

Beamforming-Based Multichannel Acoustic Echo Cancellation

Yuval Konforti M.Sc. Thesis Seminar

Supervisors: Prof. Israel Cohen and Dr. Baruch Berdugo





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Publication Info

- Y. Konforti, I. Cohen, and B. Berdugo, "Array geometry optimization for regionof-interest broadband beamforming", in Proc. IEEE International Workshop on Acoustic Signal Enhancement (IWAENC), pp. 1–5, 2022.
- Y. Konforti, I. Cohen, and B. Berdugo, "Multichannel acoustic echo cancellation with beamforming in dynamic environments", submitted to IEEE Open Journal of Signal Processing, 2023.





Background and Motivation





Acoustic Echo Cancellation

- Echo can interfere and disrupt conversations.
- Stems from the acoustic coupling between a loudspeaker and a microphone.
- Echo is eliminated with an acoustic echo canceller (AEC).
 - Usually solved by a normalized least-mean-square filter (NLMS) approach.
 - Involves a double-talk detector.





Applications







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Multichannel Acoustic Echo Cancellation

- In real environments two phenomena worsen performance:
 - Nonlinear loudspeaker distortion
 - Background noise
- Multichannel AEC can be used to mitigate these effects.



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Spatial Filtering

- A signal is propagated in space.
- The signal arrives from a certain direction.
- A sensor array is utilized. The time difference of arrival (TDOA) between the sensors is in accordance with the direction.



Beamforming - a Sum of Digital Filters



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Sensor Array Beamforming

We define the steering vector

$$\mathbf{d}(f,\cos\theta) = \begin{bmatrix} 1 & e^{-j2\pi f \tau_0 \cos\theta} & \cdots & e^{-j(M-1)2\pi f \tau_0 \cos\theta} \end{bmatrix}^T$$

- Each sensor utilizes a digital filter $H_m(f)$.
- The response is measured by the beampattern:
 - ► Angle dependent.

► Frequency dependent.

$$\mathcal{B}\left[\mathbf{h}(f), \cos\theta\right] = \mathbf{d}^{H}\left(f, \cos\theta\right)\mathbf{h}(f)$$

$$=\sum_{m=1}^{M}H_m(f)e^{j(m-1)2\pi f\tau_0\cos\theta}$$

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Multiframe Filtering

- In the short-time fourier transform (STFT) domain, speech signals are correlative between frames.
- The inter-frame correlations must be updated, as they highly vary in time.
- Multiframe and Multichannel filters were used for speech separation and noise reduction(*), yet not for AEC.

(*) E. A. P. Habets, J. Benesty, and J. Chen, "Multi-microphone noise reduction using interchannel and interframe correlations," in Proc. IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 305–308, 2012.

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(*) Z. Zhang, Y. Xu, M. Yu, S. X. Zhang, L. Chen, D. S. Williamson, and D. Yu, "Multi-channel multi-frame ADL-MVDR for target speech separation," IEEE/ACM Transactions on Audio, Speech, and Language Processing, vol. 29, pp. 3526–3540, 2021.

(*) M. Tammen and S. Doclo, "Deep multi-frame MVDR filtering for binaural noise reduction," in Proc. International Workshop on Acoustic Signal Enhancement (IWAENC), 2022.



AEC Performance Measures

- Echo-return loss enhancement (ERLE):
 - > $y_1(t)$ Echo component at reference microphone.
 - > $y_{re}(t)$ Residual echo component.
- Distortion Index (DI):
 - $u_1(t)$ Desired component at reference microphone.
 - $u_f(t)$ Filtered desired component.



 $\xi\left(t\right) = \frac{\mathrm{LPF}\left\{y_{1}^{2}\left(t\right)\right\}}{\mathrm{LPF}\left\{y_{\mathrm{re}}^{2}\left(t\right)\right\}}$

 $\nu(t) = \frac{\text{LPF}\left\{ \left[u_1(t) - u_f(t) \right]^2 \right\}}{\text{LPF}\left\{ u_1^2(t) \right\}}$

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Region-of-Interest Beamforming

- In many applications, the source location is unknown, but it can be assumed it is in a region-of-interest (ROI).
- Source localization / Direction-ofarrival (DOA) estimation may be employed to track a moving source.
- While the coefficients change, the array geometry is preserved.



Applications













Array Geometry Optimization

- Typically, the beamforming coefficients are found per given geometry. Usually symmetric geometries are considered:
 - Uniform Linear Arrays (ULAs).
 - Rectangular Arrays.
 - Uniform Circular Arrays (UCAs).
 - Uniform Concentric Circular Arrays (UCCAs).
- Differential Microphone Arrays (DMAs) produce an approximate frequency invariant (FI) response.
- The geometry of a microphone array has an important impact on beamforming performance.





Optimization Methods

- Two common methods for array geometry optimization:
 - ► Greedy-based approaches.
 - ► Genetic Algorithms.
- Greedy-based approaches find the best position for the placed microphone in each step, but the overall result may not be optimal.
- Genetic algorithms are unstable and may also converge to nonoptimal results.
- The study in (*) optimize the geometry for a ROI, but only for narrowband signals.

(*) X. Chen, C. Pan, J. Chen, and J. Benesty, "Planar array geometry optimization for region sound acquisition," in *Proc. IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, pp. 756–760, 2021.

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Beamforming Performance Measures

White Noise Gain:
$$W[h(x,\omega,\theta)] \triangleq \frac{\left| \mathbf{d}^{H}(x,\omega,\theta) \mathbf{h}(x,\omega,\theta) \right|^{2}}{\mathbf{h}^{H}(x,\omega,\theta) \mathbf{h}(x,\omega,\theta)} \mathbf{1}$$
Directivity Factor (DF):
$$\mathcal{D}[h(x,\omega,\theta)] \triangleq \frac{\left| \mathbf{d}^{H}(x,\omega,\theta) \mathbf{h}(x,\omega,\theta) \right|^{2}}{\mathbf{h}^{H}(x,\omega,\theta) \mathbf{\Gamma}(x,\omega) \mathbf{h}(x,\omega,\theta)} \mathbf{1}$$

where
$$\Gamma_{i,j}(\mathbf{x},\omega) = \frac{\sin\left(\omega\left(x_i - x_j\right)/c\right)}{\omega\left(x_i - x_j\right)/c}, \quad 1 \le i,j \le M$$

$$\blacktriangleright \text{ Directivity Index: } \mathcal{DI}_{[\omega_L,\omega_H]} \left[\mathbf{h} \left(\mathbf{x}, \omega, \theta \right) \right] \stackrel{\Delta}{=} \frac{\int_{\omega_L}^{\omega_H} \left| \mathbf{d}^H \left(\mathbf{x}, \omega, \theta \right) \mathbf{h} \left(\mathbf{x}, \omega, \theta \right) \right|^2 d\omega}{\int_{\omega_L}^{\omega_H} \mathbf{h}^H \left(\mathbf{x}, \omega, \theta \right) \mathbf{\Gamma} \left(\mathbf{x}, \omega \right) \mathbf{h} \left(\mathbf{x}, \omega, \theta \right) d\omega}$$







Multichannel Acoustic Echo Cancellation with Beamforming in Dynamic Environments





A Beamforming-Based Approach

- y(k, n)- Echo component received by the array.
- u(k,n)- Desired component received signal by the array.
- ▶ v(k,n)- Background noise received signal by the array.
- d(k,n)- total received signal by the array.

 $\hat{U}\left(k,n\right) = \mathbf{h}^{H}\left(k,n\right)\mathbf{d}\left(k,n\right)$

 $\mathbf{d}\left(k,n\right)=\mathbf{y}\left(k,n\right)+\mathbf{u}\left(k,n\right)+\mathbf{v}\left(k,n\right)$





Linear-Constraint-Minimum-Variance Beamforming

s.t.

- Theoretically enforces:
 - Complete echo cancellation.
 - No distortion.
 - Minimum residual noise.
- Ideal ERLE and DI.
- Requires accurate estimates of g(k) and q(k).

$$\mathbf{h}^{*}(k,n) = \mathbf{C}(k) \left[\mathbf{C}^{H}(k) \mathbf{C}(k) \right]^{-1} \mathbf{i}_{k}$$

 $\mathbf{h}^{*}\left(k,n\right) = \operatorname*{arg\,min}_{\mathbf{h}(k,n)} ||\mathbf{h}\left(k,n\right)||^{2}$

where $\mathbf{C}(k,n) = \left[\mathbf{q}(k),\mathbf{g}(k)\right]$

 $\mathbf{i}_{c} = [1, 0]^{T}$

The Andrew and Erna Viterbi Faculty of $\mathcal{C}_{1}\left[\mathbf{h}\left(k,n\right)\right]:\mathbf{h}^{H}\left(k,n\right)\mathbf{g}\left(k\right)=0$ $\mathcal{C}_{2}\left[\mathbf{h}\left(k,n\right)\right]:\mathbf{h}^{H}\left(k,n\right)\mathbf{q}\left(k\right)=1$



AEC Scheme

Procedure:

- 1. Steering vectors g(k) and q(k) are estimated.
- 2. Beamformer h(k, n) is designed.
- 3. Estimate $\widehat{U}(k,n)$ is produced.
- No double talk detection used.



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Utilizing Multiple Sensors - A Frame Invariant Expression

- Neglecting nonlinear loudspeaker distortion.
- Neglecting background noise.
- Assuming a static environment in the last L frames.
- Utilizing the Multiplicative Transfer Function (MTF) approximation (*).

$$D_{m}(k, n - l + 1) = G_{m}(k) X(k, n - l + 1) + Q_{m}(k) S(k, n - l + 1)$$

For any two microphones m_1 and m_2 :

$$\frac{Q_{m_1}\left(k\right)}{Q_{m_2}\left(k\right)} = \frac{D_{m_1}\left(k, n-l+1\right) - G_{m_1}\left(k\right) X\left(k, n-l+1\right)}{D_{m_2}\left(k, n-l+1\right) - G_{m_2}\left(k\right) X\left(k, n-l+1\right)}$$

(*) Y. Avargel and I. Cohen, "On multiplicative transfer function approximation in the short-time fourier transform domain," IEEE Signal Processing Letters, vol. 14, pp. 337–340, 2007.

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Utilizing Multiple Frames

For any two recent frames l_1 and l_2 :

$$\begin{split} &\frac{D_{m_1}\left(k,n-l_1+1\right)-G_{m_1}\left(k\right)X\left(k,n-l_1+1\right)}{D_{m_2}\left(k,n-l_1+1\right)-G_{m_2}\left(k\right)X\left(k,n-l_1+1\right)} = \\ &\frac{D_{m_1}\left(k,n-l_2+1\right)-G_{m_1}\left(k\right)X\left(k,n-l_2+1\right)}{D_{m_2}\left(k,n-l_2+1\right)-G_{m_2}\left(k\right)X\left(k,n-l_2+1\right)} \end{split}$$

• Quadratic elements of $G_{m_1}(k)G_{m_2}(k)$ are reduced.

A linear equation with respect to $G_{m_1}(k)$ and $G_{m_2}(k)$ is obtained:

$$\begin{split} G_{m_1}\left(k\right)\left[X\left(k,n-l_1+1\right)D_{m_2}\left(k,n-l_2+1\right)-X\left(k,n-l_2+1\right)D_{m_2}\left(k,n-l_1+1\right)\right] &+ \\ G_{m_2}\left(k\right)\left[X\left(k,n-l_2+1\right)D_{m_1}\left(k,n-l_1+1\right)-X\left(k,n-l_1+1\right)D_{m_1}\left(k,n-l_2+1\right)\right] &= \\ D_{m_1}\left(k,n-l_1+1\right)D_{m_2}\left(k,n-l_2+1\right)-D_{m_1}\left(k,n-l_2+1\right)D_{m_2}\left(k,n-l_1+1\right) \end{split}$$





A Linear Set of Equations

- Overall, for any pick of $1 \le m_1, m_2 \le M$ and $1 \le l_1, l_2 \le L$, we get such an equation.
- Uninformative equations:
 - ▶ $m_1 = m_2$
 - $\blacktriangleright \quad l_1 = l_2$
 - Swapping $m_1 \iff m_2$
 - Swapping $l_1 \iff l_2$
- Must have more equations than variables:

$$M \le \binom{M}{2} \binom{L}{2} \square \square \square \square L (L-1) (M-1) \ge 4$$

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• Must have multiple frames, multiple sensors, and $M \ge 3$ or $L \ge 3$.



A Least-Mean-Squares Approach

- Equations may contradict due to assumptions.
- A least-mean-squares (LMS) solution provides a good estimate.
- As *L* grows:
 - More equations are added.
 - The environment is assumed to be static for longer periods.
- ► As *M* grows:
 - More equations are added.
 - More variables are added.





Matrix Formulation

Inputs: M, L, $X(k, n-l+1), D_m(k, n-l+1)$ $1 < l < L \quad 1 < m < M$ Outputs: $\tilde{G}_m(k,n)$ $1 \le m \le M$ Create list M_{list} of $\binom{M}{2}$ non-repetitive pairs (m_1, m_2) Create list L_{list} of $\binom{L}{2}$ non-repetitive pairs (l_1, l_2) $\mathbf{A}(k,n) \leftarrow \mathbf{0}_{\binom{M}{2}\binom{L}{2} \times M}$ for $i = 1, 2, ..., {M \choose 2}$ do $(m_1, m_2) \leftarrow M_{\text{list}}(i)$ for $i = 1, 2, ..., {L \choose 2}$ do $(l_1, l_2) \leftarrow L_{\text{list}}(j)$ $\mathbf{A}_{\left[(i-1)\binom{L}{2}+j,m_{1}\right]}\left(k,n\right)\leftarrow X\left(k,n-l_{1}+1\right)D_{m_{2}}\left(k,n-l_{2}+1\right) X(k, n-l_2+1) D_{m_2}(k, n-l_1+1)$ $\mathbf{A}_{\left[(i-1)\binom{L}{2}+j,m_{2}\right]}\left(k,n\right)\leftarrow X\left(k,n-l_{2}+1\right)D_{m_{1}}\left(k,n-l_{1}+1\right) X(k, n-l_1+1) D_{m_1}(k, n-l_2+1)$ $\mathbf{b}_{\left[(i-1)\binom{L}{2}+j\right]}(k,n) \leftarrow D_{m_1}(k,n-l_1+1) D_{m_2}(k,n-l_2+1) -$ $D_{m_1}(k, n-l_2+1) D_{m_2}(k, n-l_1+1)$

end for

end for

$$\left[\tilde{G}_{1}\left(k,n\right),\tilde{G}_{2}\left(k,n\right),...,\tilde{G}_{M}\left(k,n\right)\right]^{T}\leftarrow\mathbf{A}^{\dagger}\left(k,n\right)\mathbf{b}\left(k,n\right)$$







Simulation Configuration

- Dynamic environment.
- Double-talk.
- Low Signal-to-Echo Ratio in 4 scenarios:
 - Speakerphone at A, talker at C.
 -17.89 dB.
 - Speakerphone at A, talker at D.
 -18.7 dB.
 - 3. Speakerphone at B, talker at C.-15.27 dB.
 - 4. Speakerphone at *B*, talker at *D*.-16.89 dB.



Results - Echo Cancellation

- a. The total received signal in the reference microphone $d_1(t)$
- b. The echo component signal in the reference microphone $y_1(t)$
- c. The desired component signal in the reference microphone $u_1(t)$
- d. The beamformer output signal $\hat{u}(t)$.
- Good echo cancellation despite significant acoustic coupling.



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Results -Performance as Function of Microphones

- a. ERLE.
- b. **DI.**
- ▶ Blue M = 2, Red M = 3,
 Yellow M = 4, Purple M = 5. L = 4.
- Significant change only between M = 2 and M = 3.
- Performance limit. Increasing M also
 increases number of variables in
 estimation process.







Results -Performance as Function of Frames

a. ERLE.

b. **DI.**

- Blue L = 2. Red L = 3.
 Yellow L = 4. Purple L = 5. M = 4.
- L = 2 are insufficient for the proposed approach.
- Performance limit. Increasing L also relies more strongly on a longer period where the environment is static.





Results - Method Comparison (*)

- a. ERLE.
- b. DI.
- The NLMS filter in (*) was trained in a period like the first segment, but with no near-end speech.
- ▶ Blue proposed. Red competing.
- Improvement in both ERLE and DI. Improvement in both ERLE
- Significant improvement after the speakerphone moves due to changing reflections. The NLMS filter is irrelevant once the reflections have changed.



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(*) H. Huang, C. Hofmann, W. Kellermann, J. Chen, and J. Benesty, "Multiframe echo suppression based on orthogonal signal decompositions," in *Proc. Speech Communication; ITG Symposium*, pp. 287–291, 2016.



Array Geometry Optimization for Regionof-Interest Broadband Beamforming





A Variable Geometry

M omnidirectional microphones are placed nonuniformly across a linear aperture A in placements x_m , m = 1, 2, ..., M.



The steering vector is

$$\mathbf{d}\left(\mathbf{x},\omega,\theta\right) = \left[e^{-j\frac{\omega}{c}x_{1}\cos\theta}, e^{-j\frac{\omega}{c}x_{2}\cos\theta}, ..., e^{-j\frac{\omega}{c}x_{M}\cos\theta}\right]^{T}$$





Coefficients Dependent on Geometry

For different geometries, the coefficients are designed differently. For a geometry x and look direction θ , we define the coefficients vector as

$$\mathbf{h}\left(\mathbf{x},\omega,\theta\right) \stackrel{\Delta}{=} \left[H_1\left(\mathbf{x},\omega,\theta\right), H_2\left(\mathbf{x},\omega,\theta\right), ..., H_M\left(\mathbf{x},\omega,\theta\right)\right]^T$$

With geometry x, the beampattern directed toward $\tilde{\theta}$ has a response at angle θ of

$$\mathcal{B}\left[\mathbf{h}\left(\mathbf{x},\omega,\tilde{\theta}\right),\theta\right] = \mathbf{d}^{H}\left(\mathbf{x},\omega,\theta\right)\mathbf{h}\left(\mathbf{x},\omega,\tilde{\theta}\right)$$





Problem Formulation

Our objective is to find the optimal array geometry x, that maximizes the worst-case directivity index, in an ROI Θ. Each beamformer, directed toward θ, must admit to the distortionless constraint, have sufficient WNG, and maintain minimal distances.

$$\mathbf{x}^{*} = \arg \max_{\mathbf{x}} \quad \min_{\theta \in \Theta} \mathcal{DI}_{[\omega_{L}, \omega_{H}]} \left[\mathbf{h} \left(\mathbf{x}, \omega, \theta \right) \right]$$

s.t.
$$\mathcal{B} \left[\mathbf{h} \left(\mathbf{x}, \omega, \theta \right), \theta \right] = 1 \quad \forall \theta \in \Theta, \forall \omega \in \Omega$$
$$\mathcal{W} \left[\mathbf{h} \left(\mathbf{x}, \omega, \theta \right) \right] \geq \delta \quad \forall \theta \in \Theta, \forall \omega \in \Omega$$
$$|x_{i} - x_{j}| \geq d_{c} \qquad \forall i, j \in [1, M], i \neq j$$
$$0 \leq x_{m} \leq A \qquad \forall m \in [1, M]$$

Not convex. Cannot be solved by convex optimization algorithms.





Formalizing a Solvable Problem - Constraints

- Consider N candidate microphone locations, Q frequencies, and P look directions.
- To guarantee number of microphones:

 $\mathcal{C}_1[\mathbf{s}]: \mathbf{s}^H \mathbf{i}_N = M$

To guarantee minimal spacing:

 $\mathcal{C}_2\left[\mathbf{s}\right]:\mathbf{s}^H U \leq \mathbf{i}_G^T$

To guarantee the distortionless constraint:

$$\mathcal{C}_{3}\left[\mathbf{h}_{\text{tot}}\left(\omega,\theta\right)\right]:\mathbf{d}_{\text{tot}}^{H}\left(\omega_{q},\theta_{p}\right)\mathbf{h}_{\text{tot}}\left(\omega_{q},\theta_{p}\right)=1$$

• To guarantee the desired WNG:

 $\mathcal{C}_{4}\left[\mathbf{h}_{\text{tot}}\left(\omega,\theta\right)\right]:\mathbf{h}_{\text{tot}}^{H}\left(\omega_{q},\theta_{p}\right)\mathbf{h}_{\text{tot}}\left(\omega_{q},\theta_{p}\right) \leq \frac{1}{\delta}$ $\forall p \in [1,P], \quad \forall q \in [1,Q]$

To guarantee beamformer use of selected placements:

$$\mathcal{C}_{5} \left[\mathbf{s}, \mathbf{h}_{\text{tot}} \left(\omega, \theta \right) \right] : \left| H_{\text{tot},i} \left(\omega_{q}, \theta_{p} \right) \right|^{2} \leq \frac{S_{i}}{\delta} \\ \forall i \in \left[1, N \right], \quad \forall p \in \left[1, P \right], \quad \forall q \in \left[1, Q \right]$$

 $\mathbf{s} = [1, 0, 1, 0, 1]^{\mathrm{T}}$

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 $\forall p \in [1, P], \quad \forall q \in [1, Q]$



Formalizing a Solvable Problem - Target

When the distortionless constraint is met, the directivity index is determined only by the denominator. Therefore, to maximize the worst-case directivity index we should minimize:

$$egin{aligned} R\left[\mathbf{h}_{ ext{tot}}\left(\omega, heta
ight)
ight] \ &= \max_{p\in[1,P]}\sum_{q=1}^{Q}\mathbf{h}_{ ext{tot}}^{H}\left(\omega_{q}, heta_{p}
ight)\mathbf{\Gamma}_{ ext{tot}}\left(\omega_{q}
ight)\mathbf{h}_{ ext{tot}}\left(\omega_{q}, heta_{p}
ight) \end{aligned}$$





Optimal Array Design

- The optimal design is found by solving a mixed-integer convex optimization problem.
- The non-zero elements of the optimal binary vector s^* yield the optimal microphone locations x^* . The non-zero elements of the optimal coefficients $h_{tot}^*(\omega, \theta)$ yield the optimal coefficients $h^*(x^*, \omega, \theta)$.

 $\min_{\mathbf{s},\mathbf{h}_{\mathrm{tot}}(\omega,\theta)}$

$\begin{array}{l} & \underset{\omega,\theta}{\mathbf{n}} & R\left[\mathbf{h}_{\text{tot}}\left(\omega,\theta\right)\right] \\ & \text{s.t.} & \mathcal{C}_{1}\left[\mathbf{s}\right], \mathcal{C}_{2}\left[\mathbf{s}\right], \mathcal{C}_{3}\left[\mathbf{h}_{\text{tot}}\left(\omega,\theta\right)\right], \\ & \quad \mathcal{C}_{4}\left[\mathbf{h}_{\text{tot}}\left(\omega,\theta\right)\right], \mathcal{C}_{5}\left[\mathbf{s},\mathbf{h}_{\text{tot}}\left(\omega,\theta\right)\right] \end{array}$

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Coefficient Post-Processing

- Since the worst-case look direction is considered, beamformers directed toward other directions may not yield the best possible directivity. To circumvent this, we introduce a post-processing scheme.
- To maximize directivity while maintaining sufficient WNG, the coefficients are found by the robust superdirective beamformer:

$$\mathbf{h}_{\epsilon}\left(\mathbf{x}^{*}, \omega, \theta\right) = \frac{\boldsymbol{\Gamma}_{\epsilon}^{-1}\left(\mathbf{x}^{*}, \omega\right) \mathbf{d}\left(\mathbf{x}^{*}, \omega, \theta\right)}{\mathbf{d}^{H}\left(\mathbf{x}^{*}, \omega, \theta\right) \boldsymbol{\Gamma}_{\epsilon}^{-1}\left(\mathbf{x}^{*}, \omega\right) \mathbf{d}\left(\mathbf{x}^{*}, \omega, \theta\right)}$$

For every frequency and look direction, ϵ is found by a bisection search:

$$0 \le \epsilon \le \frac{\lambda_1 - \sqrt{M/\delta}\lambda_M}{\sqrt{M/\delta} - 1}$$

where $\Gamma(\mathbf{x}^*,\omega) = \mathbf{Q}(\mathbf{x}^*,\omega) \mathbf{\Lambda}(\mathbf{x}^*,\omega) \mathbf{Q}^T(\mathbf{x}^*,\omega)$ is the eigenvalue decomposition such that that $\mathbf{\Lambda} = \text{diag}[\lambda_1, \lambda_2, ..., \lambda_M]$, $\lambda_1 \ge \lambda_2 \ge ... \ge \lambda_M$.





Results - Geometry

• M = 6 microphones, aperture length A = 17.5[cm], minimal distance $d_c = 0.5[cm]$. Frequencies from $f_L = 2[kHz]$ to $f_H = 6[kHz]$, look directions up to $\theta_H = 30^\circ$. Minimum WNG is $\delta = -10[dB]$. N = 40, Q = 15, P = 15.



- Some sensors close together yield high directivity, like DMAs.
- Some sensors further apart, due to WNG constraint.





Results - Directivity Index



- ▶ Blue proposed. Red ULA. Yellow dense geometry.
- Better directivity in the worst-case direction. Image of the sector o
- ▶ Better directivity in all ROI directions. ♦



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Results - WNG and Directivity Factor

TECHNION

- (a) ULA Excellent WNG, but spatial aliasing damages DF at higher frequencies.
- (b) Dense geometry Barely sufficient WNG, cannot produce high directivity due to WNG constraint.
- (c) Proposed Good WNG, good broadband directivity.





Conclusions





Research Contributions

- A novel beamforming-based multichannel AEC method was proposed.
 - Capable of operating in dynamic environments.
 - Does not require double-talk detection.
 - Robust to extremely low SER.
 - Achieves higher ERLE and lower DI compared to an existing method.
- ▶ The array geometry for broadband ROI was optimized.
 - A convex framework was used, enabling the convergence to the global optimum.
 - > A broadband frequency range was considered.
 - Sensors were placed close closer / further apart, to achieve compromise between directivity and noise robustness.

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Future Research

- Modelling nonlinear loudspeaker distortion and background noise. The LMS approach works well only if those components are relatively small.
- Developing more accurate inter-frame and inter-sensor relations. For example, instead of the MTF approximation (*) using the Cross-Multiplicative Transfer Function (CMTF) approximation (**).
- Exploring other array types. The search is limited to non-uniform linear arrays. This will enable more complex ROIs too.
- Geometry optimization by other approaches, such as learning-based methods or deep neural networks (DNNs). Our framework runtime grows considerably with more optimization variables.

(*)Y. Avargel and I. Cohen, "On multiplicative transfer function approximation in the short-time fourier transform domain," IEEE Signal Processing Letters, vol. 14, pp. 337–340, 2007.

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(**) Y. Avargel and I. Cohen, "Adaptive system identification in the short-time fourier transform domain using cross-multiplicative transfer function approximation," *IEEE Transactions on Audio, Speech, and Language Processing*, vol. 16, pp. 162–173, 2008.



Questions? Thank you!

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