

DoA-estimation based on a complex Watson kernel method

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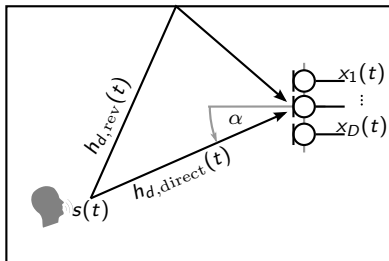
Motivation

- Direction of Arrival (DoA)-estimation is important for:
 - ▶ Speaker localization and tracking
 - ▶ Blind Source Separation (BSS)
 - ▶ Beamforming
 - ▶ Geometry calibration
 - ▶ ...

Task

- Estimate DoA from multi-channel audio signal
- Requirements:
 - ▶ Account for spatial aliasing
 - ▶ Exploit phase and level differences (microphone directivity relevant for beamforming [Gaubitch et al. 2014])
 - ▶ Derive statistical approach

Signal model



Time domain model

$$\begin{aligned}
 x_d(t) &= h_{d,direct}(t) * s(t) \\
 &+ \underbrace{h_{d,rev}(t) * s(t) + n_d(t)}_{\tilde{n}_d(t)}
 \end{aligned}$$

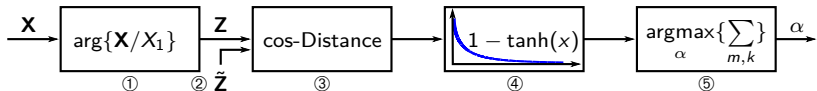
Assumption

- Reverberation is uncorrelated from direct component

Short Time Fourier Transform (STFT) model

$$\mathbf{X}(m, k) = \mathbf{H}_{direct}(k)S(m, k) + \tilde{\mathbf{N}}(m, k)$$

where m: frame index, k: frequency bin



Procedure

1. Get phase relative to 1st channel
2. Compute expected phase for candidate direction: $\tilde{Z}_d = 2\pi f(k)\tau_d(\alpha)$
3. Calculate deviation: $c = 2D - 2 \sum_{d=1}^D \cos(Z_d - \tilde{Z}_d)$
4. Reduce influence of outliers and increase spatial resolution
5. Sum up TF-slots and select maximum

Summary

- + Account for spatial aliasing
- Heuristic non-linearity

Proposed approach

Requirements

- Avoid heuristic non-linearity
- Additionally: Exploit level differences

Prepare microphone signal

- Source power influences length of $\mathbf{X}(m, k)$
- Absolute power contains no information \Rightarrow Remove absolute power

$$\mathbf{Y}(m, k) = \mathbf{X}(m, k) / \|\mathbf{X}(m, k)\|$$

- Phase and level difference due to propagation conditions are maintained
- Observations $\mathbf{Y}(m, k)$ are located on complex unit hyper-sphere

Wanted

- Distribution to model the clusters related to the source position/direction

Complex Watson probability density function

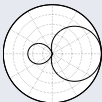
- 2 parameters: concentration κ and mode vector \mathbf{W}
- Observations \mathbf{Y} are located on complex unit hyper-sphere

$$p(\mathbf{Y}; \kappa, \mathbf{W}) = \frac{1}{c_W(\kappa)} e^{\kappa |\mathbf{W}^H \mathbf{Y}|^2}$$

Applications of complex Watson distribution

- Image recognition [Mardia et al. 1999]
- BSS [Ito et al 2013], [Tran Vu et al. 2013], [Jafari et al. 2014]

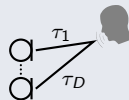
Anechoic model



ideal directivity pattern

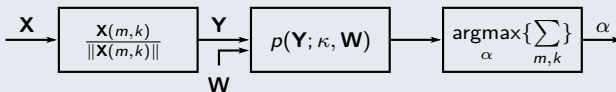
$$\tilde{\mathbf{W}}(k, \alpha) = \left[\underbrace{A_d(\alpha)}_{\text{Attenuation}} \underbrace{e^{-2\pi j f(k) \tau_d(\alpha)}}_{\text{Phase}} \right]_{1 \leq d \leq D}$$

$$\mathbf{W} = \tilde{\mathbf{W}} / \|\tilde{\mathbf{W}}\|$$



- Get impinging direction α related to the direct path
- ⇒ Find best match with anechoic model
- Candidate model can be 3D or positions

Summary

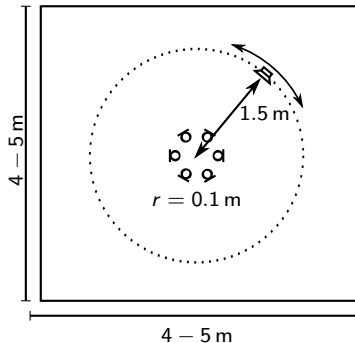


- + Exploit phase and level differences
- + Only one parameter instead of heuristic function

Experimental evaluation

Setup

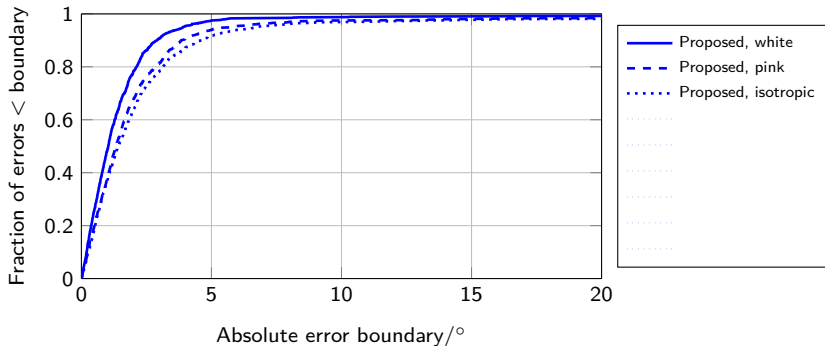
- 1000 sampled setups
- Speech samples from TIMIT database (Duration 1 sec)
- $T_{60} = 400$ ms (image method)
- Noise types (SNR = 10 dB):
 - ▶ white (spectral and spatial)
 - ▶ pink (spatial white)
 - ▶ isotropic (spectral white)



Algorithms

- [Loesch et al. 2010]
- Proposed: Waston Kernel
- Steered Response Power with Phase Transform (SRP-PHAT) [DSB01]

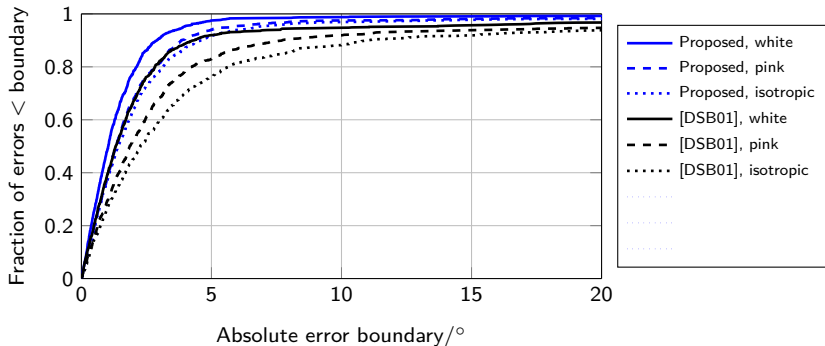
Performance for omnidirectional microphones



Conclusion

- Proposed algorithm achieved best performance in all conditions
- Smallest degradation

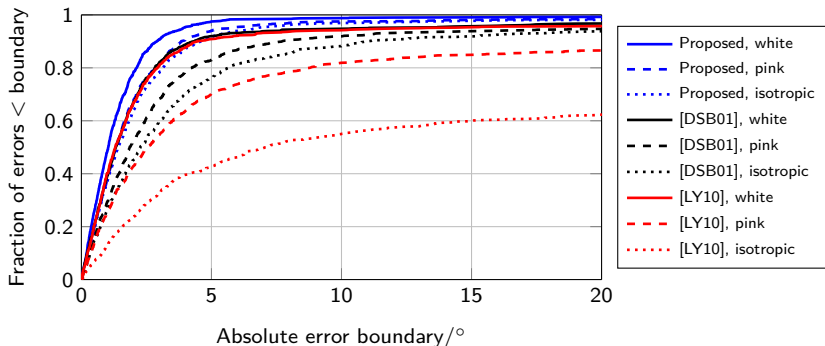
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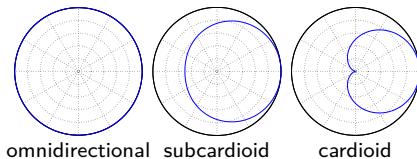
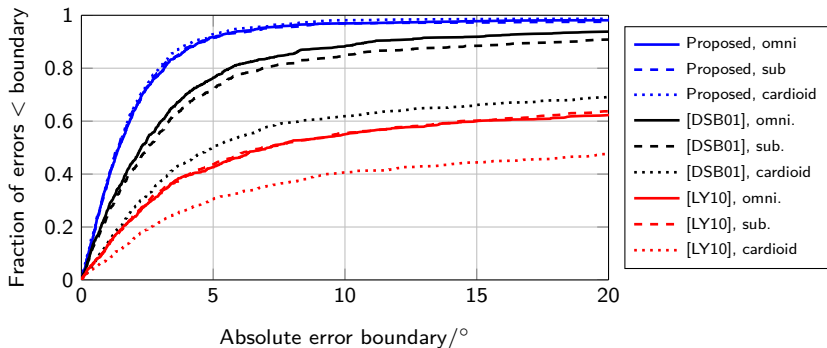
Performance for omnidirectional microphones



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Performance with given sensor directivity - isotropic noise



Conclusion

- SRP-PHAT and [LY10]:
unconsidered directivity pattern
⇒ large degradation
- Proposed algorithm: no
significant changes

Summary

- DoA (or source position) estimators:
 - ▶ Loesch
 - ▶ Watson Kernel (proposed)
- Statistically motivated (Complex Watson distribution)
- Exploits phase and level differences
 - ⇒ Avoid degradation due to pattern mismatch
- Even better performance for omnidirectional microphones
- Can be easily combined with speech presence probability (SPP) estimation

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Thank you for your attention!

Questions?

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