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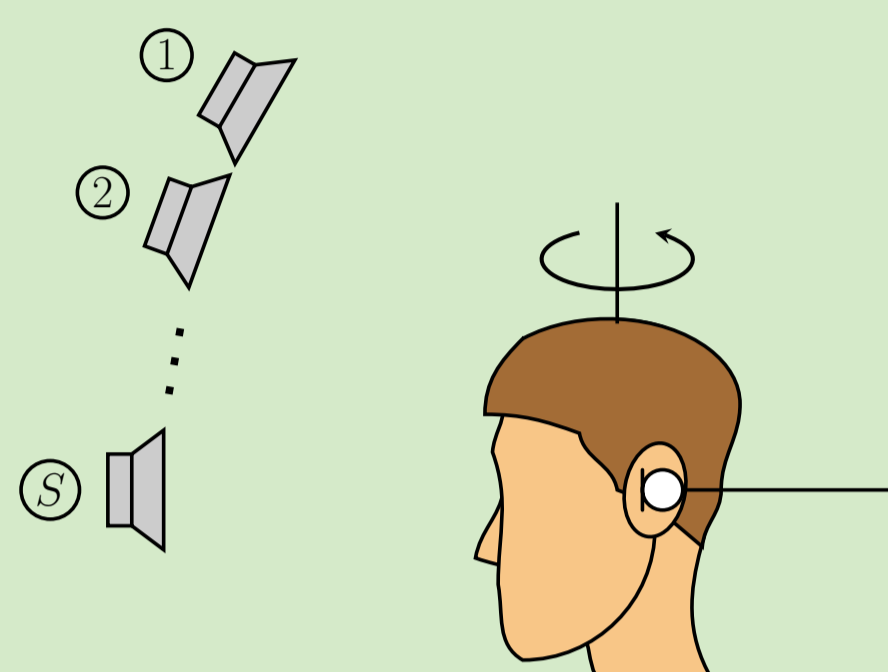
1 Introduction

Head-related transfer functions (HRTFs) / impulse responses (HRIRs):

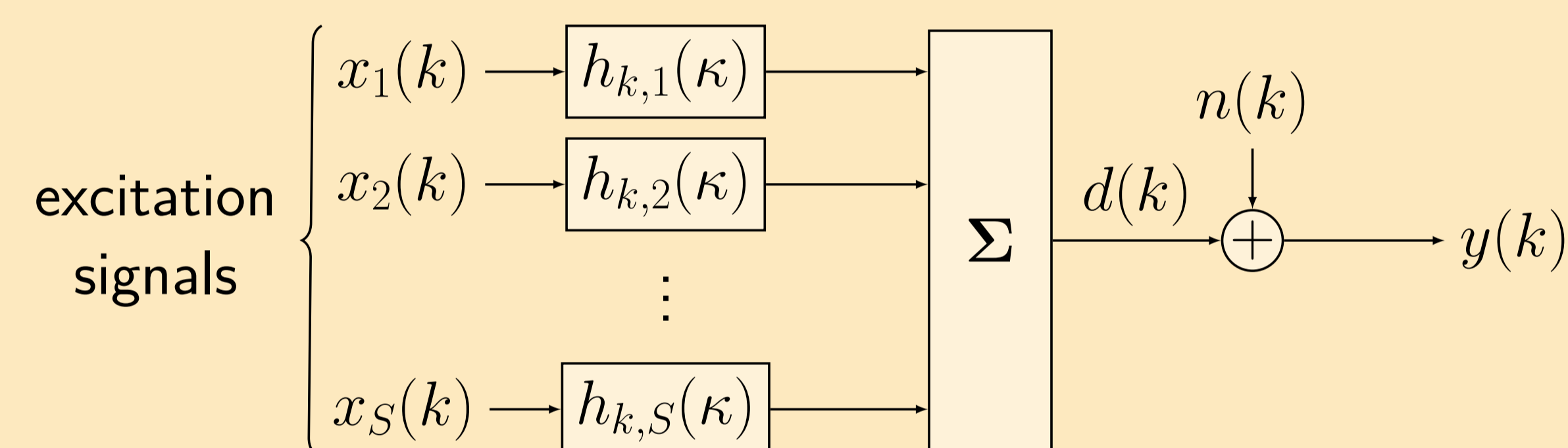
- Describe acoustic transmission between sound source and human ear
- Applications in binaural audio
- Individualization beneficial: binaural synthesis, crosstalk cancellation

Measurement of HRTFs:

- Two approaches: stop-and-go or continuous
- Goal: **faster continuous** measurements
 - Learning a system model
 - Offline processing



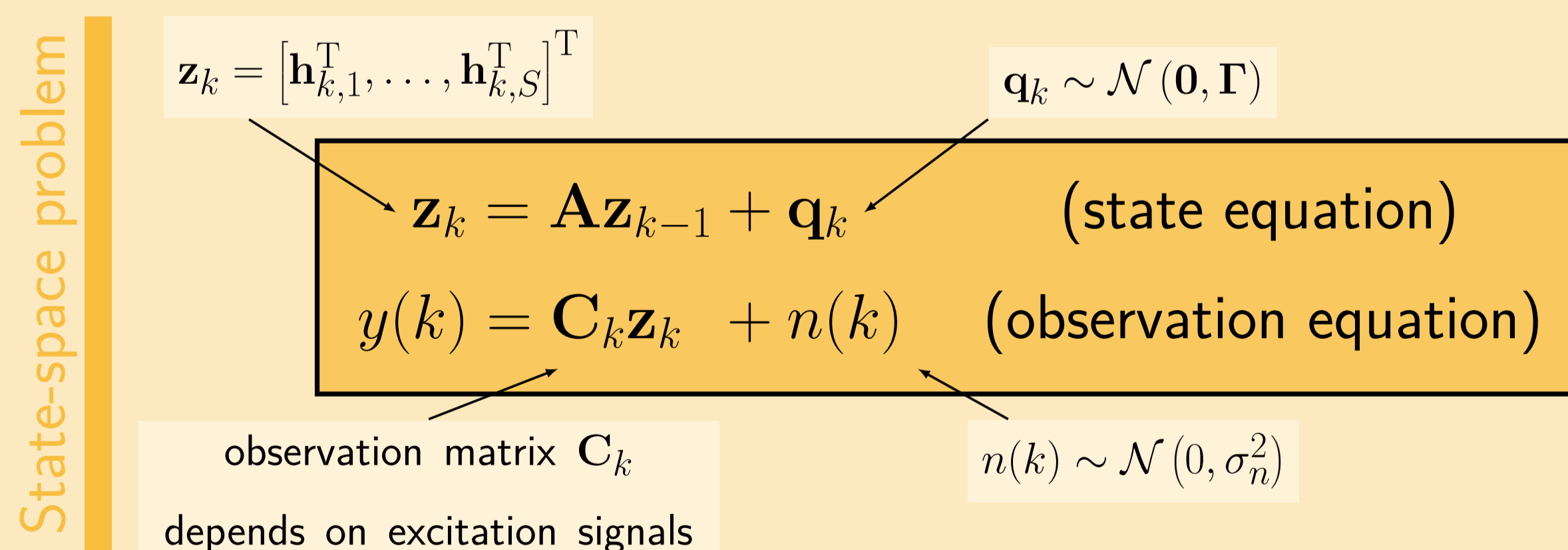
2 System and Signal Model



Task: estimate S time-variant HRIRs $h_{k,s}$ of length \tilde{L} at time k from excitation signals $x_s(k)$ and noisy microphone signal

$$y(k) = \sum_{s=1}^S \sum_{\kappa=0}^{\tilde{L}-1} x_s(k-\kappa) h_{k,s}(\kappa) + n(k)$$

► Traditional solution: NLMS algorithm [1]



- State estimation requires parameters $\theta = \{\mathbf{A}, \mathbf{\Gamma}, \sigma_n^2, \boldsymbol{\mu}_0, \mathbf{P}_0\}$, e.g.,
 - Initial state $\boldsymbol{\mu}_0$, initial error covariance \mathbf{P}_0 heuristically chosen
 - Scalar fading factor $\mathbf{A} = \gamma \mathbf{I}$ with $0 \ll \gamma \leq 1$
 - Online-estimate of diagonal process noise covariance $\mathbf{\Gamma}$ and σ_n^2

- Kalman Filter: $\boldsymbol{\mu}_k = \mathbb{E} \left\{ \mathbf{z}_k \mid \{\mathbf{x}_\ell\}_{\ell=0}^k, \{y(\ell)\}_{\ell=0}^k \right\}$

3 Proposed Measurement Method

1. Idea: exploit that **future samples** are available (offline processing)

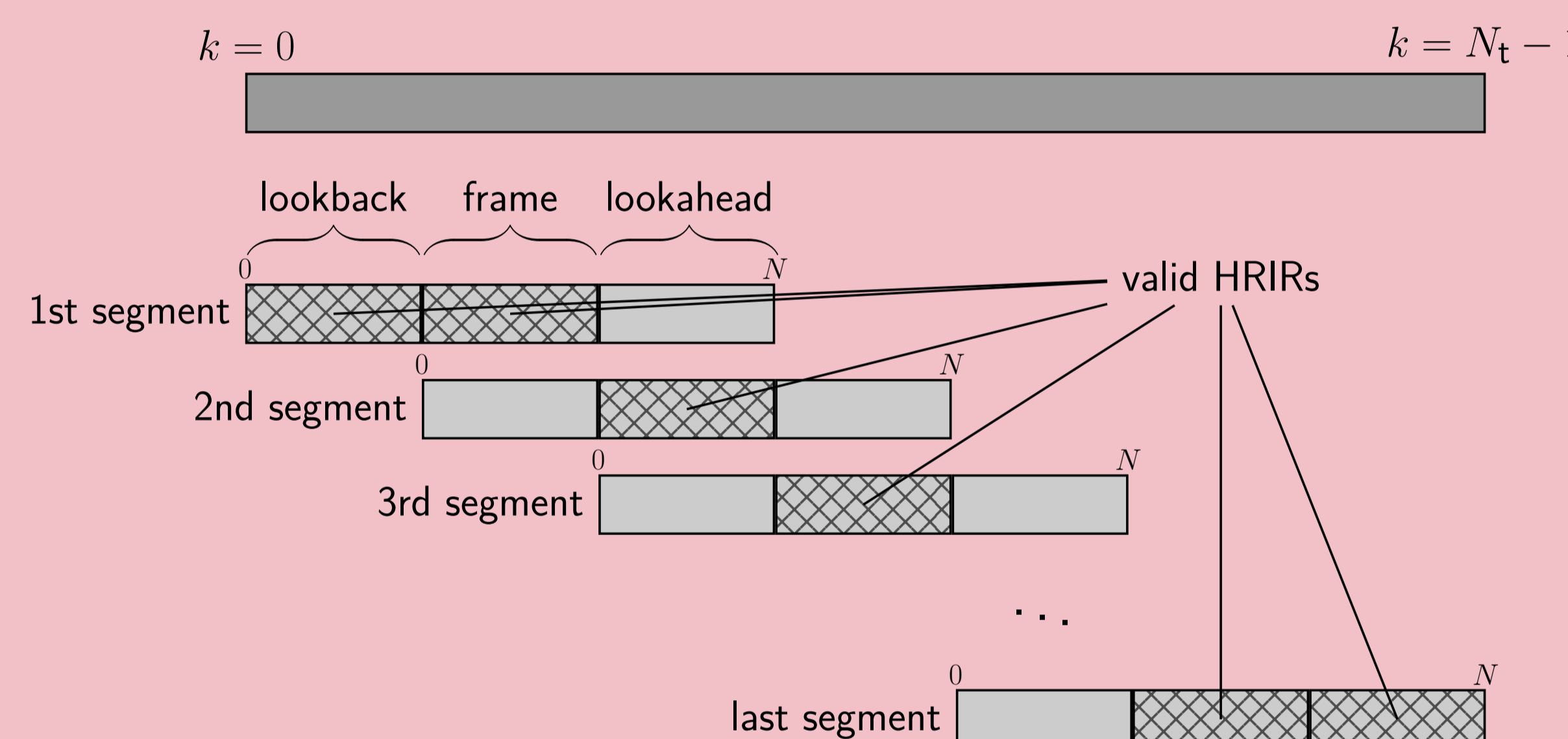
$$\rightarrow \text{Kalman Smoother: } \hat{\boldsymbol{\mu}}_k = \mathbb{E} \left\{ \mathbf{z}_k \mid \{\mathbf{x}_\ell\}_{\ell=0}^N, \{y(\ell)\}_{\ell=0}^N \right\}$$

2. Idea: **learn parameters** $\theta = \{\mathbf{A}, \mathbf{\Gamma}, \sigma_n^2, \boldsymbol{\mu}_0, \mathbf{P}_0\}$ with Expectation Maximization (EM) algorithm [2]

- Generic state transition matrix $\mathbf{A} \rightarrow$ better state prediction
 - E-step: compute expected log-likelihood
 - M-step: update parameters
- repeat for \mathcal{I} iterations
- Memory problem: sampling rate of 24 kHz, $S = 3$ loudspeakers, 8 ms estimated HRIR lengths, 20-second recording \rightarrow 1.2 TB memory \neq

3. Idea: process **small segments** to limit memory requirements

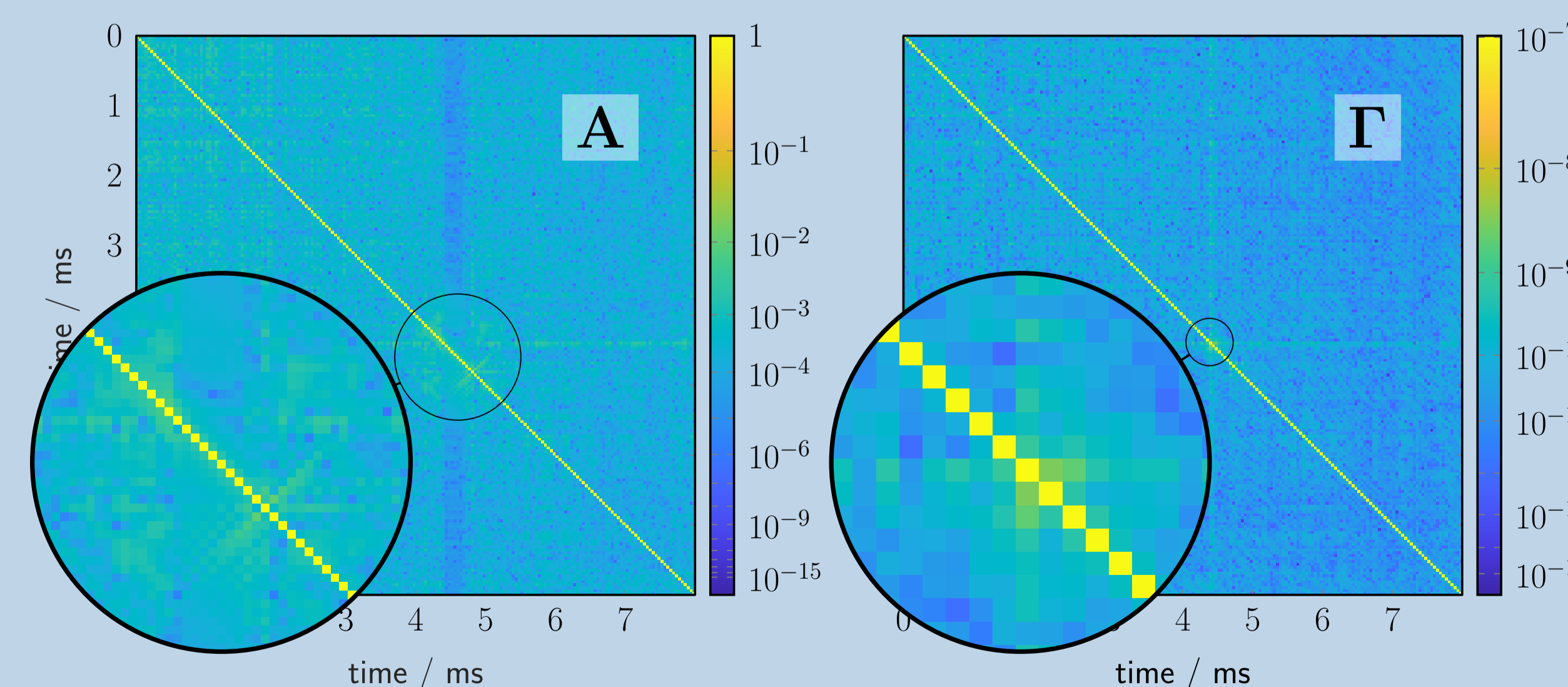
- Divide total recording of length N_t into small segments of length N
- Learn *independent* set of parameters per segment



4a Evaluation: Learned Parameters

Simulated measurements with rigid sphere "HRTFs" for different velocities

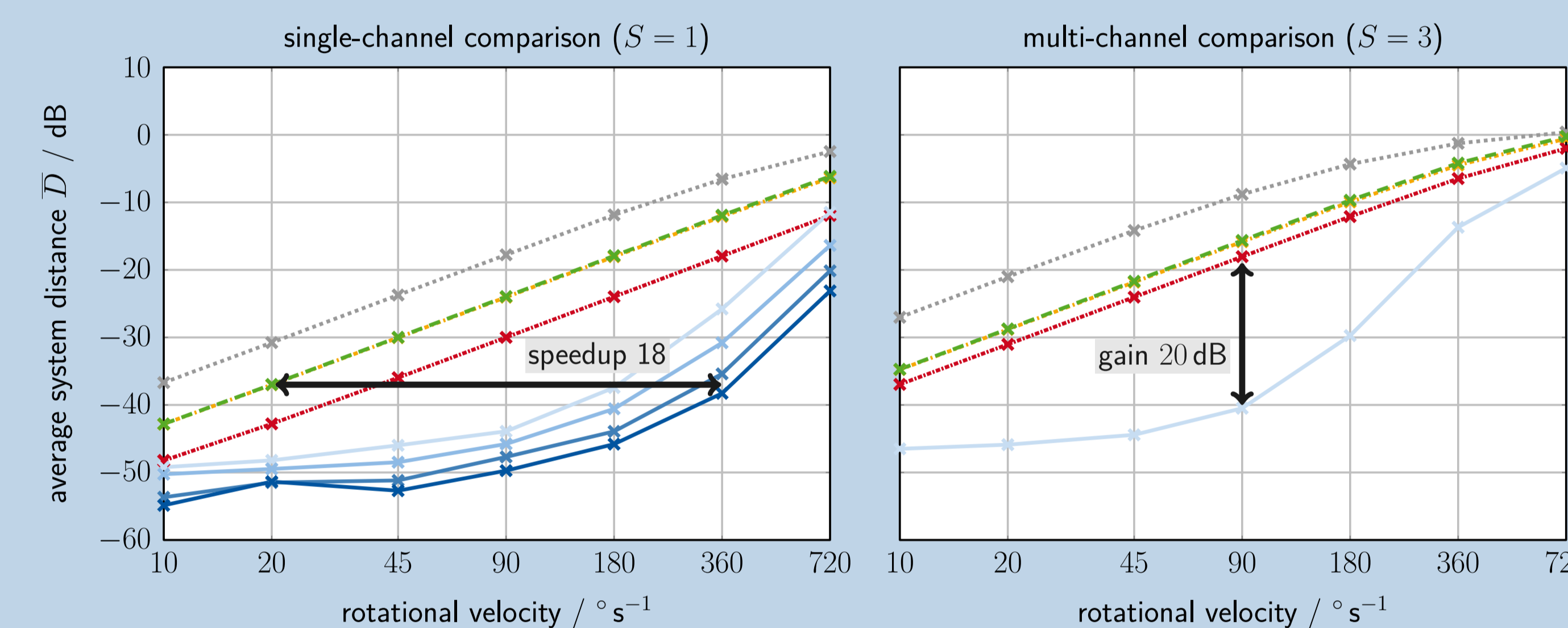
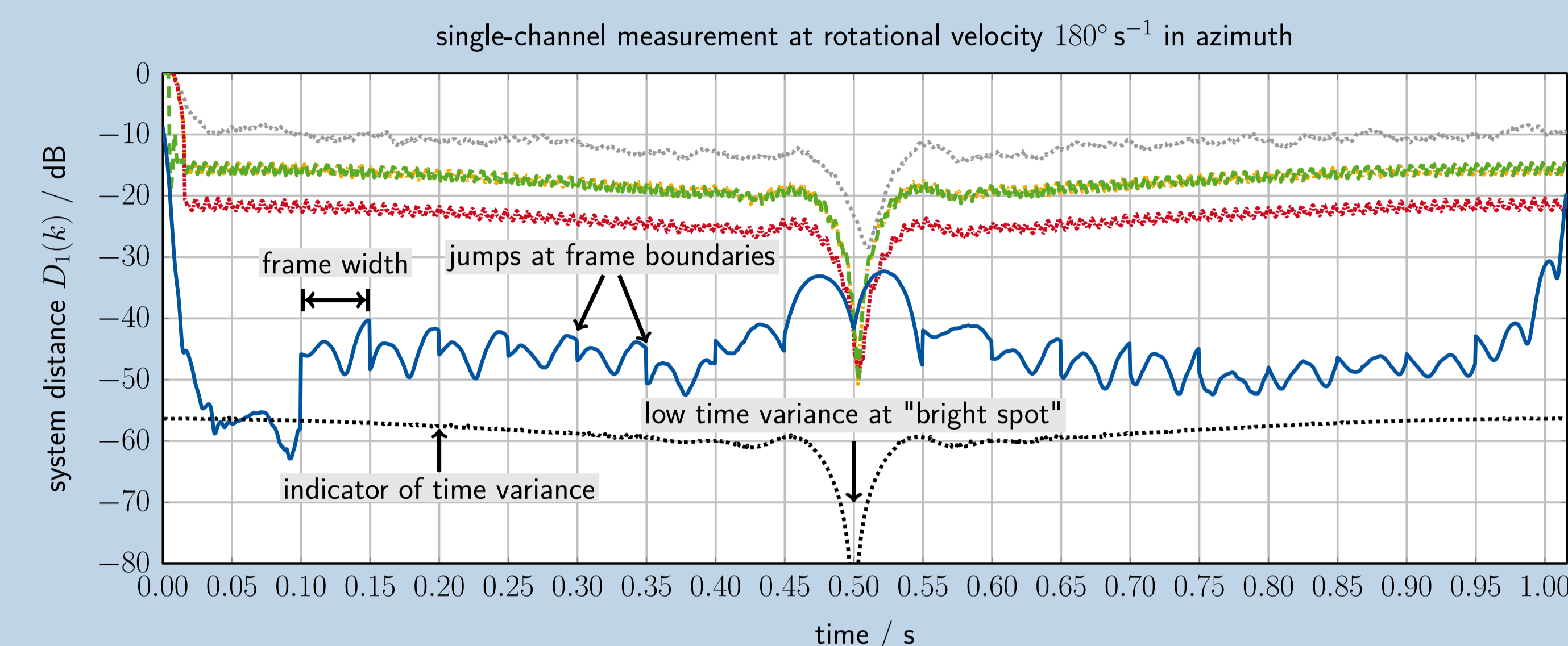
- Exemplary results (1st segment) for single-channel and $\mathcal{I} = 10$ iterations:



- Larger weights and covariances near base delay of 4.4 ms

4b Evaluation: Identification Performance

- White noise or perfect sequence (PSEQ) excitation [3]



- Open: evaluation in real HRTF measurement setting

5 Conclusions

- Learning system model with EM very beneficial for spherical head model
- **Increase quality** of HRTFs or **reduce duration** of measurement

References

- [1] G. Enzner, "3D-continuous-azimuth acquisition of head-related impulse responses using multi-channel adaptive filtering," in *IEEE Workshop on Appl. of Signal Process. to Audio and Acoust. (WASPAA)*, 2009, pp. 325–328.
- [2] C. M. Bishop, *Pattern Recognition and Machine Learning*. New York, NY, USA: Springer, 2006.
- [3] C. Antweiler and G. Enzner, "Perfect sequence LMS for rapid acquisition of continuous-azimuth head related impulse responses," in *IEEE Workshop on Appl. of Signal Process. to Audio and Acoust. (WASPAA)*, 2009, pp. 281–284.