Linear and Parametric Microphone Array Processing

Part III: Distributed Linear Spatial Processing

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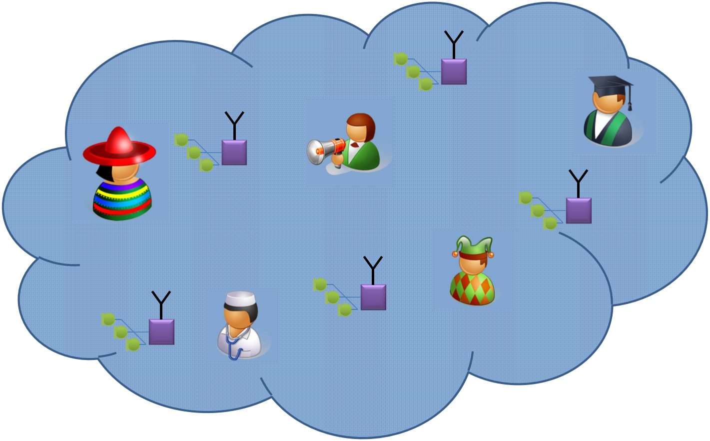
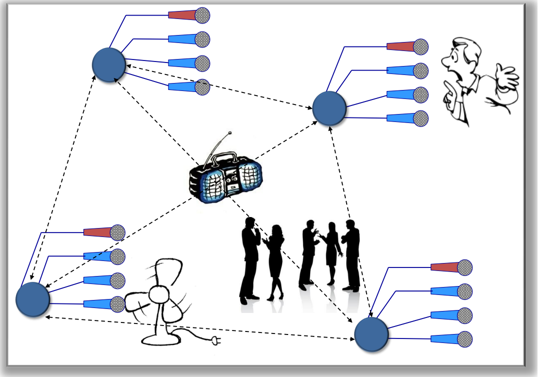
Distributed Algorithms for Microphone Arrays



Outline

Wireless Acoustic Sensor Networks (WASNs)



Advantages

- Microphones can be placed randomly, avoiding tedious calibration.
- Using more microphones improves spatial resolution.
- High probability to find microphones close to a relevant sound source.
- Improved sound field sampling.

Challenges of Distributed Beamforming

Power

- Communication bandwidth
- Computational complexity

Arbitrary Constellation

- Ad hoc
- Dynamics
- Calibration

Communication

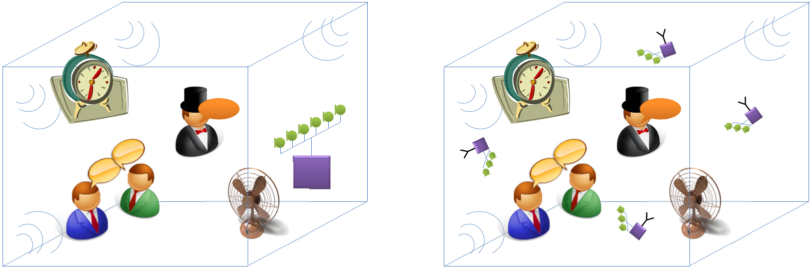
- Connectivity
- Protocol
- Capacity

Signal Processing

- Partial data
- Synchronization
- Dynamics
- Coherence

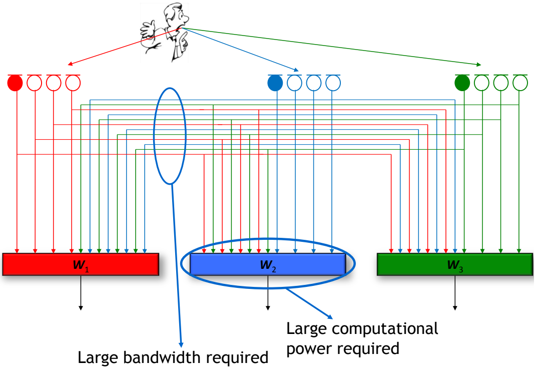
Goals

- Develop beamforming algorithms for distributed microphone constellation:
    - Ad hoc sensor networks.
    - Large volume (and many nodes).
- Robustness against randomly deployed microphones:
    - High fault percentage.
    - Arbitrary deployment of nodes.
- Applicability to Hearing Aids ([Doclo et al., 2009]; [Markovich-Golan et al., 2010]).



Centralized Network-Wide MWF (Fully Connected)

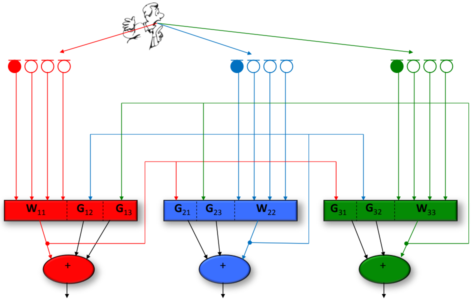
Courtesy of Dr. Alexander Bertrand, K.U.Leuven



- Each node has its own local reference microphone.
- Each node k solves network-wide MWF (here: 12-channel MWF).

Distributed Adaptive Node-specific Signal Estimation (DANSE)

Courtesy of Dr. Alexander Bertrand, K.U.Leuven



- Parameterized network-wide filter at node 1: $\mathbf{W}_1^T = \left[\mathbf{W}_{11} \; \mathbf{W}_{22}\mathbf{G}_{12} \; \ldots \; \mathbf{W}_{NN}\mathbf{G}_{1N} \right]$
- Node k adapts its filter coefficients $\mathbf{W}_{kk}$, $\mathbf{G}_{k1}, \ldots, \mathbf{G}_{kN}$ based on local MWF (here: 6-channel MWF).
- If single desired source: DANSE converges to centralized MWF

[Bertrand and Moonen, 2010a].

DANSE and Extensions

Courtesy of Dr. Alexander Bertrand, K.U.Leuven

- Can be generalized to Q desired sources [Bertrand and Moonen, 2010a].
- Small modification allows for simultaneous node updating [Bertrand and Moonen, 2010b].
- DANSE in networks with a tree topology (Tree-DANSE [Bertrand and Moonen, 2011]).
- LCMV-based DANSE (LC-DANSE [Bertrand and Moonen, 2012]).
- Robust DANSE (R-DANSE) for ill-conditioned scenarios (e.g., low-SNR nodes [Bertrand and Moonen, 2009]).
- Improved tracking using internal adaptive filters (this ICASSP [Szurley et al., 2013]).

Distributed LCMV

Formulation

- N nodes with M_n microphones.
- $\sum_{n=1}^{N} M_n = M$.
- $\mathbf{z} \triangleq \begin{bmatrix} \mathbf{z}_1^T & \cdots & \mathbf{z}_N^T \end{bmatrix}^T$.
- Closed-form LCMV necessitates the inversion of $\mathbf{\Phi}_{zz}$.
  A cumbersome task in distributed networks.

Naïve GSC Implementation

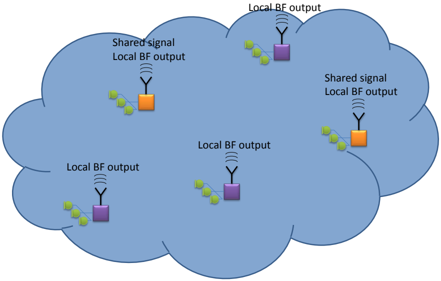
- Summation of local BFs: $y = \sum_{n=1}^{N} y_n$.
- Implement a local GSC at each node:
  - $M_n - P$ outputs of the BM at the nth node (might go negative!).
  - Total number of BM outputs: $\sum_{n=1}^{N}(M_n - P) = M - (N \times P)$.
  - $M - (N \times P) < (M - P) \Rightarrow$ degrees of freedom (DoF) lost
    $\Rightarrow$ incomplete minimization $\Rightarrow$performance degradation.

Distributed GSC

[Markovich-Golan et al., 2013a]

Overview

- Introduce P shared signals:
  - Broadcast by a subset of the nodes.
  - Retrieve degrees of freedom.
- Extended inputs at each node:
  - Local microphones plus shared signals.
  - Purely local FBF, BM, ANC.
- DGSC adaptively converges to the centralized solution.



Total of $N + P$ broadcast channels.

Nodes Connectivity

Sources "Owned" by the nth Node:

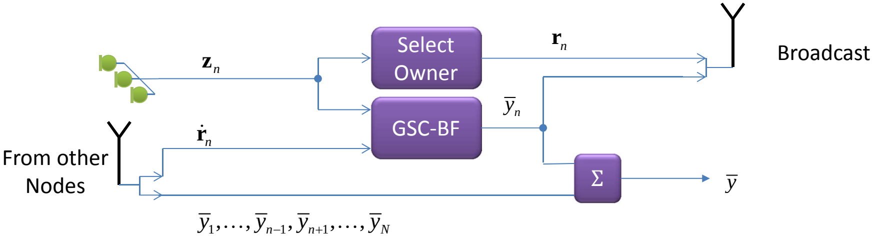
- A node n that receives the pth source with the highest SNR is declared its "owner".
- The shared signals broadcast by the nth node: $\mathbf{r}_n = \mathbf{D}_n^H \mathbf{z}_n$.
- $\mathbf{D}_n$: an $M_n \times P_n$ selection matrix.
- A shared signal (one component of $\mathbf{r}_n$) is responsible for only one source.
- Shared signals serve as a reference for RTF estimation in each node.

Extended Inputs at the nth Node

- $P - P_n$ shared signals (excluding self-owned signals): $\dot{\mathbf{r}}_n$.
- Total number of signals: $\bar{M}_n = M_n + P - P_n$.
- Signals: $\bar{\mathbf{z}}_n = \begin{bmatrix} \mathbf{z}_n^T & \dot{\mathbf{r}}_n^T \end{bmatrix}^T$.

DGSC at the *n*th Node

High Level Block-Diagram



Local & Global BF

- An $\bar{M}_n \times 1$ local GSC-BF at the nth node: $\bar{\mathbf{w}}_n$.
- Outputs of local GSC-BFs: $\bar{y}_n = \bar{\mathbf{w}}_n^H \bar{\mathbf{z}}_n; \ \forall n = 1, 2, \ldots, N$.
- Global BF: $\bar{\mathbf{w}} \triangleq \begin{bmatrix} \bar{\mathbf{w}}_1^T & \cdots & \bar{\mathbf{w}}_N^T \end{bmatrix}^T$.
- Global output (available at each node): $\bar{y} = \sum_{n=1}^N \bar{y}_n$.

Blocks of the DGSC at the nth Node

Fixed Beamformer (Local)

- $\hat{\mathbf{H}}_n$: the RTF relating the extended inputs and the shared signals.
- Build local FBF $\bar{\mathbf{q}}_n$ using only local RTFs.
- $\bar{\mathbf{q}}_n \triangleq \frac{1}{N}\hat{\mathbf{H}}_n \left(\hat{\mathbf{H}}_n^H \hat{\mathbf{H}}_n\right)^{-1} \mathbf{g} \Rightarrow \bar{\mathbf{H}}_n^H \bar{\mathbf{q}}_n = \mathbf{g}$.

Blocking Matrix (Block Diagonal)

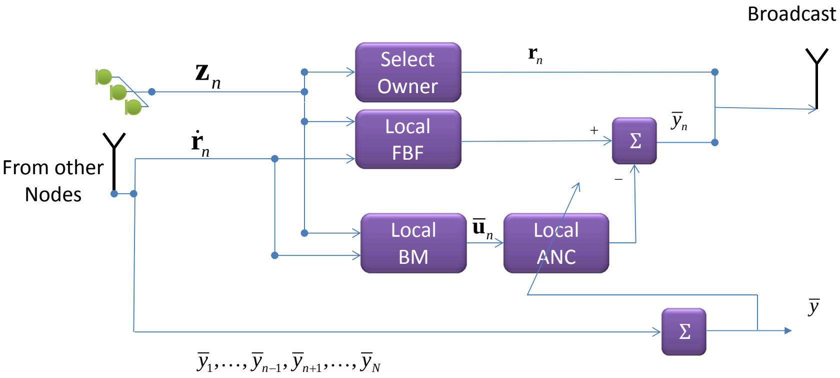
- $\bar{\mathbf{B}}_n$: $M_n \times (\bar{M}_n - P)$ BM.
- Noise references: $\bar{\mathbf{u}}_n = \bar{\mathbf{B}}_n^H \bar{\mathbf{z}}_n$
- $\sum_{n=1}^{N}(\bar{M}_n - P) = \sum_{n=1}^{N}(M_n - P_n) = M - P \Rightarrow$ DoF fully utilized.

Adaptive Noise Canceler (Local)

- Least Mean Squares: $\bar{\mathbf{f}}_n(\ell) = \bar{\mathbf{f}}_n(\ell - 1) + \mu \frac{\bar{\mathbf{u}}_n(\ell)\bar{y}^*(\ell)}{\bar{P}_{u,n}(\ell)}$.
- Power normalization $\bar{P}_{u,n}(\ell)$.

DGSC at the *n*th Node

Low Level Block-Diagram



DGSC Summary I

Features

- Distributed processing for distributed constellation.
- It is shown [Markovich-Golan et al., 2013a] that the distributed and centralized LCMV implementations identifies.
- Proof is based on: constraint set is a subspace of the M-dimensional linear space. Extending the linear space dimensions to $\bar{M}$ does not alter the sub-space.
- Local input signals selection (quasi-) fixed:
  - Original inputs.
  - Shared signals selected by the system.
  - Hence RTF estimation valid until the acoustics changes.
- The DGSC sequentially converges to the centralized solution using local ANC updates.

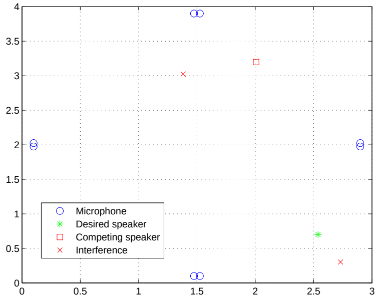
DGSC Summary II

Important Practical Considerations

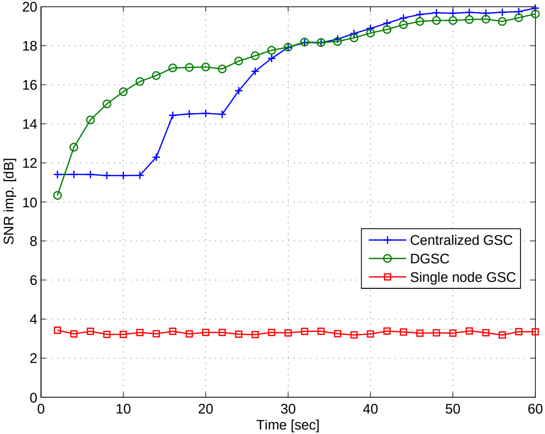
- Latency in the communication channel might require large buffering in each node.
- Owner selection is a cumbersome task if several speakers are concurrently active, since it is not clear how to identify each speaker.
- RTF can be very long for remote nodes.
- Number of nodes and constraints can dynamically change (see [Markovich-Golan et al., 2012c] for possible cure).

Scenario

- $4\text{m} \times 4\text{m} \times 3\text{m}$ room.
- Reverberation time $T_{60} = 300\text{ms}$.
- N=4 nodes.
- $M_n = 2$ microphones $\forall n$.
- Desired and competing speaker with the same level.
- 2 point source Gaussian noises, 13dB lower than the speech signals.
- Sensors noise.
- 90 Monte-Carlo experiments (sources' positions).

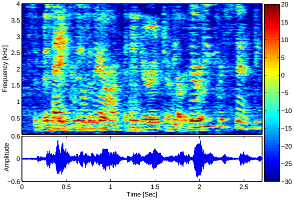


Convergence

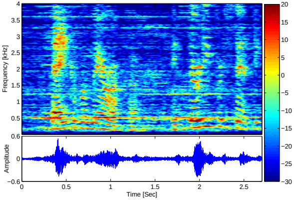


The convergence of the tested algorithms versus time.

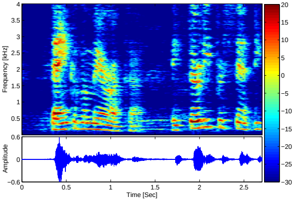
Speech Samples



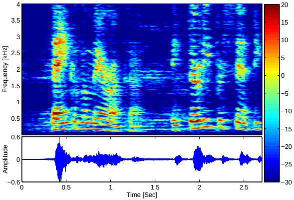
(a) Noisy at mic. #1



(b) Single node GSC



(c) Centralized GSC

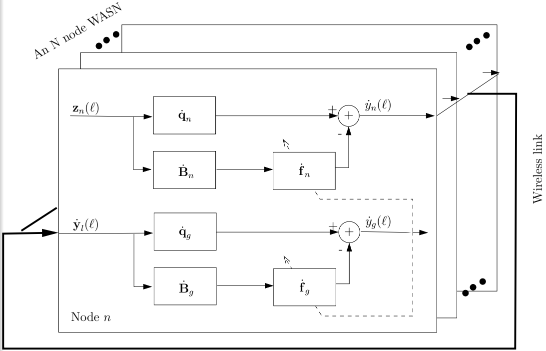


(d) Distributed GSC

Distributed single constraint GSC (DS-GSC)

[Markovich-Golan et al., 2012b]

- Two stage filtering:
  - Local filtering
  - Global filtering
- N transmission channels
- Alternating local and global filter updates
  - Iterative version
  - Time-recursive version
- Converges to the centralized TF-GSC



Blind Sampling Rate Offset Estimation and Compensation

[Markovich-Golan et al., 2012a]

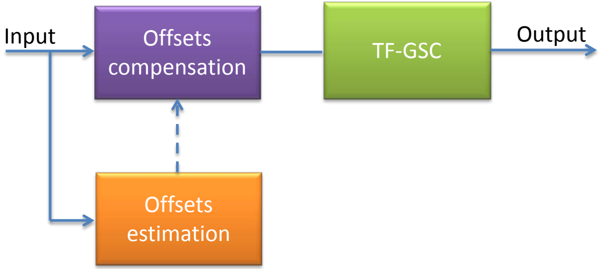
Scenario

- Fully connected N nodes network with M_n microphones at the nth node.
- Nominal sampling rate f_s.
- Sampling rate $f_{s,n} = (1 + \epsilon_n) f_s$, sampling period $T_{s,n}$ with Sampling rate offset ϵ_n.

TF-GSC [Gannot et al., 2001] with Sampling Rate Offsets

- RTF is constantly changing: signal distortion.
- ANC is constantly updating: increased noise level.
- Microphone signals are less coherent: degraded performance.

Block diagram of synchronized TF-GSC



Synchronized TF-GSC

- Sampling rate estimation: based on the phase drift of the coherence between microphones in stationary noise-only segments (in coherent frequency bands).
- Resampling with Lagrange polynomials interpolation.
- Other beamforming sync. methods: [Wehr et al., 2004]; [Ono et al., 2009]

Results

TF-GSC Algorithms

W.o. offsets; Conventional TF-GSC; Synchronized TF-GSC

Criteria

Signal to Distortion ratio (SDR); Signal to Noise (SNR)

	Without offset		With offset			
	Conventional		Conventional		Synchronized	
Q	SDR	SNR	Ex. Dist.	Ex. Noise	Ex. Dist.	Ex. Noise
1	15.0	34.3	11.2	7.7	0.0	0.0
2	14.9	27.5	11.2	4.9	0.1	0.0
3	14.6	24.5	11.5	3.4	0.4	0.1
4	14.7	23.5	11.9	2.9	0.8	0.2

Values in dB, Ex. stands for excess values

WASNs with Random Node Deployment

[Markovich-Golan et al., 2011]; [Markovich-Golan et al., 2013b]; general reading [Lo, 1964]

Scenarios

- Ad hoc sensor networks.
- Large volume (and many nodes).
- High fault percentage.
- Arbitrary deployment of nodes.

Questions

- How many nodes are required?
- What is the expected performance?
- Is there an optimal deployment?

[Kodrasi et al., 2011]

Random Beampattern using WASN

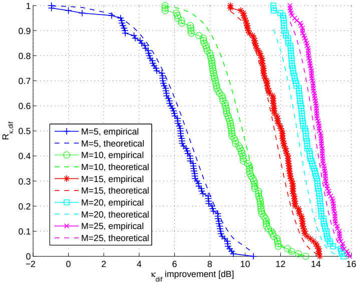
Scenario

- Desired speaker.
- Coherent or diffused noise fields.
- Microphones are randomly positioned.
- Reverberant enclosure.

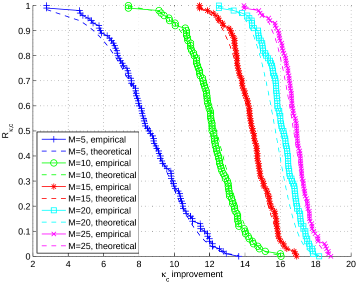
Derivation

- ATFs are complex random variables.
- Derive a model for ATFs (based on the work of Schröder [Schroeder, 1962]; [Schroeder, 1987]).
- Consider two BF performance criteria: SNR, White noise gain
- These criteria become random variables.
  - Analyze the statistics.
  - Derive reliability functions: The probability that the criteria exceed a pre-defined level.

Reliability I



(a) Diffuse Noise, SIR=30dB, residual interference dominant



(b) Coherent Noise, SIR=0dB, residual noise dominnat

$\mathrm{SINR}_{\mathrm{out}} - \mathrm{SNR}_{\mathrm{in}}$, $T_{60} = 0.4$sec, Room dimensions $4 \times 4 \times 3$m.

Special Thanks

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2. Dr. Alexander Bertrand
3. Prof. Simon Doclo
4. Prof. Walter Kellermann

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