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

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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 $\mathbf{N}_r(\mathbf{f})$ of the recording from the mixture of speech and noise $(\mathbf{S}(\mathbf{f}) + \mathbf{N}(\mathbf{f}))_r$ (1).

$$\text{SNR}(\mathbf{f}) = \frac{(\mathbf{S}(\mathbf{f}) + \mathbf{N}(\mathbf{f}))_r - \mathbf{N}_r(\mathbf{f})}{\mathbf{N}_r(\mathbf{f})} \quad (1) \quad \text{SNR}(\mathbf{f}) = \frac{\mathbf{S}_r(\mathbf{f})}{\mathbf{N}_r(\mathbf{f})} \quad (2)$$

- 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 $\mathbf{S}_r(\mathbf{f})$ (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

- 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 $\mathbf{N}_{\text{RMS}}(\mathbf{f})$ from signal $\mathbf{S}_{\text{RMS}}(\mathbf{f})$

$$\mathbf{S}_c(\mathbf{f}) = \mathbf{S}(\mathbf{f}) \frac{\mathbf{S}_{\text{RMS}}(\mathbf{f}) - \mathbf{N}_{\text{RMS}}(\mathbf{f})}{\mathbf{S}_{\text{RMS}}(\mathbf{f})}$$

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 to understanding how humans communicate in challenging environments, and help improving compensation strategies in hearing instruments.

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