2005년도 대한전자공학회 추계종합학술대회 제28권 제2호

서라운드 오디오 시스템을 위한 가상음원의 거리를 조절할 수 있는 인공잔향기

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Artificial reverberation algorithm to control distance of phantom sound source for surround audio system

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Abstract

Multi-channel artificial reverberation algorithm to control perceived direction and distance is described in this paper. In conventional algorithms using IIR filters, reverberation time is the only parameter to be controlled. Moreover, since the convolution-based conventional algorithms apply only same impulse responses, but not considering sound localization, it was not realistic enough. The new algorithm proposed in this paper utilizes early reflections segmented according to the azimuth from which direct sound comes and controls perceived direction by panning the direct sound, and controls perceived distance by adjusting Energy Decay Curve (EDC) of reverberation and gain of the direct sound. In addition, the algorithm enhances Listener Envelopment(LEV) to make late reverberation incoherent among channels.

I. Introduction

Recently, as discrete five-channel surround formats became popular, the need of new devices for the multi-channel surround formats has emerged. This need has brought many attempts to localize virtual sound source with multi-channel surround formats [13]. To perceive direction of sound source naturally, attempts to use appropriate early reflections have been made. The early reflections also play an important role for the perception of the distance of the sound source.

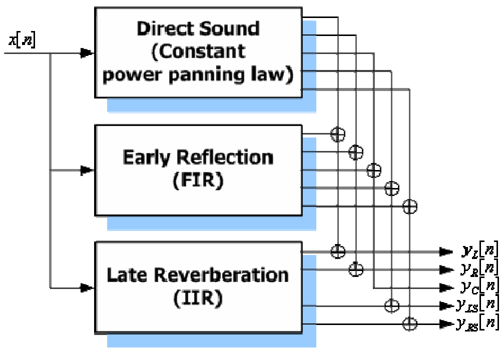

Fig. 1. Generic scheme of proposed algorithm

In [1], it is said that the most important objective measures to perceive the distance of the virtual sound source are the intensity of direct sound and the Energy Decay Curve (EDC) of reflections. In this paper, we propose a robust artificial reverberation algorithm which localizes mono sound source recorded by a spot microphone to desired position (direction and distance) by proper manipulation of early reflections. This proposed algorithm provides controllable parameters that consist of Reverberation Time (RT), perceived direction and perceived distance. Moreover, it is more efficient with reduced computing complexity. The reverberator with proposed algorithm deals with direct sound, early reflections, and late reverberation. The direct sound is localized to the desired position by constant power panning law. Once localized, manipulation of early reflections should be followed to improve naturalness and to control perceived distance of the localized direct sound. Then, the listener envelopment and the reverberance are improved by the generated late reverberation.

II. Control of Perceived Direction and Distance

2.1. Control of Perceived Direction

In the standard surround formats, direction of the virtual sound source is formed by summing localization and people perceive it. For this reason, the virtual sound source is localized by pair-wise power panning law using level difference between two adjacent loudspeakers. However, since this localization by pair-wise power panning law is not enough to satisfy naturalness of perceived direction, manipulation of the early reflections is needed.

2.2. Control of Perceived Distance

In [1], the intensity of the direct sound and the clarity of sound source are the most important measures for the perception of distance. The sound pressure attenuates in inverse-proportion to the square of distance. For this reason, the sound pressure is the most important measure for the perception of distance. In [2], the distance to the virtual sound source gets closer as EDT gets shorter. In the case of multi-channel formats, the perceived distance varies according to early reflections. To perceive distance by using conventional multi-channel reverberation algorithms and surround panpots, it is necessary to employ different impulse responses for different distances. These algorithms are inefficient because they need a lot of memories in computing process. This paper proposes to control EDT according to the distance by manipulating envelope (EDC) of impulse response of each channel at a specific distance.

It was confirmed that the sense of distance can be controlled by applying proposed algorithm through listening test. From this, it was possible to infer that the main reason of the test result is the fact that the envelope of early reflections has a form similar to that of energy decay curve even though the impulse response of each channel changes. For this reason, the sense of distance of the multi channel system can be controlled by adjusting energy decay curve. Therefore, this proposed algorithm does not need all of the impulse response database for all different distances to control perception of distance.

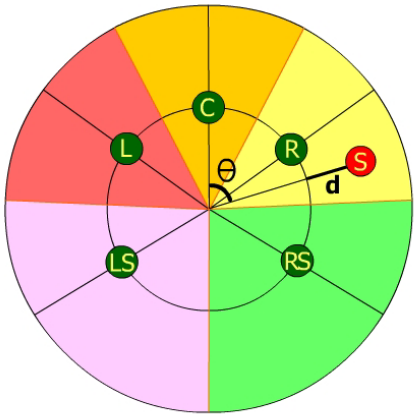

Fig. 2. Measurement of impulse responses of 5 divisions set (S : desired location of virtual sound source)

III. Measurement of Early Reflections

Direct sound and reflections are measured by microphone array method [4]. As Fig. 2. shows, impulse responses of 5 divisions set according to the position of loudspeakers are measured. This division method was performed according to the spherical coordinate of the virtual recording space. However, because all the loudspeakers are equally positioned vertically in home theater system, it divides the virtual space into 5 different regions horizontally. Since IIR filters model upon late reverberation, measured impulse responses are truncated from 0ms to 150ms. By analyzing measured impulse responses with peak detection and pair matching algorithms, we obtained a set of parameters which consists of amplitude, delay, and direction of each reflection [5,6]. This set of parameters will be stated as Reflection Parameter (RP). The direction and amplitude of each early reflection are recreated by the Constant Power Panning law (CPP) between two adjacent loudspeakers [12]. For this reason, FIR filter coefficients are obtained by using RPs. This process will be dealt with in next chapter.

IV. The Proposed Algorithm

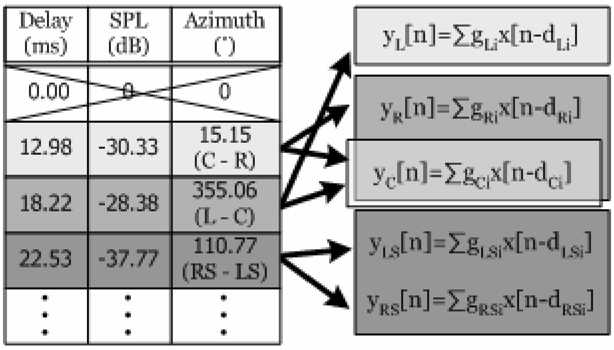

Fig. 3. FIR filter coefficients obtained by applying inverse power panning Reflection Parameters (RP)

Direct sound, early reflections, and late reverberation are all under control, since the proposed algorithm is designed for not only virtual source localization but also appropriate reverberation. As Fig. 2. describes, direct sound is localized to desired azimuth θ by CPP. For natural localization and control of perceived distance, early reflections are generated by FIR filter. When θ, the azimuth angle of the virtual sound source that we want to localize, is in the Right (R) division, the algorithm uses RPs of the representative position whose azimuth is that of loudspeaker in R division for standard surround system at a distance of r meters [11]. Hence, FIR filter coefficients are closely related with RPs of the representative position. The algorithm also compensates the difference d between the virtual source and the representative position for perceiving the distance of virtual source. In addition, artificial late reverberation filters fulfill appropriate reverberance.

FIR filter coefficients can be obtained by applying inverse power panning RPs as Fig.3. shows. The output signal of kth channel can be represented as

$$y_k[n]=\sum_{i=0}^{M_k} g_{k_i} x\left[n-d_{k_i}\right],$$

where g_{ki} coefficient of FIR filter of ith early reflection and d_{ki} is time delay of ith early reflection. Perceived distance can be controlled by adjusting energy decay curve. The gain function of relative distance $a(d_{ki})$ is obtained by comparing with energy decay curve of impulse responses at desirous distance. After multiplying gain function, the output signal of kth channel can be described as

$$y_k'[n]=\sum_{i=0}^{M_k} a\left(d_{k_i}\right) g_{k_i} x\left[n-d_{k_i}\right]$$

Late reverberation is reproduced by a set of IIR filters because of computational cost and economical cost. IIR filters are used to generate late reverberation for 5 channels. Generally, it is well-known that Listener EnVelopment (LEV) gets better as later reflections get more incoherent and that reverberant impression gets better as LEV gets better [2,7]. Hence, it is possible to reproduce affluent reverberant impression eventually by making late reverberations of each channel incoherent [8,9,10].

V. Implementation

Fig 4 shows an implemented software to adopt proposed reverberation algorithm by using Virtual Studio Technology (VST) SDK for real-time processing. The parameters to be manipulated are Reverberation Time (RT), perceived direction, and perceived distance.

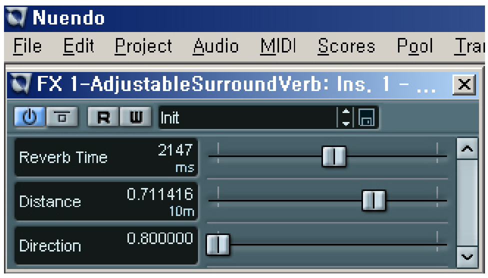

Fig. 4. Implemeted software to adopt proposed reverberation algorithm

VI. Conclusion

With the proposed algorithm, the distance and direction of sound source were perceptually controlled. The proposed algorithm measures impulse responses of 5 divisions. In further works, we will increase the number of divisions and measure impulse responses of all of the divisions. In the end, more natural early reflections with increased number of divisions are expected. In addition, compensation filters are expected to reduce coloration due to the panning law. Controlling time interval among early reflections from each distance in advance of controlling intensities of early reflections will be also helpful to obtain more solid perceived distance.

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