Data-Dependent Initialization for ECM-Based Blind Geometry Estimation of a Microphone Array Using Reverberant Speech

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https://gitlab.com/signal-processing-uol/ecm-based-microphone-array-geometry-estimation

Introduction

- Reverberant speech contains a direct and **reverberation** component
- **Coherence** of the reverberation encodes **distances** between microphones and can be estimated using expectation conditional-maximization (ECM) [1]
- ECM requires **initial estimates** of direct and reverberation power spectral densities (PSDs), which can be estimated using a **data-independent coherence** matrix and a **matched beamformer** [2]

MAIN IDEA

Replace with a **data-dependent** coherence matrix estimate and a **minimum-power distortionless response** (MPDR) beamformer to reduce the MAG estimation error

Geometry Estimation

Reverberant Speech Signal Model (STFT-Domain)

$$\underbrace{\mathbf{x}[k,l]}_{\text{reverberant speech}} = \underbrace{\mathbf{g}[k]X_{d,1}[k,l]}_{\text{direct}} + \underbrace{\mathbf{x}_r[k,l]}_{\text{reverberation}}$$

k: freq. index, l: frame index

- $\mathbf{g}[k] = \exp(-j2\pi\tau k/K)$: Relative direct transfer function (RDTF) vector

Covariance Matrices

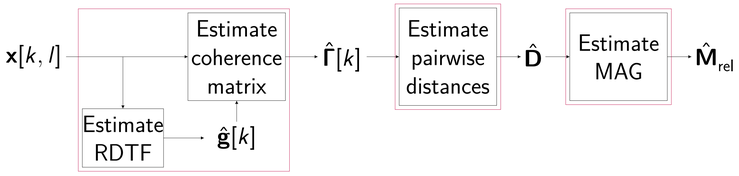
$$\underbrace{\mathbf{\Phi}_x[k,l]}_{\text{cov. of recorded speech}} = \underbrace{\phi_d[k,l]\mathbf{g}[k]\mathbf{g}^{\mathsf{H}}[k]}_{\text{cov. of direct speech } \mathbf{\Phi}_d[k,l]} + \underbrace{\phi_r[k,l]\mathbf{\Gamma}[k]}_{\text{cov. of reverberation } \mathbf{\Phi}_r[k,l]}$$

- $\mathbf{\Gamma}[k]$: coherence matrix of the reverberation
- $\phi_d[k,l]$: direct speech PSD
- $\phi_r[k,l]$: reverberation PSD

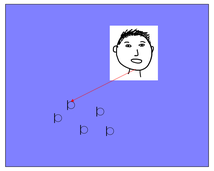
Assumptions:

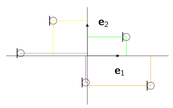
- Direct and reverberation components uncorrelated
- Reverberation spatially homogeneous sound field

Overview:



- $\mathbf{D}$: matrix of squared pairwise distances between the microphones
- $\mathbf{M}_{\text{rel}} = [\mathbf{m}_{\text{rel},1}, \mathbf{m}_{\text{rel},2}, \ldots, \mathbf{m}_{\text{rel},M}]$: MAG, i.e., relative microphone coordinates





1. Estimate the coherence matrix of the reverberation $\hat{\mathbf{\Gamma}}[k]$ using ECM [1, 2]
2. Estimate the pairwise distances (PDs) between the microphones $d_{a,b}$ by comparing estimated coherence $\hat{\Gamma}_{a,b}[k]$ with a coherence model $\gamma(k, d_{a,b}) = \operatorname{sinc}(\varepsilon\, k\, d_{a,b})$, for a specific range of frequencies [3, 2]
3. Estimate the MAG $\hat{\mathbf{M}}_{\text{rel}}$ using multidimensional scaling [3, 4]

ECM Initialization

ECM requires **initial estimates** of the direct speech PSD $\hat{\phi}_d[k,l]$ and Reverberation PSD $\hat{\phi}_r[k,l]$. These are estimated using a beamformer $\mathbf{h}[k,l]$ and an initial coherence matrix estimate $\hat{\mathbf{\Gamma}}^{(0)}[k]$ [5]

State-of-the-art [2]

$$\hat{\mathbf{\Gamma}}_{\mathbf{I}}^{(0)}[k] = \mathbf{I}, \quad k$$

$$\mathbf{h}_{\text{MVDR}}[k] = \frac{\hat{\mathbf{g}}[k]}{||\hat{\mathbf{g}}[k]||^2}$$

- $\hat{\mathbf{\Gamma}}_{\mathbf{I}}^{(0)}[k]$ is **data-independent**
- Time-invariant **matched beamformer**

Proposed

$$\hat{\mathbf{\Gamma}}_x^{(0)}[k] = \frac{1}{L}\sum_{l=1}^{L} \frac{\mathbf{x}[k,l]\mathbf{x}^{\mathsf{H}}[k,l]}{\frac{1}{M}\mathrm{Tr}(\mathbf{x}[k,l]\mathbf{x}^{\mathsf{H}}[k,l])}$$

$$\mathbf{h}_{\text{MPDR}}[k,l] = \frac{(\hat{\mathbf{\Phi}}_x^{(0)})^{-1}[k,l]\hat{\mathbf{g}}[k]}{\hat{\mathbf{g}}^{\mathsf{H}}[k](\hat{\mathbf{\Phi}}_x^{(0)})^{-1}[k,l]\hat{\mathbf{g}}[k]}$$

- $\hat{\mathbf{\Gamma}}_x^{(0)}[k]$ is **data-dependent**
- Time-varying **MPDR beamformer**

Experimental Evaluation

Framework and Acoustical Parameters

- Simulated acoustic scenarios using RIR generator [6]
- 50 acoustic scenarios (random 5 s speech signal, array location & geometry, and speech source location) simulated per experiment
- 6 × 6 × 2.4 m room with equally reflective walls and DRR = 0 dB (at ref. mic.)
- 32 ms frame length with 50% overlap between frames

Experiment 1:

Analysis of the median PD error $\epsilon_{\text{PD}} = \frac{|\hat{d}_{1,2} - d_{1,2}|}{d_{1,2}}$

$M = 2$ microphones $d_{1,2} = 10$ cm apart and frequency range varied between (0, 4] kHz

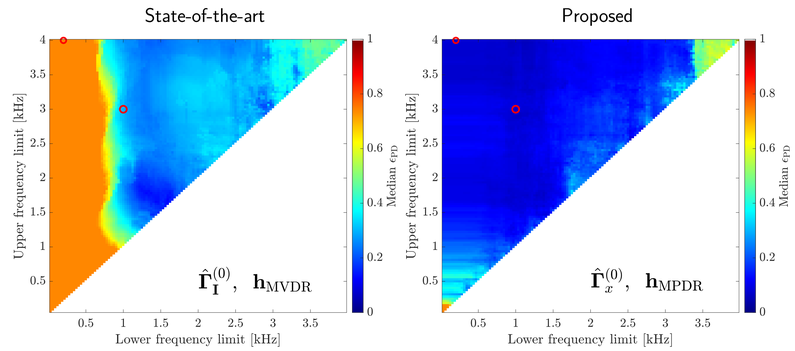
Experiment 2:

Analysis of the normalized MAG error $\epsilon_{\mathbf{m}_m} = \frac{||\hat{\mathbf{m}}_m - \mathbf{m}_m||_2}{\text{CL}}$

$M = 6$ microphones randomly located within cubes with cube lengths (CLs) 10, 20, ... 50 cm

Results

Experiment 1:

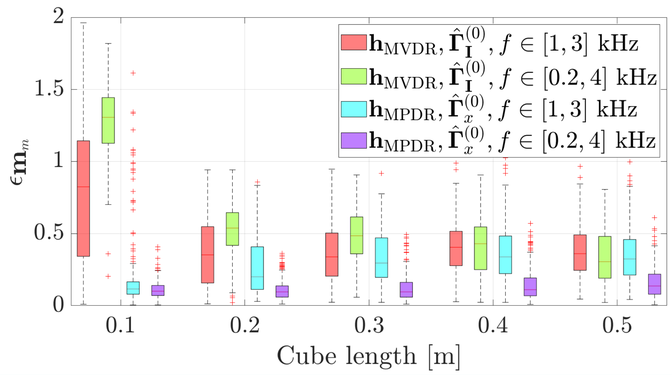


- Red circles indicate the considered frequency ranges in experiment 2

Conclusions

- X Proposed initialization greatly reduces PD errors at low frequencies
- X Proposed initialization allows a fixed lower frequency limit to be used which is **independent of the distances** between microphones

Experiment 2:



Conclusions

- X Proposed initialization significantly reduces MAG errors for all array sizes
- X MAG error is relatively **constant** for all considered array sizes when using proposed initialization

Future Work

- Comparison of estimation methods of the coherence matrix of the reverberation
- Integration/combination with blind MAG estimation approaches based on time-difference of arrivals (TDoAs)

References

[1] X.-L. Meng and D. B. Rubin, "Maximum likelihood estimation via the ECM algorithm: A general framework," Biometrika, vol. 80, no. 2, pp. 267–278, 1993.

[2] O. Schwartz, A. Plinge, E. Habets, and S. Gannot, "Blind microphone geometry calibration using one reverberant speech event," in Proc. IEEE Workshop Appl. Signal Process. Audio. Acoust., 2017, pp. 131–135.

[3] I. McCowan, M. Lincoln, and I. Himawan, "Microphone array shape calibration in diffuse noise fields," Proc. IEEE Trans. Audio, Speech, Language Process., vol. 16, no. 3, pp. 666–670, 2008.

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[5] H. Ye and R. DeGroat, "Maximum likelihood DOA estimation and asymptotic Cramér-Rao bounds for additive unknown colored noise," IEEE Trans. Speech Audio Process., vol. 43, no. 4, pp. 938–949, 1995.

[6] E. Habets, RIR-Generator, available at https://github.com/ehabets/RIR-Generator.