Improving Blind Source Separation Performance By Adaptive Array Geometries for Humanoid Robots

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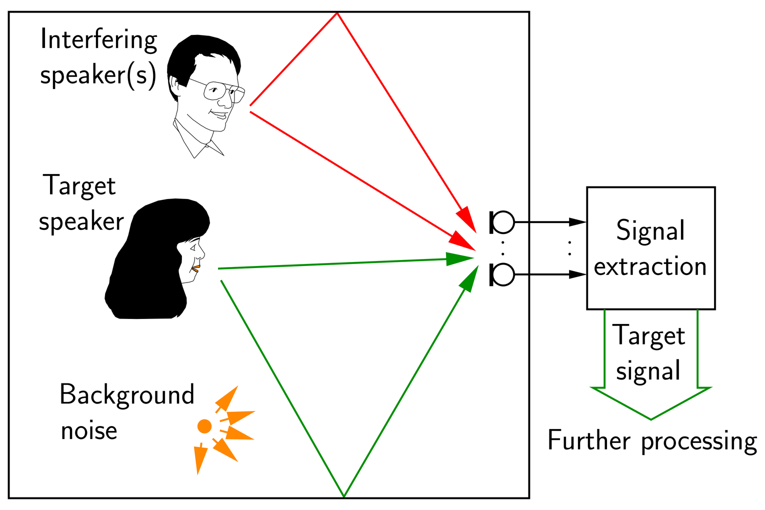
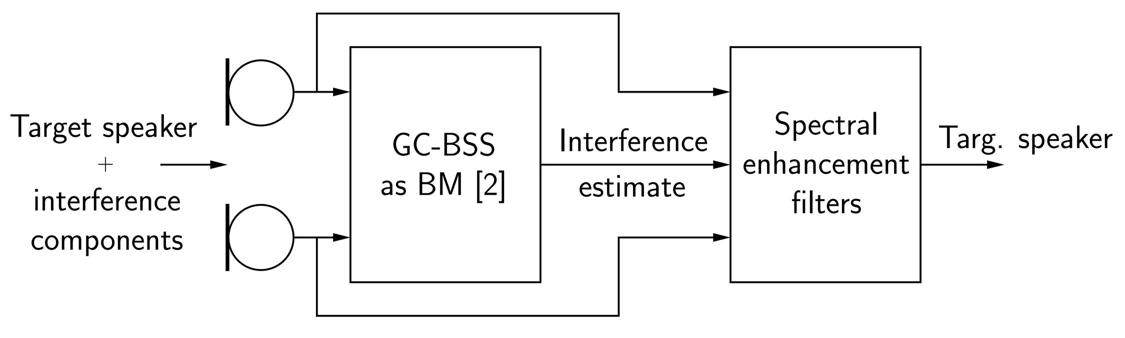
1. Introduction

Signal extraction scenario:



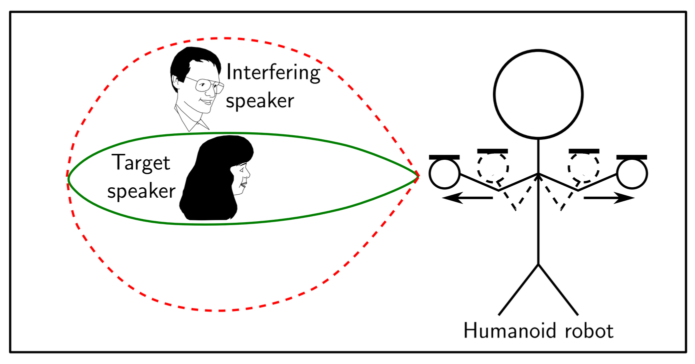
Goal: Extraction of desired source

Blind signal extraction (BSE) scheme [1]:



- Geometrically-constrained BSS (GC-BSS) acts as blocking matrix to estimate interference components
- Wiener-type spectral enhancement filters derived from interference estimate
- Extraction performance based on quality of interference estimate

Idea:



- Place microphones on movable limbs of a humanoid robot
- Use movable microphones to adapt microphone array topology to improve target signal suppression of GC-BSS
- Improved target signal suppression of GC-BSS leads to higher signal extraction performance of BSE [1]

2. Geometrically-constrained BSS

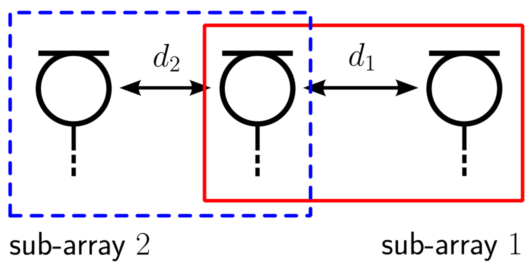
Cost function of GC-BSS [2]:

$$J_{\text{total}}(\mathbf{W}) = J_{\text{BSS}}(\mathbf{W}) + \gamma_{\text{C}} J_{\text{C}}(\mathbf{W}),$$

- $\mathbf{W}$: Matrix containing demixing filter weights
- $J_{\text{BSS}}(\mathbf{W})$: Generic cost function of TRINICON BSS [3]
- Minimization of $J_{\text{BSS}}(\mathbf{W})$ yields statistically independent output channels
- $J_{\text{C}}(\mathbf{W})$: Penalty term to force spatial null towards direction of target source in one output channel
- γ_{C}: Weight to control importance of penalty term, typically $0.4 < \gamma_{\text{C}} < 0.6$
- Direction of arrival ϕ_{t} of target source required for $J_{\text{C}}(\mathbf{W})$

3. Proposed Array Topology Adaptation Algorithm

Main concept:



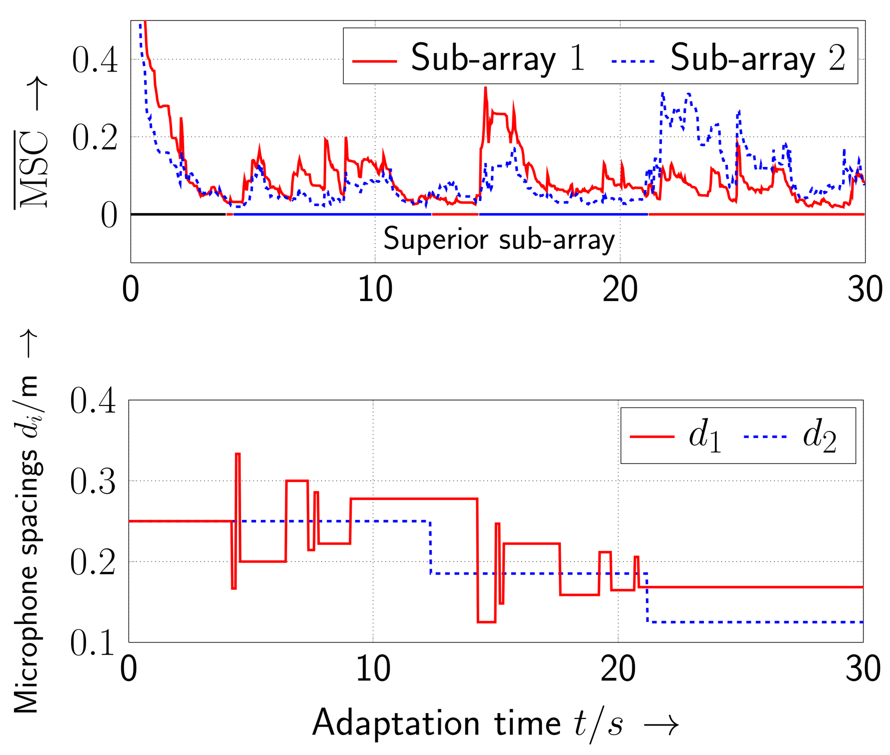
- Configuration of microphone array into two sub-arrays
- Use inferior sub-array to create new competitor for superior sub-array
- Adaptation of microphone spacing d_{inf} of inferior sub-array in dependence of microphone spacing d_{sup} of superior sub-array

$$d_{\text{inf}} = a_n d_{\text{sup}},\ n \in \{1, 2, 3, \ldots\},$$

with

$$a_n = \left(\frac{1+n}{2+n}\right)^{(-1)^{n+1}} = \left\{\frac{2}{3}, \frac{4}{3}, \frac{4}{5}, \frac{6}{5}, \frac{8}{7}, \ldots\right\}$$

Example of array topology adaptation:



- $\phi_{\text{t}} = 0°$, $\phi_{\text{int}} = 40°$, $T_{60} = 100\,\text{ms}$, Source distance: $1.0\,\text{m}$

4. Experiments

Evaluation measure:

- Weighted Magnitude Squared Coherence (MSC)

$$\overline{\text{MSC}} = \frac{1}{\sum_{\nu=0}^{\nu_{\max}} W(\nu)} \sum_{\nu=0}^{\nu_{\max}} W(\nu) \frac{|S_{y_1 y_2}(\nu)|^2}{S_{y_1 y_1}(\nu) \cdot S_{y_2 y_2}(\nu)},$$

with

$W(\nu)$: Window function

$S_{y_1 y_2}$, $S_{y_1 y_2}$, $S_{y_1 y_2}$: Auto- and cross-power spectrum densities of output channels

- Segmental target suppression gain

$$\Delta\overline{\text{TS}}_{\text{seg}} = \overline{\text{TS}}_{\text{seg},y_2} - \overline{\text{TS}}_{\text{seg},x}\ \text{dB},$$

with

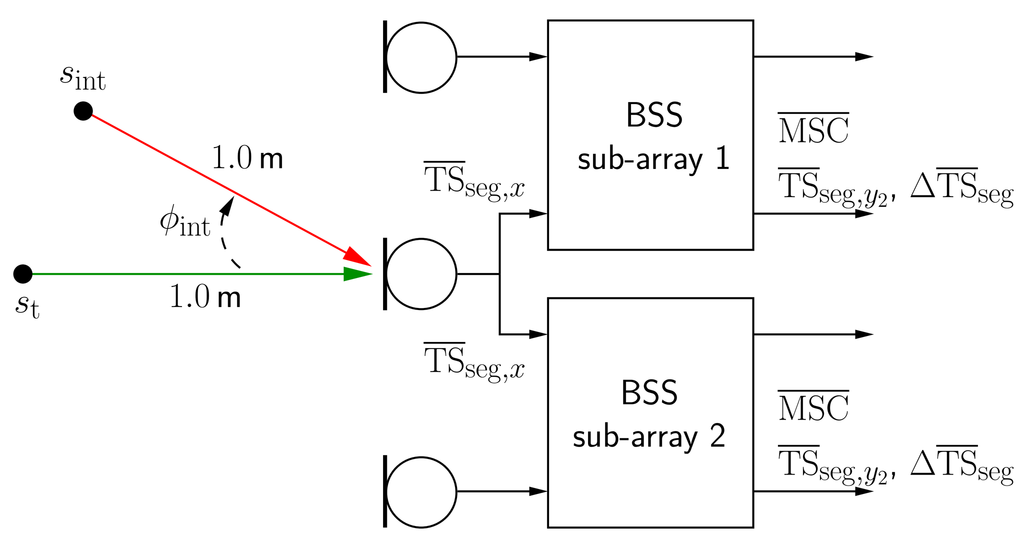
$$\overline{\text{TS}}_{\text{seg},y_2} = \frac{1}{K_{\text{S}}} \sum_{m=1}^{K_{\text{S}}} \left(10\log_{10}\left(\frac{\sum_{\kappa=1}^{N_{\text{S}}} y_{2,\text{int}}^2(\kappa + mN_{\text{S}})}{\sum_{\kappa=1}^{N_{\text{S}}} y_{2,\text{t}}^2(\kappa + mN_{\text{S}})}\right)\right)\ \text{dB}$$

$\overline{\text{TS}}_{\text{seg},x_2}$: Analogously to $\overline{\text{TS}}_{\text{seg},y_2}$

$y_{2,\text{t}}$, $y_{2,\text{int}}$: Target and interference components in second output channel

$N_{\text{S}} = 256, K_{\text{S}}$: Block length and number of blocks used to compute $\overline{\text{TS}}_{\text{seg},y}$ and $\overline{\text{TS}}_{\text{seg},y_2}$

Experimental Setup:



- $T_{60} = 100\,\text{ms}$, Source distance: $1.0\,\text{m}$
- $\phi_{\text{t}} = 0°$, $\phi_{\text{int}} \in \{20°, 40°, 60°, 80°\}$
- $f_s = 16\,\text{kHz}$, signal length: $60\,\text{s}$, Topology adaptation phase: $30\,\text{s}$
- Initial microphone spacings $d_i = 0.25\,\text{m}$
- Room impulse responses simulated using image method by Allen and Berkley [4]
- BSS: filter length $L = 512$, block length $N = 2L$

Results:

Adaptation	ϕ_{int}	$20°$	$40°$	$20°$	$80°$	Avg.
No	$\Delta\overline{\text{TS}}_{\text{seg}}$ [dB]	8.01	8.76	9.96	10.07	**9.20**
Yes	$\Delta\overline{\text{TS}}_{\text{seg}}$ [dB]	8.49	11.56	10.94	12.02	**10.75**

5. Conclusions

- Proposed array topology adaptation algorithm leads to improved separation, and thus, higher target source suppression performance
- By using array topology adaptation for humanoid robots, BSE algorithm [1] is expected to yield better signal extraction performance

References:

[1] K. Reindl, Y. Zheng, A. Schwarz, S. Meier, R. Maas, A. Sehr, and W. Kellermann, "A Stereophonic acoustic signal extraction scheme for noisy and reverberant environments," Computer Speech and Language (CSL), vol. 27, no. 3, pp. 726–745, May 2012.

[2] Y. Zheng, K. Reindl, and W. Kellermann, "BSS for improved interference estimation for blind speech signal extraction with two microphones," Int. Workshop on Comp. Advances in Multi-Sensor Adapt. (CAMSAP), Aruba, Dutch Antilles, pp. 253–256, Dec. 2009.

[3] H. Buchner, R. Aichner, and W. Kellermann, "TRINICON: A versatile framework for multichannel blind signal processing," IEEE Int. Conference on Acoustics, Speech and Signal Processing (ICASSP), vol. 3, pp. iii - 889-92, May 2004.

[4] J.B. Allen and D.A. Berkley, "Image method for efficiently simulating small-room acoustics," Journal Acoustic Society of America (JASA), vol. 65, no. 4, pp. 943–1086, April 1979.

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