

Alternating Least-Squares-Based Microphone Array Parameter Estimation for A Single-Source Reverberant and Noisy Acoustic Scenario

Changheng Li and Richard C. Hendriks

PRELIMINARIES

Motivation

- Spatial filtering techniques depend on microphone array parameters: source RTF, source PSDs, late reverberation PSDs and ambient noise PSDs.
- Existing estimators either assume some of these parameters known or suffer from high computational cost.
- A joint estimator at a low complexity is in demand.

Signal model

- For a single source reverberant and noisy scenario (STFT domain),

$$\mathbf{y}(l, k) = \mathbf{a}(l, k)s(l, k) + \mathbf{r}(l, k) + \mathbf{v}(l, k).$$
- The noisy covariance matrix (omitting frequency bin index):

$$\Phi_y(l) = \underbrace{\phi_s(l)\mathbf{a}\mathbf{a}^H}_{\text{Late reverb}} + \underbrace{\phi_\gamma(l)\Gamma}_{\text{Ambient noise}} + \underbrace{\phi_v(l)\Psi}_{\text{Ambient noise}}$$

- \mathbf{a} is constant for N sequential time frames, Γ and Ψ are time-invariant known spatial coherence matrices.
- Goal: estimate \mathbf{a} , $\phi_s(l)$, $\phi_\gamma(l)$ and $\phi_v(l)$ using $\hat{\Phi}_y(l)$, given Γ and Ψ for $l = 1, \dots, N$.

Minimize the model mismatch error using the sum of Frobenius norms:

$$\arg \min_{\mathbf{a}, \phi_s(l), \phi_\gamma(l), \phi_v(l)} \sum_{l=1}^N \left\| \hat{\Phi}_y(l) - \phi_s(l)\mathbf{a}\mathbf{a}^H - \phi_\gamma(l)\hat{\Gamma} - \phi_v(l)\hat{\Psi} \right\|_F^2.$$

Reparameterization to avoid ambiguous issue: $\tilde{\mathbf{a}} = \frac{\mathbf{a}}{\sqrt{\mathbf{a}^H \mathbf{a}}}$ and $\tilde{\phi}_s(l) = \phi_s(l)\mathbf{a}^H \mathbf{a}$.

Alternating estimation

For given $\tilde{\mathbf{a}}$, estimate the PSDs $\tilde{\phi} = \{\tilde{\phi}_s(l), \tilde{\phi}_\gamma(l), \tilde{\phi}_v(l)\}$ by $\tilde{\phi}(l) = \tilde{\Phi}_{\tilde{\mathbf{a}}}^{-1}\tilde{\mathbf{b}}(l)$, with

$$\tilde{\Phi}_{\tilde{\mathbf{a}}} = \begin{bmatrix} 1 & \hat{\mathbf{a}}^H \hat{\Gamma} \hat{\mathbf{a}} & \hat{\mathbf{a}}^H \hat{\Psi} \hat{\mathbf{a}} \\ \hat{\mathbf{a}}^H \hat{\Gamma} \hat{\mathbf{a}} & \text{tr} \left\{ \hat{\Gamma}^H \hat{\Gamma} \right\} & \text{tr} \left\{ \hat{\Gamma}^H \hat{\Psi} \right\} \\ \hat{\mathbf{a}}^H \hat{\Psi} \hat{\mathbf{a}} & \text{tr} \left\{ \hat{\Gamma}^H \hat{\Psi} \right\} & \text{tr} \left\{ \hat{\Psi}^H \hat{\Psi} \right\} \end{bmatrix}, \quad \tilde{\mathbf{b}}(l) = \begin{bmatrix} \hat{\mathbf{a}}^H \hat{\Phi}_y(l) \hat{\mathbf{a}} \\ \text{tr} \left\{ \hat{\Gamma}^H \hat{\Phi}_y(l) \right\} \\ \text{tr} \left\{ \hat{\Psi}^H \hat{\Phi}_y(l) \right\} \end{bmatrix}.$$

For given $\tilde{\phi}$, $\hat{\mathbf{a}} = \text{principal eigenvector of } \sum_{l=1}^N \hat{\phi}_s(l) [\hat{\Phi}_y(l) - \hat{\phi}_\gamma(l)\hat{\Gamma} - \hat{\phi}_v(l)\hat{\Psi}]$.

Robust PSDs constraints

Upper bounds:

$$\tilde{\phi}_s(l) \leq \min_m \left\{ \frac{\hat{\Phi}_{y_{m,m}}(l)}{|\hat{a}_m|^2} \right\}, \quad \phi_\gamma(l) \leq \min_m \left\{ \frac{\hat{\Phi}_{y_{m,m}}(l)}{\hat{\Gamma}_{m,m}} \right\}, \quad \phi_v(l) \leq \min_m \left\{ \frac{\hat{\Phi}_{y_{m,m}}(l)}{\hat{\Psi}_{m,m}} \right\}.$$

Lower bounds (avoid negative estimates):

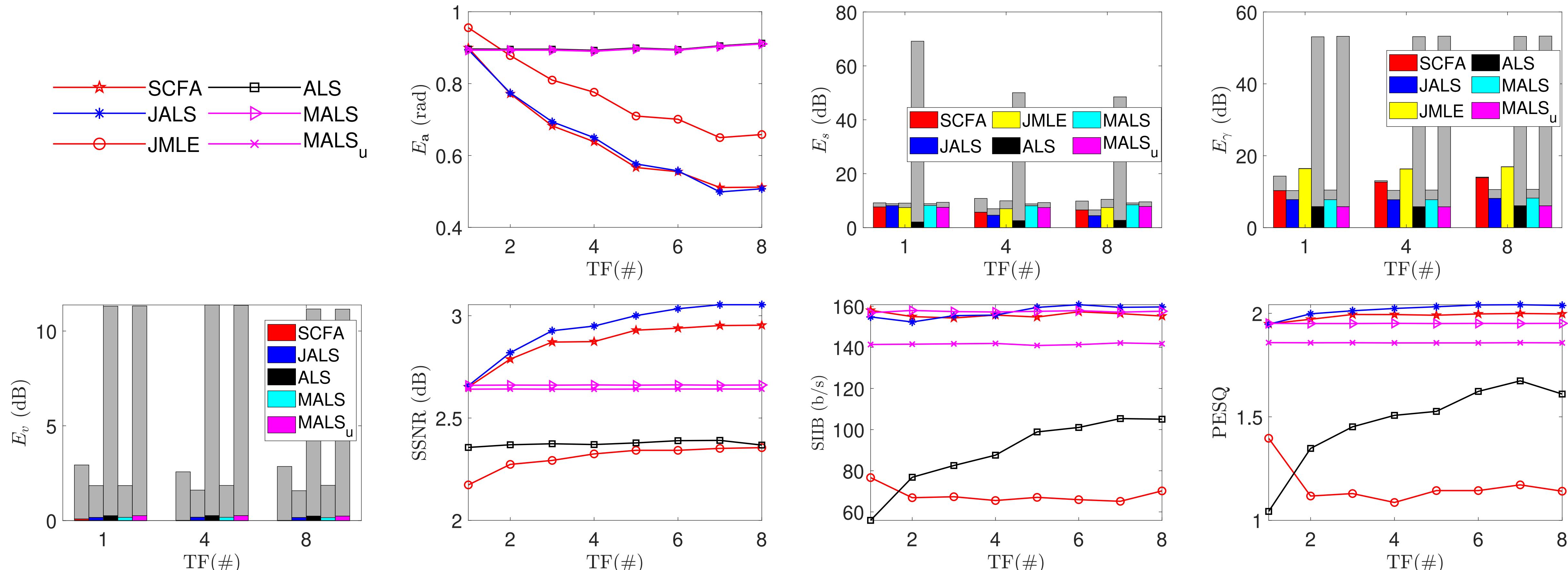
e.g. if $\hat{\phi}_\gamma(l) > 0$ while $\hat{\phi}_s(t) \leq 0$, we lower bound $\tilde{\phi}_s(l)$ by

$$\hat{\phi}_\gamma(l) \min_{t=1, \dots, l} \frac{\hat{\phi}_s(t)}{\hat{\phi}_\gamma(t)} \leq \tilde{\phi}_s(l).$$

EXPERIMENTS

$$E_a = \frac{1}{L(K/2+1)} \sum_{l=1}^L \sum_{k=1}^{K/2+1} \arccos \left(\frac{|\mathbf{a}^H(l, k)\hat{\mathbf{a}}(l, k)|}{\|\mathbf{a}^H(l, k)\|_2 \|\hat{\mathbf{a}}(l, k)\|_2} \right) \text{ (rad)}, \quad E_i = 10 \log_{10} \left(\phi_i / \hat{\phi}_i \right)$$

method	SCFA[1]	ALS[2]	MALS	JMLE	JALS
Normalized run time	154.65	6.27	5.7	1.66	1



[1] A. I. Koutrouvelis, R. C. Hendriks, R. Heusdens, and J. Jensen, "Robust joint estimation of multimicrophone signal model parameters," *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, vol. 27, no. 7, pp. 1136–1150, 2019.

[2] M. Tammen, S. Doclo, and I. Kodrasi, "Joint Estimation of RETF Vector and Power Spectral Densities for Speech Enhancement Based on Alternating Least Squares," in *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 2019, pp. 795–799.