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REPRODUCTION OF SOUND FIELD USING A VIRTUAL LOUDSPEAKER ARRAY SYSTEM

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Abstract
Traditional sound fields firstly establish a sound field by using two loudspeakers. Right now, placing several groups of loudspeakers around listeners has been proposed; however, the optimal placement of loudspeakers is required for the best listening perception. This paper aims at solving the placement issue by means of sound field reappearance when an optimal placement environment is unavailable. Both interaural time difference and interaural intensity difference are employed as the principle of wave field synthesis. Using the proposed method, the generated virtual loudspeakers can relieve loudspeakers from the restriction of specific placement requirements for optimal listening perception. Finally, tests of the virtual loudspeakers are performed to show the effect of the proposed algorithm.

Keywords
Duplex Theory, Virtual Loudspeaker, Optimal Loudspeaker Placement Algorithm, Re-production of Sound Field.

1. Introduction

The traditional stereo research is first developed by Blumlein[1]. As time goes by, people pursue a virtual reality stereo system and pay more attention to the auditory environment. The integrated and multi-level environment is thus established by means of placing several groups of loudspeakers. For example, the Dolby Digital or DTS surrounding technology are built upon this multi-speaker environment. However, both Dolby Digital and DTS require optimal placements of multi-channel loudspeakers which require specific angles between loudspeakers and listeners in order to achieve a better sound field. Problems arise when the optimal placement may not be available due to the inherent environment settings. The perceived sound field may degrade consequently.

This paper aims to solve the placement issue by means of sound field reappearance[2-5]. A brief description of the proposed method is as follows. First, interaural time difference (ITD) and interaural intensity difference (IID) are employed using the principle of wave field synthesis. Then we analyze the sound field generated by virtual loudspeakers. Next step is to obtain sound signals of real loudspeakers such that the produced sound field is close to the sould field generated by the virtual loudspeakers by means of minimizing least square errors. Lastly, verification tools are used to verify the location of the produced virtual loudspeakers by detection of signal magnitude.

2. Reproduction of sound field

2.1 Synthesis theory

The so-called reproduction of sound field is to create a new sound field, that we feel the sound distance and direction being the same as the original sound field. Human hearing elements detect sound source location based on intensity difference and time difference, and these elements are the auditory localization cues which are called interaural time difference(ITD) and interaural intensity difference(IID). This is also known as the duplex theory, which was proposed by Lord Rayleigh[6]. Duplex theory has a disadvantage which is that the sound field can only be synthesized in either the front or back side since sound fields of the front and back are the same. To cope with this problem, techniques using multi-group loudspeakers surrounding the listener are used in sound field reappearance and can regenerate omnidirectional sound fields.

2.2 Optimal loudspeaker placement algorithm

In this paper, we use a wave field synthesis algorithm based on ITD and IID. We produce virtual sound sources when the environment disallow placing loudspeakers at their optimum positions. As memtioned above, using duplex theory one condition must be satisfied for real loudspeakers layout, which is that the sound field of real loudspeakers must surround with the listener. The problem is solved in [4] which was proposed by Foo and Hawksford in 1988. Our goal in this research is to generate a sound field such that the listener can perceive correct sound localization even when the listener shifts his position or turns his head. Since it is difficult to reproduce the whole sound field physically, we then focus on the sould field within the listening area which is a relative small range near the listener.

Assume the sound source is generating spherical waves, its amplitude is inversely proportional to the distance of the sound source to the listener. Eq. (1) is a transfer function from a sound source to arbitrary points in the sound field.

$$H = \frac{1}{d}\delta(n-\phi) \qquad (1)$$

where d is the distance from the source to the listener, δ(n) is the Dirac delta function, and ϕ is the phase difference. The sound field can then be expressed as the following convolution

$$SF = s(n) * H \, , \qquad (2)$$

where SF is the sound field, and s(n) is the sound source signal. In this paper, we assume the number of real loudspeakers is 4, the number of virtual loudspeakers is 1, and the number of reference points is 72. Let SF_v, SF_r be the sound fields generated by virtual and real loudspeakers respectively. We would like SF_r to be as close to SF_v as possible. We solve the least square problem in order to minimize the square error:

$$error = \left\| SF_v - SF_r \right\|_2^2 \qquad (3)$$

Using (2), we decompose sound fields as the product of the signal of sound source and the transfer function. That is, $SF_v = VG, SF_r = RH$, where $\vec{V} \in R^{1\times 1}$ and $\vec{R} \in R^{1\times 4}$ are the vectors of input signals at virtual and real sound sources, and $G \in R^{1\times 72}$ and $H \in R^{4\times 72}$ are the matrices of transfer functions from virtual and real sound sources to the reference point, respectively. Elements of V, R, G, H are shown in the following:

$$\begin{aligned} &V = Be^{-j\omega t}, \qquad R = Ee^{-j\omega(t-\phi)}, \\ &G_{lr} = \frac{1}{|x_l - l_r|} e^{-j\omega\left(t - \frac{|x_l - l_r|}{c}\right)}, \qquad H_{lv} = \frac{1}{|x_l - l_v|} e^{-j\omega\left(t - \frac{|x_l - l_v|}{c}\right)} \end{aligned} \qquad (4)$$

In the above, l_v is virtual sound source location, l_r is real sound source location, $B \in R^{1\times 1}$ and $E \in R^{1\times 4}$ are vectors of amplitude of virtual and real loudspeakers respectively, c is velocity of sound, x_l is the reference point, and ω is anglular velocity of the sound source.

Through linearization processes on the virtual sound field, we have

$$V = RA \qquad (5)$$

where matrix A is described as follows:

$$A = HG^T\left(GG^T\right)^{-1} \qquad (6)$$

The inverse of GG^T can be computed by singular value decomposition[7]. Using (5), we obtain the mathematical formulation of the time and intensity difference between the generated virtual loudspeakers and the desired virtual loudspeakers. Finally, we substitute the time and intensity difference into Eq. (2) and minimize (3) to get the signals for virtual loudspeakers.

Our algorithm is as follows. The virtual sound source signal is inputted into the transfer function from virtual loudspeakers to arbitrary points in sound field. Signals of real loudspeakers are formulated by multiplying the virtual sound field with an inversed transfer function from real loudspeakers to arbitrary points in the sound field, which is the matrix A in (6). We then calculate the signal for the real loudspeakers so that minimum square error between the real sound field and the virtual sound field is achieved within the listening area near the listener. This process is called the optimal loudspeaker placement algorithm. By utilizing this algorithm, one can perceived the optimal sound field in any environment in the sense of minimum square error.

3. Experimental results

To verify the generated virtual sound source location, we use a directional microphone and adopt the bisection search for the direction in which the received signal has the maximum strength. Experimental equipment includes a set of real loudspeakers and a microphone. The real loudspeakers are Logitech Z5500, and the microphone is audio-technica AT 9944. Experimental layout is shown in Fig. 1(a). We synthesize multiple virtual loudspeakers through computer simulation, and also check the result with different listening area and different locations and angles of virtual loudspeakers. To check whether the reproduction of sound field is correct or not, we then carry out the verification procedures stated above.

When the environment forbids the optimal loudspeakers layout placement, we then use the optimal virtual loudspeaker algorithm. The placements of virtual loudspeakers are not confined by environment, and we can thus perceive better sound field. Fig. 1(b) shows the optimal virtual loudspeaker algorithm used in real environment.

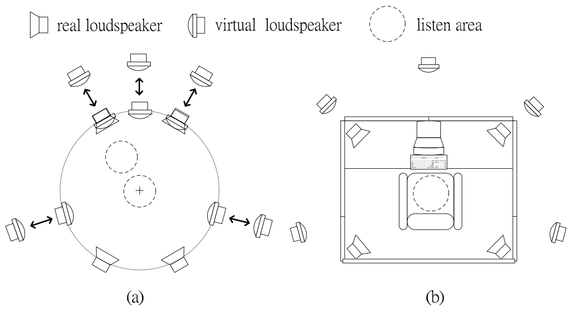

Figure 1 – (a) Experimental layout (b) The optimal virtual loudspeaker algorithm is used in real environment.

3.1 Directivity test

The experiment layout is stated as the following. The number of reference points is 72, and these reference points are placed at distances of 7.5, 12.5, 17.5 cm between reference points and the origin with each reference point separated by 15 degrees relateive to the origin. The number of real loudspeakers is four, which are placed at 60, 120, 240, 300 degrees with distance of 1 meter relative to the origin. We test the virtual loudspeakers at every 15 degree and the distance is 1 meter between the origin and virtual loudspeaker.

The origin is placed at the center of real loudspeakers array. We compose the signal of virtual loudspeakers through optimal loudspeaker placement algorithm, and we obtain the minimum error sum shown in Fig. 2(a), in which the x axis is the angular degree of virtual loudspeaker location and the y axis is the percentage error of sound field signal at the reference points. From Fig. 2(a), the error can be as large as 50% at 0 and 180 degrees that sound field reappearance could not be accurate.

Virtual loudspeakers are produced by decomposing the signal of virtual loudspeakers into real loudspeakers array. We examine the directivity of virtual loudspeakers by using verification tools in real environment. We discover that the error degree is smaller than 15 degree. The result is shown in Fig. 2(b). It should be noted that large sound field error does not immediately imply larger directivity error. Although the sound field cannot be accurately synthesized, the virtual sound source can be synthesized with similar directivity. From Fig. 2(b), the directivity test shows that the virtual loudspeaker is detected at the correct bearings in most cases and the maximum error is 15 degrees. Therefore, we can produce virtual loudspeakers array of each type through optimal loudspeaker placement algorithm.

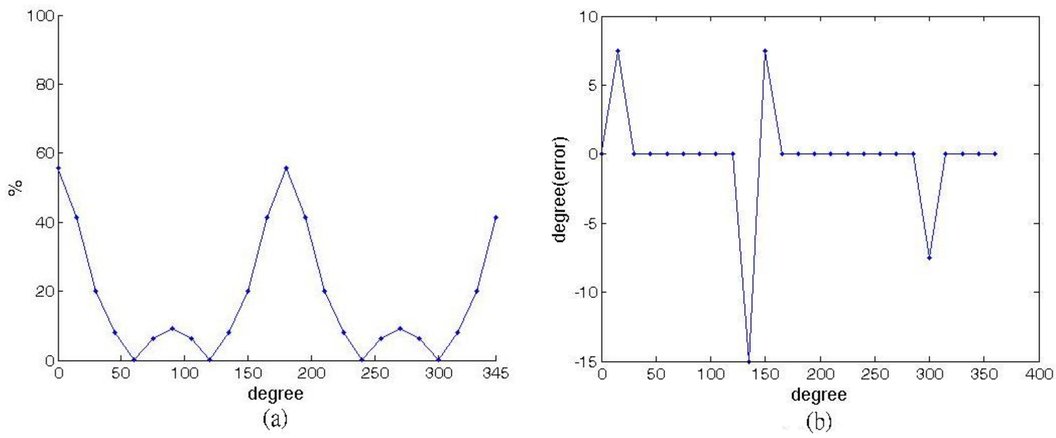

Figure 2 – (a) Minimum error of directivity test between sound field of virtual and real loudspeakers (b) Error degree of directivity test of virtual loudspeaker in real environment

3.2 The car audio environment: directivity test at driver's seat

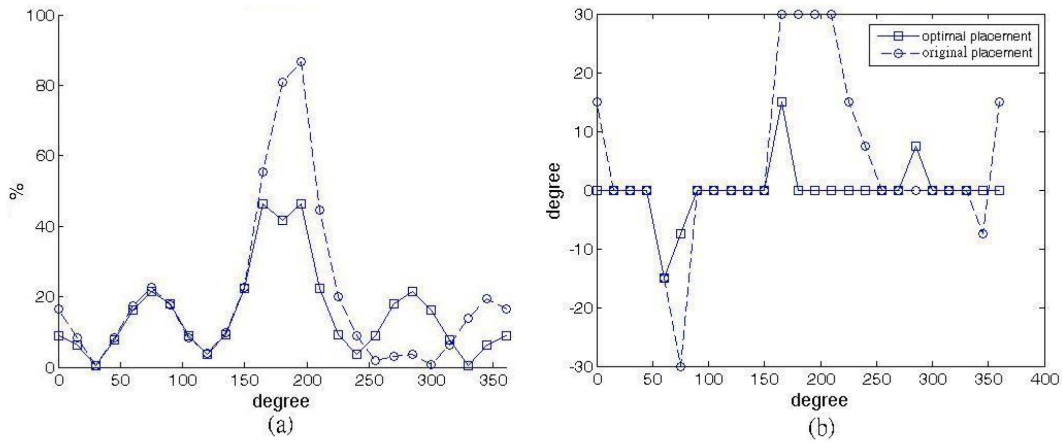

Figure 3 – (a) Simulation error of directivity test between sound field of virtual and real in car-audio environment (b) Placement error of directivity test of virtual loudspeaker in car-audio environment

Here we study a problem when the listener is not located at the center of real loudspeakers array. Consider the case of car audio environment in which the driver sits on the left-hand side. Other layout is the same as the previous section. In Fig. 3(a), dashed line shows the simulation error more is than 80% from 180 to 210 degree. In Fig. 3(b), dashed line shows the actual difference of directivity. In this figure, the error has highest value between 165 and 195 degrees. In this situation, the layout of sound field reappearance is worse when listeners approach to one side of the loudspeaker layout. This is due to a high error in some section of the sound field.

To solve the problem of sound field reappearance as listeners approaching to one side, we change real loudspeaker placement form the rear pillar to the center pillar. We place two loudspeakers at 50 cm from the origin at 0 and 180 degree and two loudspeakers at 100 cm from the origin at 60 and 120 degree. In Fig. 3(a), dashed line showed simulation error with more than 40% from 165 to 195 degree, but the error has been reduced largely comparing with the original placement In Fig. 3(b), dashed line showed the bearing error in the modified speaker placement scenario. We can see that the error is smaller than 15 degrees and the directivity error also can be reduced. From the above discussion, a change in placement layout may have large impact on the result.

4. Summary

We proposed a method that can reproduce a virtual loudspeakers array using real loudspeakers array. In addition, we can preserve the sound field of virtual loudspeakers by synthesizing sound signals for real loudspeakers. The resultant virtual loudspeakers can be shown to have similar time and spatial characteristics to real loudspeakers. Therefore, we can get a better sound field by the proposed optimal loudspeaker layout when listener is not located at the center of real loudspeaker array.

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