Joint Dereverberation and Noise Reduction Using Beamforming and a Single-Channel Speech Enhancement Scheme

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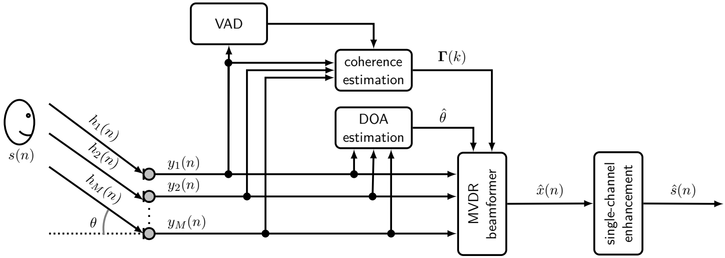
Introduction

- Overview of the proposed system
- Design of the MVDR beamformer
  - DOA estimated using MUSIC
  - Estimated noise covariance
- Single-channel enhancement scheme
  - Combination and optimization of published estimators
- Results
  - Objective measures
  - MUSHRA scores
  - WER using a baseline recognizer

1. Proposed System

Overview



- Beamformer: towards estimated direction of arrival (DOA)
- Single-channel enhancement: based on statistical estimators
  - Late reverberant spectral variance (LRSV)
  - Noise spectral variance (NSV)
  - Speech spectral variance (SSV)

2. MVDR Beamformer

With $Y_m(k,\ell)$ the STFT of the input signal in the m-th microphone we define

$$\mathbf{Y}(k,\ell) = [Y_1(k,\ell)\ Y_2(k,\ell)\ \ldots\ Y_M(k,\ell)]^T$$

The output $\hat{X}(k,\ell)$ of the beamformer is obtained as

$$\hat{X}(k,\ell) = \mathbf{W}_\theta^H(k)\mathbf{Y}(k,\ell)$$

where

$$\mathbf{W}_\theta(k) = \frac{\mathbf{\Gamma}^{-1}(k)\mathbf{d}_\theta(k)}{\mathbf{d}_\theta^H(k)\mathbf{\Gamma}^{-1}(k)\mathbf{d}_\theta(k)}$$

- Noise coherence matrix: $\mathbf{\Gamma}(k) \rightarrow$ estimated using a VAD.
- Steering vector: $\mathbf{d}_\theta(k) \rightarrow$ from $\hat{\theta}$ using a far-field assumption.

Estimation of noisefield coherence

- Noise periods identified with a VAD
  - Comparison between the long-term spectral envelope and the average noise spectrum
- $\mathbf{\Gamma}(k)$ is estimated using detected noise-only frames
- Alternatively, a theoretically diffuse noise field is used:

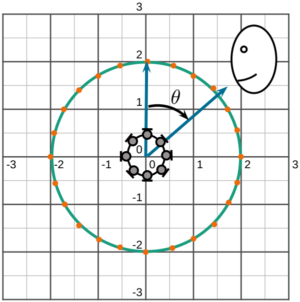
$$\mathbf{W}_\theta(k) = \frac{(\mathbf{\Gamma}_{\mathrm{diff}}(k)+\varrho(k)\mathbf{I}_M)^{-1}\mathbf{d}_\theta(k)}{\mathbf{d}_\theta^H(k)(\mathbf{\Gamma}_{\mathrm{diff}}(k)+\varrho(k)\mathbf{I}_M)^{-1}\mathbf{d}_\theta(k)}$$

with $\varrho(k)$ a constraint such that

$$\mathbf{W}_\theta^H(k)\mathbf{W}_\theta(k) \leq \mathrm{WNG}_{\mathrm{max}} = 10 \text{ dB}$$

Ramirez, J., Segura, J.C., Benitez, C., de la Torre, A., and Rubio, A., *Efficient voice activity detection algorithms using long-term speech information*, 2003.

DOA Estimation



$$\hat{\theta} = \underset{\theta}{\text{argmax}} \frac{1}{K} \sum_{k_{\text{low}}}^{k_{\text{high}}} U_\theta(k, \ell),$$

where $U_\theta(k, \ell)$ is the MUSIC pseudo-spectra:

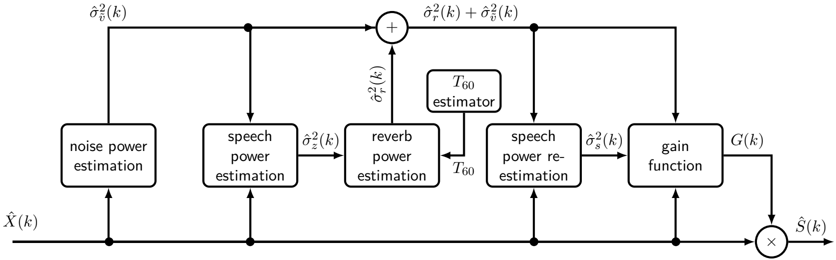
$$U_\theta(k, \ell) = \frac{1}{\mathbf{d}_\theta^H(k)\boldsymbol{E}(k,\ell)\boldsymbol{E}^{\mathrm{H}}(k,\ell)\mathbf{d}_\theta(k)}$$

$$\boldsymbol{E}(k, \ell) = [\boldsymbol{e}_{Q+1}(k, \ell) \ldots \boldsymbol{e}_M(k, \ell)]$$

with $\boldsymbol{e}_m$ denoting eigenvectors of the covariance matrix of $\mathbf{Y}(k, \ell)$.

Schmidt, R., *Multiple emitter location and signal parameter estimation*, 1986.

Overview

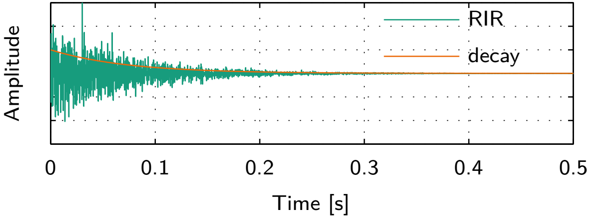


- $\sigma^2_{\tilde{v}}(k,\ell)$ estimated using Minimum Statistics
- $\sigma^2_s(k,\ell)$ estimated using Cepstral Smoothing
- $\sigma^2_r(k,\ell)$ estimated using Lebart's approach

Martin, R., *Noise Power Spectral Density Estimation Based on Optimal Smoothing and Minimum Statistics*, 2001.
Breithaupt, C., Gerkmann T. and Martin, R., *A Novel A Priori SNR Estimation Approach Based on Selective Cepstro-Temporal Smoothing*, 2008.
Eaton, J., Gaubitch, N.D., Naylor, P.A., *Noise-robust reverberation time estimation using spectral decay distributions with reduced computational cost*, 2012.
Lebart, K., Boucher J.M. and Denbigh, P., *A new method based on spectral subtraction for speech dereverberation*, 2013.

LRSV estimation

- RIR modeled as Gaussian noise with decay $\Delta = \frac{3 \ln 10}{T_{60} f_s}$



- Representing the variance of the reverberant speech as:
  - $\sigma_z^2(k,\ell) = \sigma_r^2(k,\ell) + \sigma_s^2(k,\ell)$
- Leads to the estimator
  - $\hat{\sigma}_r^2\,(k,\ell) = \mathrm{e}^{-2\Delta T_d f_s} \sigma_z^2(k, \ell - T_d/T_s)$

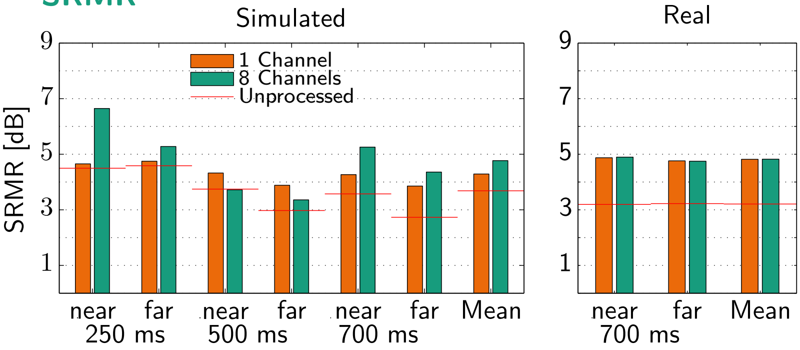
Lebart, K., Boucher J.M. and Denbigh, P., *A new method based on spectral subtraction for speech dereverberation*, 2001.

Gain function

- The output $\hat{X}(k,\ell)$ of the beamformer contains the anechoic speech, remaining noise and spatially filtered reverberation
  - $\hat{X}(k,\ell) = S(k,\ell) + \tilde{V}(k,\ell) + R(k,\ell)$
- We aim to compute a real gain such that:
  - $\hat{S}(k,\ell) = G(k,\ell)\hat{X}(k,\ell)$
- Computation of $G(k,\ell)$ using an MMSE estimation of the speech amplitude based on a super Gaussian speech model.

Breithaupt, C., Krawczyk, M., and Martin, R., *Parameterized MMSE spectral magnitude estimation for the enhancement of noisy speech*, 2008.

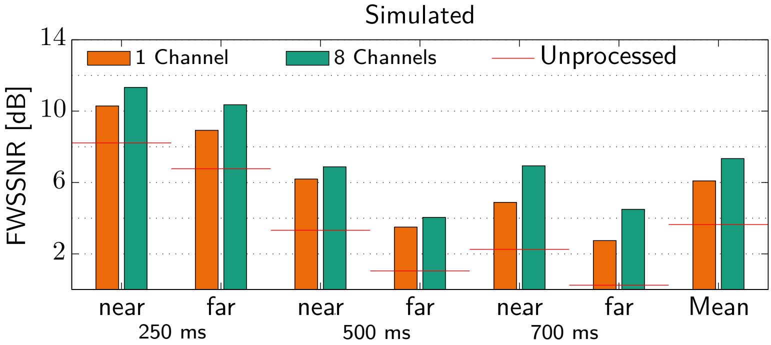
SRMR



- Illustrates dereverberation performance in all condition
- Better dereverberation achieved by multichannel, except for T_{60}=500 ms

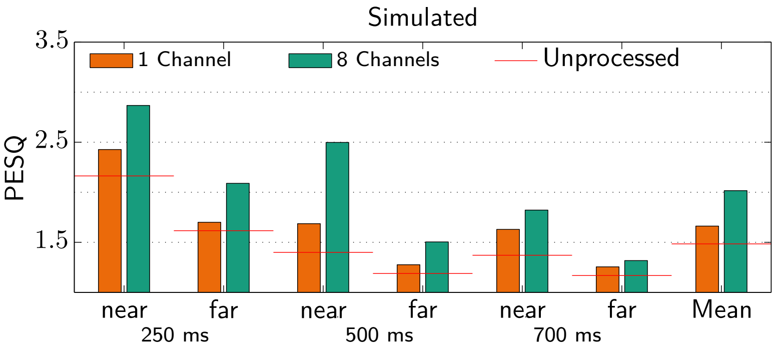
4. Objective Measures

FWSSNR



- Illustrates noise reduction in all condition
- Beamforming step advantageous for the noise reduction

PESQ



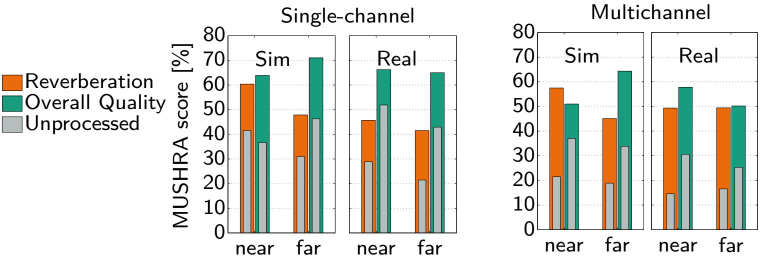
- Improvement of PESQ score in all condition illustrate the overall improvement in speech quality

5. Subjective Tests

MUSHRA test

Intermediate results of the subjective test ran by the organizers:

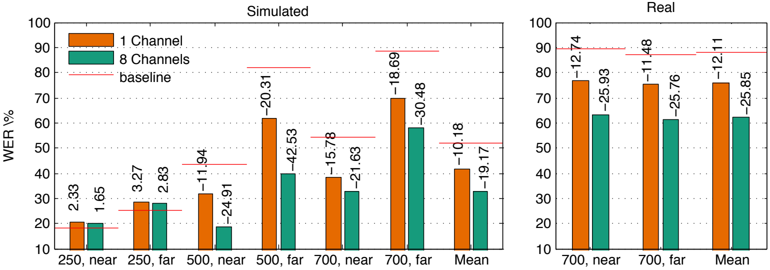
- Tests carried out separately for 1 and 8 channels



- Improvement for all tested condition
- Higher improvement of the overall quality

Word Error Rate

- Baseline recognizer provided by the organizers
- Using pre-trained models on clean data



7. Conclusion

- System based on combination of MVDR beamformer and spectral enhancement
- All parameters are blindly estimated
- Speech enhancement achieved in all conditions in terms of:
  - Objective measures
  - Subjective tests
  - Word error rate

Thank you very much for your attention

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Questions ?

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