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Improved methods for measuring early reflections from Five-channel room impulse response using newly introduced Peak-Detecting algorithm

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Abstract

When we measure the acoustical properties of a room using multiple microphone system, it is important to grasp exact time delay of the early reflections from impulse response pair. But it is often very difficult to identify the early reflections in natural shape, because a waveform may be deformed due to the characteristics of a sound source loudspeaker, microphone and reflected wall and overlapping of plural waveform.

In this paper to obtain more accurate and enough early reflections, we propose the brand-new five-channel sound receiving system and introduce peak-detecting algorithm. The system has microphones mounted at the origin and four points of a regular tetrahedron. The newly introduced peak-detecting algorithm can show exact peak position in each channel, in spite of deformation due to reflected walls, loudspeaker and microphone.

KEYWORDS: Spatial information, Early reflection, Sound field, Room acoustics

Introduction

Besides reverberation times and sound pressure levels, difference of spatial information, especially in early reflection periods is very important because we often get a quite different acoustic impression in the sound fields which have about the same reverberation time and sound pressure level.

Many researchers have conducted various studies to obtain spatial information.

Yamazaki and Ito obtained impulse responses from four synchronous microphones system on the rectangular coordinate and utilized correlation technique to find pairs of each microphone. However, the correlation method depends upon the similarities of waveforms, where waveforms are often hardly separated if they overlap each other. And sampling frequency is high enough in order to increase accuracy, as a necessary consequence the amount of data and calculation time increases.

Sekiguchi, Kimura and Hanyuu used four microphones, which are installed at the apex of a regular tetrahedron and utilized deconvolution method to improve measuring accuracy for waveform analysis. In order to use this method, the sound source speaker has identical characteristic through all direction. Otherwise deconvolution is confined to a few reflections, which have identical characteristic, and other reflections are just distorted. Also, because impulse response of loudspeaker is generally mixed phase, deconvolution is not easy to performed. Although perfect deconvolution can be performed, influence of the reflecting walls remains to serious problem for recognizing peak position.

The purpose of previously described techniques is to find exact peak position of each reflections from four synchronous impulse responses. Here, we will newly introduce peak-detecting algorithm satisfying this purpose and overcoming the demerits of other technique.

Five-channel sound receiving system

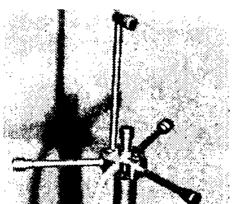


Fig. 1 Receiving system structure

About the sound-receiving device, we applied Sekiguchi, Kimura and Hanyuu's fundamental structure as shown in Fig.1. But the spaces between each microphone is much reduced, thus lowering the possibility of other sound waves being mixed in when one sound wave is passing through. In order to obtain uniform spatial information in all directions wherever practicable, each of four microphones is installed at the apex of a regular tetrahedron. In addition, one microphone is installed at the center of gravity and it is used as basis for finding pairs of each peaks from 5 channel reflectogram.

Peak–Detecting algorithm

fig. 4 Speaker

In order to calculate a virtual sound source position, it is necessary to obtain time and level accurately when direct or reflected wave front has just passed through each microphone. In natural shape, exact position of direct or reflected peak is hard to be identified as shown in Fig. 2 because of deformation due to reflected walls, loudspeaker and microphone.

Accordingly, we use peak-detecting algorithm to find exact position of direct and reflected peak. The process of peak-detecting algorithm is as follows:

1. Early period impulse response(about 100ms) is interpolated 10 times to increase time resolution. We make use of cubic spline interpolation.
2. Envelop of that interpolated impulse response is obtained by using Hilbert transform.

$$E[n] = \sqrt{\left(s[n]^2 + \check{s}[n]^2 \right)} \text{ , } \check{s}[n] \text{ denotes the Hilbert transform of } s[n]$$

By using this, insignificant details of a reflectogram are removed beforehand.

3. Difference of envelop is calculated, and when the sign of difference change from positive sign to negative sign, we recognize that point as peak.
4. Difference of peak series itself is calculated, and then same process of 3) is conducted until expecting number of dominant peak is showed.

In this paper, the iteration number is decided considering speaker response.

The peak-detecting algorithm processed result is shown as Fig. 3 in the last page.

Calculation and Visualization of coordinates of direct and virtual image source

The measurement was performed at the applied acoustics laboratory. The receiving system is shown in Fig. 1. The distance between microphones on each apex is 10cm. The received signal was obtained by a personal computer through A/D converter in sound card, while source sound was transmitted through D/A converter in the same sound card in a synchronized manner; sampling frequency is 48KHz, and MLS(Maximum-Length sequences) is used as source signal. MLS method has the superior noise and distortion immunity than the periodic pulse method, which was used by previous papers.

As a source loud speaker we used pseudo omni-directional speaker, which is shown at Fig.4. Theoretically any loud speaker can be used if we use peak-detecting algorithm, but we use that speaker to excite room through all direction.

Channel 5 is used as a basis for searching direct and reflected peak pairs. The microphone of channel 5 is located inside of other microphones in any case. So peaks in other channel are within specific samples. The number of specific samples 'p' is

$$p = \frac{\text{distance between origin and each apex}}{\text{speed of sound}} \times sampling rate$$

calculated as follows;

Where distance between origin and each apex = 5.5cm

Speed of sound = 343m/s

Sampling rate = 480000, this is due to 10 times interpolation.

So p is approximately 80.

The five channel response after using peak detecting algorithm is shown at Fig.5 in the last page.

The direct and virtual sound positions and amplitude is shown at Fig.6. The origin is receiving point, the centers of circles are the position of direct and virtual image source, the radius of circles are energy of that source .

Conclusion

We newly introduced five-channel receiving system and peak-detecting algorithm. By using it, we could find exact position of direct and reflected peak in 5-channel reflectogram. It can be applied into various sound field, even in case of small room like inside of car, in which the time difference between successive reflections is so short that most of early reflections cannot be identified.

References

1. Yamazaki, Y. & Ito, T. "Measurement of spatial information in sound field by closely located 4-point microphone method." *J. Acoustics. Soc. Jpn. (E)*, 10(2), 101-110 (1989)
2. Sekiguchi, K., Kimura, S. & Hanyuu, T. "Analysis of Sound Field on Spatial Information Using a Four-Channel Microphone System Based on Regular Tetrahedron Peak Point Method" *Applied Acoustics* 37 305-323(1992)
3. H. Kuttruff, *ROOM ACOUSTICS Third Edition* (Elsevier applied science, London and New York, 1991)

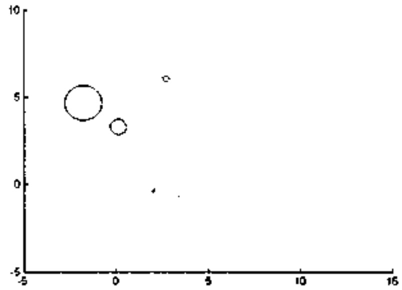


Fig. 6 Direct and virtual sound positions and amplitude

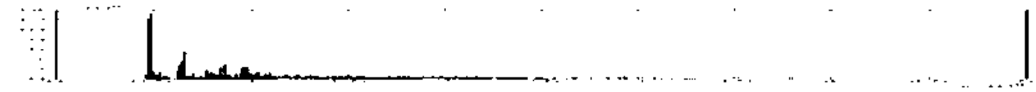
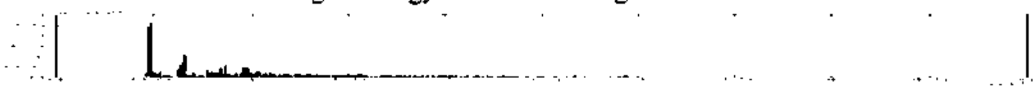
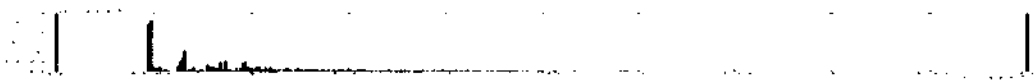


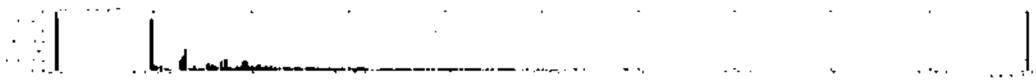
Fig. 2 Energy vs. time reflectogram



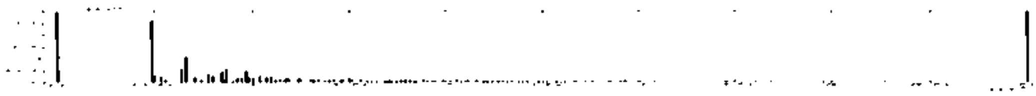
(a) Envelop of interpolated impulse response



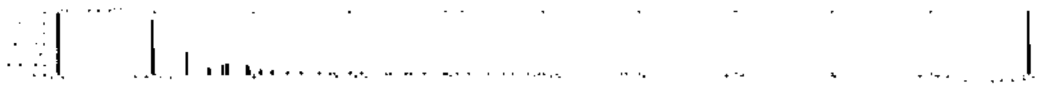
(b) Iteration 1



(c) Iteration 2

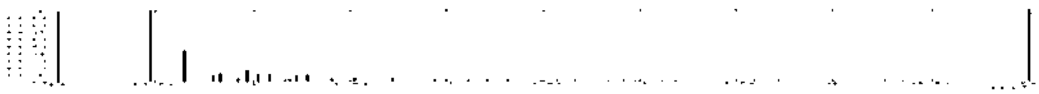


(d) Iteration 3

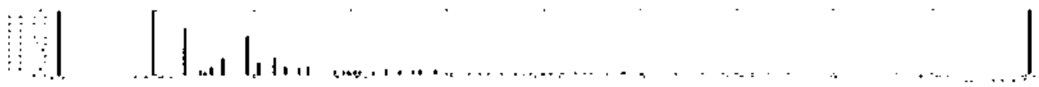


(e) Iteration 4

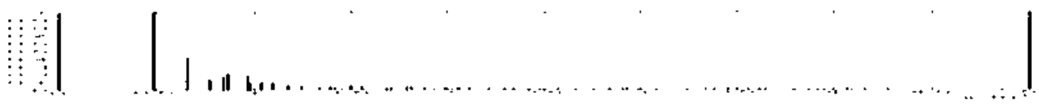
Fig. 3 Peak-detecting algorithm process



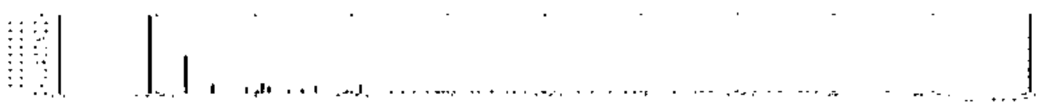
(a) Channel 1



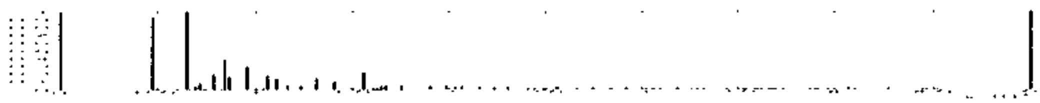
(b) Channel 2



(c) Channel 3



(d) Channel 4



(e) Channel 5

Fig. 5 Five channel response after using peak detecting algorithm