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MEASUREMENT OF SOUND TRANSMISSION BY HALL DOORS

Y. Soeta, K. Ohtori, K. Fujii and Y. Ando

Graduate School of Science and Technology, Kobe University, Rokkodai, Nada, Kobe 657-8501, *Japan. E-mail*: 979*d*870*n@kobe-u.ac.jp*

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Acoustical properties of sound transmission through to all doors were investigated by measuring and calculating temporal and spatial factors in sound fields based on the model of the auditory–brain system (see reference [4]). To describe acoustical properties of the effect of doors, measurements were carried out obtaining five orthogonal factors. These are (1) sound pressure level (SPL), (2) the magnitude of the interaural cross-correlation (IACC), (3) subsequent reverberation time (T_{sub}), (4) interaural time delay (τ_{IACC}), and (5) the width of the IACC function (W_{IACC}). In order to describe subjective responses in the receiving room, not only the difference of SPL but also other factors, for example IACC, and T_{sub}, can be measured at the same time.

1. INTRODUCTION

A number of methods have been used to measure noise transmission indoors. As these are in terms of sound pressure levels (SPLs) alone, however, the acoustical properties extracted do not sufficiently correspond to those required for evaluation in psychological and physiological terms [1]. Various aspects of acoustical properties need to be clarified to describe the effect of noise on human beings. When designing the sound fields of a concert hall, temporal and spatial factors must be considered together to achieve the most preferable overall result. In this research, a method which is for the evaluation of concert halls [2–4] is applied to evaluating the sound-insulating qualities of doors. The following orthogonal factors were measured [2–4]: (1) sound pressure level (SPL); (2) the magnitude of the interaural cross-correlation (IACC); and (3) subsequent reverberation time (T_{sub}). The following two factors were derived: (4) interaural time delay (τ_{IACC}); and (5) the width of the IACC function (W_{IACC}). τ_{IACC} is related to the perceived location of a sound source, and W_{IACC} is related to the apparent source width (ASW) [5].

A method which uses a maximum-length sequence (MLS) signal [6–8] was applied to obtain a clear impulse response. The MLS signal used during this measurement is called pseudo-random white noise.

2. METHOD

2.1. MEASUREMENT PROCEDURE

The acoustical measurements were performed in a community hall in Ichikawa, a town in Japan. Figure 1 shows the floor plan of the community hall. A loudspeaker, S, and

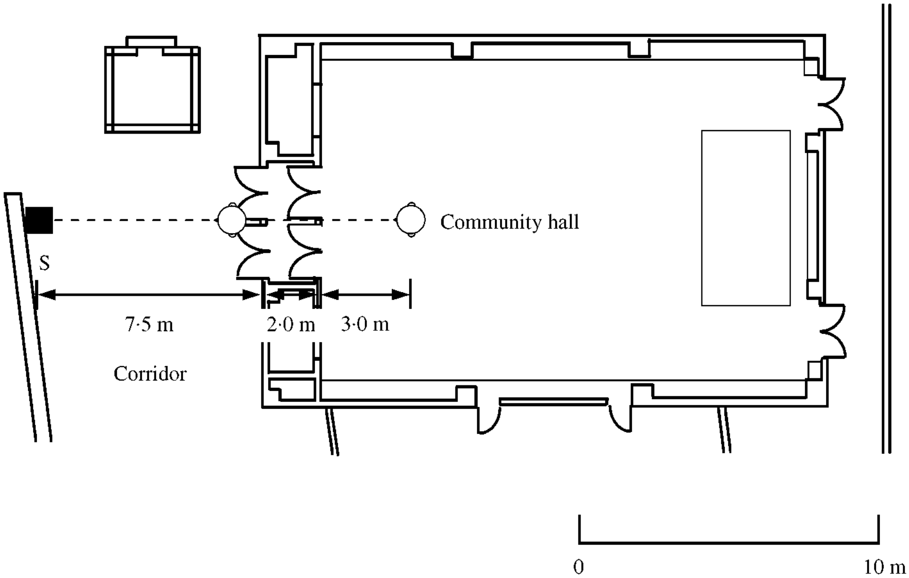

Figure 1. Plan of the community hall investigated. A loudspeaker was set as a sound source S. The person with condenser microphones placed at ear-entrances was applied to each receiving point.

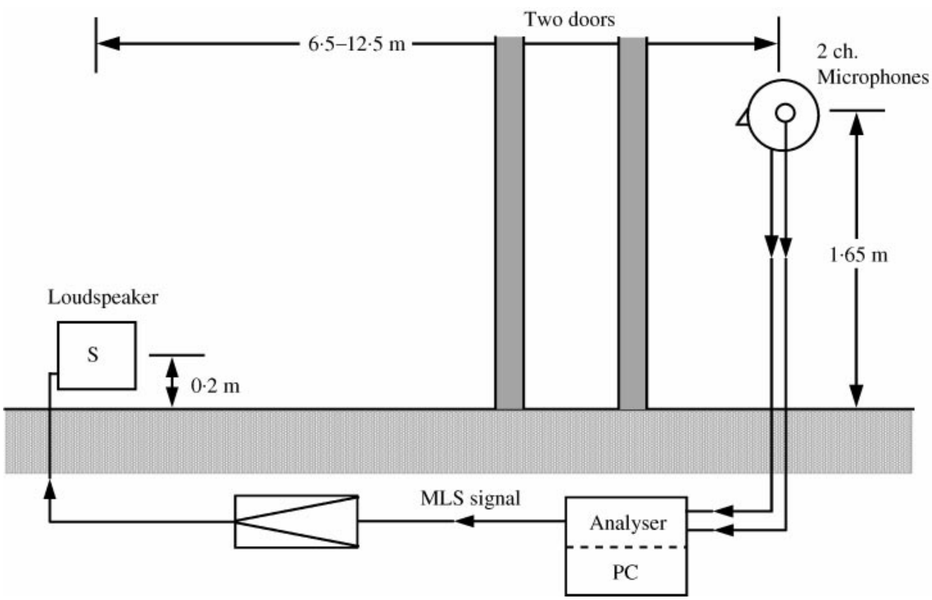

Figure 2. Schematic illustration of measurement with a loudspeaker and microphones mounted on a person's head.

microphones arranged as shown in Figures 1 and 2. The center of S was 0·2 m above the floor. The two condenser microphones were mounted on the ears of a person who faced the loudspeaker during the measurements and were 1·65 m high. The MLS signal was radiated from the sound source S. The A-weighted SPL was about 100 dB at the 6·5 m reference point. To examine the effect of doors, the acoustical measurements were performed following four conditions: (1) both sides of doors open; (2) doors on the community hall side open; (3) doors on the corridor side open; and (4) both sides of doors closed.

The three physical factors and two derived factors mentioned above were analyzed by using binaural impulse responses. Impulse responses were obtained by A/D converting the signals from the microphones and then analyzing the digital representations by using a fast Hadamard transform (FHT). After obtaining binaural impulse responses, physical factors were calculated for each 1/1 octave-band center frequency between 125 Hz and 4 kHz and for the all-pass band. The signal's duration was 1·4 s, and a sampling frequency of 48 kHz was used. In order to improve the signal-to-noise ratio, the sequence of signals was repeated twice.

The impulse responses can be analyzed by applying the Hadamard-matrix-conversion technique to the MLS signal to an operation which involves only the addition [6,7]. The impulse response of a linear system is obtained by cross-correlating the input and output signals. Impulse responses, $h_l(t)$ and $h_r(t)$ for the left and right ears, respectively, were analyzed for each condition.

2.2. METHOD OF ANALYZING ACOUSTICAL FACTORS

A binaural impulse response h_{jl} and h_{jr} [9] was calculated for each acoustical factor listed above. The index j indicates the samples of the MLS at a time interval σ ($j = 0, 1, \ldots, L-1$).

2.2.1. *Relative* SPL

The SPL at the listening position was obtained relative to the SPL at the reference point. The SPL at each ear is given by the autocorrelation function $\Phi_{ll,rr}(\tau)$ at $\tau = 0$ of the impulse responses $h_{jl,r}$, so that

$$\Phi_{ll,rr}(0) = \sum_{j=0}^{L-1} h_{jl,r}^2. \tag{1}$$

If $h_{ji,r} \neq 0$, the relative SPL is defined as

$$SPL = 10\log_{10}\frac{\sqrt{\Phi_{ll}(0)\Phi_{rr}(0)}}{\Phi^{(ref)}(0)}, \tag{2}$$

where

$$\Phi^{(ref)}(0) = \sqrt{\Phi_{ll}^{(ref)}(0)\Phi_{rr}^{(ref)}(0)}. \tag{3}$$

Here, $\Phi^{(ref)}(0)$ is the geometrical mean of the auto-correlation function of binaural impulse responses at $\tau = 0$, at the reference point, as indicated by equation (3).

2.2.2. *Subsequent reverberation time* (T_{sub})

The decay curve is obtained by integrating the square of the impulse responses [10]. The subsequent reverberation time, T_{sub}, was obtained by fitting a regression line to the curves for initial decay of 10 dB after the arrival of direct sound. The results for T_{sub} are represented as the value averaged over both ears. T_{sub} is defined by the time taken to reach 60 dB attenuation along this regression line, so that

$$T_{sub} = k\sigma, \tag{4}$$

where σ is the time between samples and k is the number of samples until the regression line reaches 60 dB decay.

2.2.3. *Factors of the interaural cross-correlation function* ($IACC$, τ_{IACC} *and* W_{IACC})

The normalized IACC function is

$$\phi_{lr}(\mathrm{j}\sigma) = \frac{\Phi_{lr}(\mathrm{j}\sigma)}{\sqrt{\Phi_{ll}(0)\Phi rr(0)}}, \tag{5}$$

where the values of $\Phi_{ll}(0)$ and $\Phi_{rr}(0)$ represent the auto-correlation functions ($\tau = 0$) of the impulse responses at the left and right ears respectively. The $\Phi_{lr}(\mathrm{j}\sigma)$ is the cross-correlation of the impulse responses at the two ears. The three factors IACC, τ_{IACC} and W_{IACC} are expressed as described below

(1) The magnitude of the interaural cross-correlation function is

$$IACC = |\phi_{lr}(\tau)|_{max}, \quad |\tau| \leqslant 1 \text{ ms}. \tag{6}$$

This quantity provides a significant indicator of the degree to which sound sources appear diffuse, as well as subjective preferences regarding in the sound field [2]. IACC represents the degree to which the incident sound waves on the two ears are similar.

(2) The interaural delay time, the time at which the IACC is decided, is the τ_{IACC}. When τ_{IACC} is zero, a frontal sound-source image and well-balanced sound field are usually perceived.

(3) The width of the IACC function W_{IACC} is defined as the interval of delay time 10% below the IACC. W_{IACC} is a significant indicator of factor related to the ASW. It is noteworthy that the ASW can be derived from the IACC and W_{IACC} [4].

3. RESULTS AND DISCUSSION

3.1. RELATIVE SPL

Impulse response for the left ear obtained at a reference point is shown in Figure 3(a). Two conditions of impulse responses for the left ear obtained at a distance 12·5 m from the sound source are illustrated in Figure 3(b) and 3(c).

The measured relative SPLs for 1/1 octave band frequencies between 125 Hz and 4 kHz are shown in Figure 4. The SPL is attenuated as more doors are closed. When one side of doors are closed, the SPL is attenuated by about 20 dB. When both side of doors are closed, the SPL is attenuated by 33 dB.

3.2. SUBSEQUENT REVERBERATION TIME (T_{sub})

The measurements of T_{sub} are shown in Figure 5. When either side of the doors is closed, the value of T_{sub} becomes longer. T_{sub} was longest with both sides of the doors closed. This increase of T_{sub} is considered to be caused for two reasons. First, multiple reflections from walls in the small space enclosed by the both sides of doors arrive at receiving points through the frontal door as subsequent reverberation components. Second, T_{sub} is defined by the time taken to reach 60 dB attenuation along a regression line to the curves for initial decay of 10 dB after the arrival of direct sound. In this case, first reflection is more attenuated by both doors than the other conditions.

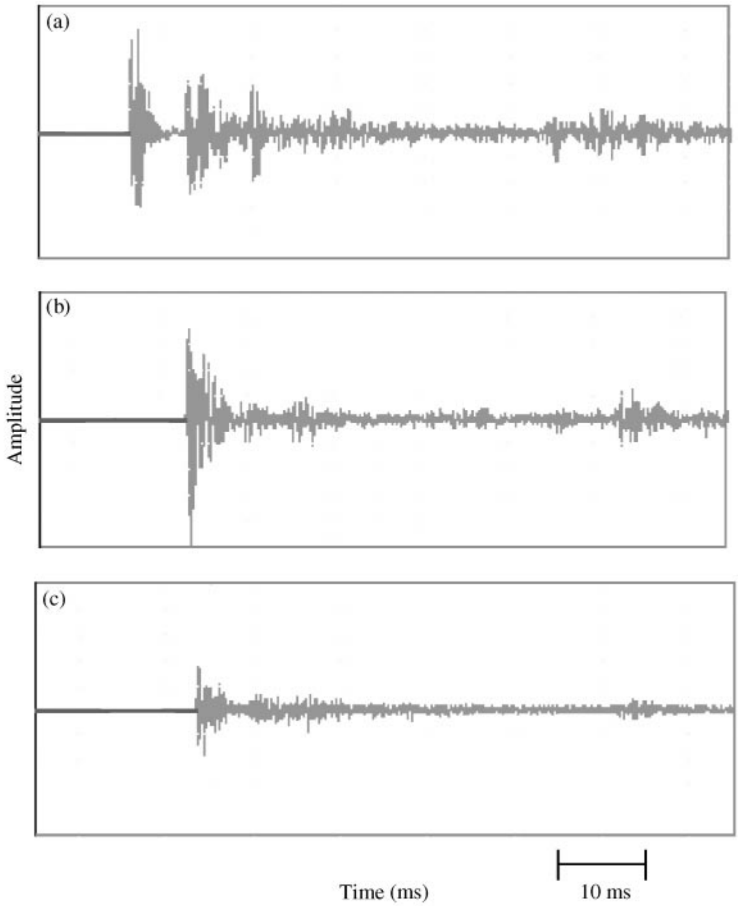

Figure 3. Normalized impulse responses for the left ear: (a) reference point; (b) both sides of doors open; (c) doors on the corridor side open.

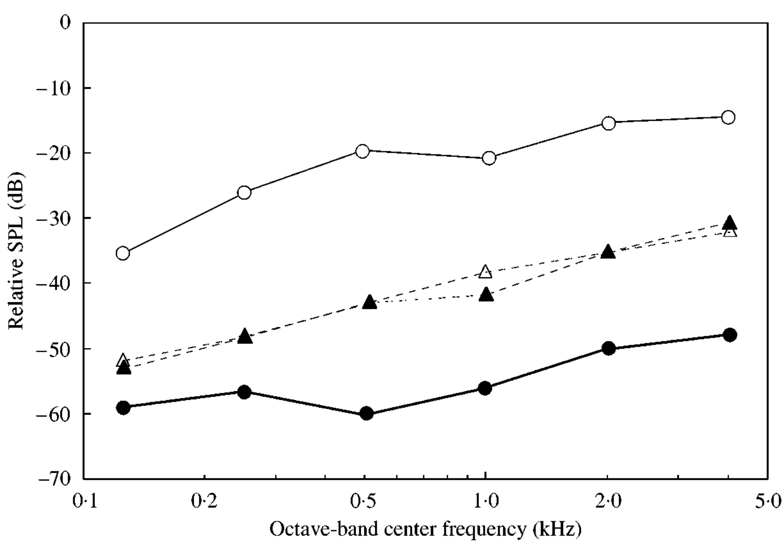

Figure 4. SPL as a function of the center frequency of the 1/1 octave band: (○), Both sides of doors open; (△), doors on the community hall side open; (▲), doors on the corridor side open; (●), both sides of doors closed.

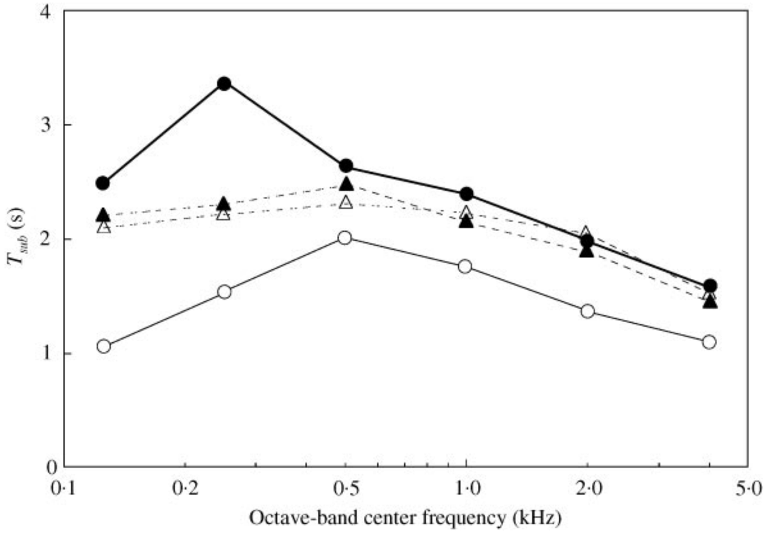

Figure 5. Subsequent reverberation time as a function of the center frequency of the 1/1 octave band: (○), Both sides of doors open; (△), doors on the community hall side open; (▲), doors on the corridor side open; (●), both sides of doors closed.

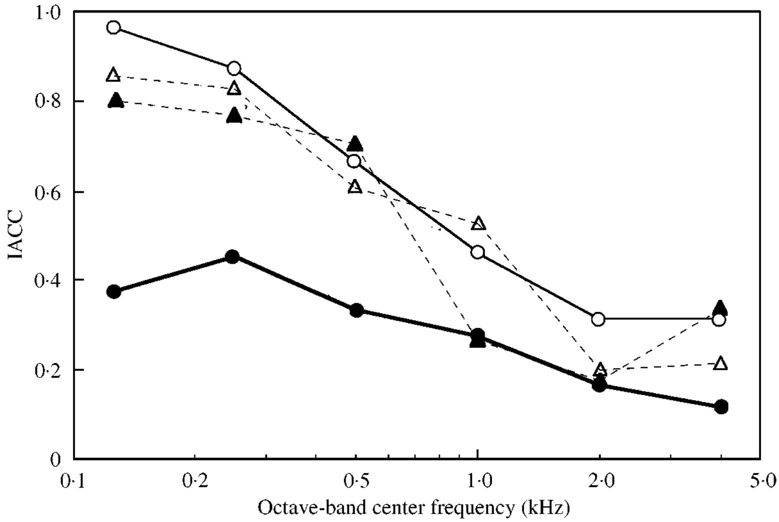

Figure 6. IACC as a function of the center frequency of the 1/1 octave band: (○), Both sides of doors open; (△), doors on the community hall side open; (▲), doors on the corridor side open; (●), both sides of doors closed.

3.3. FACTORS OF THE INTERAURAL CROSS-CORRELATION FUNCTION (IACC, τ_{IACC} AND W_{IACC})

The measurements of IACC are shown in Figure 6. The most effective and widely accepted factor in subjective preference judgment is the IACC. In general, dissimilar sound signals at the two ears (a low IACC) are preferable. The shape of the concert hall can be best designed by minimizing the IACC, and thus maximizing the subjective preference, of listeners at each seat for any kind of sound source. It is interesting that the IACC is low at lower frequencies below 500 Hz when both sides of doors are closed. τ_{IACC} at the all-pass

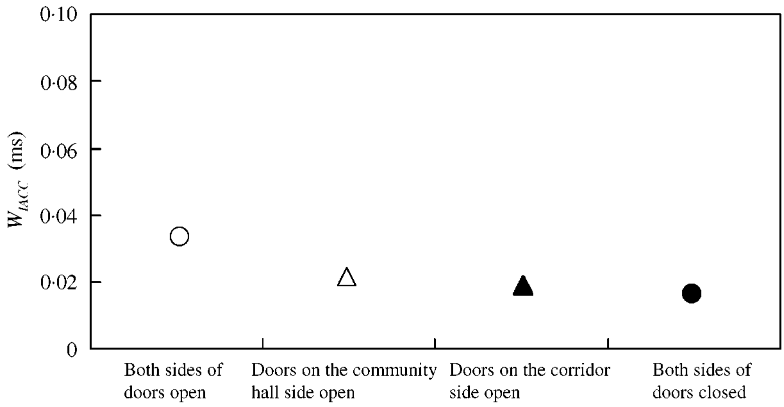

Figure 7. Results for W_{IACC} from measurements.

band never equals zero except when both sides of doors are closed. Since the IACC is low, however, a clear direction cannot be perceived for the noise source. The W_{IACC} at the all-pass band was almost constant around 0·02–0·03 as shown in Figure 7. This indicates that the ASW should be perceived as equal under all four conditions.

4. CONCLUSION

In order to describe subjective responses in the receiving room, not only the difference of SPL but also other factors, for example IACC, and T_{sub} can be measured at the same time. When both sides of doors are closed, the IACC was lower in the lower-frequency range. When doors are closed, the value of T_{sub} becomes longer.

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