

Room Impulse Response Interpolation from a Sparse Set of Measurements Using a Modal Architecture



RESEARCH AUDIO

Objectives

- Characterizing the entire sound field of a room utilizing a sparse dataset of Room Impulse Responses (RIRs) measured at different locations.
- Accurate spatial interpolation of perceptually relevant low frequency modes in rooms with simple geometries having non-rigid walls.
- Useful for real-time interpolation and extrapolation in Augmented Reality (AR) applications.

Room Modes

- Solutions to the 3D wave equation are standing waves, or room modes.
- The RIR can be characterized by a sum over M modes, whose complex amplitudes, γ , are functions of space, whereas frequencies and dampings, ω and α , determine the temporal response:

$$h(x, y, z, t) = \sum_{m=1}^{M} \gamma_m(x, y, z) \exp\left[(j\omega_m - \alpha_m)\right]$$

• Complex mode amplitudes are the solution to the homogeneous Helmholtz equation |1|:

$$\gamma_m(\mu) = C_{\mu_m} \exp(jk_{\mu_m}\mu) + D_{\mu_m} \exp(-jk_{\mu_m}\mu);$$

$$k_{\mu_m} \to \text{Wave number}; \ C_{\mu_m}, D_{\mu_m} \to \text{Constants};$$

• We want to estimate the unknown wave numbers and constants for each mode from a set of RIR measurements at different locations in the room.

Mode Estimation

- RIRs measured at different positions are time-aligned and averaged.
- Common poles (mode frequencies and amplitudes) calculated from the averaged RIR with subband ESPRIT [2].
- Mode amplitudes estimated with linear least squares.

Non-linear Optimization

• For 2D interpolation with measurements at L locations,

$$egin{bmatrix} \hat{\gamma}_m(x_1,y_1) \ \hat{\gamma}_m(x_2,y_2) \ dots \ \hat{\gamma}_m(x_L,y_L) \end{bmatrix} = egin{bmatrix} oldsymbol{u}_{m_1} \ oldsymbol{u}_{m_2} \ dots \ oldsymbol{u}_{m_2} \ dots \ oldsymbol{D}_{1m} \ oldsymbol{C}_{2m} \ oldsymbol{U}_{m_L} \end{bmatrix} egin{matrix} C_{1m} \ oldsymbol{D}_{1m} \ oldsymbol{D}_{2m} \ oldsymbol{U}_{m_L} \end{bmatrix} \ egin{matrix} oldsymbol{U}_{m_2} \ dots \ \ do$$

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 $\mu \in (x, y, z).$

 $\boldsymbol{u}_{m_{l}} = \begin{bmatrix} e^{-j(k_{x_{m}}x_{l}+k_{y_{m}}y_{l})} & e^{j(k_{x_{m}}x_{l}+k_{y_{m}}y_{l})} \\ e^{-j(k_{x_{m}}x_{l}-k_{y_{m}}y_{l})} & e^{j(k_{x_{m}}x_{l}-k_{y_{m}}y_{l})} \end{bmatrix}$ $\hat{\boldsymbol{\gamma}}_m = \boldsymbol{U}_m(k_{x_m},k_{y_m}) \boldsymbol{c}_m.$

• Find optimal parameters for the first optimization [3].

Algorithm 1 Sequential optimization

Require: $0 \le k_{x_m}, k_{y_m} \le \frac{\omega_m + j\alpha_m}{c} \forall m$ for $m = 1 \cdots M_c$ do Initialize $k_{x_m} = k_{y_m} = \frac{\omega_m + j\alpha_m}{\sqrt{2}c}$ repeat $oldsymbol{c}_{m_i} = oldsymbol{U}_{m_{i-1}}^{*\dagger}oldsymbol{\gamma}_m$ $\hat{oldsymbol{\gamma}}_{m_i} = oldsymbol{U}^*_{m_{i-1}} oldsymbol{c}_{m_i}$ $J(k_{x_{m_i}}, k_{y_{m_i}}) = ||20 \log_{10}(\boldsymbol{\gamma}_m \oslash \hat{\boldsymbol{\gamma}}_{m_i})||_2^2$ $k_{x_{m_i}}^*, k_{y_{m_i}}^* = \arg\min_{k_{x_m}, k_{y_m}} J$ until convergence end for

FDTD Simulations



Shoebox room with different materials

• Shoebox room of dimensions $3 \times$ • Non-rectangular room with no $2 \times 3 \text{ m}^3$ with different admittances (K_d) on the walls - front and back wall $K_d = 0.9$, left and right wall $K_d = 0.8$, floor and ceiling $K_d =$ 0.99.

parallel walls, 3 m on the longest edge in each direction, made of the same materials and having tilted walls at an angle of 9.4° in the x, zdirections.

- Omni-directional point source and virtual microphones placed in a rectangular grid on the xy plane at a height of 1.7 m.
- Microphone grid resolution is d = 0.2 m. Maximum mode frequency corresponding to M_c is $\frac{c}{2d} = 866$ Hz.

| M_{c} | modes | with | sequential |
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Mode at 58 Hz and 62 Hz, measured.



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Room with tilted walls.



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Results



Mode at 58 Hz and 62 Hz, fit.

• Number of microphones varied from 5 to 50. 100 trials ran for each set, microphones placed in different randomized configurations in each trial.

• MSSIM - Mean structural similarity between measured and fit mode shapes. • AMSDE - Absolute mean spectral difference error. Absolute difference between the frequency responses (averaged over all measurement points) and all configurations) of the measured and modeled RIR, expressed in dB.



References

[1] Y. Haneda, Y. Kaneda, and N. Kitawaki, "Common-acoustical-pole and residue model and its application to spatial interpolation and extrapolation of a room transfer function," IEEE Trans. on Speech and Audio Process., vol. 7, no. 6, pp. 709–717, 1999. [2] C. Kereliuk, W. Herman, R. Wedelich, and D. J. Gillespie, "Modal analysis of room impulse responses using subband ESPRIT,"

[3] P. Ainsleigh and J. George, "Modeling exponential signals in a dispersive multipath environment," in *Proceedings of IEEE Int.*