Comparison of Generalized Sidelobe Canceller Structures Incorporating External Microphones for Joint Noise and Interferer Reduction

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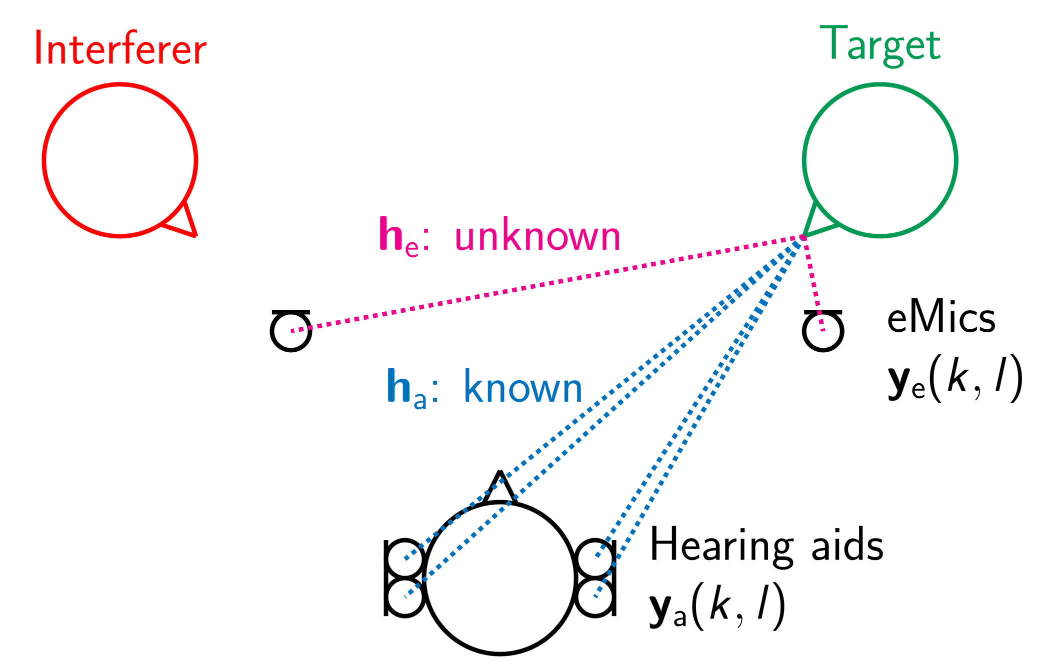
Problem Statement

- **Noise** and **interfering speaker** reduce intelligibility of target speaker
- Exploit **external microphones (eMics)** in conjunction with hearing aid microphones for speech enhancement
- MPDR-based beamforming to suppress undesired sources
  → **RTF vector of target speaker is required** to steer beamformer [1]
- **Blind estimation of target RTF vector** is difficult when interfering speaker is present
  → assume RTF vector for hearing aids to be known (e.g., frontal direction)
- **RTFs for eMics** are missing → need to be estimated

IN THIS POSTER

- Reduce noise and interferer by means of generalized sidelobe canceller (GSC) structures incorporating eMics [2]
- Pre-process local (and external) microphones to improve SIR and estimate external target RTFs more accurately
- Comparison of several GSC structures

Configuration and Notation



Signal model in STFT domain (hearing aid and external microphones):

$$\mathbf{y}(k,l) = \begin{bmatrix}\mathbf{y}_a(k,l)\\ \mathbf{y}_e(k,l)\end{bmatrix} = \underbrace{\mathbf{x}(k,l)}_{\text{desired}} + \underbrace{\mathbf{i}(k,l)}_{\text{interferer}} + \underbrace{\mathbf{n}(k,l)}_{\text{noise}}$$

Using relative transfer function (RTF) vectors: $\mathbf{x} = \mathbf{h}X_1, \quad \mathbf{i} = \mathbf{b}I_1, \quad \mathbf{h} = \begin{bmatrix}\mathbf{h}_a\\ \mathbf{h}_e\end{bmatrix}$

Accessibility of Information

- $\mathbf{R}_y = \phi_x \mathbf{h}\mathbf{h}^H + \phi_i \mathbf{b}\mathbf{b}^H + \mathbf{R}_n$ → assume that $\mathbf{R}_n$ can be estimated (e.g., VAD)
- Assume relative position of target speaker with respect to to hearing aids to be known:
  → local target RTF vector $\mathbf{h}_a$ known
  → external target RTF vector $\mathbf{h}_e$ and interferer RTF vector $\mathbf{b}$ unknown
  → to incorporate eMics in GSC structures $\mathbf{h}_e$ needs to be estimated

RTF Vector Estimation

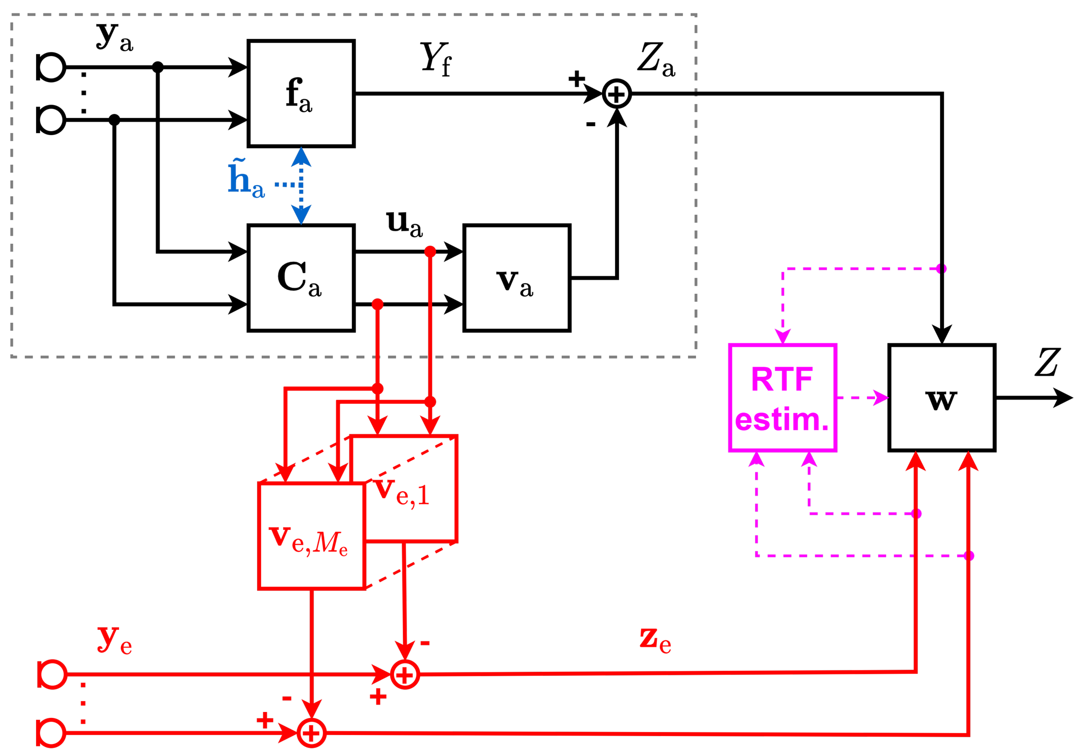
State-of-the-art RTF vector estimator **covariance whitening (CW)** [3]:

Whiten $(\mathbf{R}_y - \mathbf{R}_n)$: $\mathbf{R}_y^w = \phi_x \mathbf{R}_n^{-1/2}\mathbf{h}\mathbf{h}^H\mathbf{R}_n^{-H/2} + \phi_i \mathbf{R}_n^{-1/2}\mathbf{b}\mathbf{b}^H\mathbf{R}_n^{-H/2}$ → Principal eigenvector of $\mathbf{R}_y^w$: $\mathbf{v}_{\max} = \mathcal{P}\{\mathbf{R}_y^w\}$ → De-whiten and normalize: $\hat{\mathbf{h}}^{CW} = \frac{\mathbf{R}_n^{1/2}\mathbf{v}_{\max}}{\mathbf{e}_1^T\mathbf{R}_n^{1/2}\mathbf{v}_{\max}}$

Problem of Blind RTF Vector Estimation

$\mathbf{R}_y^w$ is rank-2 due to interfering speaker
→ CW will give **biased RTF vector estimate!**
→ Dependence on multi-channel signal-to-interferer ratio (SIR)

GSC Structures



1. Local GSC (L-GSC) [4, 5]

- Only uses hearing aid microphones (gray box)
- Exploits a-priori RTF vector $\tilde{\mathbf{h}}_a$

1. Fixed beamformer $\mathbf{f}_a$ → speech ref. Y_f
2. Blocking matrix $\mathbf{C}_a$ → noise-and-interferer refs. $\mathbf{u}_a$
3. Filter $\mathbf{v}_a$ → reduces correlation between Y_f and $\mathbf{u}_a$ to create output Z_a

2. GSC with External Speech References (GSC-ESR)

- **Novelty: Change MVDR [2] to MPDR implementation** to cancel interferer (complete diagram)
- Pre-process eMic signals $\mathbf{y}_e$ by noise-and-interferer refs. $\mathbf{u}_a$ and filters $\mathbf{v}_{e,m_e}$
- Enhanced local output Z_a and enhanced eMic signals $\mathbf{z}_e$ lead to higher mean SIR and better estimation of external RTF vector $\mathbf{h}_e$
  → used in joint beamformer $\mathbf{w}$

3. GSC with External References (GSC-ER)

- Simplified version of GSC-ESR (complete diagram without filters $\mathbf{v}_{e,m_e}$)
- No pre-processing of eMics
  → $\mathbf{v}_{e,m_e} = \mathbf{0}$
- Allows to assess benefit of pre-processing

Experimental Evaluation

- Reverberant recordings ($T_{60} \approx 350$ ms)
- 4 head-mounted microphones on a dummy head + 2 eMics
- Male target (35° to the right), female interferer (35° to the left)

Conditions

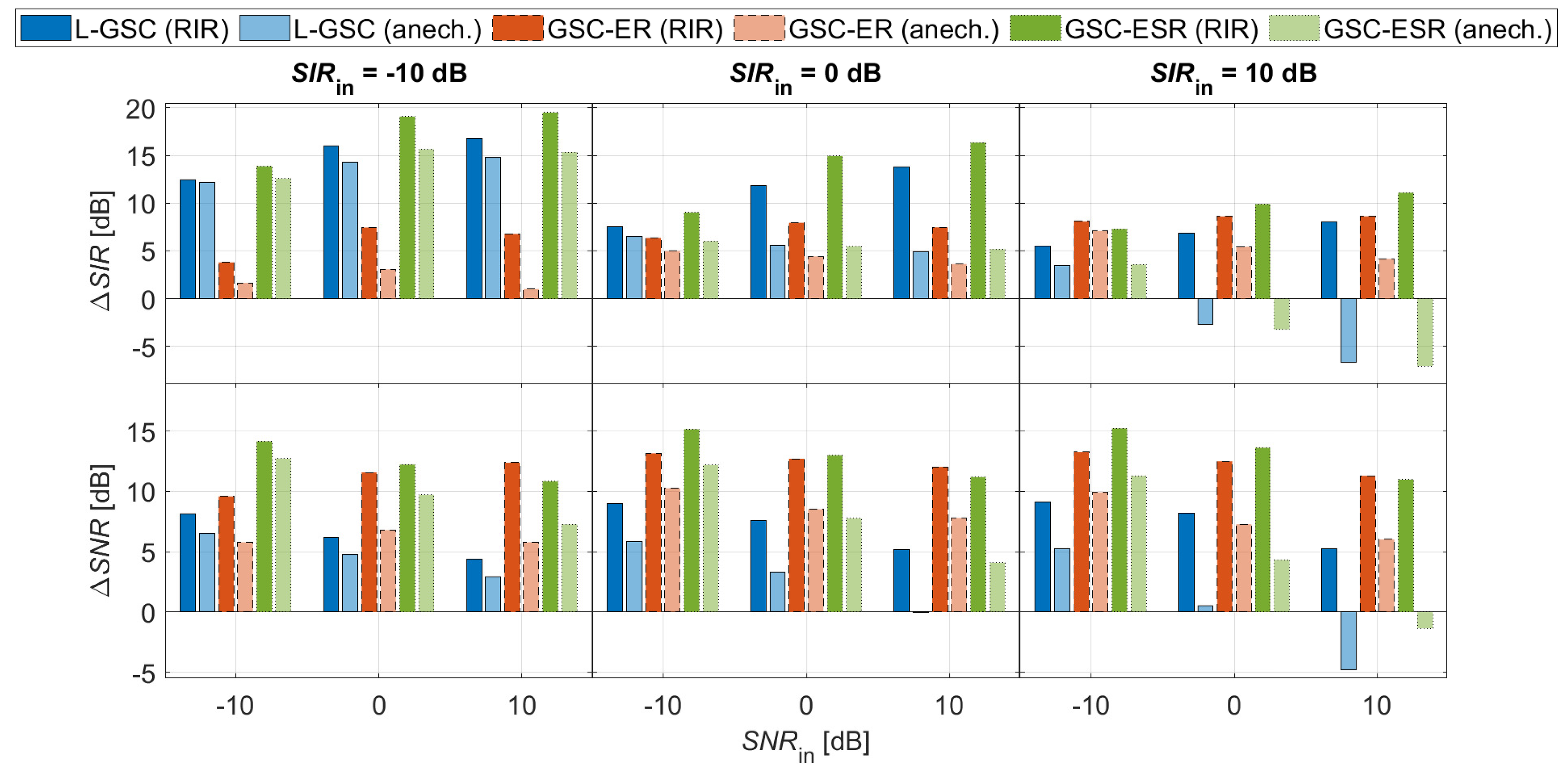
- $SIR_{in} = [-10, 0, 10]$ dB, $SNR_{in} = [-10, 0, 10]$ dB
- Two different a-priori RTF vectors $\tilde{\mathbf{h}}_a$:
  1. Reverberant RTF from measured target RIR
  2. Approximation from anechoic database [6]

Implementation and Framework

- Batch implementation
- 64 ms frame length with 50% overlap, sqrt-Hann window

Results

Evaluation of SNR improvement ΔSNR and SIR improvement ΔSIR



- Including eMics leads to better performance than processing hearing aids alone
- **Anechoic RTF vector** leads to overall lower scores than reverberant RTF vector
- **At high SIR:**
  - Using reverberant RTF vector: GSC-ESR and GSC-ER perform similarly (both better than L-GSC)
  - Using anechoic RTF vector: GSC-ESR performs worse than GSC-ER
  → target cancellation in eMic signals due to speech leakage in $\mathbf{u}_a$
- **At low SIR:** GSC-ESR outperforms L-GSC and GSC-ER in terms of noise and interferer reduction

Conclusions

✓ GSC-ESR outperforms L-GSC and GSC-ER in difficult conditions
  → Advantage of pre-processing eMics signals

✗ Sensitivity towards RTF vector mismatches
  → Especially for GSC-ESR at high SIR and SNR

Next Steps:
Analytical expression for performance of GSC structures

References

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[2] R. Ali, G. Bernardi, T. van Waterschoot, and M. Moonen, "Methods of extending a generalized sidelobe canceller with external microphones," IEEE/ACM Trans. on Audio, Speech, and Language Processing, vol. 27, pp. 1349–1364, Sep. 2019.

[3] S. Markovich-Golan and S. Gannot, "Performance analysis of the covariance-whitening and the covariance-subtraction methods for estimating the relative transfer function," in Proc. European Signal Processing Conference, Rome, Italy, Sep. 2018, pp. 544–548.

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[6] H. Kayser, S. D. Ewert, J. Anemüller, T. Rohdenburg, V. Hohmann, and B. Kollmeier, "Database of multichannel In-Ear and Behind-The-Ear Head-Related and Binaural Room Impulse Responses," Eurasip Journal on Advances in Signal Processing, vol. 2009, p. 10 pages, 2009.