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Published in:
Acoustical Society of America. Journal

Link to article, DOI:
10.1121/1.5101769

Publication date:
2019

Document Version
Publisher's PDF, also known as Version of record

Citation (APA):
Mansour, N., Marschall, M., May, T., Westermann, A., & Dau, T. (2019). A Method for Conversational Signal-to-Noise Ratio Estimation in Real-World Sound Scenarios. *Acoustical Society of America. Journal*, *145*, 1873. https://doi.org/10.1121/1.5101769

A Method for Conversational Signal-to-Noise Ratio Estimation in Real-World Sound Scenarios

Naim Mansour[1], Marton Marschall[1], Adam Westermann[2], Tobias May[1], Torsten Dau[1]

[1]Hearing Systems Section, Technical University of Denmark [2]Widex A/S, Lynge, Denmark

1. Introduction

- Analysis of conversational signal-to-noise ratios (SNRs) measured in real-world scenarios can provide insights into communicative strategies and difficulties, and guide development of hearing devices [1].
- Measuring SNRs accurately and realistically is challenging in typical recording conditions, where only a mixture of sound sources is captured. Typical single-channel methods [2] rely on subtracting estimates of noise in a frame $\mathbf{N_r(f)}$ of the recording from the mixture of speech and noise $\mathbf{(S(f)+N(f))_r}$ (1).

$$\mathrm{SNR}(f) = \frac{(S(f)+N(f))_r - N_r(f)}{N_r(f)} \quad (1) \qquad \mathrm{SNR}(f) = \frac{S_r(f)}{N_r(f)} \quad (2)$$

- A novel in-situ estimation method is proposed, where the speech signal of a person in natural conversation is captured by a cheek-mounted microphone, free-field adjusted, and then convolved with a measured impulse response to estimate the clean speech receiver component $\mathbf{S_r(f)}$ (2).
- The method is analyzed using in-situ recordings of a real-world workspace meeting and compared to the single-channel technique in terms of its resulting SNR distribution.

2. Method

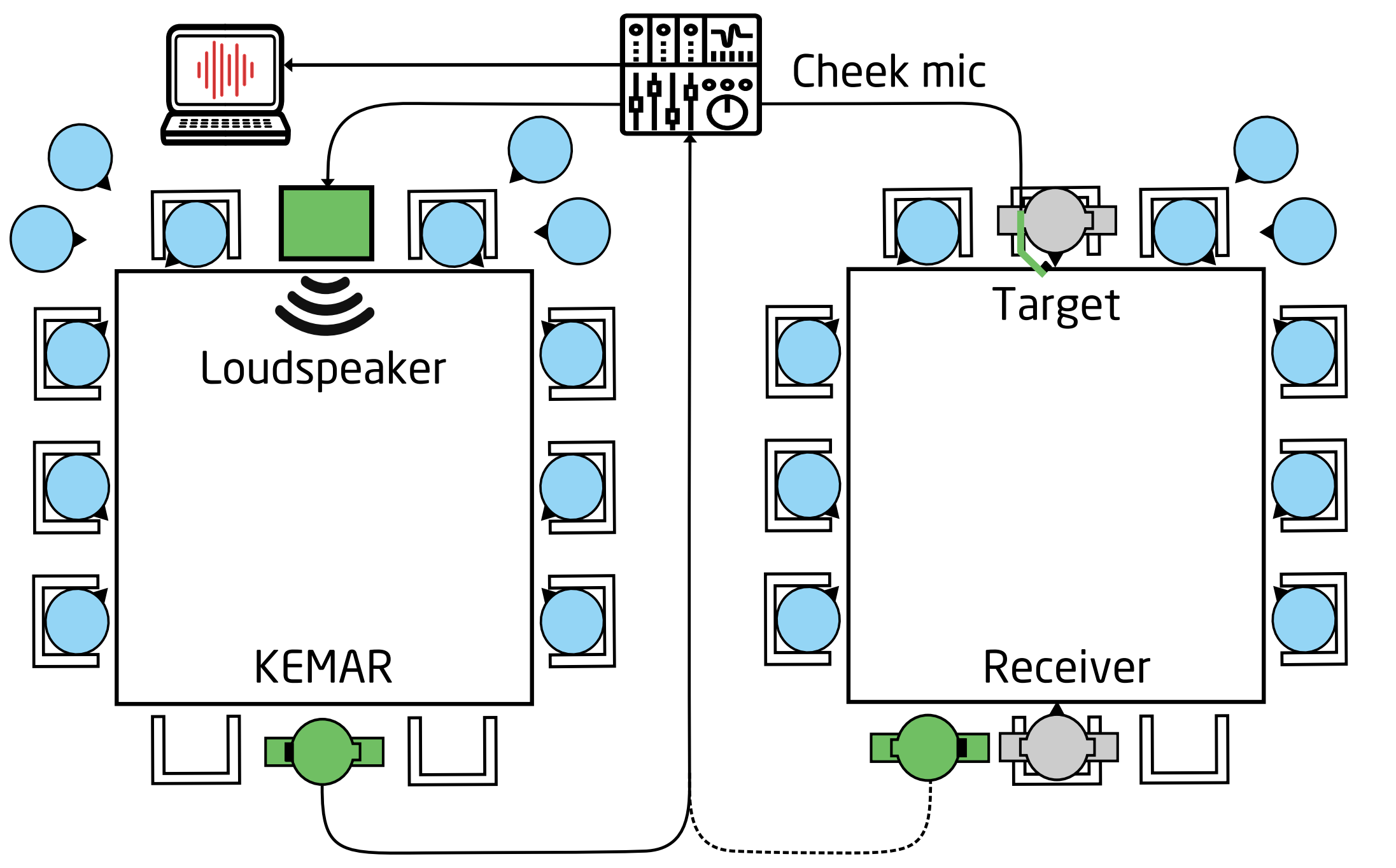
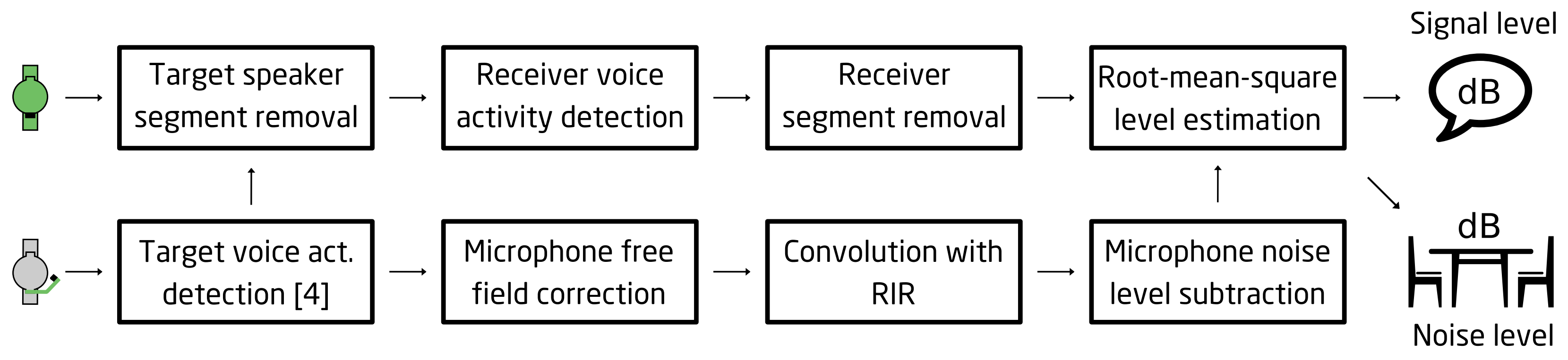


Figure 1. Left: Room impulse response recording - Right: Conversation recording (top-down view)

Controlled recording of workspace meeting

- Realistic enactment of a typical workspace meeting in a corporate office meeting room
- Conversation between two normal-hearing (NH) people seated across a square conference table, in a background (BG) of 10 NH talkers conversing in pairs about work-related topics
- Room impulse response (RIR) between target and receiver position recorded by KEMAR [3] manikin (left ear), while BG talkers were quiet
- Cheek microphone (CM) captures source speech, KEMAR (right ear) noise at receiver

SNR estimation



Target speaker segment removal → Receiver voice activity detection → Receiver segment removal → Root-mean-square level estimation → dB Signal level

Target voice act. detection [4] → Microphone free field correction → Convolution with RIR → Microphone noise level subtraction

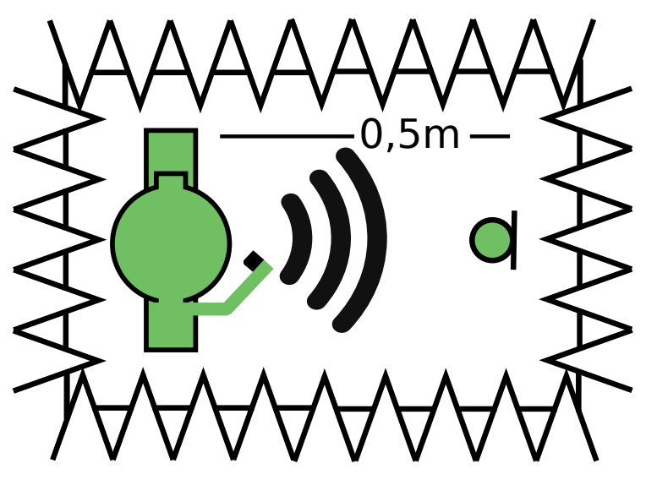
dB Noise level

- Frame-based processing, with 10s frame length, and 90% overlap between frames
- Broad-band SNR based on root-mean-square (RMS) power within each frame

Voice activity detection (VAD) & segment removal

- Energetic VAD [4] to separate speech from background in CM & KEMAR
- Target speaker segment removal in KEMAR recording through binary mask from CM VAD

Microphone free-field correction



- Transfer function between CM and reference microphone to obtain free-field acoustical conditions in the target speech signal
- Recorded in anechoic chamber with KEMAR producing white noise at a level of 90dB, and reference microphone at a distance of 0.5m

Convolution with room impulse response

- CM signal convolved with appropriately scaled RIR recorded between target & receiver

Microphone noise level subtraction

- VAD-driven CM RMS level reduction, subtracting BG noise power in frame $\mathbf{n_{RMS}(f)}$ from signal $\mathbf{s_{RMS}(f)}$

$$s_c(f) = s(f)\,\frac{s_{RMS}(f) - n_{RMS}(f)}{s_{RMS}(f)}$$

3. Results

Free-field correction & room impulse response

- IR computed from 15s exponential frequency sweep, FFC low-pass filtered at 10kHz

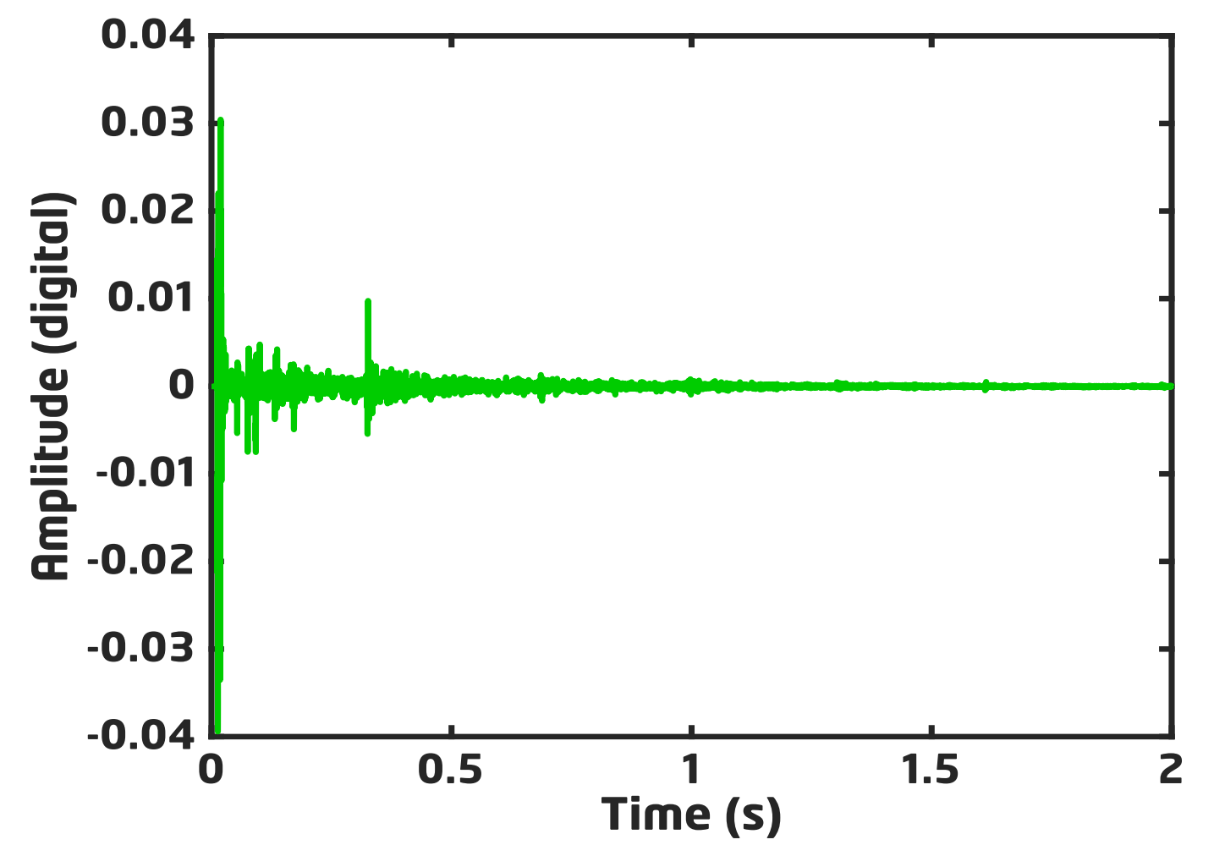


Figure 3. Impulse response

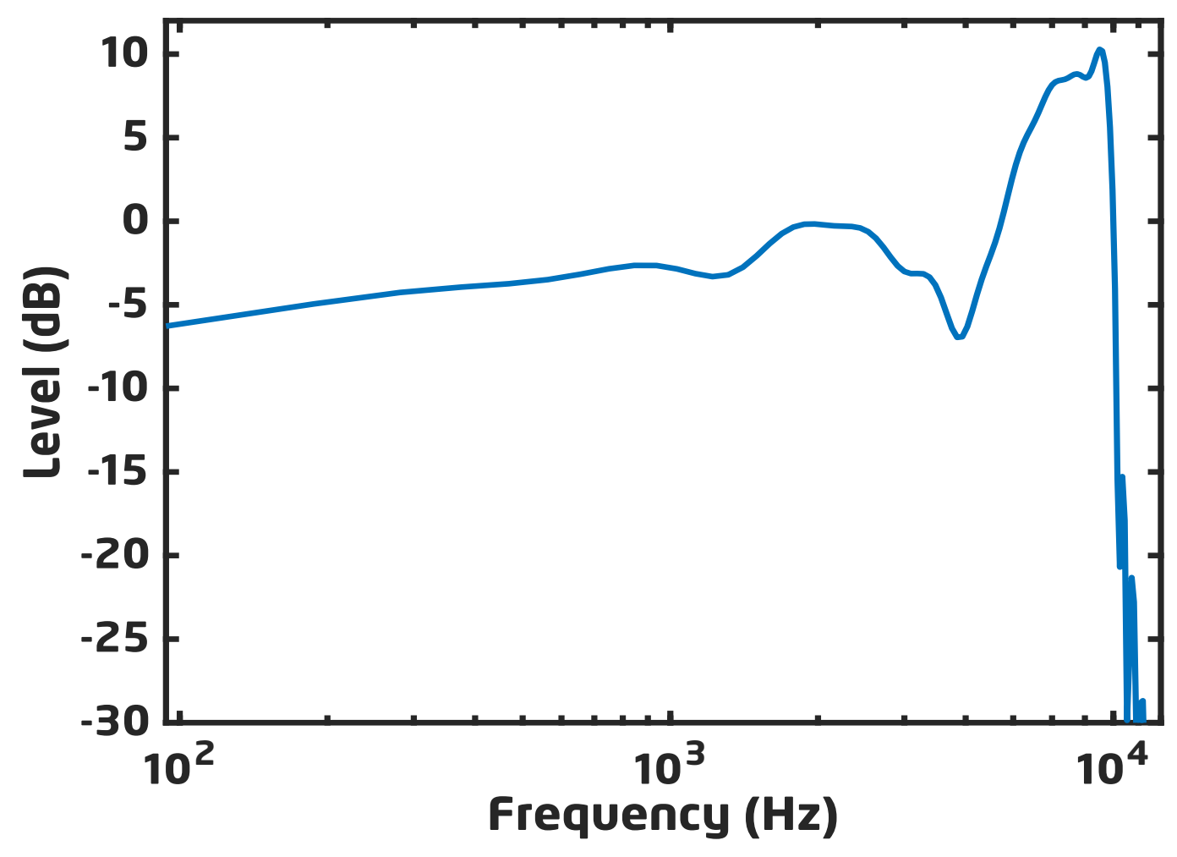


Figure 4. Free-field correction

Speech and background levels

- Derived from 6-minute recording, temporal progression and level distributions are shown

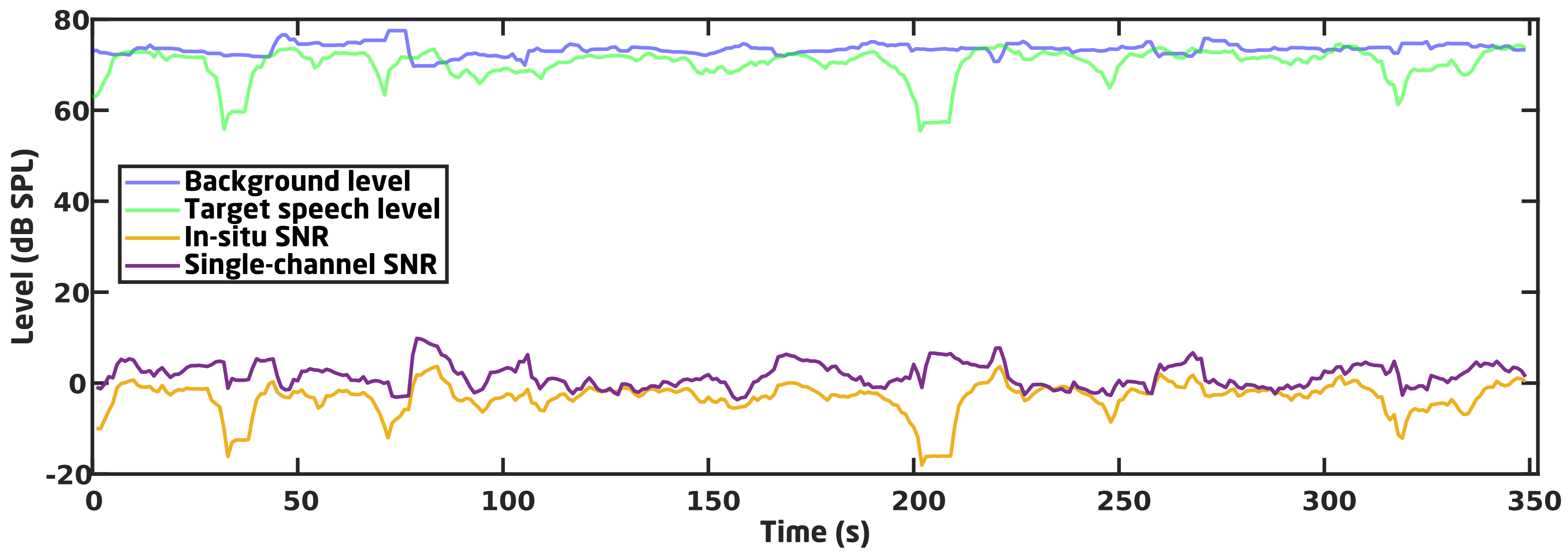


Figure 5. Levels and SNR for a 6-minute workspace meeting recording

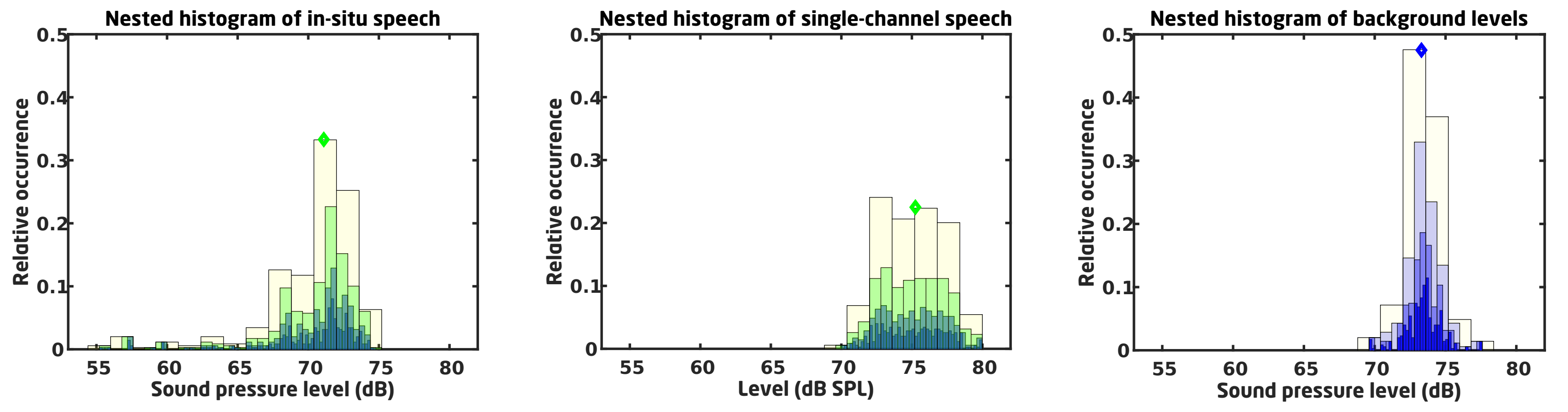


Figure 6. Nested level histograms of in-situ speech, single-channel speech and background levels

Signal-to-noise ratios

- Computed according to respective in-situ (Eqn. 2) and single-channel (1) SNR equations

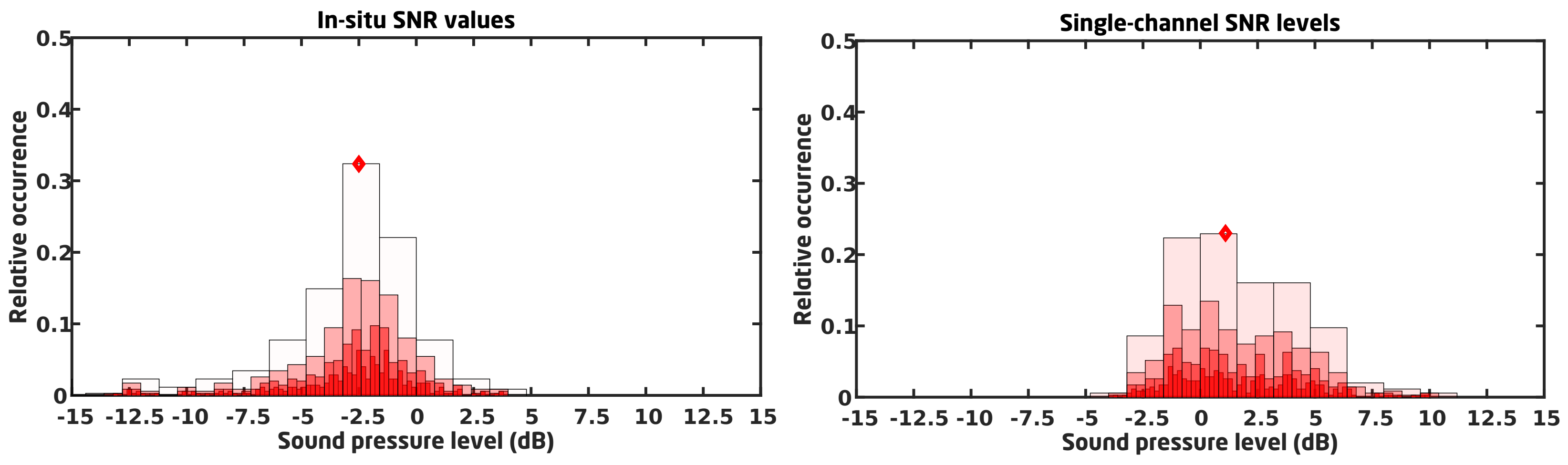


Figure 7. Nested histograms of SNR for in-situ (left) and single-channel (right) method

- Time-averaged dynamic range:
  In-situ speech: **55-74 dB** Single-channel speech: **70-80 dB** Background: **70-76 dB**
- Median values, both methods:
  In-situ speech: **71 dB** Single-channel speech: **75.2 dB** Background: **73.5 dB**
  In-situ SNR: **-2.5 dB** Single-channel SNR: **1.7 dB**

4. Discussion & Summary

- A high temporal-resolution measurement technique for speech and background levels allows tracking of in-situ conversational SNR, even at negative values.
- The wide dynamic range found for in-situ speech levels is likely due to natural speech pauses and turn-taking during conversation.
- The obtained in-situ SNRs are lower than the corresponding single-channel SNRs, likely due to speech signal being more accurately tracked.
- The 4.2 dB difference between the median in-situ and single-channel SNR indicates a potential overestimation of SNRs with traditional techniques. This is likely due to level effects and correlations between the speech and noise when both are present.
- The in-situ approach requires the availability of a CM signal and suitable RIR recordings.
- The proposed SNR estimation method can accurately characterize in-situ SNRs. This may contribute tounderstanding how humans communicate in challenging environments, and help improving compensation strategies in hearing instruments.

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Contact

Naim Mansour
E-mail: naiman@dtu.dk
Phone: +45 60 56 66 57
Web: www.naim-mansour.be