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Blind directivity estimation of a sound source in a room using a surrounding microphone array

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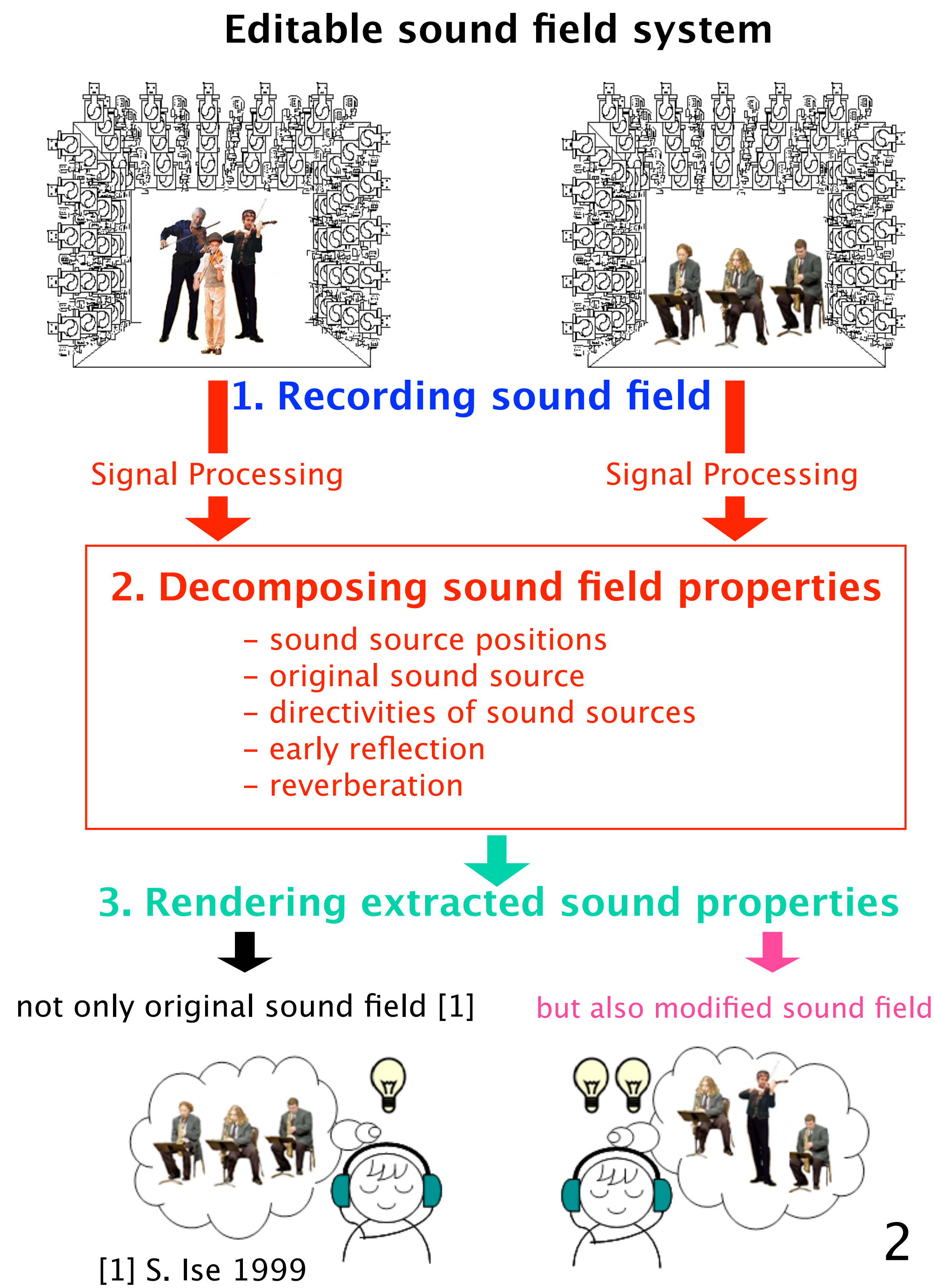
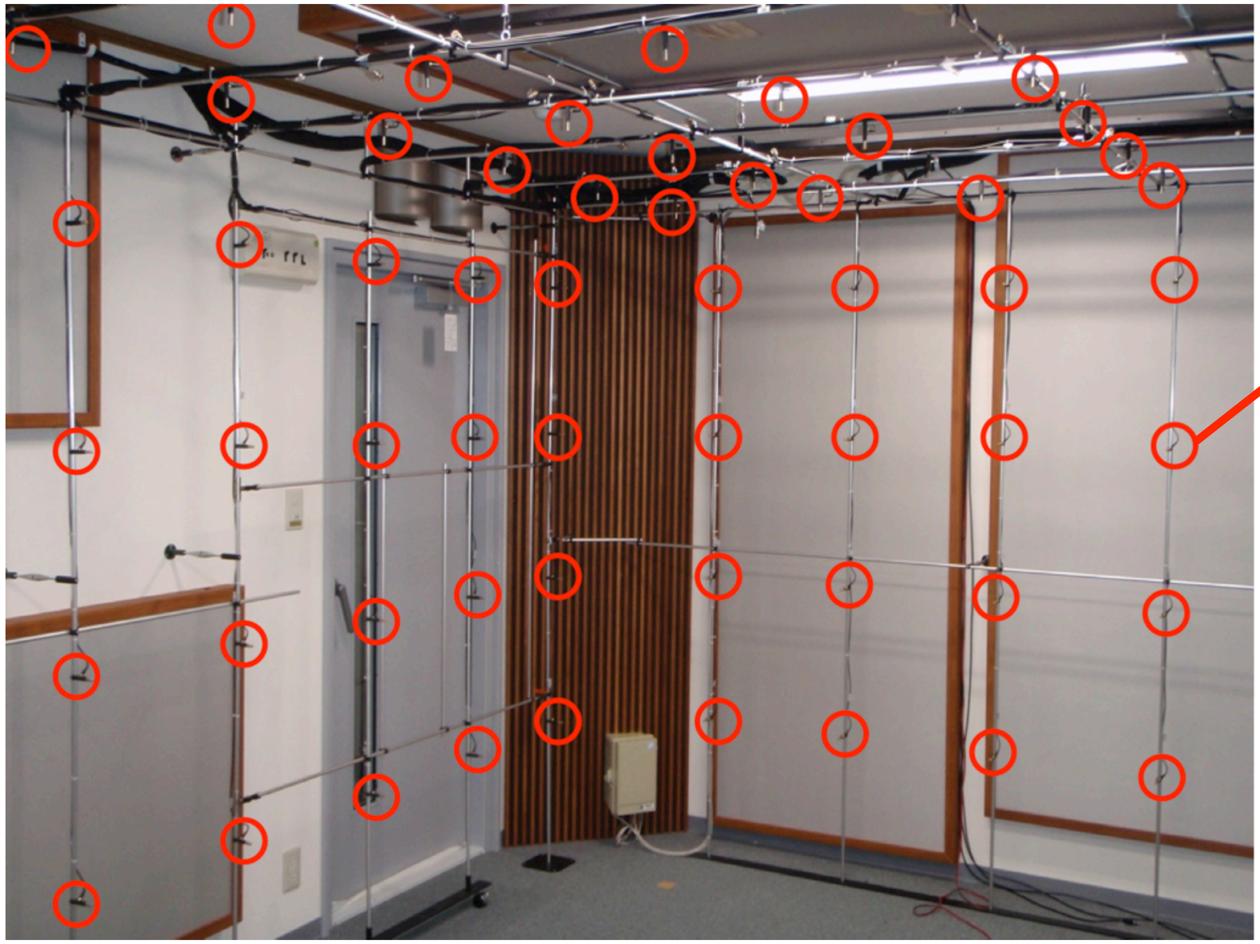
Introduction: editable sound field system

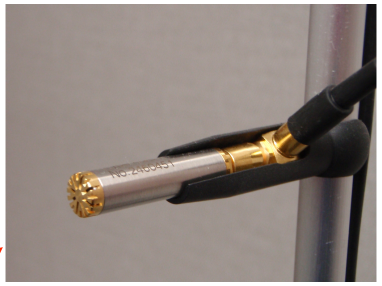
- Sensor network technologies in near future
  - multi-channel microphone array system
    - not only recording sound field
    - but also extracting sound field properties
- Contribution of Editable sound field system
  - Decomposing sound field properties accurately from recording sound information
  - Rendering extracted sound properties
    - not only original sound field
    - but also modified sound field



Design of recording sound field system

- Surrounding 157 microphone array   T. Okamoto *et al.* 2007
  - 157 microphones is installed on all four walls and the ceiling
    - All microphones are installed 30 cm inside from all four walls and the ceiling
    - They are separated from each other by 50 cm
  - Enable synchronous recording of 157 channels at the sampling frequency of 48 kHz with the linear PCM audio format using 4 PCs and 14 units of A/D controlled by PureData





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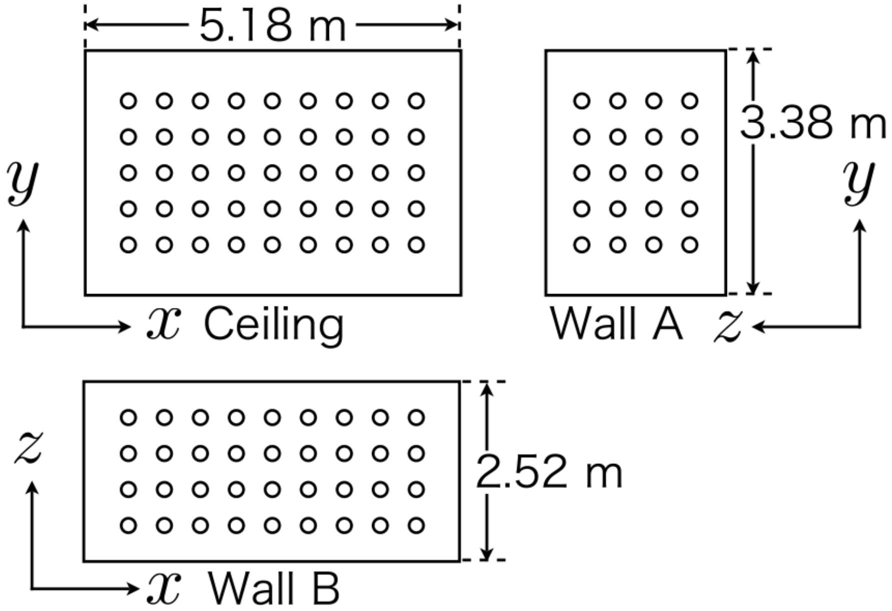
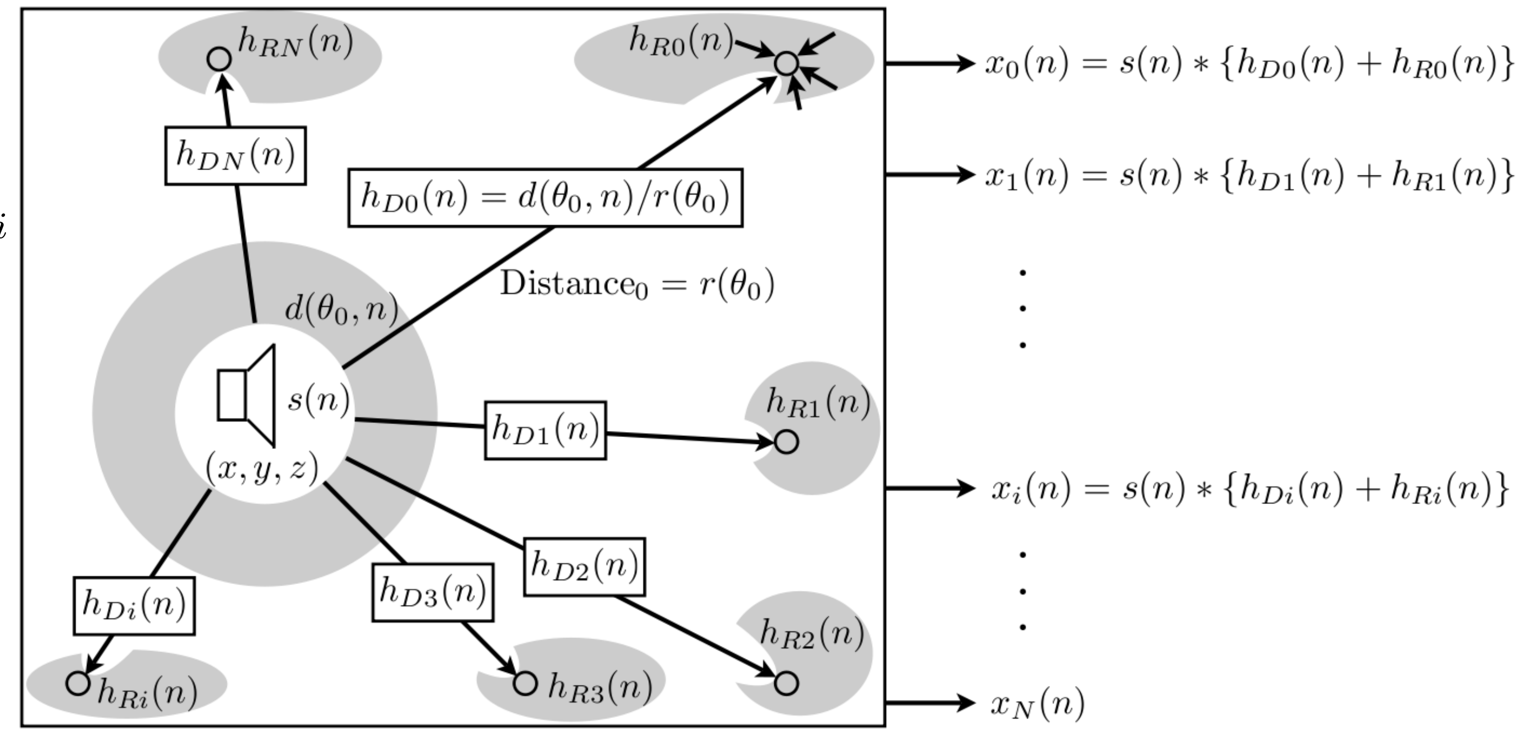


Research Aim: estimation of directivity of a sound source in a reverberant environment

- Directivity of sound source
  - Sound sources in actual environments have no omni-directional feature
  - but they do have directivity in radiation
    - human voice, instrumental sound and so on...
  - it is important to consider the directivity of a sound source according to a listening point when synthesizing a high-definition 3D sound field
- Previous studies for estimating directivity of sound source
  - In almost researches, directivity was recorded only in anechoic enviromnent
    - e.g., R. Jacques *et al.* 2005, B. FG Katz *et al.* 2007 and M. noisternig *et al.* 2009
  - Nakadai *et al.* was estimated only front direction of the sound source in a reverberant environment (2005)
- Our research aim
  - Developing method for estimating and decomposing all-around directivity of a sound source in a reverberant environment along with information of the estimated source position and the original sound signal

Proposed 1: Definition of Directivity model of a sound source in a room

- SIMO (Single-Input Multiple-Output) model including directivity of a sound source
  - $s(n)$: Source signal
  - $d(\theta_i, n)$: each directivity of a direction θ_i
  - $h_{Ri}(n)$: each reverberant component
  - $r(\theta_i)$: each distance between the source and each microphone i
  - $x_i(n)$: each observed signal at microphone i

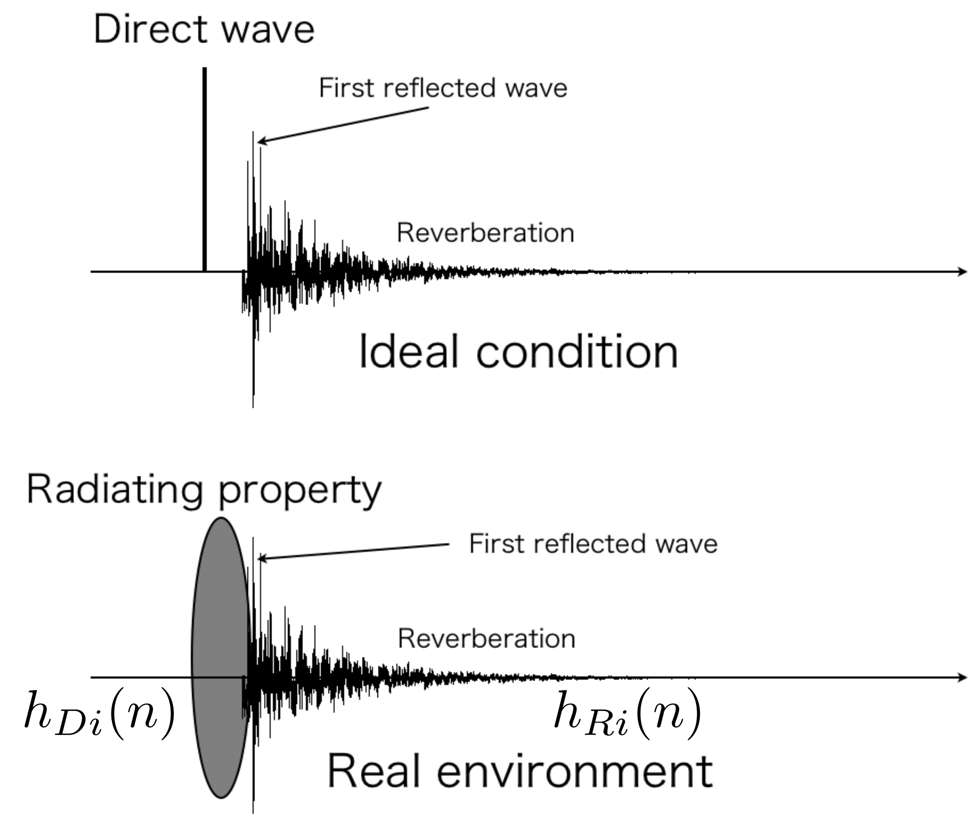


$$x(n) = \sum_{k=0}^{\infty} s(n) * d(\theta_k, n) * h(\theta_k, n) = s(n) * \sum_{k=0}^{\infty} \{d(\theta_k, n) * h(\theta_k, n)\}$$

$$x(n) = s(n) * \{d(\theta_0, n) * h(\theta_0, n) + \sum_{k=1}^{\infty} d(\theta_k, n) * h(\theta_k, n)\}$$

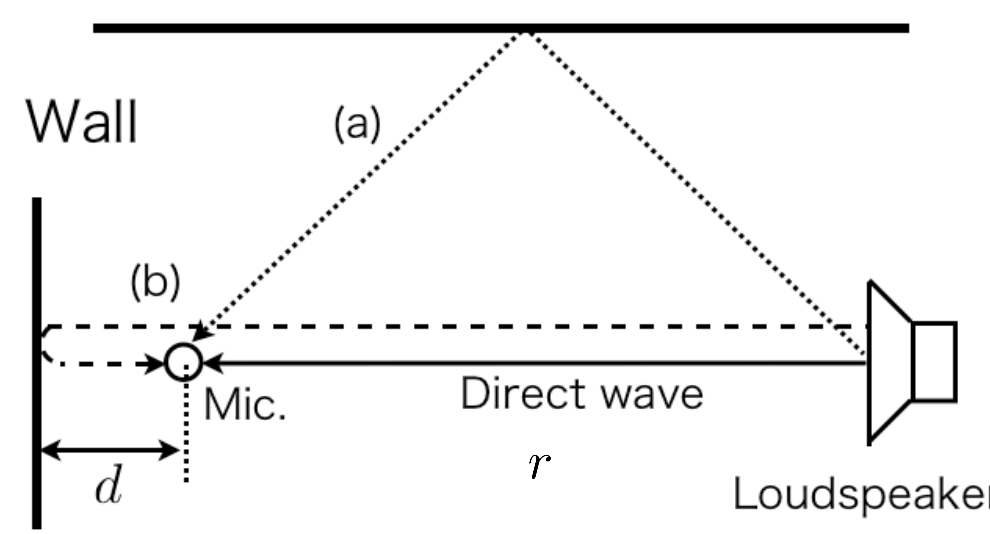
$$\begin{aligned} x(n) &= s(n) * \left\{ d(\theta_0, n) \cdot \frac{1}{r(\theta_0)} + d(\theta_0, n) * h'(\theta_0, n) + \sum_{k=1}^{\infty} d(\theta_k, n) * h(\theta_k, n) \right\} \\ &= s(n) * \{h_D(n) + h_R(n)\} \\ &= s(n) * h(n) \end{aligned}$$

Proposed 2: Decomposition of room transfer function of directivity and reverberant component



- **Relationship between directivity and reverberant component**
  - If sound source is omni-directional feature
    - direct sound component is only first response
  - When sound source has a directivity
    - directivity component is first to several samples responses

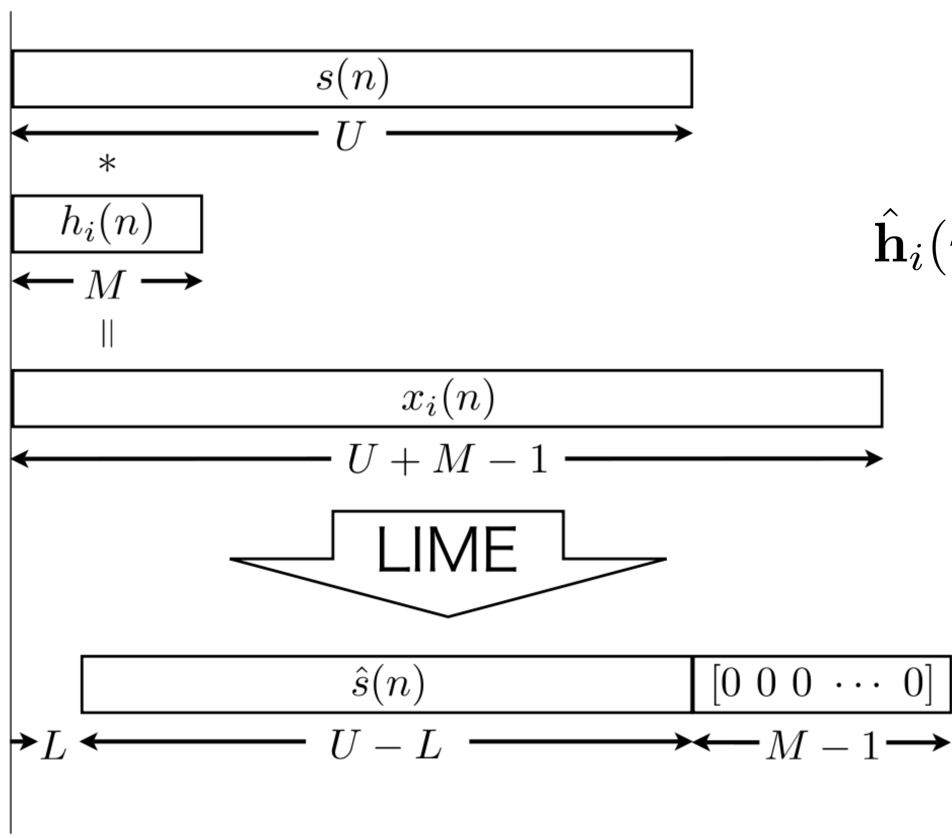
- **Decomposition of directivity and reverberant component**
  1. Estimating total impulse response
     - Blind identification or dereverberation method
  2. $h_{Di}(n)$ can be extracted as the early response from the first response to time $t = 2d/c$
     - cutting impulse response as the first t response
  3. Amplitude correction corresponding to each distance $d_i(n) = r_i h_{Di}(n)$
     - each distance is estimated from estimated sound source position



(a): First reflected wave (oblique-incidence)
(b): First reflected wave (head-on incidence)

Estimating room impulse response using estimated sound source signal

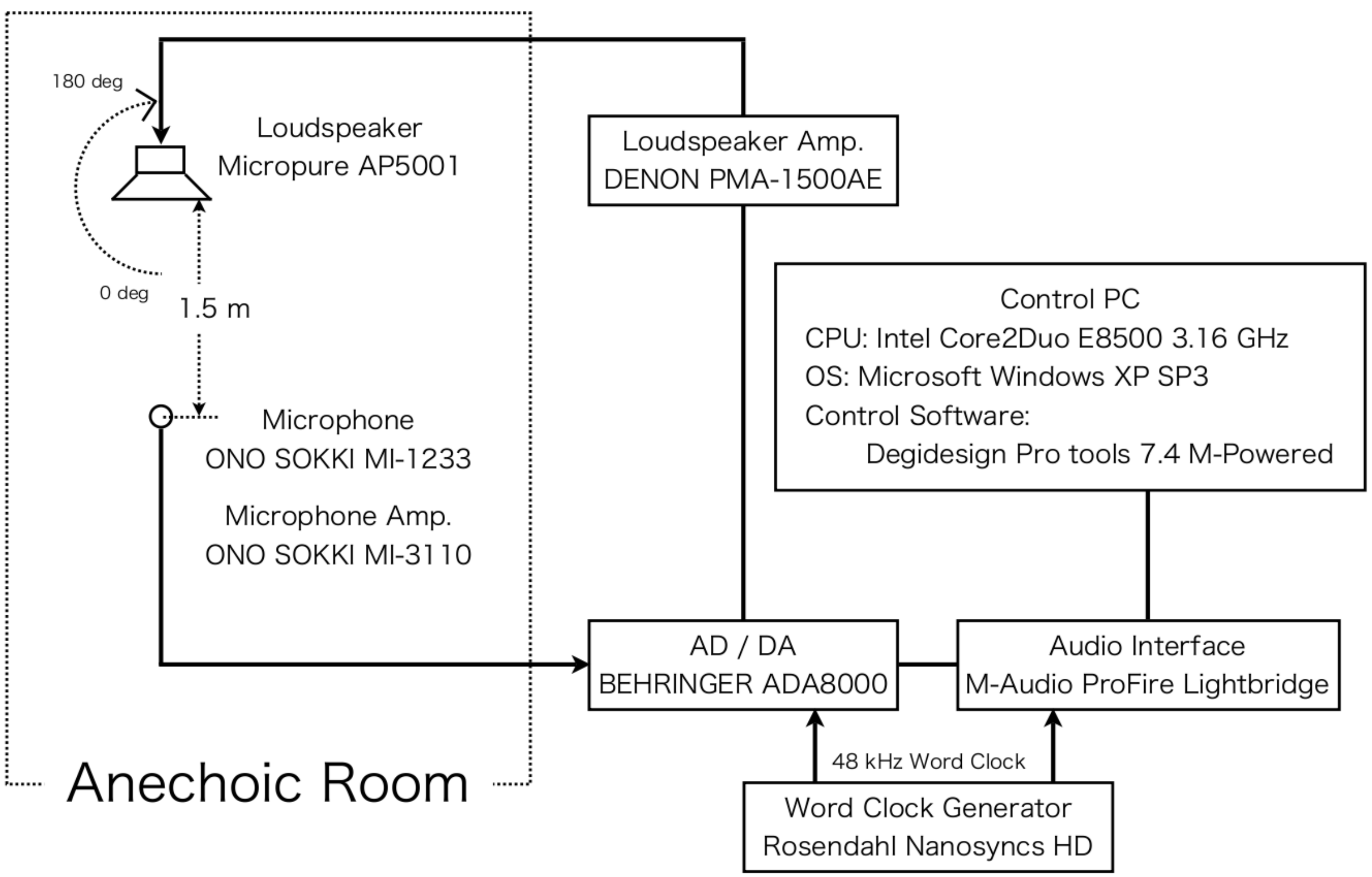
- Estimating room impulse response
  - Blind identification is difficult problem
  - Estimating room impulse response using estimated sound source and received signals
    - Sound source signal can be estimated by a dereveberation method
    - We proposed a dereverberation algorithm, White-LIME

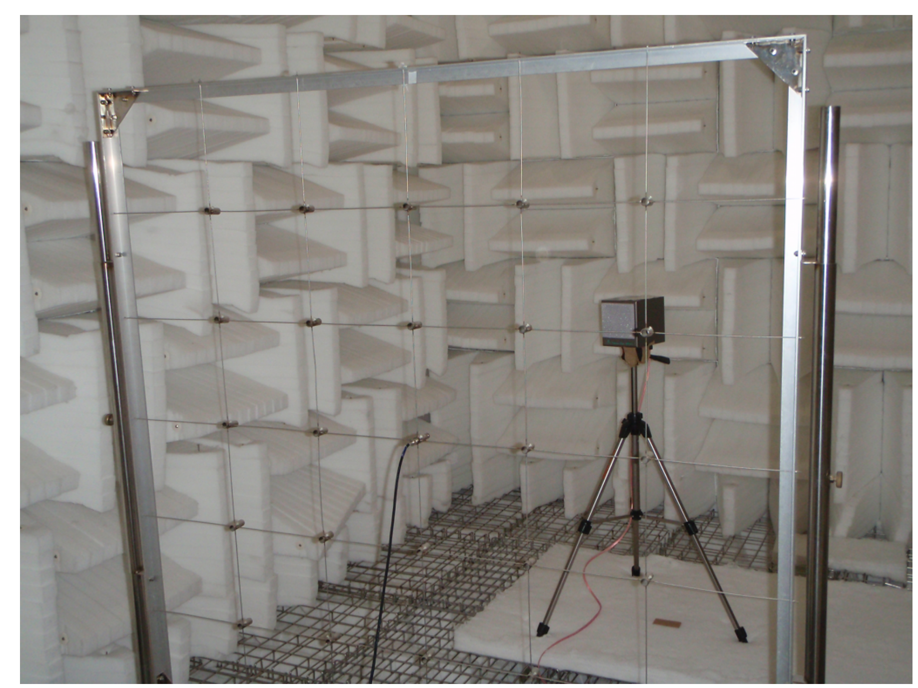


$$\hat{\mathbf{h}}_i(n) = (\mathrm{E}\{\hat{\mathbf{s}}(n)\hat{\mathbf{s}}^{\mathrm{T}}(n)\})^{-1}\mathrm{E}\left[\begin{pmatrix} x_i(n)s(n) \\ x_i(n-1)s(n-1) \\ \vdots \\ x_i(n-M+1)s(n-M+1) \end{pmatrix}\right]$$

Directivity measurement in an anechoic chamber

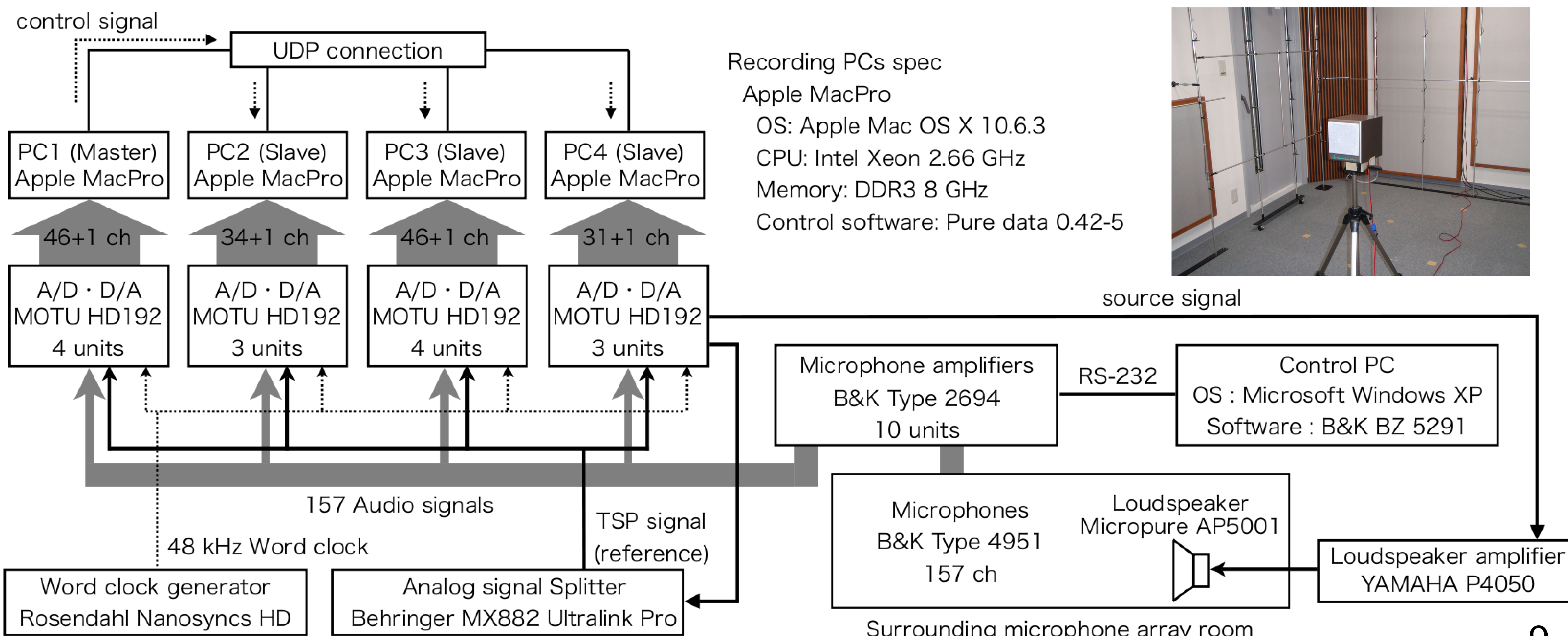
- Directivity measurement as impulse response
  - 1-way loudspeaker (Micropure; AP5001)
  - Measurements were taken from 0 deg to 180 deg in horizontal direction with a clockwise rotation by 15 deg in an anechoic chamber
  - Time Stretched Pulse (TSP) signal was used for measurement
    - The impulse responses for 13 directions were measured





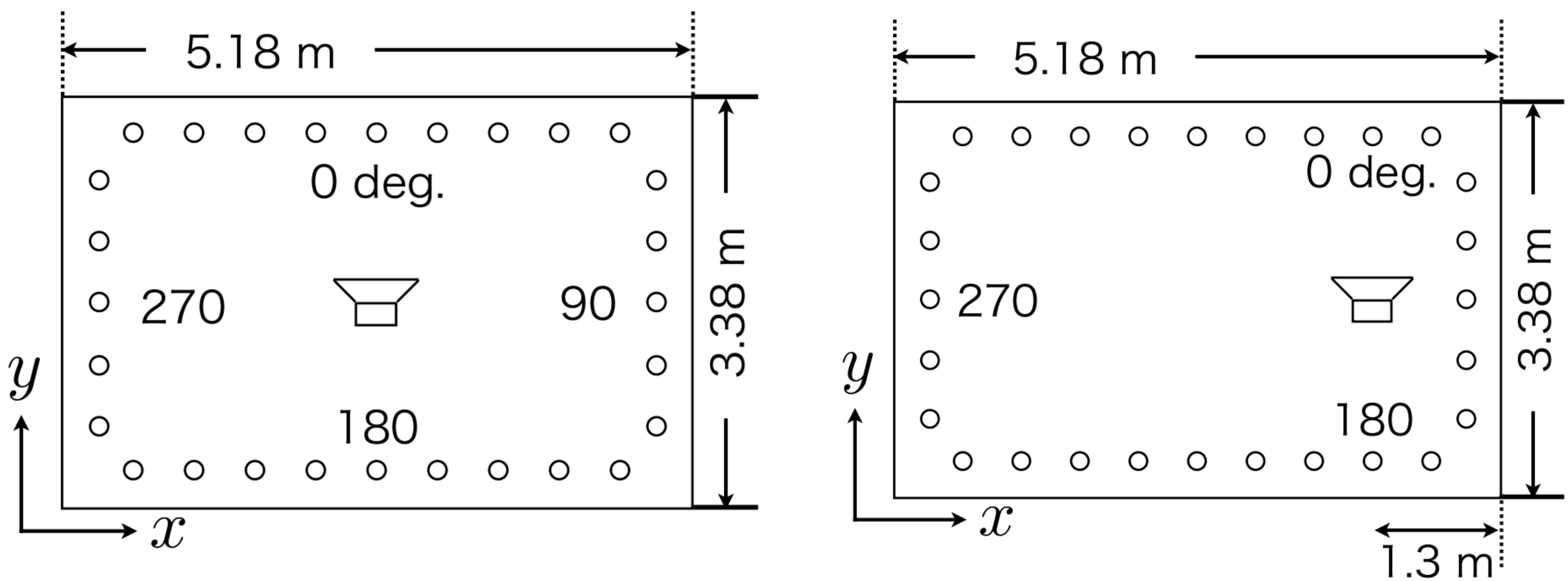
Room impulse response measurement in a room

- Measurement of room impulse response which includes both directivity of a sound source and reverberant component
  - Sound signal and loudspeaker are the same as an anechoic measurement
  - Measurement was taken by using surround 157-microphone array



Recording conditions

- Two patterns of the arrangements of the loudspeaker
  - 28 microphones (z = 1.0 m) in the room
  - The reverberation time of this room was about 0.15 s
  - The order of the room impulse responses was 7200

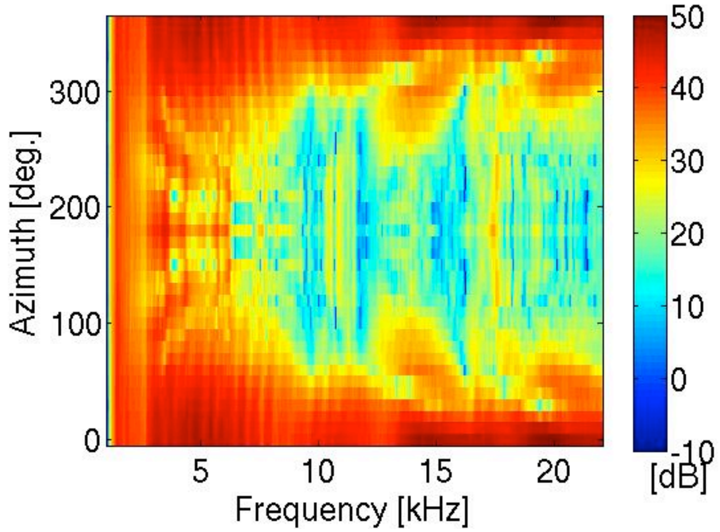


Pattern A Pattern B

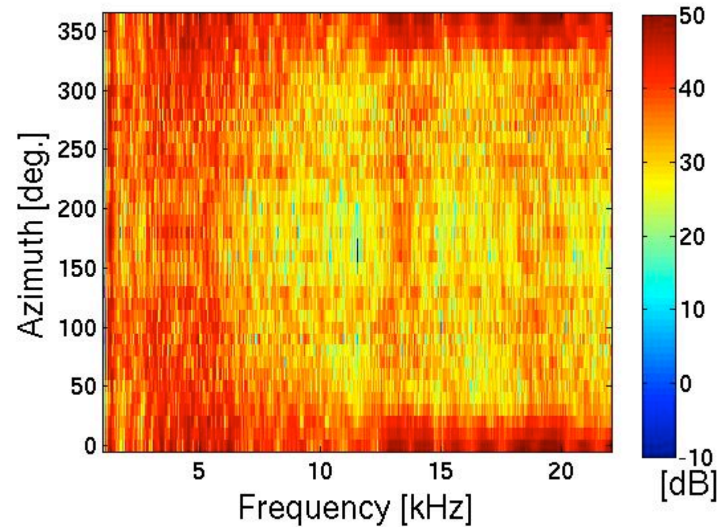
Simulations

- Calculating 157 received signals $x_i(n)$
  - Convolving the source signal $s(n)$ to each measured impulse response $h_i(n)$
  - The source signal was a musical piece (2.7 s) from RWC music database
  - The measured impulse responses were downsampled 48 kHz to 44.1 kHz
    - The order of all responses was shortened from 7200 to 6615
- Estimating impulse responses
  - The estimated source signal $\hat{s}(n)$ was obtained using White-LIME
  - Each estimated impulse response $\hat{h}_i(n)$ was calculated from each observed signal $x_i(n)$ and the estimated source signal $\hat{s}(n)$
    - The average of the score of the SDR of each original response $\hat{h}_i(n)$ was 62.6 dB
    - The estimated responses can be inferred accurately
- Decomposing directivity from room impulse responses
  - The distance between each microphone and the wall was 30 cm
  - The clipping length of each estimated response was approximately $44{,}100 \times 2 \times 0.6/340 \approx 78$ taps
  - Amplitude correction was taken from estimated sound source position and each microphones

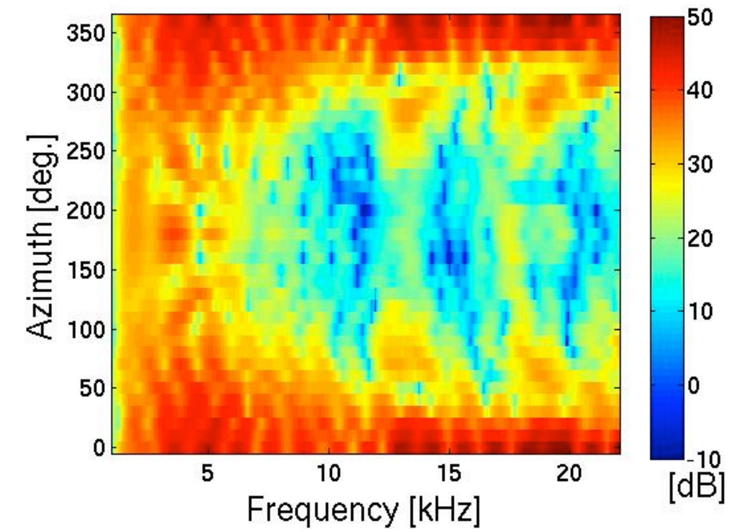
Results of Amplitude-frequency characteristics



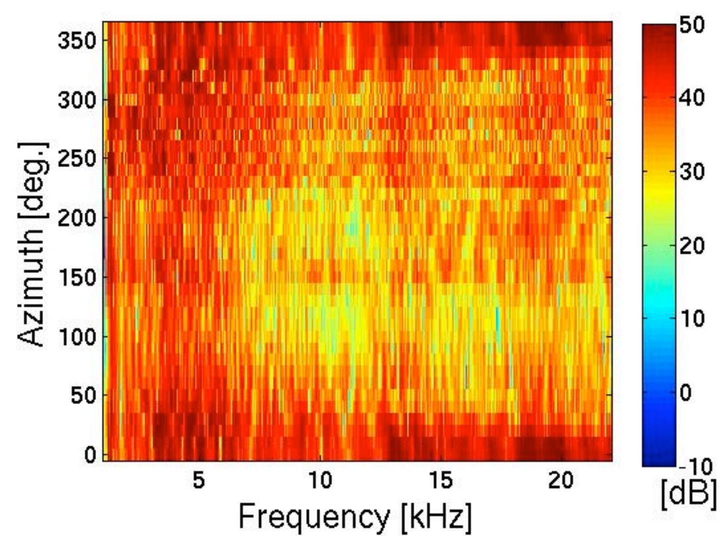
Original directivity



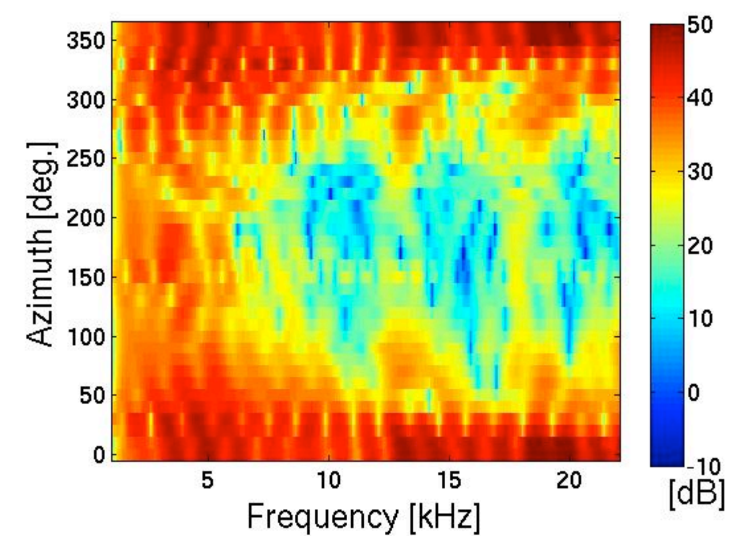
Measured impulse response (A)



Estimated directivity (A)

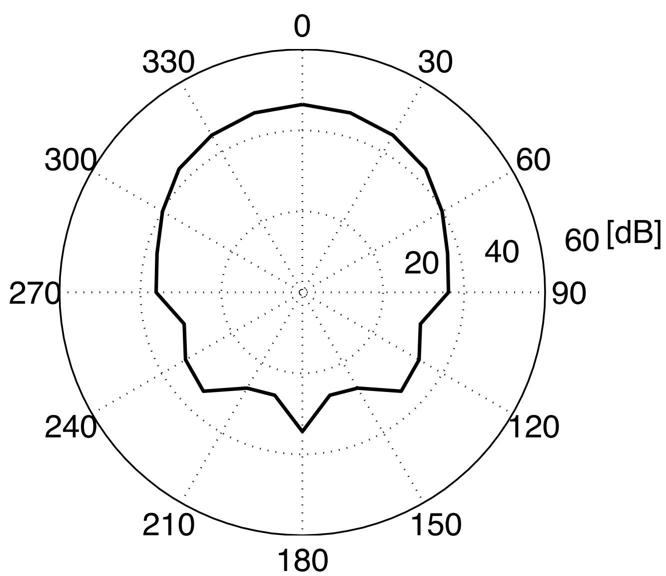


Measured impulse response (B)

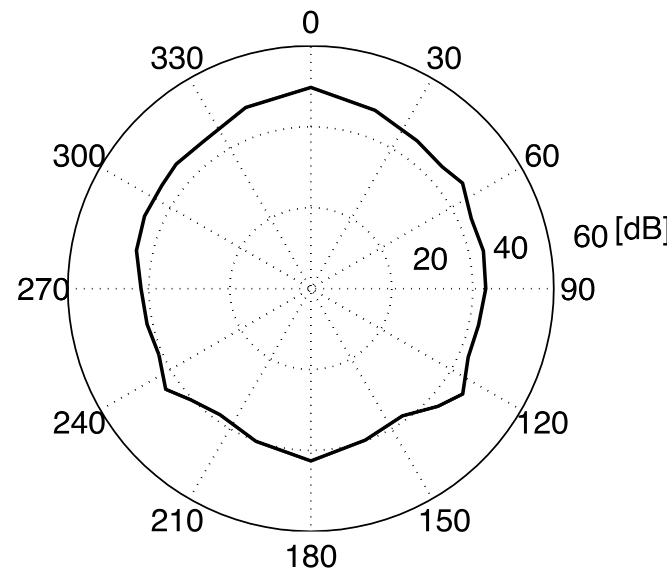


Estimated directivity (B)

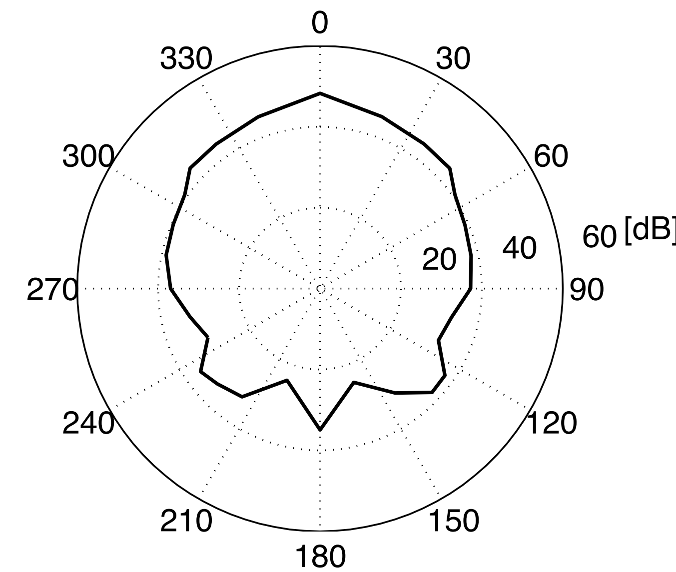
Results of 2 kHz in the 1/3 octave-band



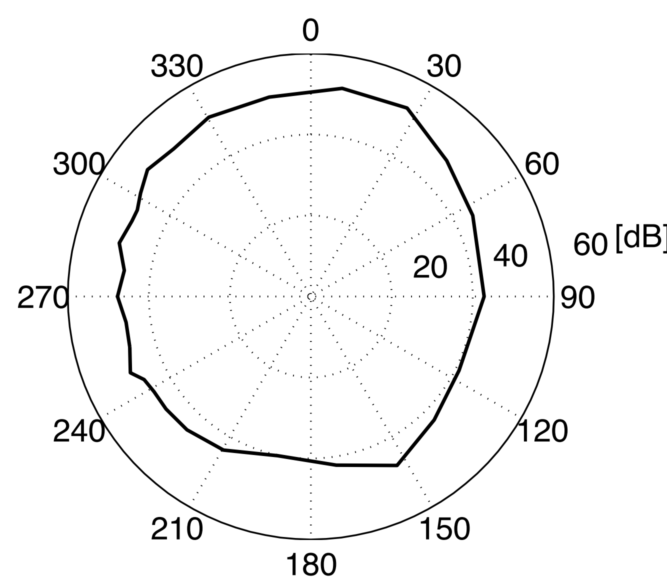
Original directivity



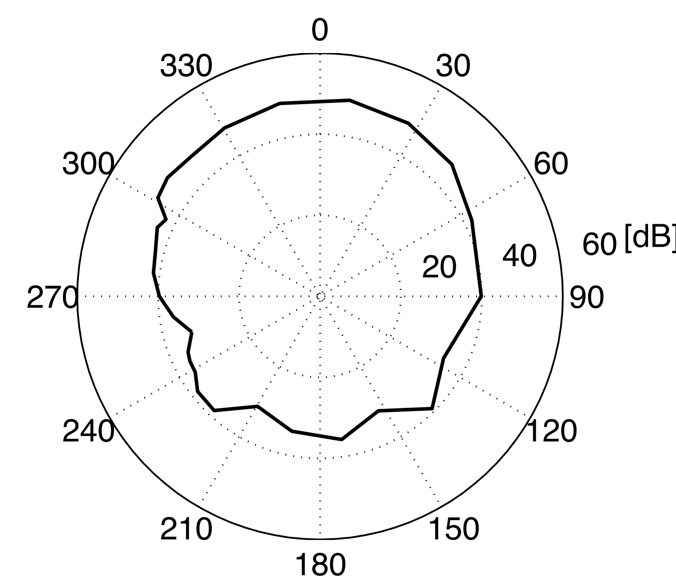
Measured impulse response (A)



Estimated directivity (A)



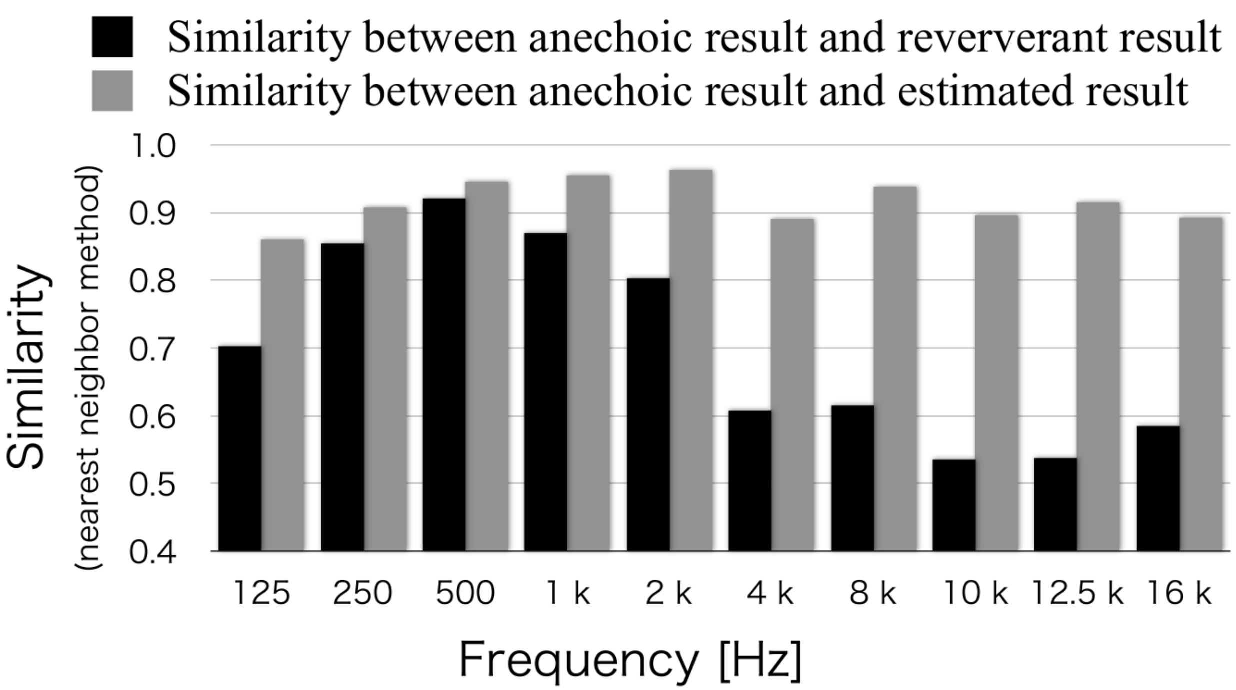
Measured impulse response (B)



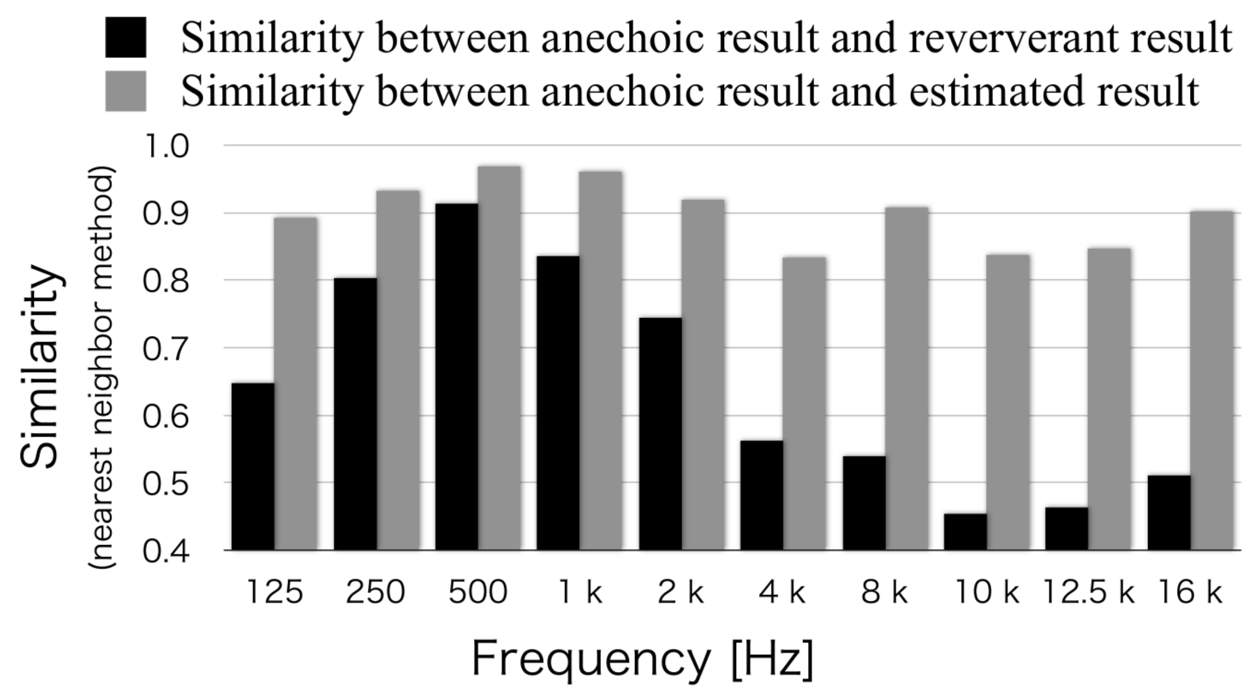
Estimated directivity (B)

Results of similarity analysis

- Confirming the effectiveness of our proposed method
  - Similarity based on the nearest neighbor method
    - Pattern vector $P(f) = (x_{\theta_1}(f), x_{\theta_2}(f). \cdots, x_{\theta_N}(f))$
    - Similarity $S(P_1(f), P_2(f)) = 1.0 - \frac{|P_1(f) - P_2(f)|}{|P_1(f)|}$



Pattern A



Pattern B

  - Our proposed method can estimate directivity patterns that closely resemble real ones up to around 16 kHz

Concluding remarks

- Conclusions
  - Estimating directivity of a sound source in a reverberant environment
    - Using information of the estimated source position and the original sound signal
    - Cutting estimated impulse responses based on the distance between each microphone and nearest wall
    - The simulation results demonstrate the availability of our proposed method
- Future works
  - Our method presents a problem in estimating the sound source directivity when the walls and microphones are close to each other
    - Widening the applicability of our pro- posed method is development of a method extracting the early part from an impulse response with overlapped reflections
  - Estimating directivity of a sound source not only 2-dimensional direction but also 3-dimensional direction for 3D sound field analysis and synthesis