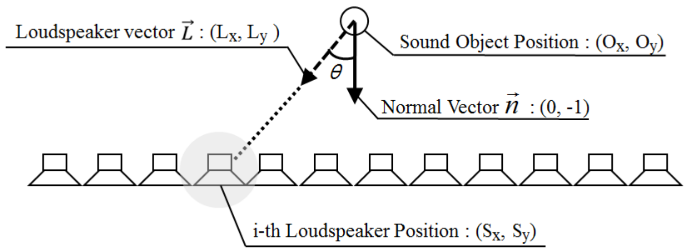

Figure 2. Sound source and related vectors

II. RENDERING SOFTWARE AND ALGORITHM

A. Coefficient Calculator

To render the virtual sound source, the delay and gain should be obtained for each channel. The delay and gain can be obtained from a loudspeaker vector and normal vector, as shown in Figure 2. In our proposed system, loudspeaker directivity is ignored to reduce the calculation complexity and simplify the system. Using the coordinates and angle described

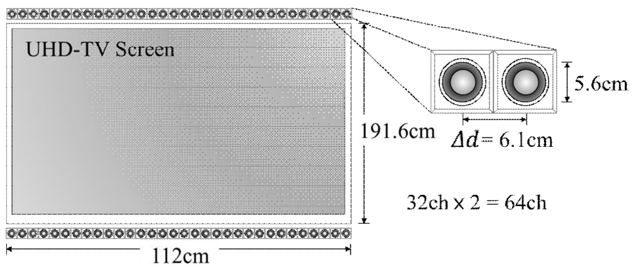

Figure 3. Dimension information of two-layer loudspeaker array system

in Figure 2, the final equation for implementing the gain and delay of i-th channel can be described as below.

$$D(i) = \frac{\sqrt{(O_X - S_X)^2 + (O_Y - S_Y)^2}}{c \cdot F_S}$$
$$G(i) = \frac{\cos(\theta)}{\sqrt{(O_X - S_X)^2 + (O_Y - S_Y)^2}} \quad (0)$$

where c is the speed of sound, F_S is the sampling frequency, and $\cos(\theta)$ is obtained through the inner product described below.

$$\cos(\theta) = \frac{|\vec{L}||\vec{n}|}{\vec{L} \bullet \vec{n}} \quad (0)$$

Once the delay and gain are obtained, these coefficients are transmitted to the render software module, which is described in Figure 1.

B. *Renderer*

To generate the wave field, the gain and delay of each loudspeaker channel are obtained. The output of the rendered signal can be described as below

$$x_{out}(n,i) = G(i) \cdot x_{in}(n - D(i)) \quad (0)$$

where $x_{out}(n,i)$ is i-th loudspeaker channel output with sample index n. By multiplying the gain and delaying the original source signal, a wave field is generated from the source position described in Figure 2.

III. Implementation and Hardware

A. *Spatial alisasing*

The diameter and interval of the loudspeaker are crucial for the performance of the sound reproduction because a discrete loudspeaker cannot reproduce an ideal wave field in reality [3]. This artifact caused by a discrete loudspeaker generates spatial aliasing over a certain frequency constraint. The frequency constraint can be described as follows:

$$F_a = \frac{c}{2 \cdot \Delta d} \quad (0)$$

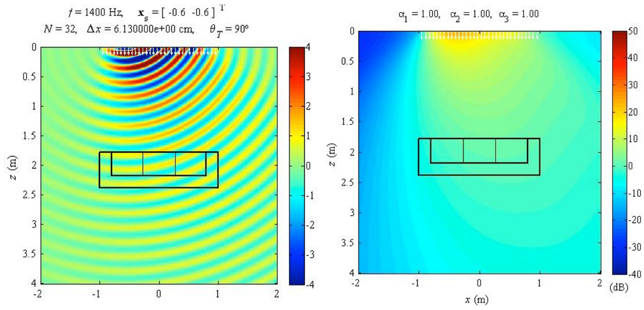

Figure 4. Simulation results of virtual sound source
(Left: wave field simulation, Right: SPL simulatoion)

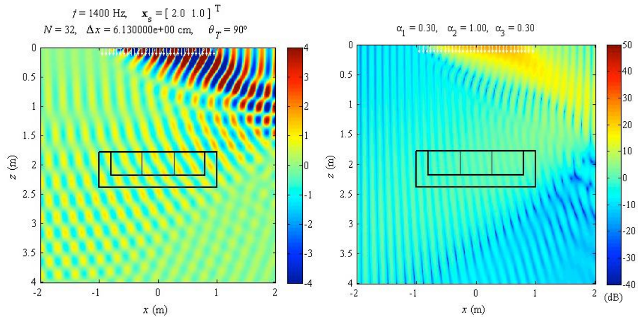

Figure 5. Simulation results of wall reflection
(Left: wave field simulation, Right: SPL simulatoion)

where Δd is the interval between loudspeakers. The frequency component of the sound source that has a frequency over F_a is unable to be expressed by controlling the discrete array of loudspeakers. Therefore, we filter the frequency over F_a and apply a rendering process for only the frequency component that is below F_a. On the other hand, a frequency component that is over F_a is emitted through a loudspeaker without applying a rendering process. Thus, we obtain the overall output signal with fewer artifacts.

B. *Hardware Dimensions*

The dimensions of the hardware are described in Figure 3. The two-layer loudspeaker array is placed on both the top and bottom of the TV screen. The transducers of the loudspeakers are placed with an equal interval of Δd. A total of 32 of loudspeakers are placed in each layer. These two layers provide a vertical sound image for listeners with a panning effect. Therefore, the proposed system is able to express a multichannel surround format with a vertical channel set-up.

IV. Simulation

Because a wave field is not visible, a simulation is an effective way to determine how a wave field is formed after a reflection. Therefore, a simulation can be effective way to analyze how a wave field propagates. The left side of Figure 4 describes how a wave field propagates when a virtual source is placed beyond the loudspeaker. The right-side of Figure 4 describes the Sound Pressure Level (SPL) of the rendered virtu

-al source. The left side of Figure 5 describes how a wave field propagates when a virtual source is reflected from the right wall. Once a wave field is reflected from the wall, the wave field is formed as if it propagated from the right side. The right side of Figure 5 describes the SPL of the reflected virtual source.

V. CONCLUSION

As 3D video technology develops, there is a need to develop better audio reproduction systems. Our proposed two-layer loudspeaker array system is a good solution for the upcoming home theater environment. Figure 6 describes the versatility of our proposed system. Both the traditional 5.1ch system and our proposed 7.1ch system can express a multichannel format which has a vertical sound image such as 22.2ch. Furthermore, the proposed system is able to render an object-based sound format. Since our proposed system can place a virtual sound source anywhere in the room, an object-based sound format is best input source for our proposed system to enjoy surround or 3D sound.

Also, further study is needed to overcome the restrictions of the room environment. If the room is half closed, the surround channel cannot be expressed using a reflection method. Therefore, an alternative method is needed to overcome the restriction in room dimensions. In addition, there is a strong need to develop an automatic setup technique to maximize the reflection performance.

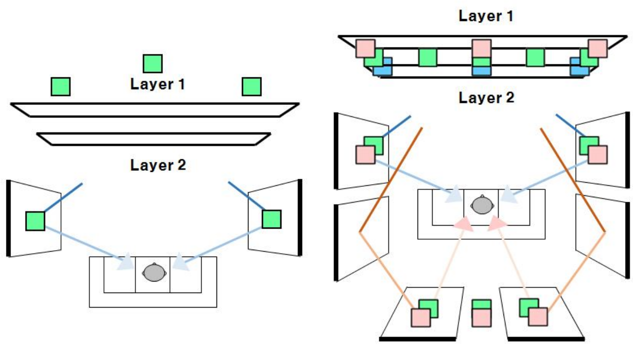

Figure 6. Reproduction of multichannel surround sound.

ACKNOWLEDGMENT

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