Sound Field Recording and Reproduction Using Small Number of Microphones and Loudspeakers

Shoichi Koyama

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About me

➢ **Shoichi Koyama, Ph.D.**

- 2007 B.E. and 2009 M.S. degrees from UTokyo
- 2009 – 2014: Nippon Telegraph and Telephone Corp.
- 2014: Ph.D. (Inf Sci&Tech) from UTokyo
- 2014 – present: Assistant Prof. (Research Associate) at UTokyo
- 2016 – present: Visiting researcher at Paris Diderot University (Paris7) / Institut Langevin

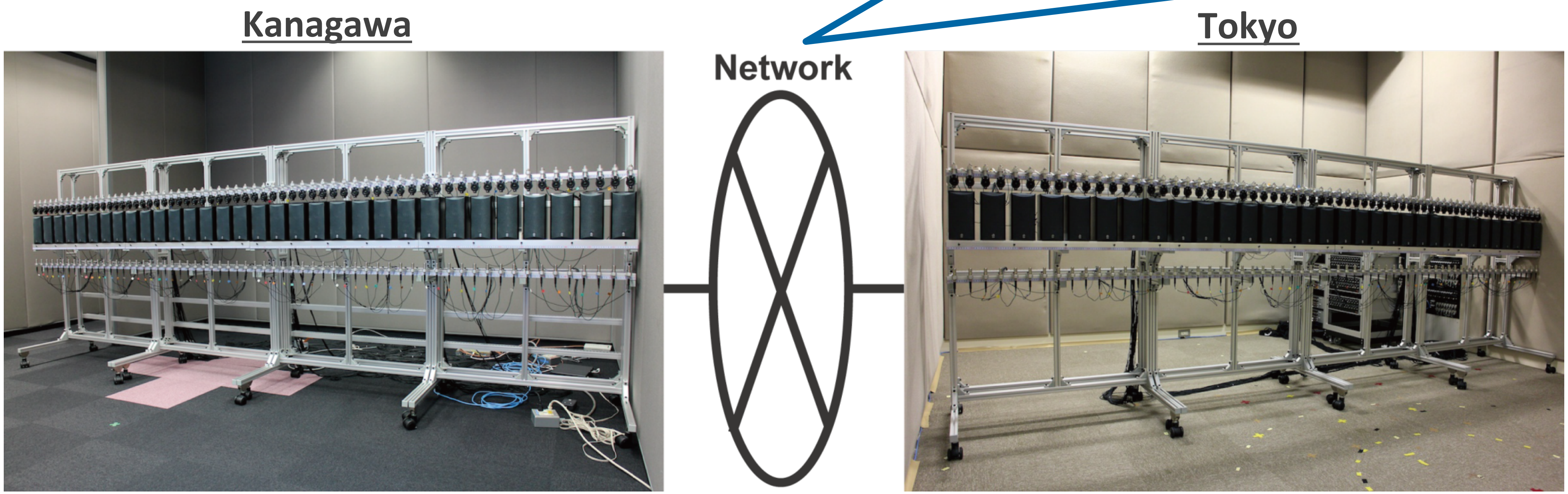
UTokyo and Institut Langevin

- **The University of Tokyo (Tokyo, Japan):**
  - Our lab has just started on Apr. 2014
  - 5 staffs including Prof. Hiroshi Saruwatari and 9 students
  - Audio, speech, and music signal processing

- **Institut Langevin (Paris, France):**
  - Research institute for "*physical waves*" including optics and acoustics
  - Working with Prof. Laurent Daudet
  - Staying for 2 years (until Mar. 2018)

Real-time Sound Field Transmission System

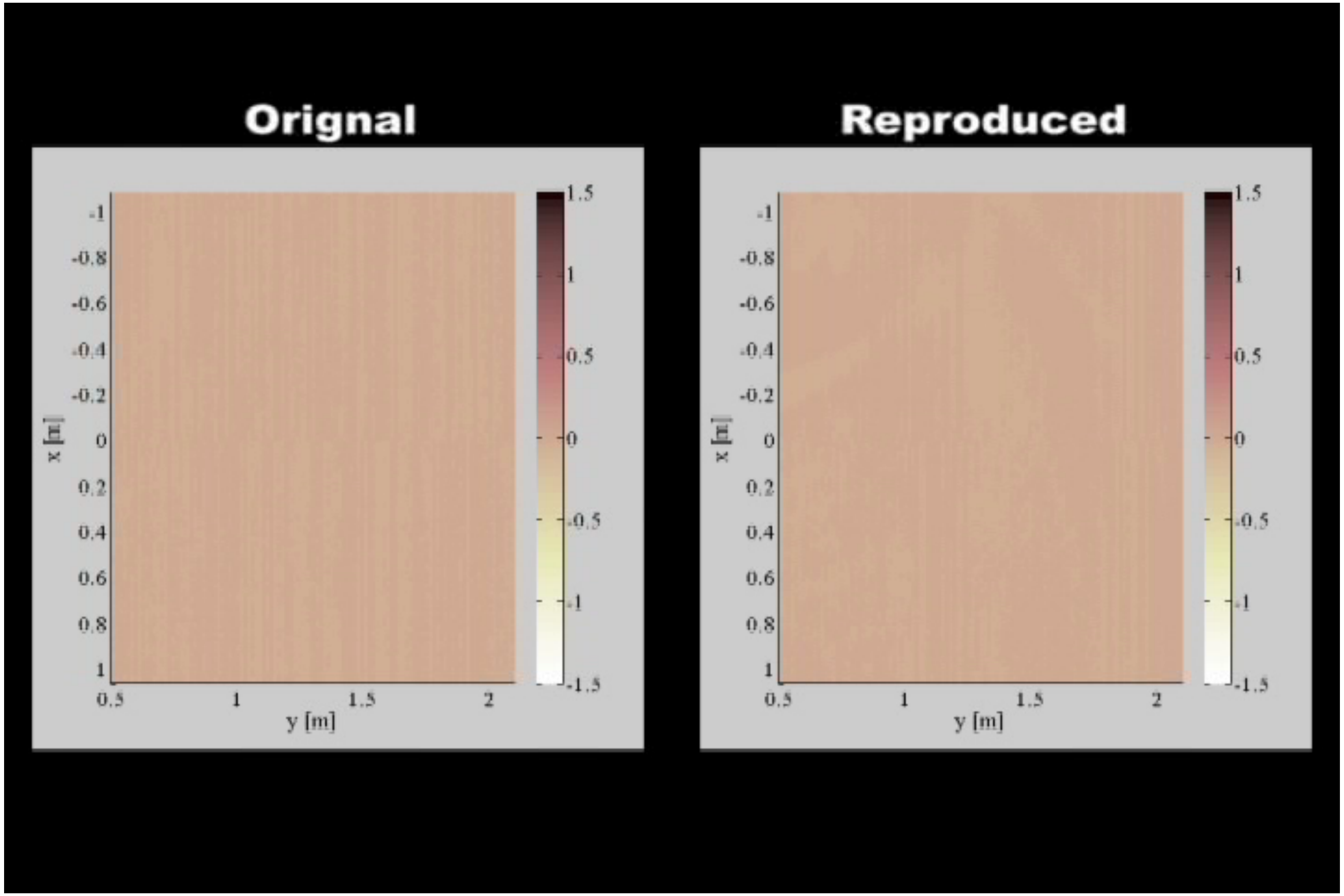
System developed while I worked at NTT [Koyama+ IEICE Trans 2014]



- Loudspeakers (for high freq.): 64, 6cm intervals
- Loudspeakers (for low freq.): 32, 12cm intervals
- Microphones: 64, 6cm intervals
- Array size: 3.84 m
- Sampling freq.: 48 kHz, Delay: 152 ms

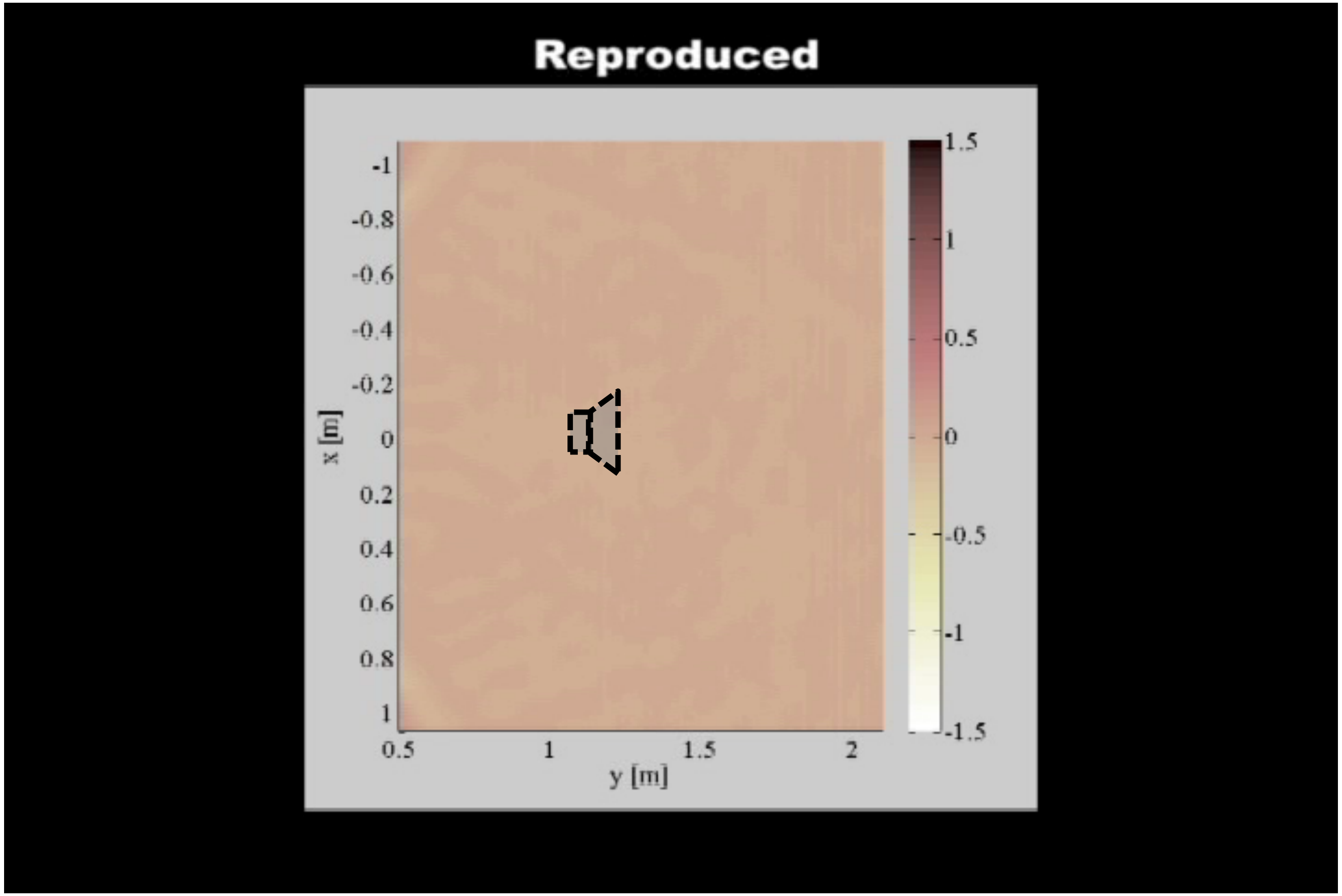
Visualization of Reproduced Sound Field

- Source signal: Low-passed pulse (0 – 2.6kHz)
- Source: Loudspeaker, Position: (-1.0, -1.0, 0.0) m



Visualization of Reproduced Sound Field

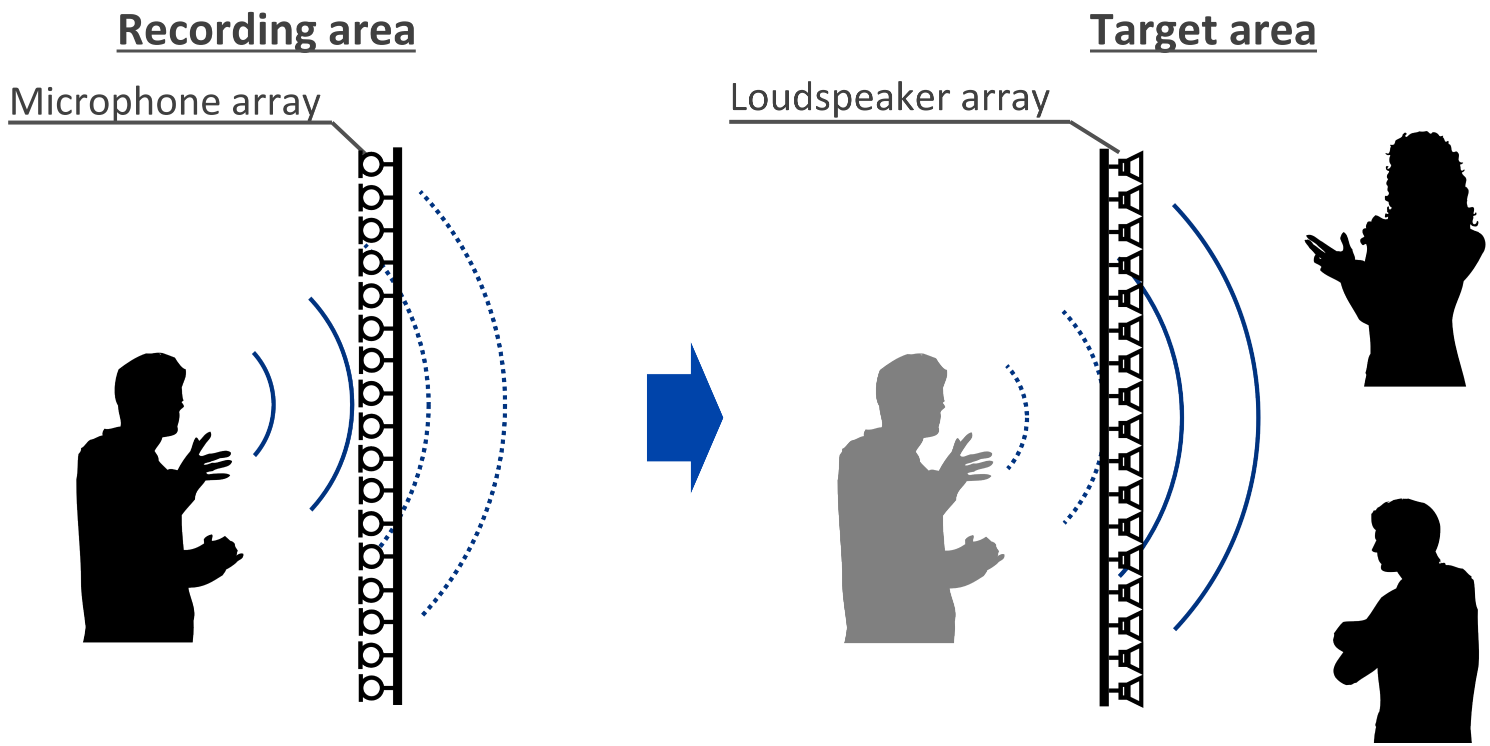
- Source signal: Low-passed pulse (0 – 2.6kHz)
- Source: Loudspeaker, Position: (0.0, -1.0, 0.0) m, 2.0 m forward shift



[Koyama+ IEEE TASLP 2012]

Today's Topic

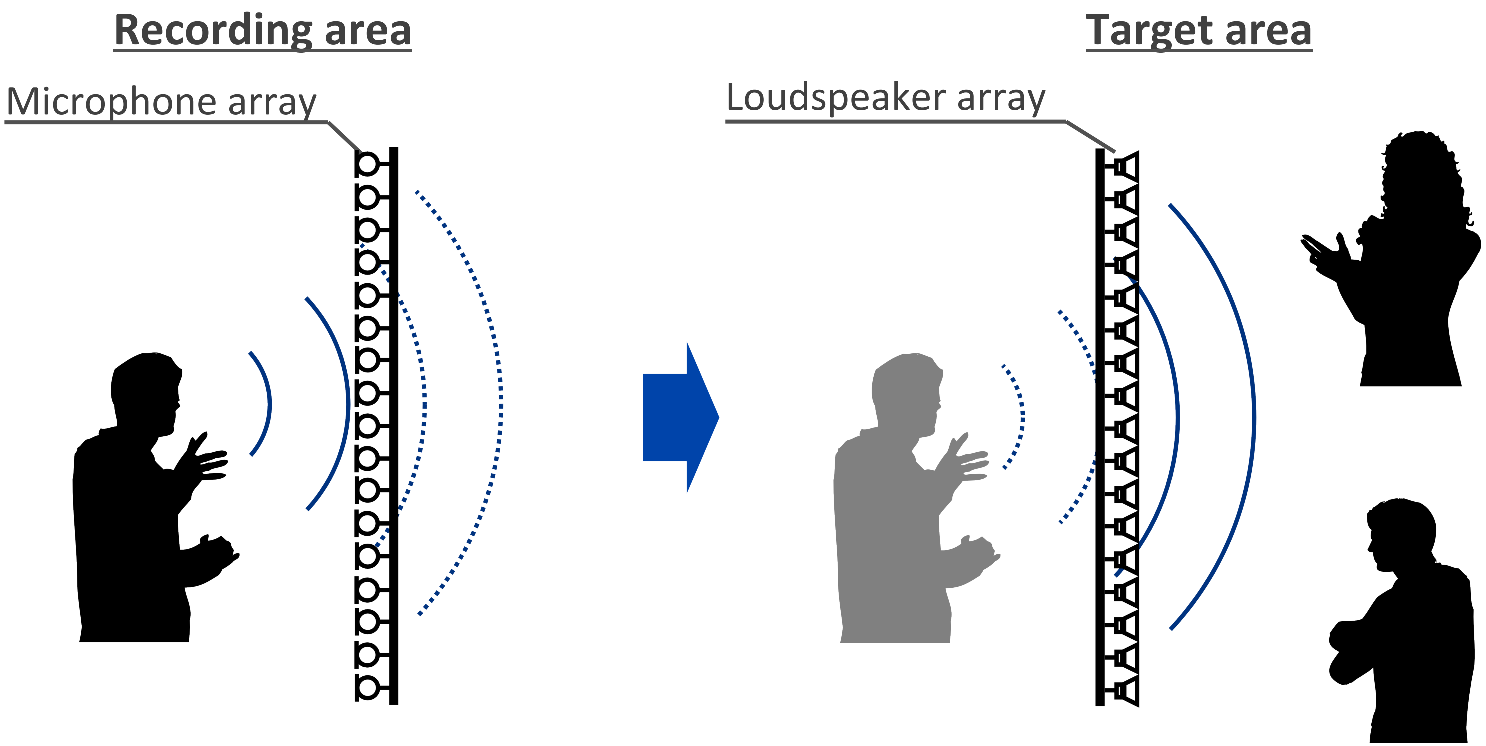
How to reduce microphones and loudspeakers in recording and reproduction?



- Insufficient number of array elements: ***spatial aliasing artifacts***
  - Low reproduction accuracy at high frequencies
  - Inaccurate frequency characteristics (*coloration effect*)

Today's Topic

How to reduce microphones and loudspeakers in recording and reproduction?



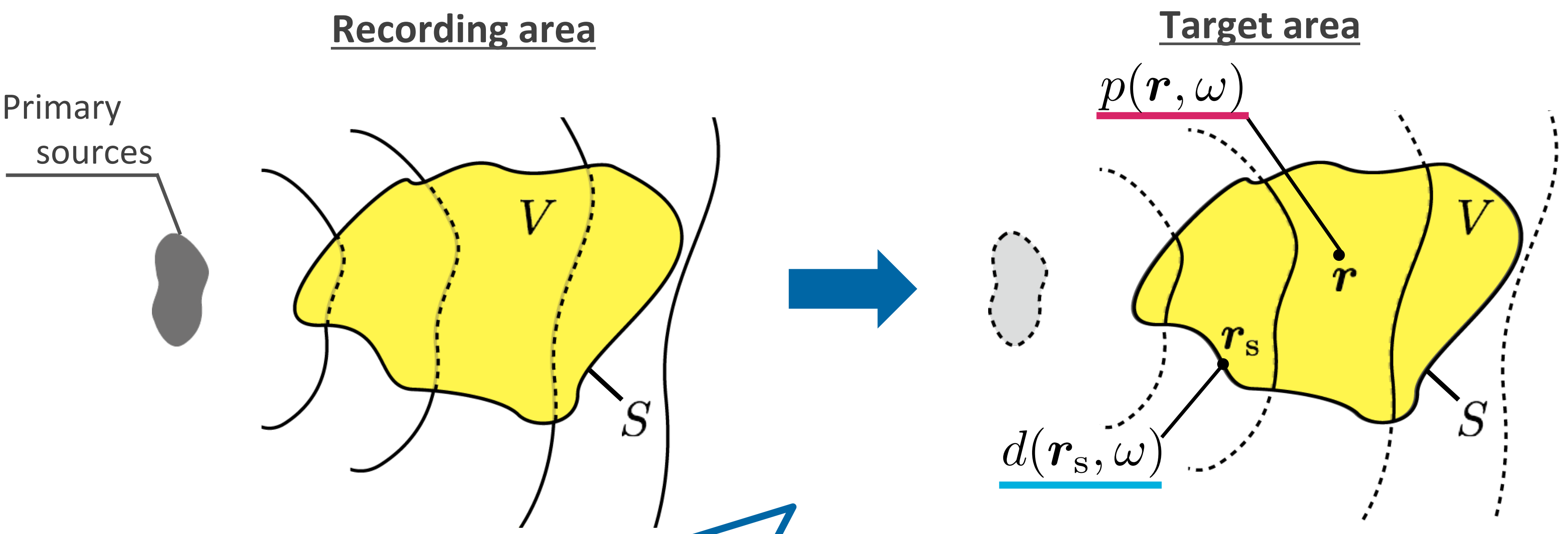
Improve reproduction accuracy using prior information

- **Reduction of the number of microphones** [Koyama+ ICASSP2014, 2015]
  - Improve reproduction accuracy within predefined near-field source area
- **Reduction of the number of loudspeakers** [Ueno+ ICASSP2017, HSCMA2017]
  - Improve reproduction accuracy within predefined listening area

REDUCTION OF THE NUMBER OF MICROPHONES

Sound Field Recording and Reproduction

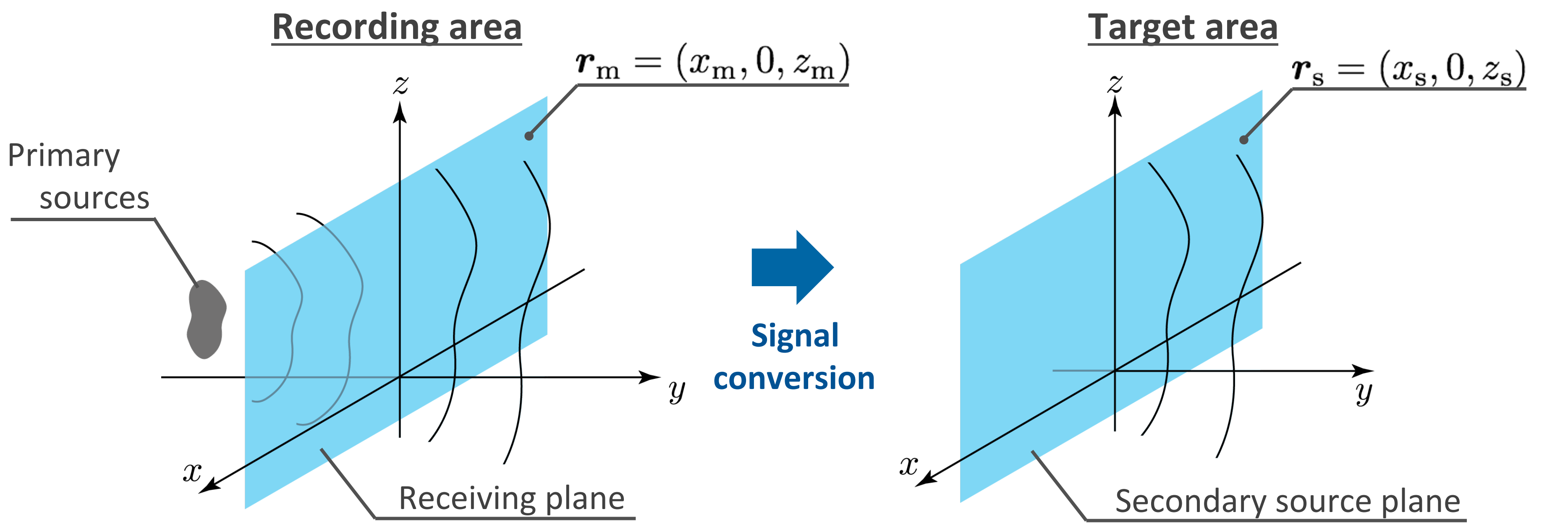
Obtain driving signals $d(\boldsymbol{r}_{\mathrm{s}}, \omega)$ of secondary sources (= loudspeakers) arranged on S to reconstruct desired sound field inside V



When sound pressures at multiple positions are only known in the recording area, typical strategy to obtain the driving signals is <u>plane wave or harmonic decomposition of the captured sound field</u>

Conventional: WFR filtering method

[Koyama+ IEEE TASLP 2013]



$$p(\mathbf{r}_{\rm m}, \omega) = \sum_n b_n(\omega)\varphi_n(\mathbf{r}_{\rm m}, \omega) \quad \Rightarrow \quad d(\mathbf{r}_{\rm s}, \omega) = \sum_n c_n(\omega)\varphi_n(\mathbf{r}_{\rm s}, \omega)$$

Received signals: $p(\mathbf{r}_{\rm m}, \omega)$

Plane wave: $\varphi_n(\mathbf{r}_{\rm m}, \omega)$

Driving signals: $d(\mathbf{r}_{\rm s}, \omega)$

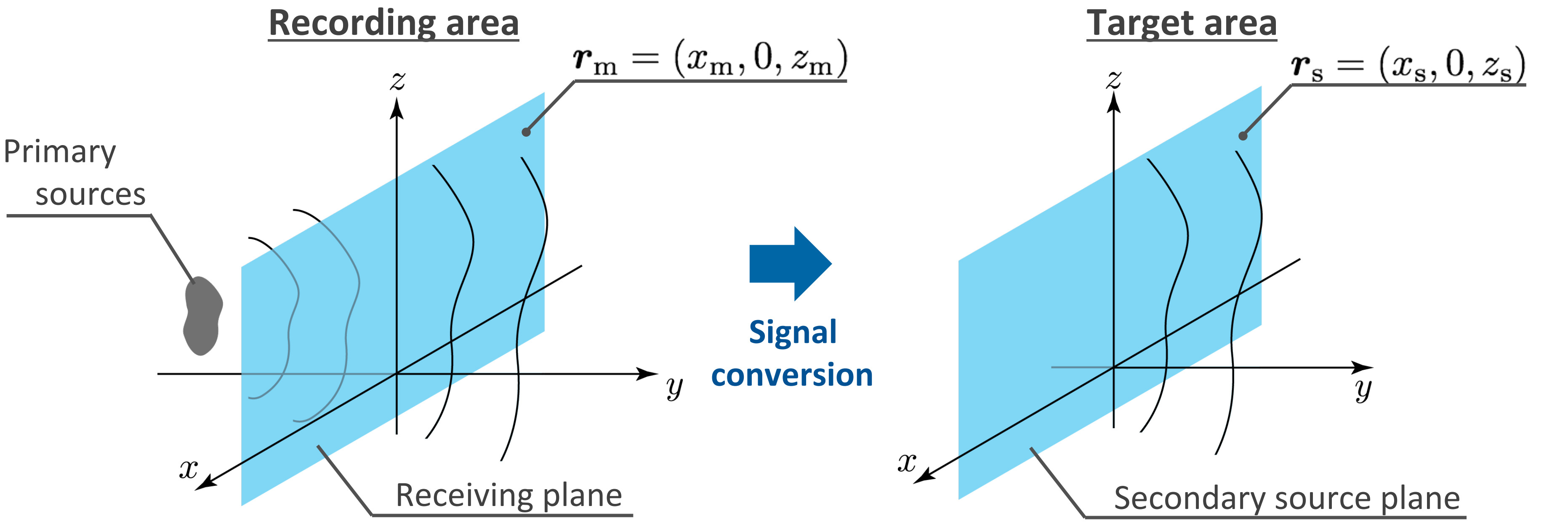
Plane wave: $\varphi_n(\mathbf{r}_{\rm s}, \omega)$

Each plane wave determines entire sound field

Signal conversion based on plane-wave decomposition

Conventional: WFR filtering method

[Koyama+ IEEE TASLP 2013]



$$p(\mathbf{r}_{\mathrm{m}}, \omega) = \sum_n b_n(\omega) \varphi_n(\mathbf{r}_{\mathrm{m}}, \omega) \Rightarrow d(\mathbf{r}_{\mathrm{s}}, \omega) = \sum_n c_n(\omega) \varphi_n(\mathbf{r}_{\mathrm{s}}, \omega)$$

Received signals: $p(\mathbf{r}_{\mathrm{m}}, \omega)$ — Plane wave: $\varphi_n(\mathbf{r}_{\mathrm{m}}, \omega)$ — Driving signals: $d(\mathbf{r}_{\mathrm{s}}, \omega)$ — Plane wave: $\varphi_n(\mathbf{r}_{\mathrm{s}}, \omega)$

Each plane wave determines entire sound field

Spatial aliasing artifacts due to plane wave decomposition ➡ Significant error above spatial Nyquist freq of microphone array

Sound Field Decomposition in Recording

$$\underbrace{p(\mathbf{r},\omega)}_{\text{Received signals}} = \sum_n b_n(\omega)\underbrace{\varphi_n(\mathbf{r},\omega)}_{\text{Basis function}}$$

Received signals

Basis function

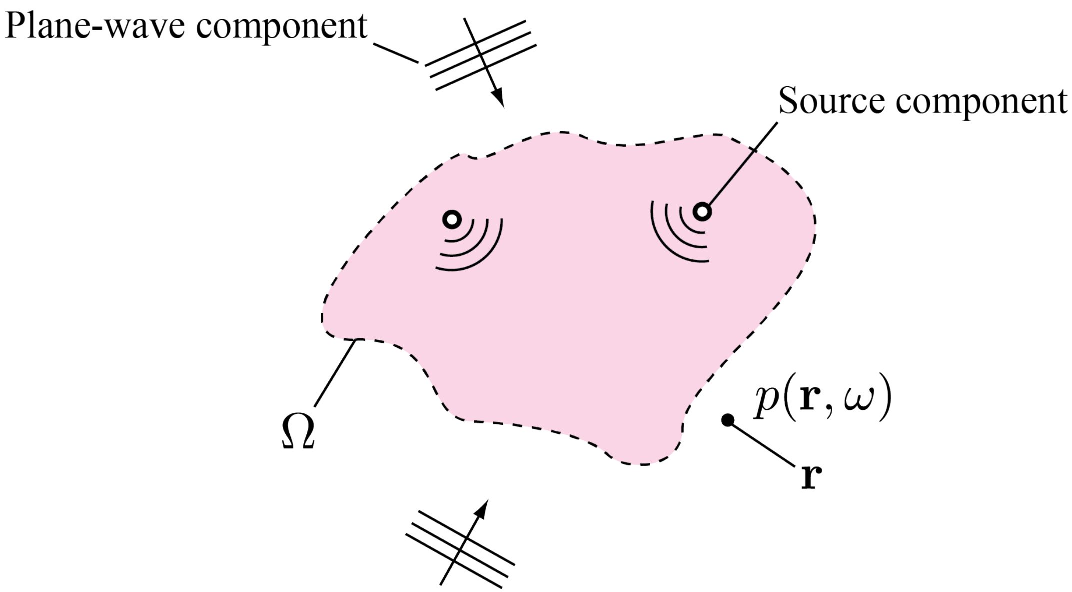
- Plane wave / harmonic decomposition suffers from spatial aliasing artifacts because many basis functions are used
- Observed signals should be represented by a few basis functions for accurate interpolation of sound field
- Appropriate basis function may be close to pressure distribution originating from near-field sound sources
- To obtain driving signals of loudspeakers, basis functions must be elementary solutions of Helmholtz equation (e.g. Green functions)

➡ ***Sound field decomposition***

Sound field decomposition into elementary solutions of Helmholtz equation is necessary

Generative model of sound field

[Koyama+ ICASSP 2014]



- **Inhomogeneous Helmholtz eq.**

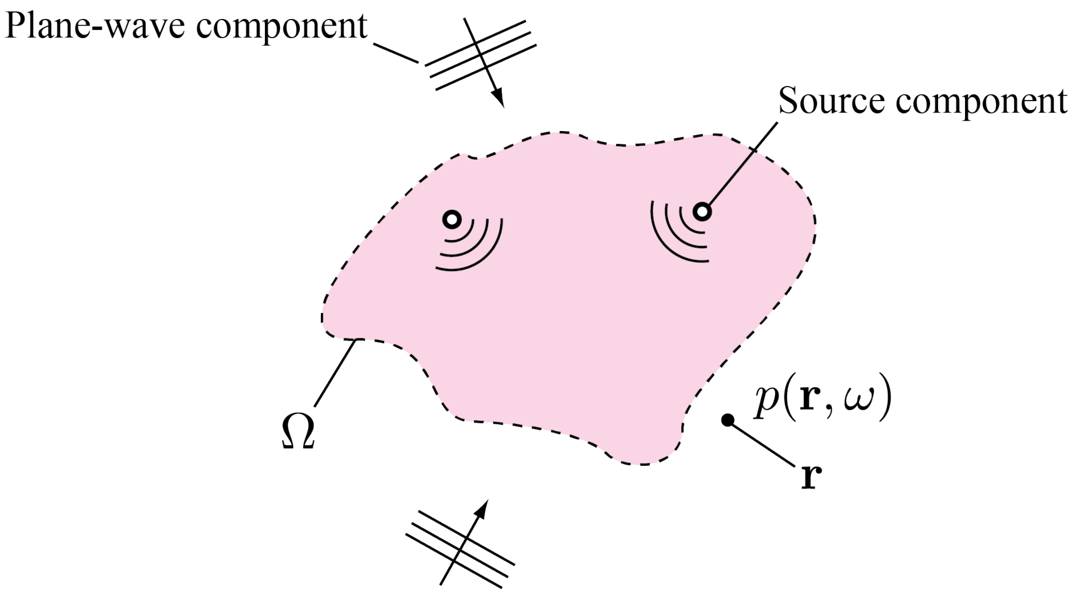
Distribution of source components

$$\begin{cases} \left(\nabla^2 + k^2\right) p(\mathbf{r}, \omega) = -Q(\mathbf{r}, \omega) \\ \text{Unknown boundary condition on room boundary} \end{cases}$$

Sound field consisting of near-field source and far-field plane-wave components

Generative model of sound field

[Koyama+ ICASSP 2014]



- **Inhomogeneous Helmholtz eq.**

$$p(\mathbf{r}) = p_{\mathrm{P}}(\mathbf{r}) + p_{\mathrm{H}}(\mathbf{r})$$

Sum of particular + homogeneous solutions

$$= \int_{\mathbf{r}' \in \Omega} Q(\mathbf{r}') G(\mathbf{r}|\mathbf{r}') d\mathbf{r}'$$

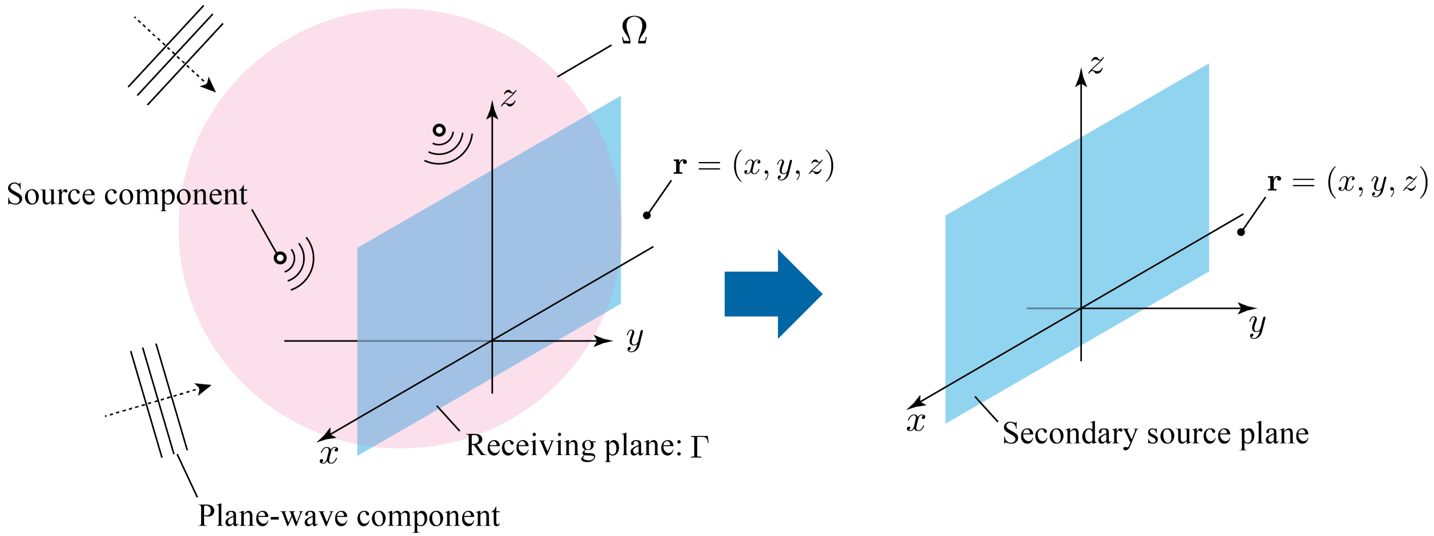
Green's function

$$+ \frac{1}{4\pi^2} \int_{-\infty}^{\infty} dk_x \int_{-\infty}^{\infty} dk_z P_{\mathrm{H}}(k_x, k_z) e^{j(k_x x + k_y y + k_z z)}$$

Plane wave

Generative model of sound field

- **Observe sound pressure distribution on plane Γ**



- **Conversion into driving signals**

$$d(\mathbf{r}) = \frac{\partial}{\partial y} p(\mathbf{r}) \Big|_{y=0}$$

$$= \int_{\mathbf{r}' \in \Omega} Q(\mathbf{r}') \frac{\partial G(\mathbf{r}|\mathbf{r}')}{\partial y} \Big|_{y=0} d\mathbf{r}' + \frac{1}{4\pi^2} \int_{-\infty}^{\infty} dk_x \int_{-\infty}^{\infty} dk_z j k_y P_{\mathrm{H}}(k_x, k_z) e^{j(k_x x + k_z z)}$$

Synthesize monopole sources [Spors+ AES Conv. 2008]

Direct source components

Reverberant components

Applying WFR filtering method [Koyama+ IEEE TASLP 2013]

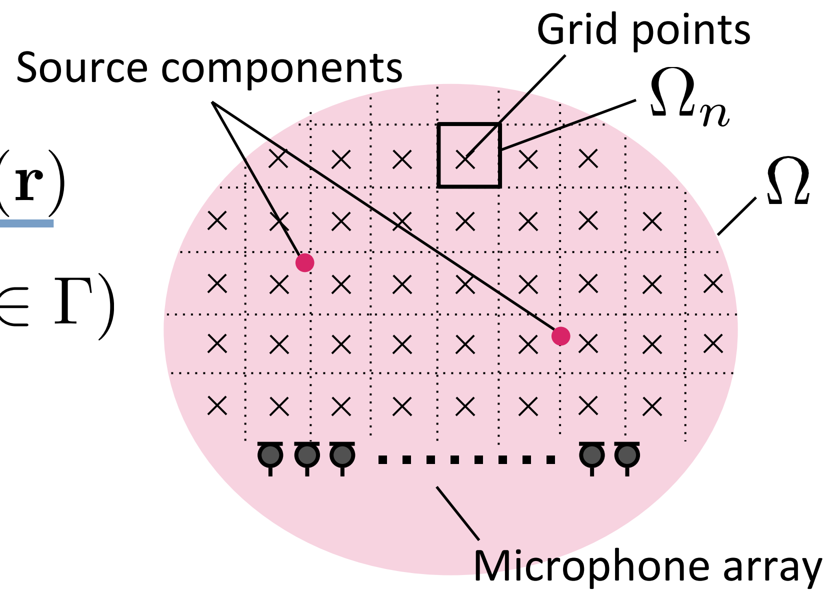
Decomposition into two components can lead to higher reproduction accuracy above spatial Nyquist freq

Sparse sound field representation

- **Sparsity-based signal decomposition**

$$p(\mathbf{r}) = \int_{\mathbf{r}' \in \Omega} Q(\mathbf{r}') G(\mathbf{r}|\mathbf{r}') d\mathbf{r}' + p_{\mathrm{H}}(\mathbf{r})$$

$$(\mathbf{r} \in \Gamma)$$



Discretization

Dictionary matrix of Green's functions

Reverberant components

$$\mathbf{y} = \mathbf{D}\mathbf{x} + \mathbf{z}$$

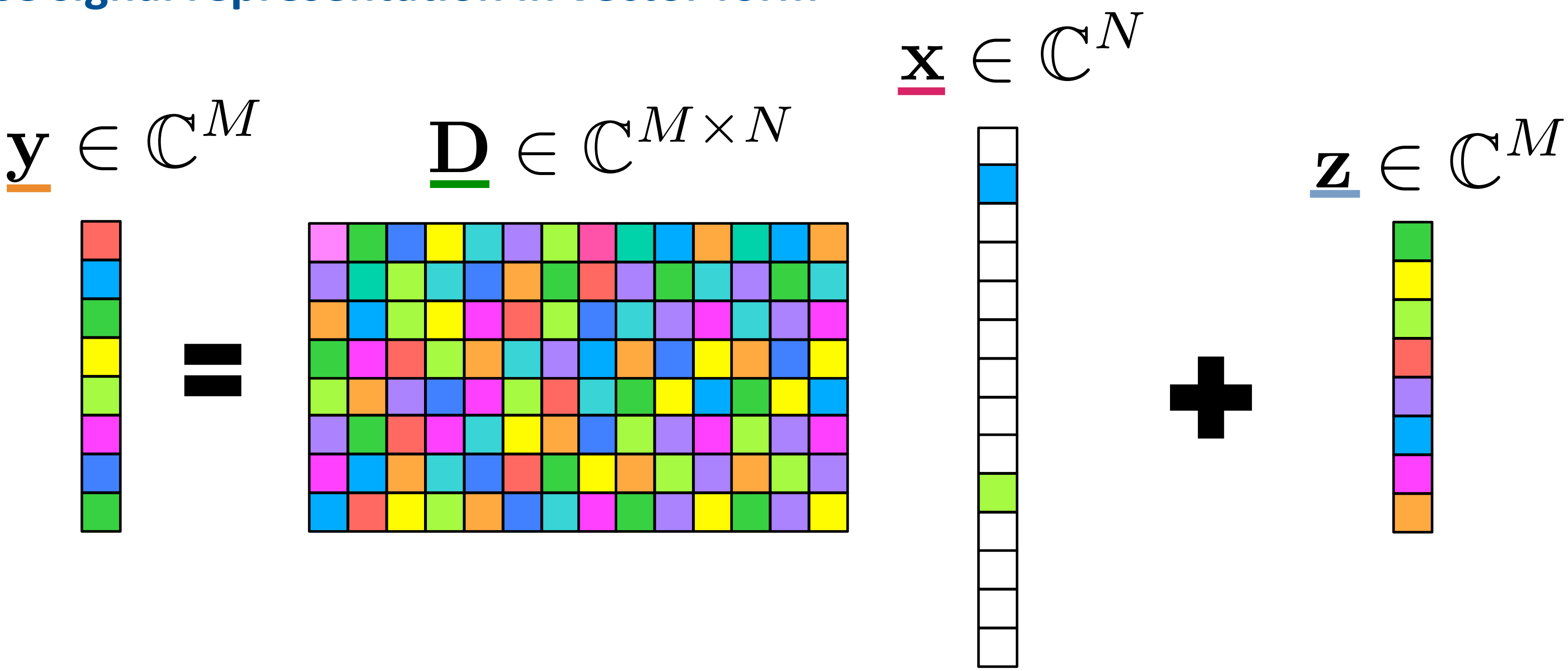
Observed signal

Distribution of source components

A few elements of $\mathbf{x}$ has non-zero values under the assumption of spatially sparse source distribution

Sparse signal decomposition

- **Sparse signal representation in vector form**



- **Signal decomposition based on sparsity of $\mathbf{x}$**

$$\text{minimize } \|\mathbf{x}\|_p^p \quad (p \leq 1)$$

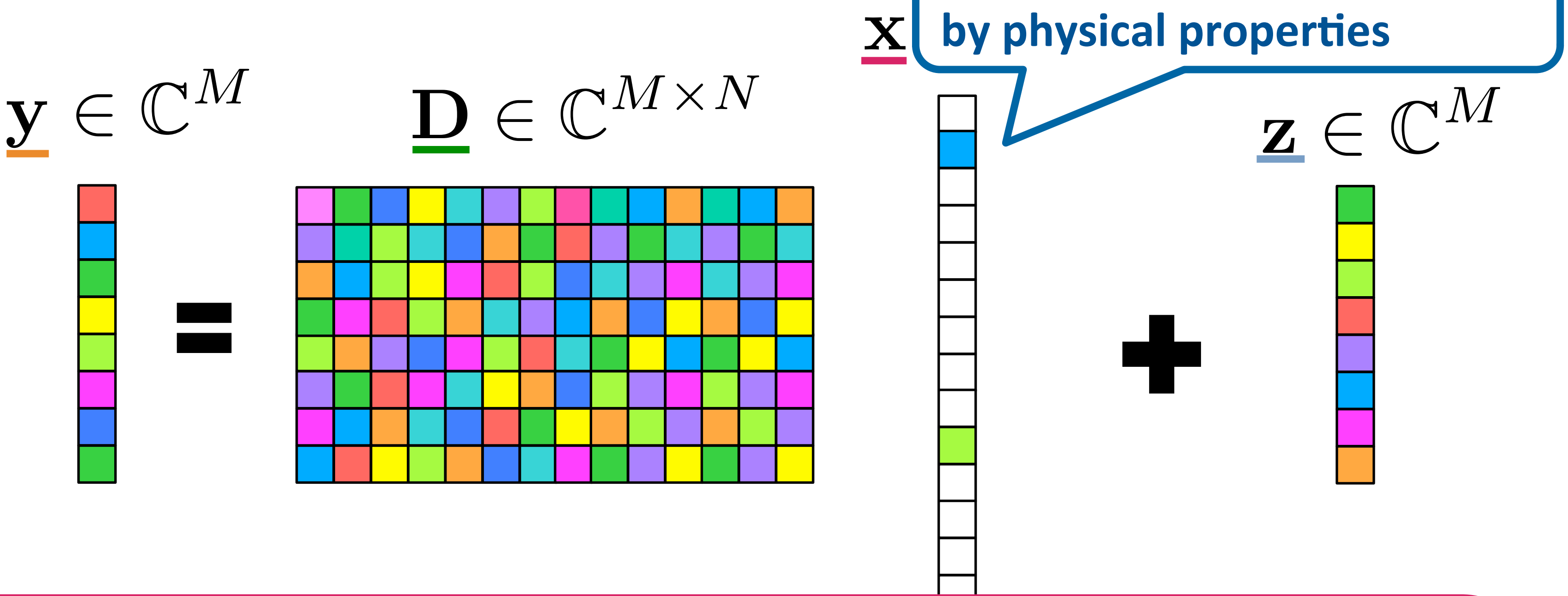
$$\text{subject to } \|\mathbf{y} - \mathbf{D}\mathbf{x}\|_2^2 < \epsilon$$

Minimize ℓ_p -norm of $\mathbf{x}$

Group sparsity based on physical properties

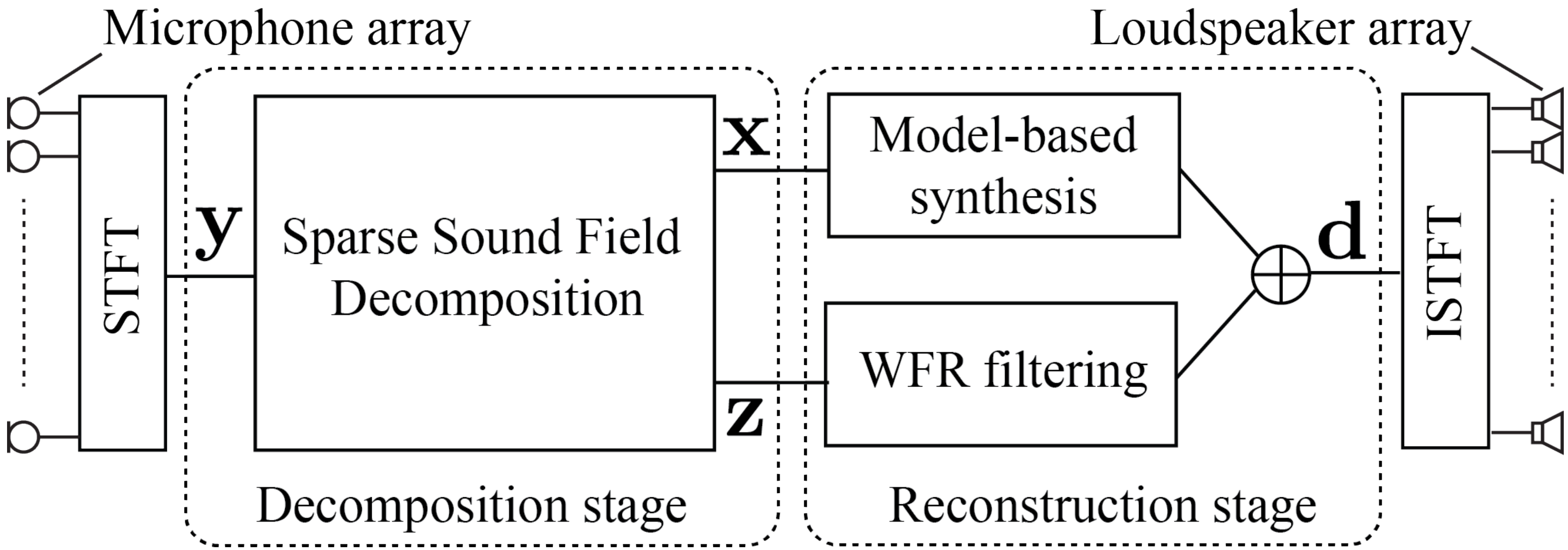
- **Sparse signal representation in vector form**

Structure of sparsity induced by physical properties



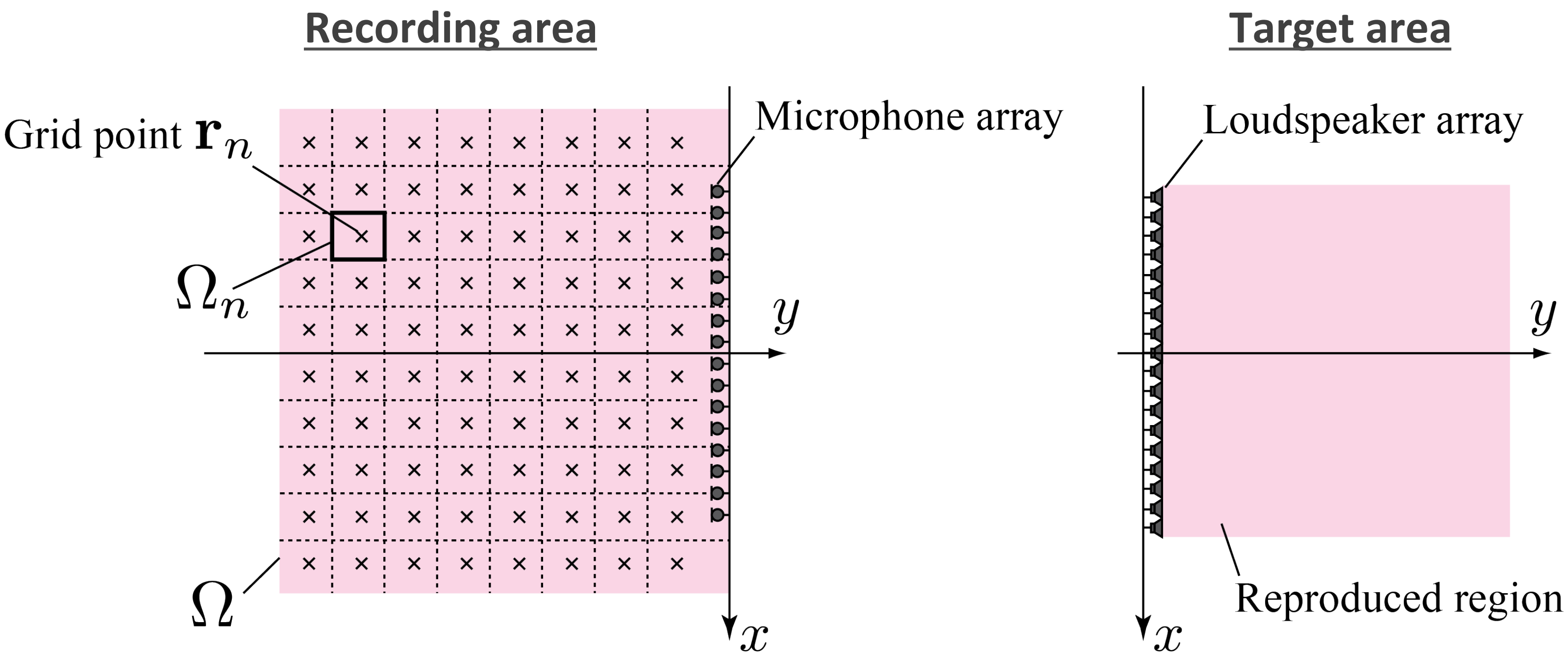
- **Group sparse signal models for accurate and robust decomposition**
  - Multiple time frames
  - Temporal frequencies
  - Multipole components
- **Decomposition algorithm extending FOCUSS** [Koyama+ ICASSP 2015]

Block diagram of signal conversion



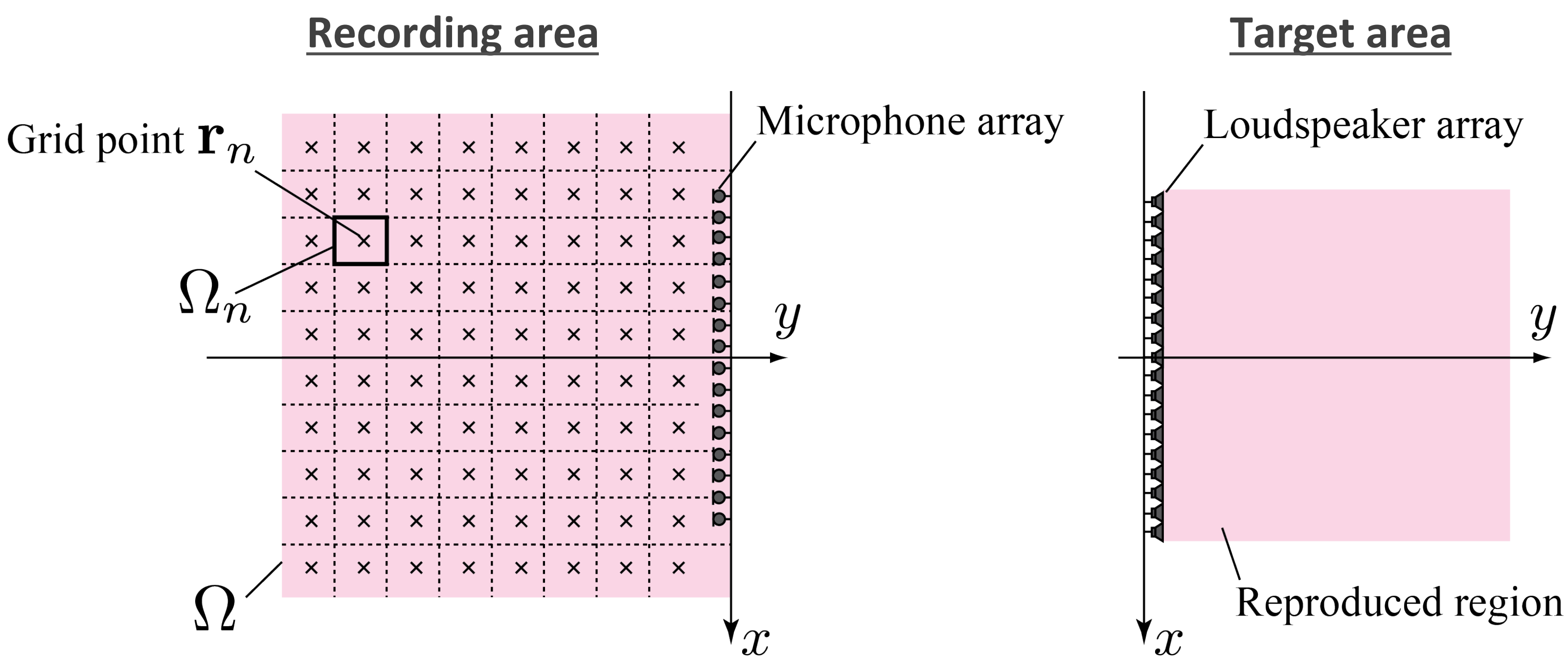
- **Decomposition stage**
  - Group sparse decomposition of $\mathbf{y}$
- **Reconstruction stage**
  - $\mathbf{x}$ and $\mathbf{z}$ are respectively converted into driving signals
  - $\mathbf{d}$ is obtained as sum of two components

Simulation Experiment



- Proposed method (**Proposed**), method based on sparse circular harmonics decomposition (**CH**) WFR filtering method (**WFR**), and Sound Pressure Control method (**SPC**) were compared
- 32 microphones (0.06 m intervals) and 48 loudspeakers (0.04 m intervals)
- Ω : Rectangular region of 2.4x2.4 m, Grid points: (0.01 m, 0.02 m) intervals
- Source directivity: unidirectional
- Source signal: single frequency sinewave

Simulation Experiment



- **Signal-to-distortion ratio of reproduction (SDRR)**

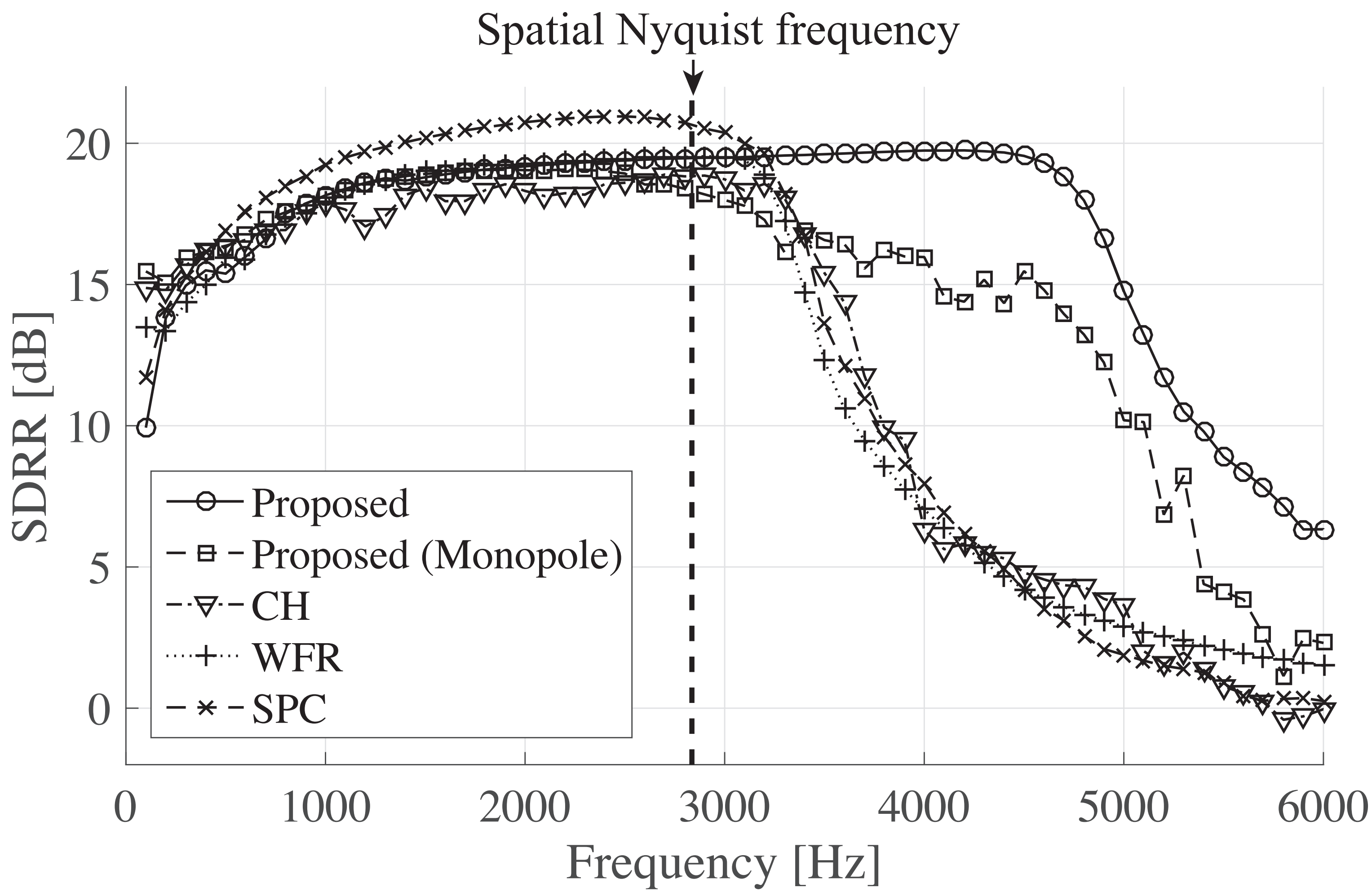
$$\mathrm{SDRR} = 10\log_{10}\frac{\sum_i\sum_j\sum_k |\bar{p}_{\mathrm{org}}(x_i, y_j, t_k)|^2}{\sum_i\sum_j\sum_k |\bar{p}_{\mathrm{org}}(x_i, y_j, t_k) - \bar{p}_{\mathrm{rep}}(x_i, y_j, t_k)|^2}$$

Original pressure distribution

Reproduced pressure distribution

Frequency vs. SDR

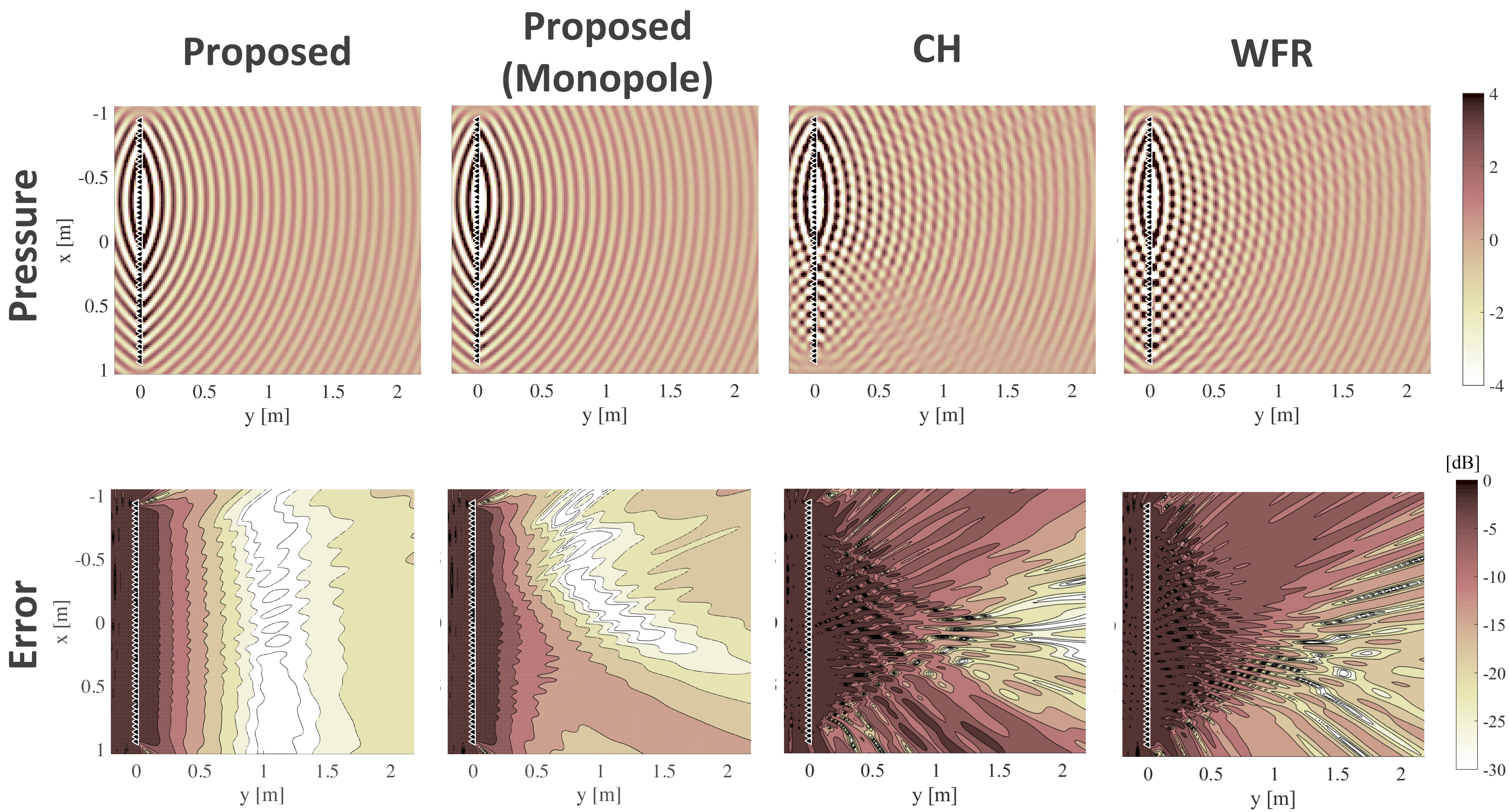
- Source location: (-0.32, -0.84, 0.0) m



SDRRs above spatial Nyquist frequency were improved

Reproduced sound pressure distribution (4.0 kHz)

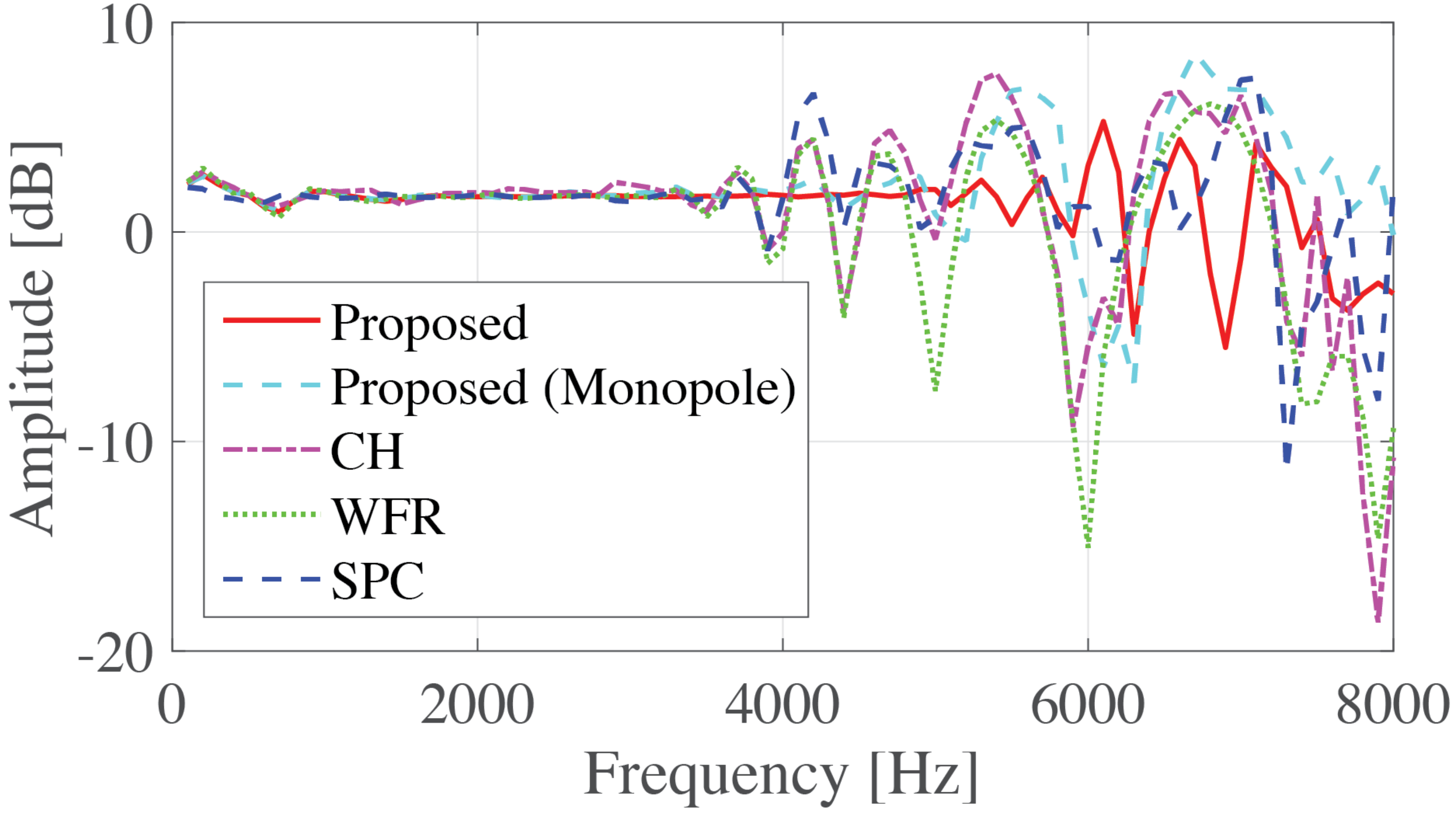
- Source location: (-0.32, -0.84, 0.0) m



SDRR: 19.7 dB | 16.0 dB | 6.3 dB | 7.0 dB

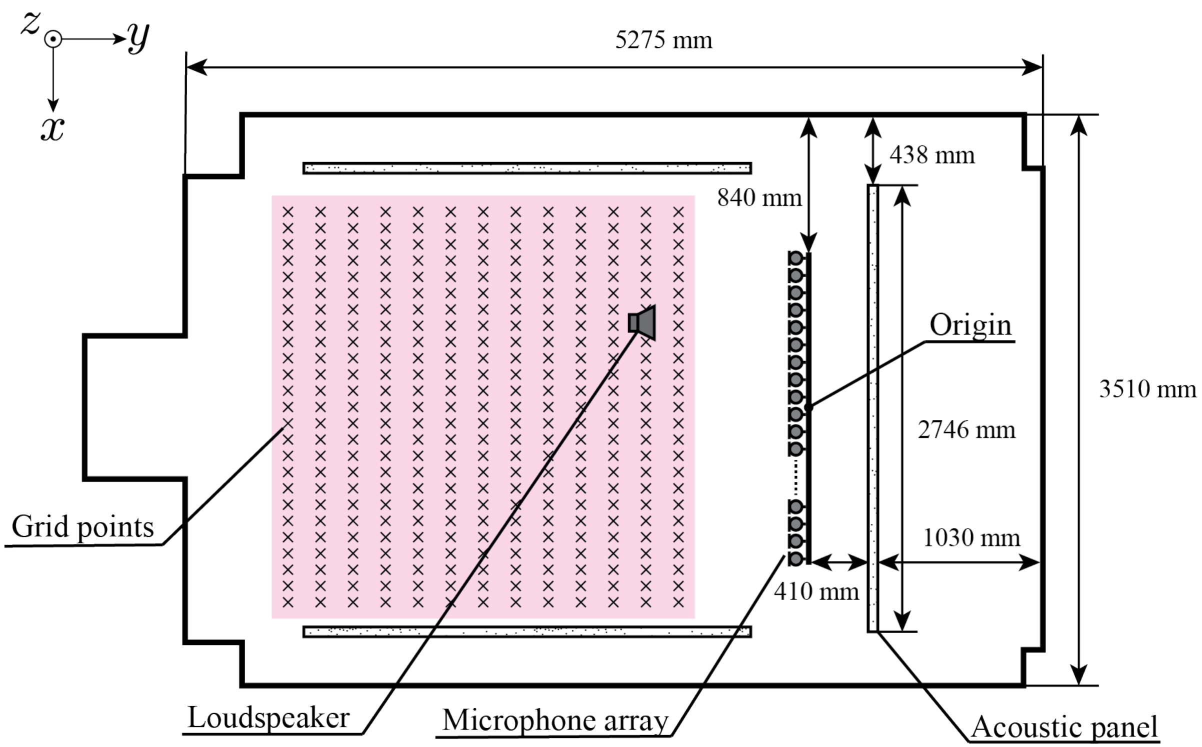
Frequency response of reproduced sound field

➢ Frequency response at (0.0, 1.0, 0.0) m



Reproduced frequency response was improved

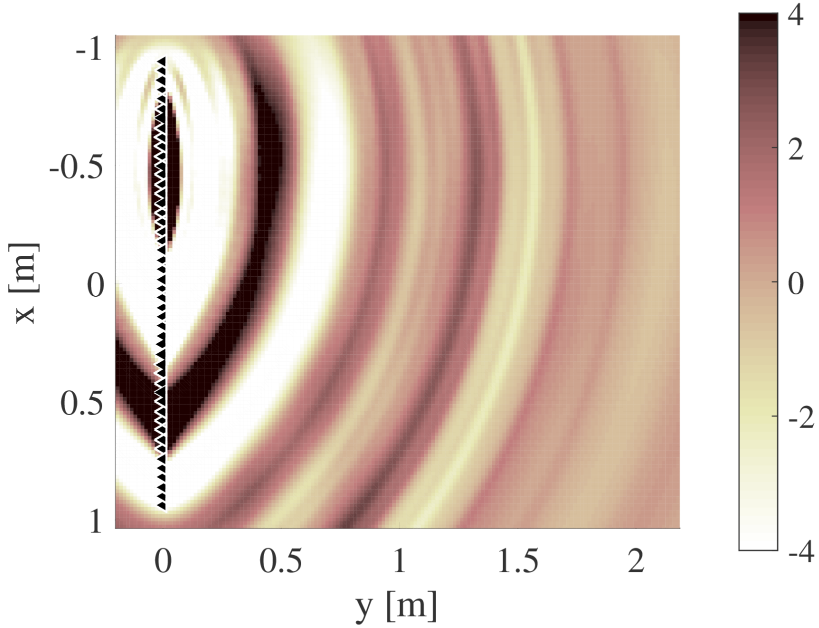
Experiments using real data



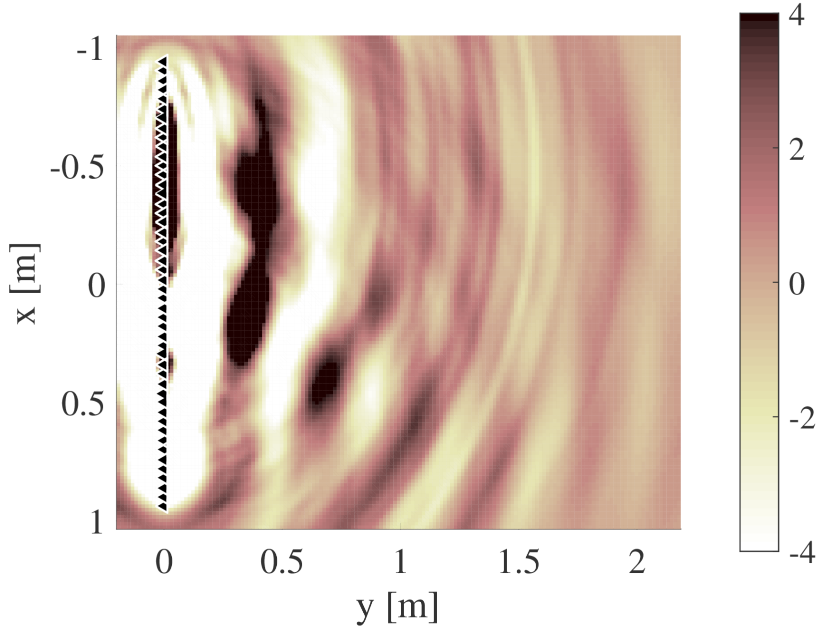
- Proposed method (**Proposed**) and WFR filtering method (**WFR**), were compared
- Same experimental setting as the previous one
- Reproduced region was simulated as free field
- Source signal: speech

Reproduced sound pressure distribution

Proposed



WFR



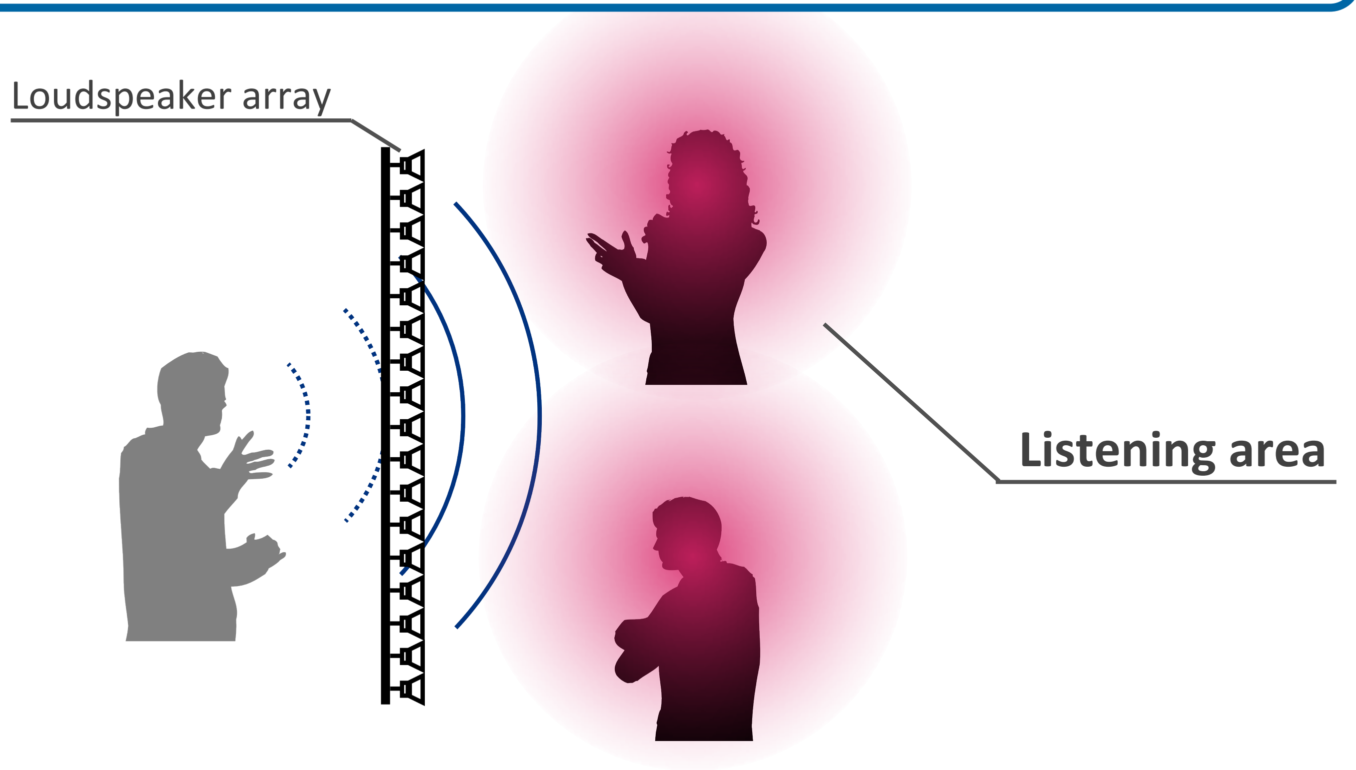
Reduction of The Number of Microphones

- Conventional plane wave decomposition is suffered from spatial aliasing artifacts
- Sound field representation using near-field source and plane wave components
- Sound field decomposition based on spatial sparsity of near-field source components
- Group sparsity based on physical properties of sound field
- Experimental results indicated that reproduction accuracy above spatial Nyquist frequency can be improved

REDUCTION OF THE NUMBER OF LOUDSPEAKERS

Listening-area-informed sound field reproduction

Highly-accurate sound field reproduction exploiting prior information on listening area



- Probability distribution on the listeners' position is given
- High reproduction accuracy within the listening area is achieved

Problem statement

- Sound field synthesized by L discrete secondary sources in 2D

$$p_{\mathrm{syn}}(\mathbf{r}) = \underline{\mathbf{h}(\mathbf{r})}^{\mathsf{T}} \underline{\mathbf{d}}$$

Transfer function

$$\underline{\mathbf{h}(\mathbf{r})} = [h(\mathbf{r}, \mathbf{r}_1), \cdots, h(\mathbf{r}, \mathbf{r}_L)]$$

Driving signals

$$\underline{d(\mathbf{r})} = [d(\mathbf{r}_1), \cdots, d(\mathbf{r}_L)]$$

- Expectation minimization of pressure error

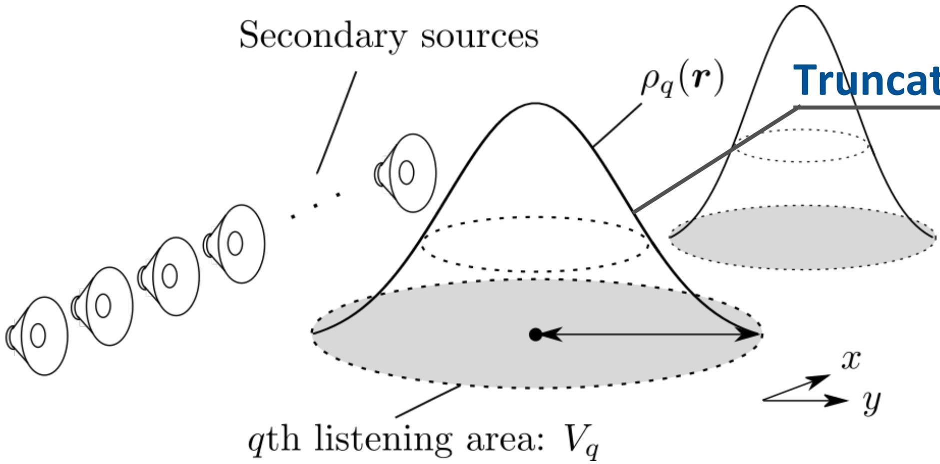
$$\underset{\mathbf{d}}{\mathrm{minimize}} \int_{\mathbf{r} \in V} \underline{\rho(\mathbf{r})} \left| \underline{\mathbf{h}(\mathbf{r})}^{\mathsf{T}} \underline{\mathbf{d}} - p_{\mathrm{des}}(\mathbf{r}) \right|^2 d\mathbf{r}$$

Given desired pressure

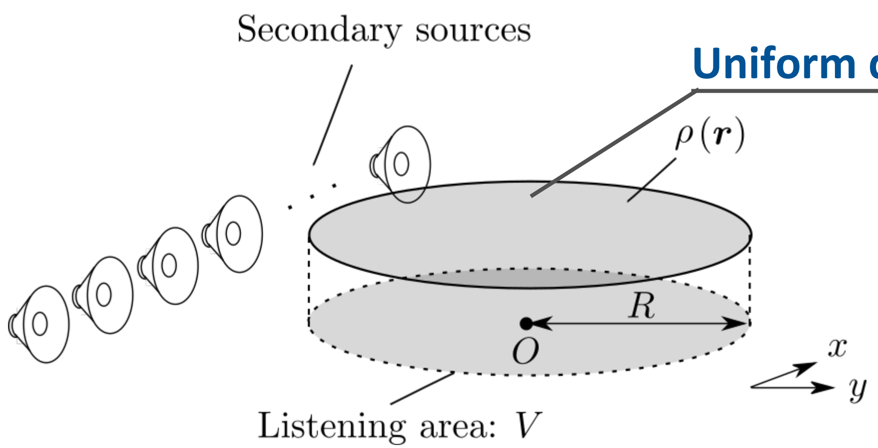
Prior distribution on the listeners' position inside listening area V

Problem statement

- $\rho(\mathbf{r})$ is a mixture of truncated Gaussian distributions [Ueno+ HSCMA 2017]



- $\rho(\mathbf{r})$ is a uniform distribution in single or multiple circular areas [Ueno+ ICASSP 2017]



Harmonic Expansion of Objective Function

- Objective function

$$\mathcal{J} = \int_{\mathbf{r}\in V} \rho(\mathbf{r}) \left| \mathbf{h}(\mathbf{r})^{\mathsf{T}} \mathbf{d} - p_{\mathrm{des}}(\mathbf{r}) \right|^2 d\mathbf{r}$$

Difficult to minimize analytically due to multiple integrals

- Circular harmonic expansion

Basis functions: $\varphi_m(\mathbf{r}) = J_m(kr)e^{jm\phi}$ ($\mathbf{r} = (r, \phi)$: Polar coordinates)

$$\begin{cases} \mathbf{h}(\mathbf{r}) \simeq \begin{bmatrix} c_{-M,1} & \cdots & c_{M,1} \\ \vdots & \ddots & \vdots \\ c_{-M,1} & \cdots & c_{M,L} \end{bmatrix} \begin{bmatrix} \varphi_{-M}(\mathbf{r}) \\ \vdots \\ \varphi_M(\mathbf{r}) \end{bmatrix} = \mathbf{C}^{\mathsf{T}} \boldsymbol{\varphi}(\mathbf{r}) \\ p_{\mathrm{des}}(\mathbf{r}) \simeq \begin{bmatrix} b_{-M} & \cdots & b_M \end{bmatrix} \begin{bmatrix} \varphi_{-M}(\mathbf{r}) \\ \vdots \\ \varphi_M(\mathbf{r}) \end{bmatrix} = \mathbf{b}^{\mathsf{T}} \boldsymbol{\varphi}(\mathbf{r}) \end{cases}$$

Harmonic Expansion of Objective Function

- Objective function

$$\mathcal{J} = \int_{\mathbf{r}\in V} \rho(\mathbf{r}) \left| \mathbf{h}(\mathbf{r})^{\mathsf{T}} \mathbf{d} - p_{\mathrm{des}}(\mathbf{r}) \right|^2 d\mathbf{r}$$

Approximation using circular harmonic expansion

$$\begin{cases} \mathbf{h}(\mathbf{r}) \simeq \mathbf{C}^{\mathsf{T}} \boldsymbol{\varphi}(\mathbf{r}) \\ p_{\mathrm{des}}(\mathbf{r}) \simeq \mathbf{b}^{\mathsf{T}} \boldsymbol{\varphi}(\mathbf{r}) \end{cases}$$

$$\mathcal{J} \simeq (\mathbf{C}\mathbf{d} - \mathbf{b})^{\mathsf{H}} \underbrace{\left\{ \int_{\mathbf{r}\in V} \rho(\mathbf{r}) \boldsymbol{\varphi}(\mathbf{r})^* \boldsymbol{\varphi}(\mathbf{r})^{\mathsf{T}} d\mathbf{r} \right\}}_{\mathbf{W}} (\mathbf{C}\mathbf{d} - \mathbf{b})$$

Integrals including $\rho(\mathrm{r})$ and $\varphi(\mathrm{r})$ are only required to be calculated

Harmonic Expansion of Objective Function

➢ Each element of $\mathbf{W}$ can be analytically calculated as

$$
\begin{aligned}
w_{m,n} &= \int_{\mathbf{r}\in V} \rho(\mathbf{r})\varphi_m(\mathbf{r})^* \varphi_n(\mathbf{r}) d\mathbf{r} \\
&= \int_{\mathbf{r}\in V} J_m(kr)e^{-jm\phi} J_n(kr)e^{jn\phi} d\mathbf{r} \\
&= \int_0^R J_m(kr)J_n(kr)r dr \int_0^{2\pi} e^{j(n-m)\phi} d\phi \\
&= \delta_{m,n}\pi R^2 \left\{ J_m(kR)^2 - J_{m-1}(kR)J_{m+1}(kR) \right\}
\end{aligned}
$$

(m,n) element of $\mathbf{W}$

$\mathbf{W}$ is a diagonal matrix with positive value

- ➢ **$\mathbf{W}$ can be analytically calculated for uniform distribution**
- ➢ **Similar results can be obtained for truncated Gaussian distribution**

Optimal Driving Signals

- Objective function using circular harmonic expansion

$$\mathcal{J} \simeq (\mathbf{C}\underline{\mathbf{d}} - \mathbf{b})^{\mathsf{H}} \underbrace{\left\{ \int_{\mathbf{r}\in V} \underline{\rho(\mathbf{r})}\,\underline{\boldsymbol{\varphi}(\mathbf{r})}^{*}\,\underline{\boldsymbol{\varphi}(\mathbf{r})}^{\mathsf{T}} d\mathbf{r} \right\}}_{\underline{\mathbf{W}}} (\mathbf{C}\underline{\mathbf{d}} - \mathbf{b})$$

- **Optimal driving signals**

$$\underline{\hat{\mathbf{d}}} = \left(\mathbf{C}^{\mathsf{H}}\underline{\mathbf{W}}\mathbf{C} + \lambda\mathbf{I}\right)^{-1} \mathbf{C}^{\mathsf{H}}\underline{\mathbf{W}}\mathbf{b}$$

- **Objective function is simply minimized and optimal driving signals can be obtained**

Relationship with mode-matching method

- **Proposed method**

$$\hat{\mathbf{d}} = \left(\mathbf{C}^{\mathsf{H}}\mathbf{W}\mathbf{C} + \lambda\mathbf{I}\right)^{-1}\mathbf{C}^{\mathsf{H}}\mathbf{W}\mathbf{b}$$

- **Circular harmonics are optimally weighted based on prior information**

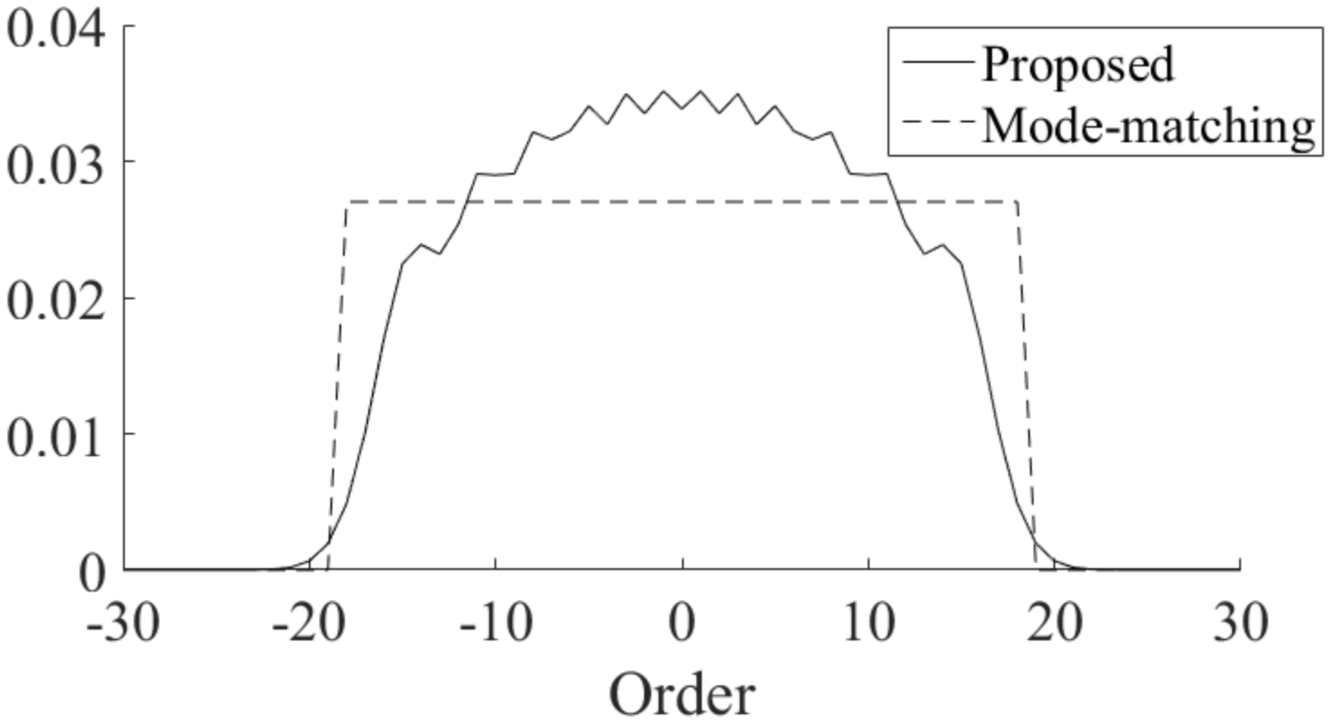
- **Mode-matching method: least squares solution of** $\mathbf{C}\mathbf{d} = \mathbf{b}$

$$\hat{\mathbf{d}} = \left(\mathbf{C}^{\mathsf{H}}\mathbf{C} + \lambda\mathbf{I}\right)^{-1}\mathbf{C}^{\mathsf{H}}\mathbf{b}$$

- **Circular harmonics have to be truncated at appropriate order**
- **Truncation at $M = \lceil kR \rceil$ is empirically known to give high performance within circular region of radius R**

Relationship with mode-matching method

- Weight on circular harmonics when k = 36.9 rad/m and R = 0.4 m



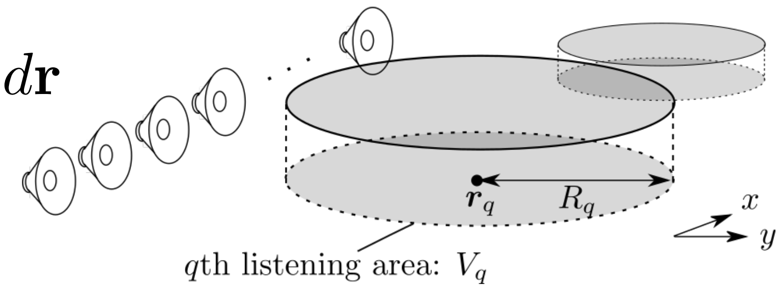
Optimal weight on circular harmonics based on prior information

Extension to Multiple Listening Areas

- Objective function

$$\mathcal{J} = \sum_{q=1}^{Q} \int_{\mathbf{r} \in V_q} \rho(\mathbf{r}) \left| \mathbf{h}(\mathbf{r})^{\mathsf{T}} \mathbf{d} - p_{\mathrm{des}}(\mathbf{r}) \right|^2 d\mathbf{r}$$

*q*th listening area (*q = 1, ..., Q*)



Approximation using circular harmonic expansion

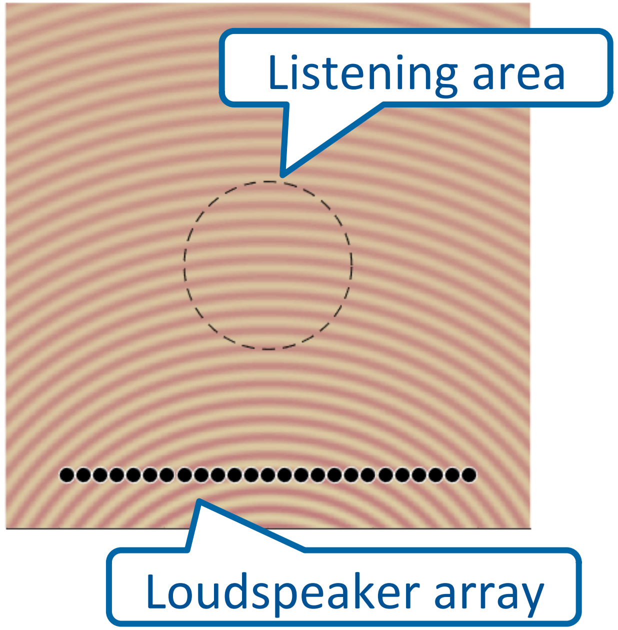
$$\mathcal{J} \simeq \sum_{q=1}^{Q} \left(\mathbf{C}^{(q)} \mathbf{d} - \mathbf{b}^{(q)} \right)^{\mathsf{H}} \mathbf{W}^{(q)} \left(\mathbf{C}^{(q)} \mathbf{d} - \mathbf{b}^{(q)} \right)$$

- **Optimal driving signals**

$$\hat{\mathbf{d}} = \left(\sum_{q=1}^{Q} \mathbf{C}^{(q)\mathsf{H}} \mathbf{W}^{(q)} \mathbf{C}^{(q)} + \lambda \mathbf{I} \right)^{-1} \sum_{q=1}^{Q} \mathbf{C}^{(q)\mathsf{H}} \mathbf{W}^{(q)} \mathbf{b}^{(q)}$$

Simulation Experiment

- Array geometry
  - Linear: 25 loudspeakers, 0.16 m intervals
  - Circular: 64 loudspeakers, 2.0 m radius
- Desired sound field: cylindrical wave
- Listening area: two circular areas
- Compared method:
  - **Proposed**
  - **MM**: Mode-matching method
  - **CD**: Continuous distribution method (WFS/HOA) [Spors+ 2008, Poletti 2005]
  - **CD w/ BL**: CD with band limitation [Ahrens+ 2009, 2011]
- Evaluation: Signal-to-Distortion Ratio of Reproduction (SDRR)



Original pressure distribution

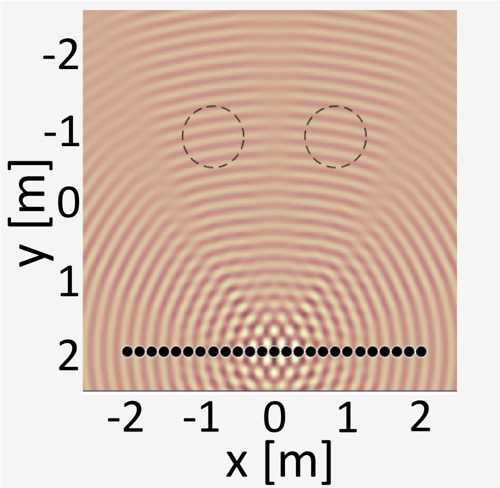
$$\mathrm{SDRR} = 10 \log_{10} \frac{\sum_i \sum_j \sum_k |\bar{p}_{\mathrm{org}}(x_i, y_j, t_k)|^2}{\sum_i \sum_j \sum_k |\bar{p}_{\mathrm{org}}(x_i, y_j, t_k) - \bar{p}_{\mathrm{rep}}(x_i, y_j, t_k)|^2}$$

Reproduced pressure distribution

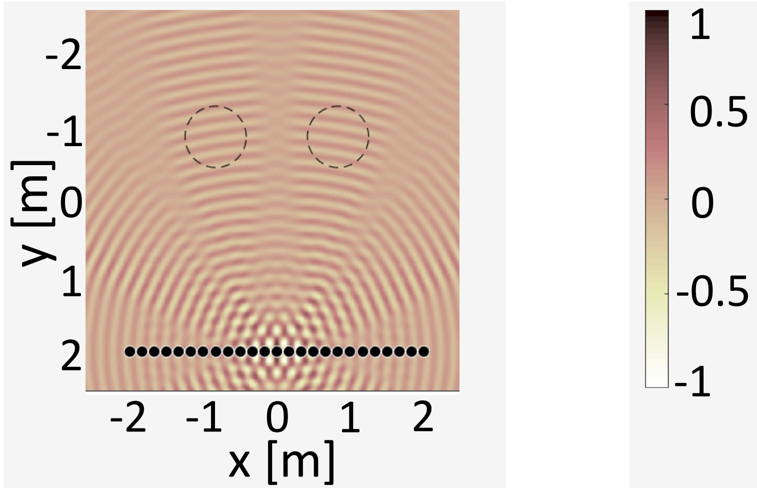
Pressure distribution (2 kHz, linear array)

Sound pressure distribution

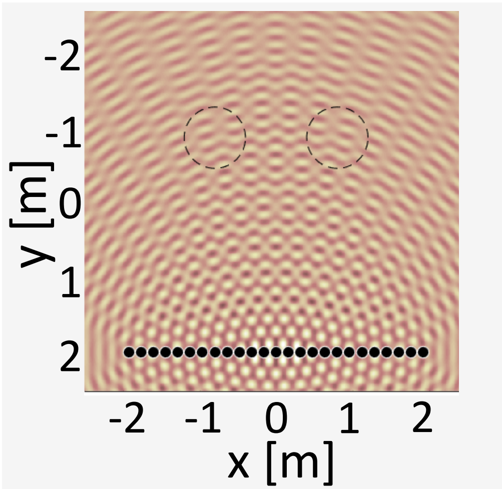
Proposed (SDRR = **68.51** dB)



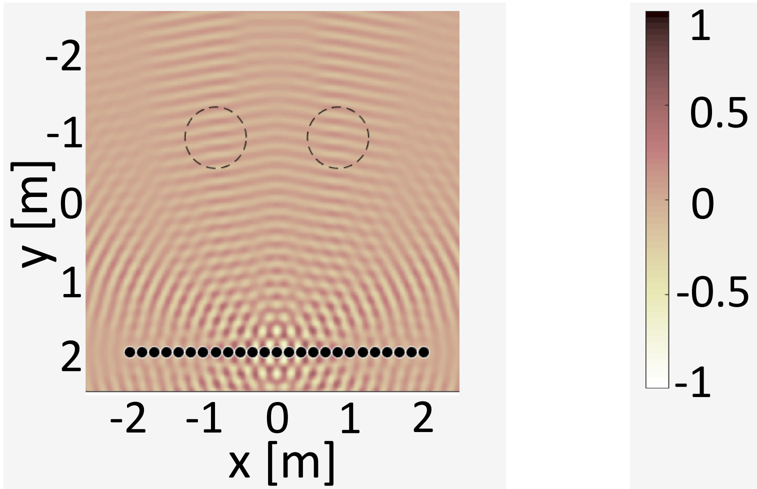
MM (SDRR = **60.42** dB)



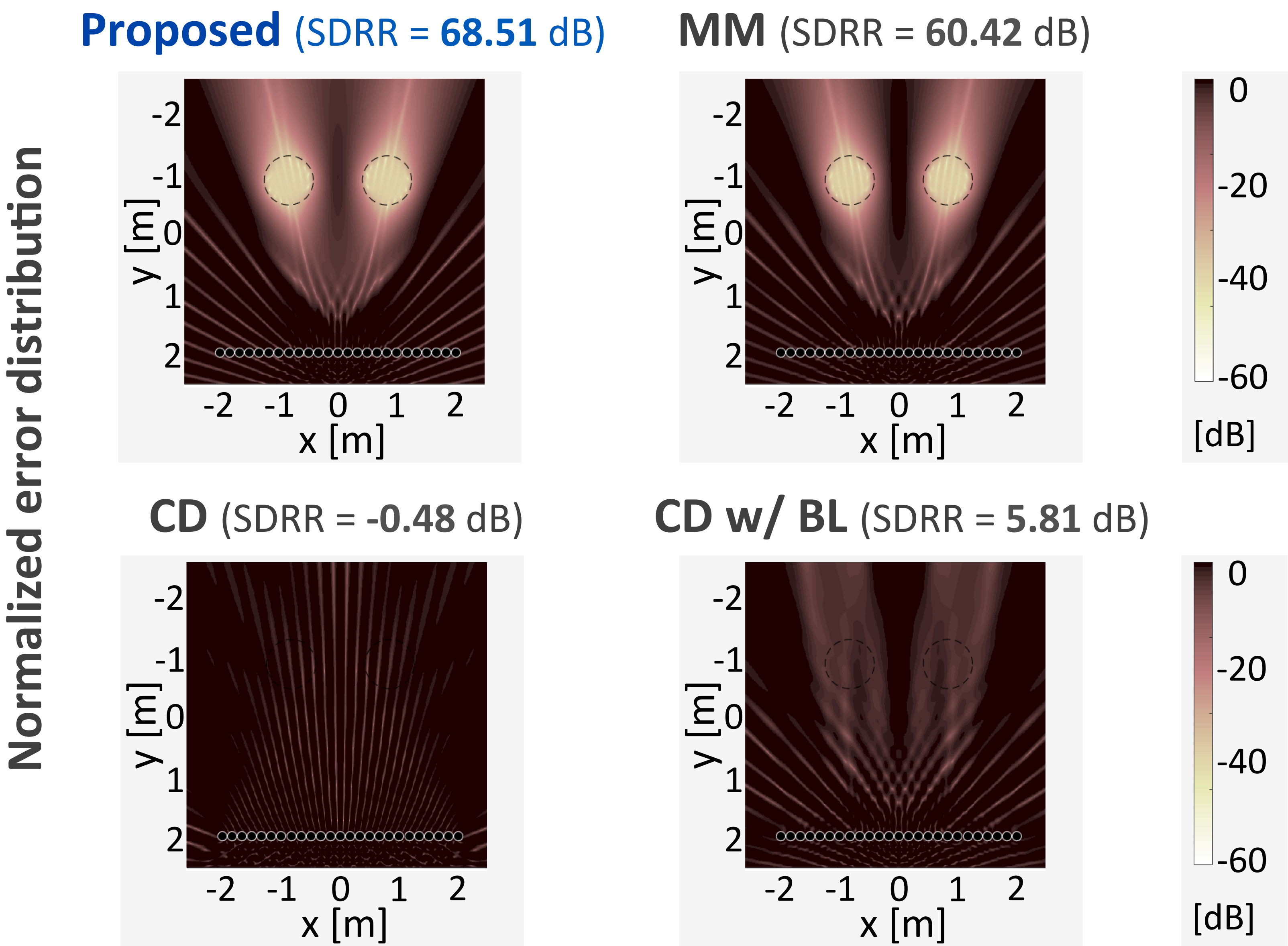
CD (SDRR = **-0.48** dB)



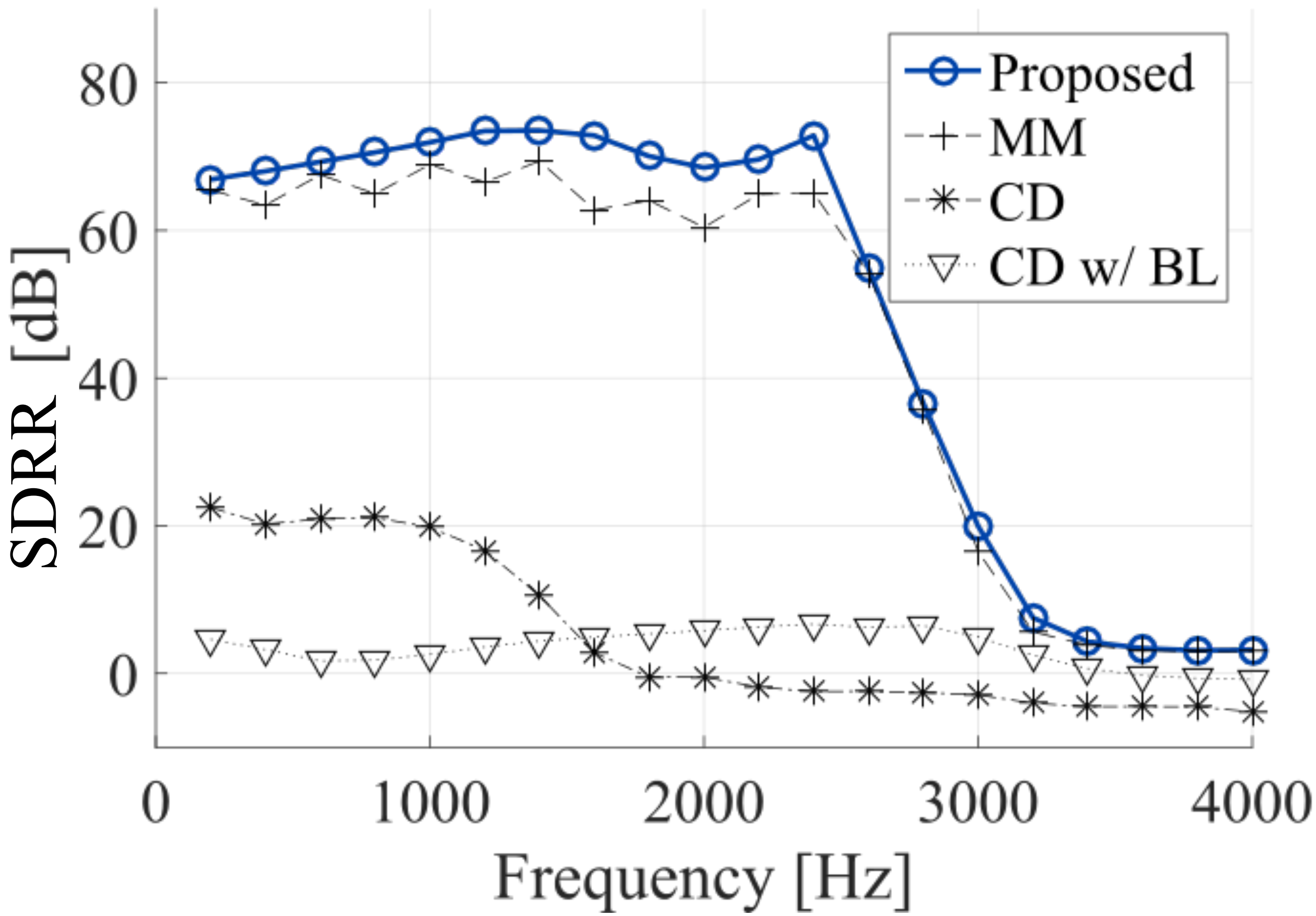
CD w/ BL (SDRR = **5.81** dB)



Error distribution (2 kHz, linear array)



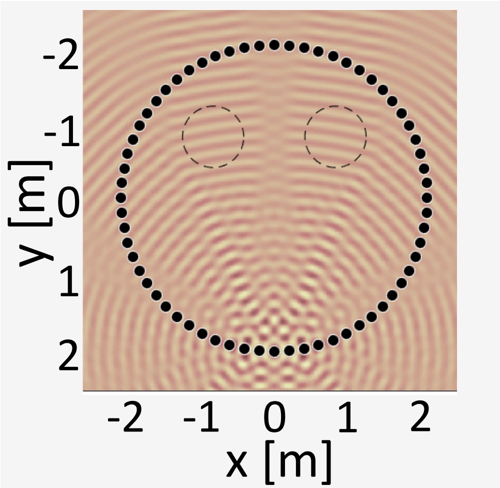
Frequency vs. SDRR (linear array)



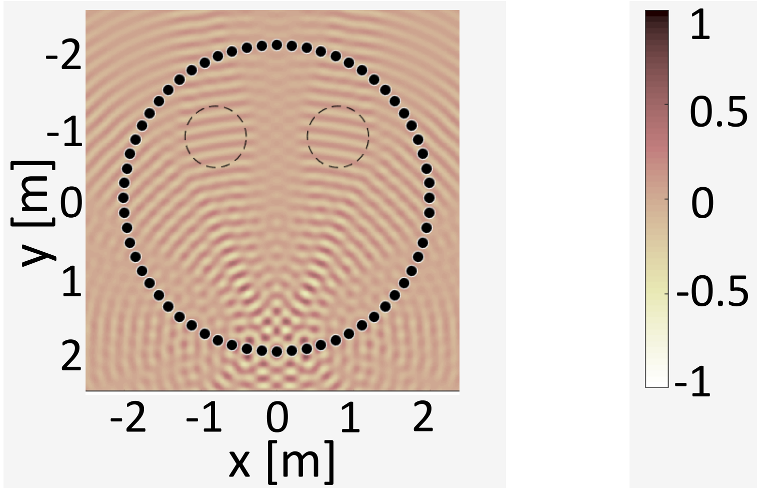
Pressure distribution (2 kHz, circular array)

Sound pressure distribution

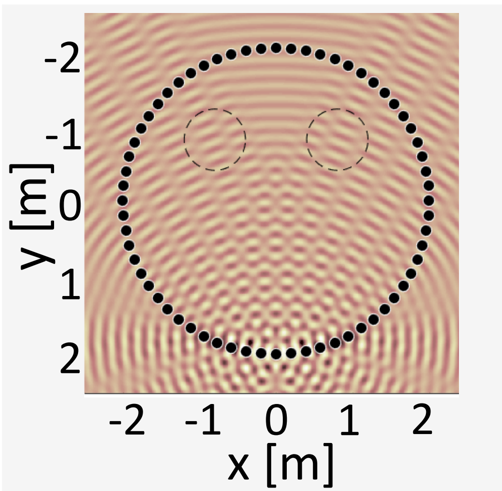
Proposed (SDRR = **66.58** dB)



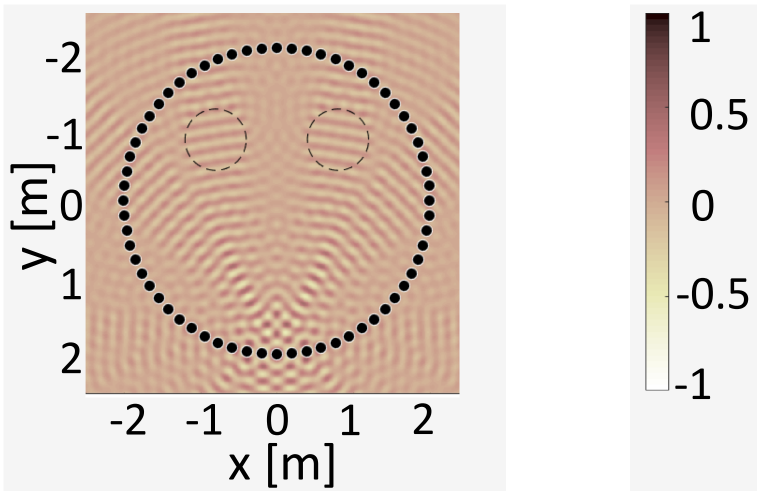
MM (SDRR = **39.63** dB)



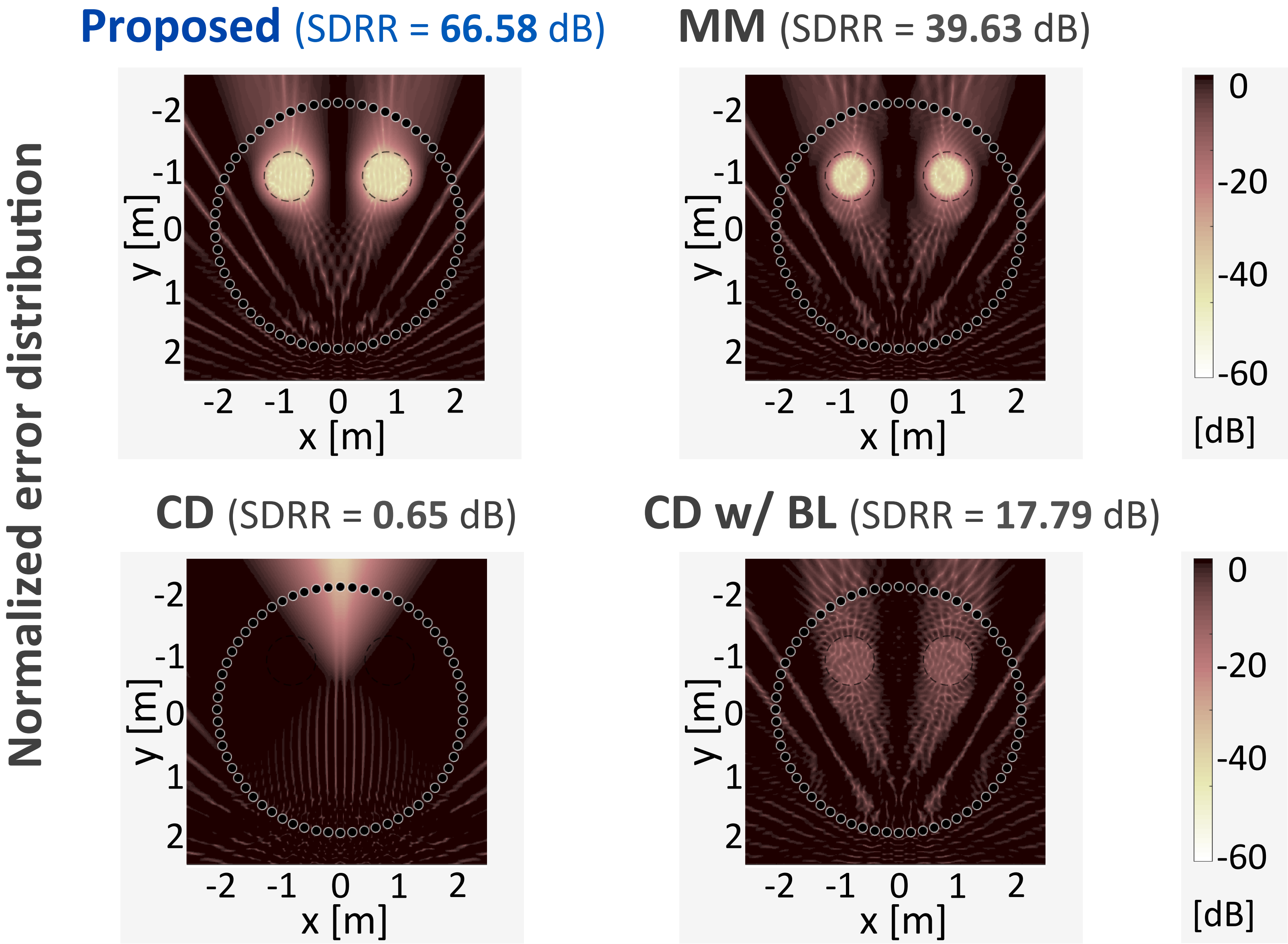
CD (SDRR = **0.65** dB)



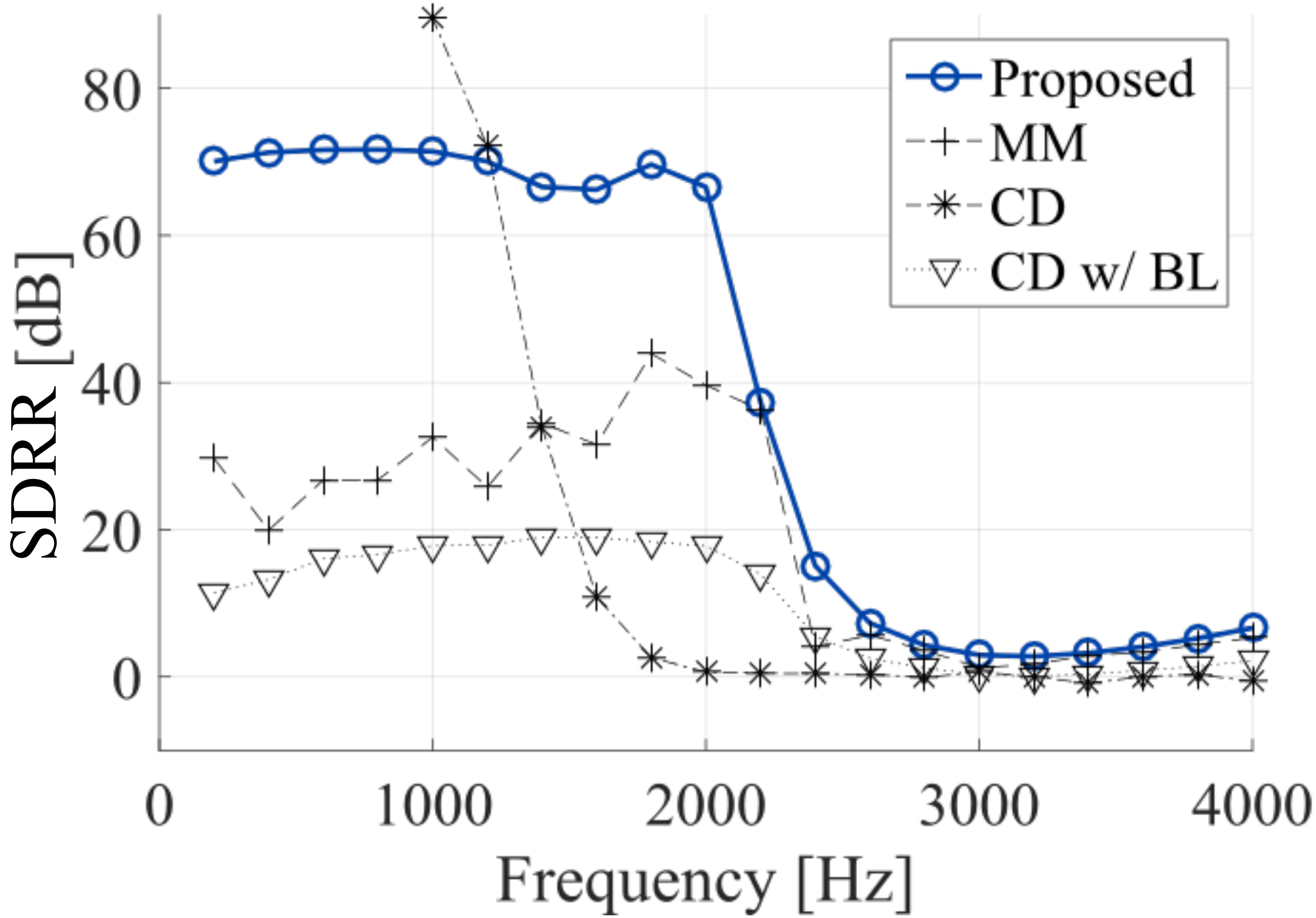
CD w/ BL (SDRR = **17.79** dB)



Error distribution (2 kHz, circular array)



Frequency vs. SDRR (circular array)



Reduction of The Number of Loudspeakers

- Sound field reproduction exploiting prior information on listening area
- Objective function is formulated as expectation minimization of spatial squared error inside listening areas
- Optimal driving signals are obtained by circular harmonic expansion
- Optimal weighting of circular harmonics can be analytically calculated based on prior information
- Experimental results indicated that high reproduction accuracy can be achieved by using the proposed method

Conclusion

- **Reduction of microphones and loudspeakers in sound field recording and reproduction**
  - Improve reproduction accuracy exploiting prior information
  - Near-field source area and source sparsity for reducing microphones
  - Probability distribution on listeners' position for reducing loudspeakers
  - These two methods can be combined

Thank you for your attention!

Related publications

- S. Koyama, *et al*. "Effect of multipole dictionary in sparse sound field decomposition for super-resolution in recording and reproduction," *Proc. ICSV*, 2017 (to appear).
- N. Murata, S. Koyama, et al. "Spatio-temopral sparse sound field decomposition considering acoustic source signal characteristics," *Proc. IEEE ICASSP*, 2017.
- N. Ueno, S. Koyama, and H. Saruwatari, "Listening-area-informed sound field reproduction based on circular harmonic expansion," *Proc. IEEE ICASSP*, 2017.
- N. Ueno, S. Koyama, and H. Saruwatari, "Listening-area-informed sound field reproduction with Gaussian prior based on circular harmonic expansion," *Proc. HSCMA*, 2017.
- S. Koyama and H. Saruwatari, "Sound field decomposition in reverberant environment using sparse and low-rank signal models," *Proc. IEEE ICASSP*, 2016.
- N. Murata, S. Koyama, *et al.* "Sparse sound field decomposition with multichannel extension of complex NMF," *Proc. IEEE ICASSP*, 2016.
- S. Koyama, *et al.* "Sparse sound field decomposition using group sparse Bayesian learning," *Proc. APSIPA ASC*, 2015.
- N. Murata, S. Koyama, *et al.* "Sparse sound field decomposition with parametric dictionary learning for super-resolution recording and reproduction," *Proc. IEEE CAMSAP*, 2015.
- S. Koyama, *et al*. "Structured sparse signal models and decomposition algorithm for super-resolution in sound field recording and reproduction," *Proc. IEEE ICASSP*, 2015.
- S. Koyama, *et al.* "Sparse sound field representation in recording and reproduction for reducing spatial aliasing artifacts," *Proc. IEEE ICASSP*, 2014.