JOINT ESTIMATION OF THE ROOM GEOMETRY AND MODES WITH COMPRESSED SENSING



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April 19, 2018

Content Atlas



[1] R. Mignot, G. Chardon and L. Daudet, "Low Frequency Interpolation of Room Impulse Responses Using Compressed Sensing," in IEEE/ACM Transactions on Audio, Speech, and Language Processing, vol. 22, no. 1, pp. 205-216, Jan. 2014. helena.peictukuljac@epfl.ch [2] H. Peic Tukuljac, H. Lissek, and P. Vandergheynst, "Localization of sound sources in a room with one microphone," SPIE, Wavelets and Sparsity XVII, vol. 10394, 2017. [3] R. Boulandet, "Tunable Electroacoustic Resonators through Active Impedance Control of Loudspeakers", PhD thesis, EPFL, Lausanne, 2012

 Δu

Spatio-Temporal Sampling Relations

 Δt

ω

Sampling steps:

Sampling frequencies:

$$=\frac{2\pi}{\Delta t} \quad \varphi_x = \frac{2\pi}{\Delta x} \quad \varphi_y = \frac{2\pi}{\Delta y} \quad \varphi_z = \frac{2\pi}{\Delta y}$$

[3]

 Δx

Two points of view on spatial sampling:



Connected plane wave sparsity and compressed sensing in 2014

 Δz

Choosing Δt is easy: [1]

$$\omega \geq 2\omega_{\rm cutoff}$$

View #2: Courant–Friedrichs–Lewy condition[4]

If a wave is moving across a discrete spatial grid and we want to compute its amplitude at discrete time steps of equal duration, then this duration must be less than the time for the wave to travel to *adjacent grid points*

$$\varphi_x^2 + \varphi_y^2 \le \frac{\omega^2}{c^2} \quad \Leftrightarrow \frac{c\Delta t}{\Delta x_i} \le \frac{1}{\sqrt{2}}$$
$$\varphi_x^2 + \varphi_y^2 + \varphi_z^2 \le \frac{\omega^2}{c^2} \quad \Leftrightarrow \frac{c\Delta t}{\Delta x_i} \le \frac{1}{\sqrt{3}}$$

Can we introduce a parametric solution and some assumptions that will reduce the complexity of the problem?

[1] H. Nyquist, "Certain topics in telegraph transmission theory," Transactions of the American Institute of Electrical Engineers, vol. 47, no. 2, pp. 617–644, April 1928.

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Wave equation and its parametrized solution

$$\Delta p(t, \mathbf{X}) - \frac{1}{c^2} \frac{\partial^2}{\partial^2 t} p(t, \mathbf{X}) = 0$$

$$p(t, \mathbf{X}) = \sum_{q \in \mathbb{I}} A_q \Phi_q(\mathbf{X}) g_q(t)$$

Spatial dependency:

$$\Phi_q(\mathbf{X}) \approx \sum_{r=1}^R a_{q,r} e^{j\mathbf{k}_{q,r} \cdot \mathbf{X}}$$

T

Temporal dependency:

$$= e^{jk_qct}$$
 $k_q = \frac{\omega_q - j\xi_q}{c}$

$$p(t, \mathbf{X}) = \sum_{q, r} \alpha_{q, r} e^{j(k_q ct + \mathbf{k}_{q, r} \cdot \mathbf{X})}$$

The goal: Fast and efficient **parameter learning** from a *small* number of microphone measurements

$$p(\mathbf{t}, \mathbf{X}) = \sum_{q, r} \alpha_{q, r} e^{j((\omega_q - j\xi_q)\mathbf{t} + \mathbf{k}_{q, r} \cdot \mathbf{X})}$$

 $g_q(t)$

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A set of assumptions for a well-posed problem

What natural **assumptions** can we introduce to reduce the complexity of the mathematical model at a low cost of approximation losses?

Assumption #1: simple (rectangular) room geometries $L_x \times L_y \times L_z$

Axial modes imply room shape

800

1000

1200



Assumption #2: lightly damped rooms



$$\xi \ll \omega \quad |{f k}| pprox \left|rac{\omega}{c}
ight| \ = \sqrt{k_x^2 + k_y^2 + k_z^2} \, {
m Spherical Search space}^{
m Spherical space}$$

Separability of spatial and temporal parameter estimation:

2. *spatial*: estimate the direction of ${f k}$





[1] Kuttruff, H. and Mommertz, E., "Room Acoustics", Springer Berlin Heidelberg, Berlin, Heidelberg(2013).

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ac @epfl.ch [2] R. Mignot, G. Chardon and L. Daudet, "Low Frequency Interpolation of Room Impulse Responses Using Compressed Sensing," in IEEE/ACM Transactions on Audio, Speech, and Language Processing, vol. 22, no. 1, pp. 205-216, Jan. 2015 [3] H. Peic Tukuljac, H. Lissek, and P. Vandergheynst, "Localization of sound sources in a room with one microphone," SPIE, Wavelets and Sparsity XVII, vol. 10394, 2017.

Three points of view on room modes

	Plane waves	Pressure isosurfaces	Wave vectors in search space $r = \frac{\omega}{c}$
<i>x</i> -axial mode			2.0 1.5 1.0 0.5 1.0 1.5 1.0 1.5 1.0 1.5 1.0 1.5 1.0 1.5 1.0 1.5 1.0 1.5 1.0 1.5 1.0 1.5 1.0 1.5 1.0 1.5 1.0 1.5 1.0 1.5 1.5 1.5 1.5 1.5 1.5 1.5 1.5
<i>xy</i> -tangential mode			^{2.0} 1.5 1.0 0.5 -2.0 1.5 -2.0 1.5 -2.0 K[X] 05 1.0 1.5 -2.0 -0.5 -1.0 -1.5 -2.0 -0.5 -1.0 -1.5 -2.0 -0.5 -1.5 -2.0 -0.5 -1.5 -1.5 -1.5 -1.5 -1.5 -1.5 -1.5 -1
oblique mode			-2.0_1.5_1.0_0.5_0.0_1.5_1.0_0.5_1.0_0.5_1.0_0.5_1.5_0_0.5_0.5_0.5_0.5_0.5_0.5_0.5_0.5_0.5

Focusing on the low frequencies and room mode assignment



ReSEMblE algorithm



procedure RESEMBLE(R, X)

for $i_l \in \{1, ..., N_l\}$ do

Separate the measurements with f_p : $\mathbf{R} = \mathbf{R}^l + \mathbf{R}^h$.

step 1: estimate $(\omega_{i_l}, \xi_{i_l})$ from $\mathbf{R}_{i_l}^l$

Temporal dictionary:

 $\theta[i] = e^{\xi_n[i]t} e^{j\omega_n[i]t}$

ReSEMblE algorithm



```
procedure RESEMBLE(\mathbf{R}, \mathbf{X})

Separate the measurements with f_p: \mathbf{R} = \mathbf{R}^l + \mathbf{R}^h.

for i_l \in \{1, ..., N_l\} do

step 1: estimate (\omega_{i_l}, \xi_{i_l}) from \mathbf{R}_{i_l}^l

step 2: estimate \mathbf{k}_{i_l} from \mathbf{r}_{i_l}^l

step 3: compute new residual \mathbf{R}_{i_l+1}^l

end for

Recover the room size \tilde{L}_x, \tilde{L}_y, \tilde{L}_z from basic axial
```

recover the room size D_x , D_y , D_z from basic tank room modes and form the regular wave vector grid. for $i_h \in \{N_l + 1, ..., N\}$ do step 1: get ω_{i_h} and \mathbf{k}_{i_h} from the wave vector grid step 2: estimate ξ_{i_h} from $\mathbf{R}_{i_h}^h$ step 3: compute new residual $\mathbf{R}_{i_h+1}^h$ end for Estimate the expansion coefficients $\{\alpha\}_{n=1,v=1}^{N,V}$ using least square approach. end procedure



[1] J. A. Tropp, A. C. Gilbert and M. J. Strauss, "Simultaneous sparse approximation via greedy pursuit," *Proceedings. (ICASSP '05). IEEE International Conference on Acoustics, Speech, and Signal Processing, 2005.*, 2005, **helena.peictukuljac@epfl.ch** pp. v/721-v/724 Vol. 5. doi: 10.1109/ICASSP.2005.1416405

Structured group sparsity

Separability of spatial and temporal parameter estimation:

1. temporal: estimate ω and build a ball $r = \frac{\omega}{c}$

2. *spatial*: estimate the direction of ${f k}$





ReSEMblE algorithm

Imply the wave vectors from the reconstructed room shape:

$$(n_x, n_y, n_z) \in \mathcal{N}_0^3 \setminus (0, 0, 0)$$

$$k_x = n_x \frac{\pi}{\tilde{L}_x} \qquad \qquad k_y = n_y \frac{\pi}{\tilde{L}_y} \qquad \qquad k_z = n_z \frac{\pi}{\tilde{L}_z}$$

After applying the *high* part of the algorithm, the Pearson correlation coefficient that the approximation is good (e.g. 82% for only 19-microphone setting and $f_c = 200$ Hz), but it should be further improved once the deviation of the wave vectors is efficiently characterized

procedure RESEMBLE(R, X) Separate the measurements with f_p : $\mathbf{R} = \mathbf{R}^l + \mathbf{R}^h$. for $i_l \in \{1, ..., N_l\}$ do step 1: estimate $(\omega_{i_l}, \xi_{i_l})$ from $\mathbf{R}_{i_l}^l$ step 2: estimate \mathbf{k}_{i_l} from $\mathbf{r}_{i_l}^l$ step 3: compute new residual $\mathbf{R}_{i_{1}+1}^{l}$ end for Recover the room size $\tilde{L}_x, \tilde{L}_y, \tilde{L}_z$ from basic axial room modes and form the regular wave vector grid. for $i_h \in \{N_l + 1, ..., N\}$ do step 1: get ω_{i_h} and \mathbf{k}_{i_h} from the wave vector grid step 2: estimate ξ_{i_h} from $\mathbf{R}_{i_h}^h$ step 3: compute new residual $\mathbf{R}_{i_{k+1}}^{h}$ end for Estimate the expansion coefficients $\{\alpha\}_{n=1}^{N,V}$ using least square approach. end procedure



In the spirit of open research and acoustic data augmentation

⑦ 7 commits	¥ 1 branch	© 0 roleases	11	1 contributor	
Branch: master +		Create new file Uploa	d files Find file	Close or download •	
Helle Add files vie upload			Latest comm	nit anazza: 28 days ago	
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matlab_code	Add tiles via upload	29 days ago			
🖿 readme_images	Add this via upload	a month ago			
README.md	Update README.md			29 days ago	
UE README.md					

JOINT ESTIMATION OF ROOM GEOMETRY AND MODES



February 8, 2018

Preview

Room Impulse Response measurements of a rectangular room

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About the measurements

This archive contains the data for a Gitlab hosted project (https://github.com/epfl-

pass filtering, downsampling and truncation in time domain

Its2/joint_estimation_of_room_geometry_and_modes). This archive allows users to extract the Room impulse responses (RIRs) measurements for a real rectangular room. A total of 132 measurements are included. Furthermore, users have the option to perform several post processing steps on the RIRs such as filtering, downsampling and truncation in time domain. Please refer to the guidelines pdf for more information.

1 of 2 - + Automatic Zoom=



aset Open Access

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Publication date: February 8, 2018 DOI: DOI 10.5281/zenodo.1169161 Keyword(s): ICASSE rectangular room Meetina: ICASSP 2018, Calgary, Alberta, Canada, 15–20 April 2018 License (for files): Creative Commons Attribution 4.0 Versions Version 1.0 10.5281/zenodo.1169161 Feb 8, 2018 Cite all versions? You can cite all versions by using the DOI 10.5281/zenodo.1169160. This DOI represents all versions and will always resolve to the latest one. Read more

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[1] <u>https://github.com/epfl-lts2/joint estimation of room geometry and modes</u> [2] <u>https://zenodo.org/record/1169161#.WquZD-jwbmF</u>

The room used for measurements (top) and the positions of the microphones

and speaker for the measurements (right)

 Measurements are done in 12 groups, each group contains 11 microphones placed at different positions. This gives a total of 132 microphone measurements.

- The original data file is TF data.mat which contains 132 transfer functions corresponding to



[1] R. Mignot, G. Chardon and L. Daudet, "Low Frequency Interpolation of Room Impulse Responses Using Compressed Sensing," in *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, vol. 22, no. 1, pp. 205-216, Jan. 2014 helena.peictukuljac@epfl.ch [2] H. Peic Tukuljac, H. Lissek, and P. Vandergheynst, "Localization of sound sources in a room with one microphone," SPIE, Wavelets and Sparsity XVII, vol. 10394, 2017. [3] R. Boulandet, "Tunable Electroacoustic Resonators through Active Impedance Control of Loudspeakers", PhD thesis, EPFL, Lausanne, 2012