

# Blind Speech Separation Exploiting Temporal and Spectral Correlations Using 2D-HMMS

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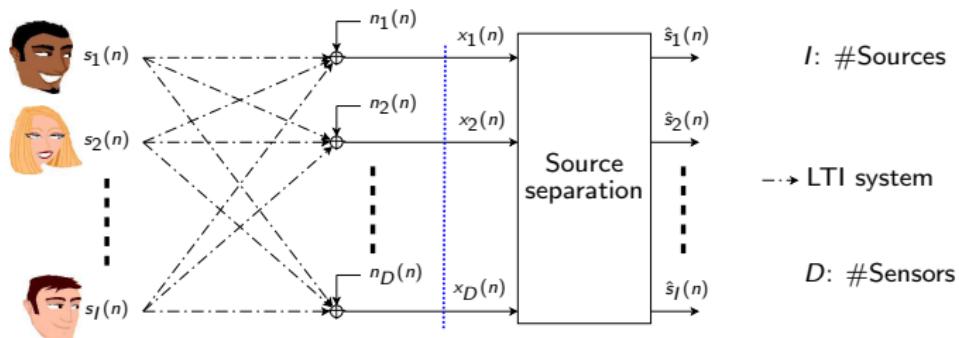
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Computer Science, Electrical  
Engineering and Mathematics

Communications Engineering  
Prof. Dr.-Ing. Reinhold Häb-Umbach



# Frequency domain blind source separation



## Short-time Fourier-transform (STFT) domain mixing model

$$\mathbf{X}(m, k) \approx \sum_{i=1}^I \mathbf{H}_i(k) S_i(m, k) + \mathbf{N}(m, k)$$

where  $m$ : frame index,  $k$ : frequency bin

# Sparseness-based BSS

## Generative model in STFT domain

- At most one source dominant in each time-frequency slot  $(m, k)$
- Generative model:

$$\mathbf{X}(m, k) = \begin{cases} \mathbf{N}(m, k) & \text{if } Z(m, k) = 0 \\ \mathbf{H}_i(k)S_i(m, k) + \mathbf{N}(m, k) & \text{if } Z(m, k) \in \{i; i = 1, \dots, I\} \end{cases}$$

- $Z(m, k)$  hidden random variable indicating which source is active

## Goal: Extract $S_i(m, k)$ solely from $\mathbf{X}(m, k)$

- By computation of source activity probability

$$P(Z(m, k) | \mathbf{X}(m, k))$$

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- $Z(m, k)$  hidden random variable indicating which source is active
- Temporal and spectral correlations in  $\mathbf{X}(1..M, 1..K)$  present

## Goal: Extract $S_i(m)$ solely from $\mathbf{X}(1..M, 1..K)$

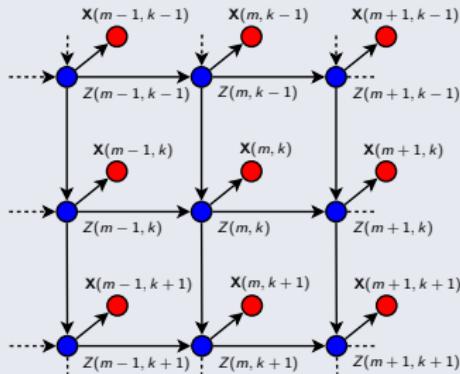
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$$P(Z(m, k) | \mathbf{X}(1..M, 1..K))$$

using temporal and spectral correlations!

# Capturing temporal and spectral correlations

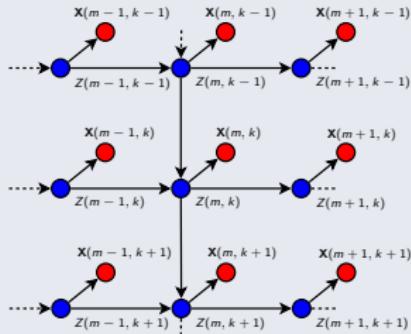
## 2D Hidden Markov Model



- Modeling correlations among adjacent time-frequency slots
- **Problem:** Exact inference is computational intractable!
  - ▶ **Solution:** Alternate inference in time and frequency direction

# Decoding of a 2D-HMM

## Modified forward-backward algorithm (FBA) in frequency direction



- Ignoring vertical dependencies in all other columns
- Modified forward-backward algorithm (FBA):

$$\nu\alpha(m, k) = 1 \propto \nu\mathbf{T}^T (\nu\alpha(m, k-1) \circ \mathbf{o}(m, k-1) \circ \nu\mathbf{u}(m, k-1))$$

$$\nu\beta(m, k) = \pi \propto \nu\mathbf{T} (\nu\beta(m, k+1) \circ \mathbf{o}(m, k+1) \circ \nu\mathbf{u}(m, k+1))$$

$$\nu\gamma(m, k) = 1 \propto \mathbf{o}(m, k) \circ \nu\mathbf{u}(m, k) \circ \nu\alpha(m, k) \circ \nu\beta(m, k)$$

vertical forward prediction variable:  $\nu\alpha(m, k)$ ,  
prior probabilities:  $\pi$ ,

observation likelihood:  $\mathbf{o}(m, k)$ ,

vertical backward variable:  $\nu\beta(m, k)$ ,  
vertical transition matrix:  $\nu\mathbf{T}$ ,

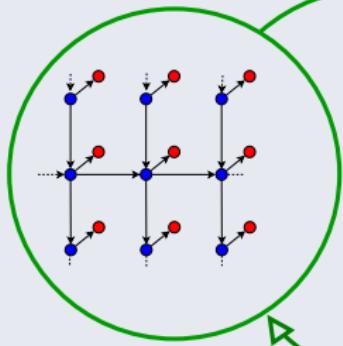
vertical junction variable:  $\nu\mathbf{u}(m, k)$

# Decoding of a 2D-HMM

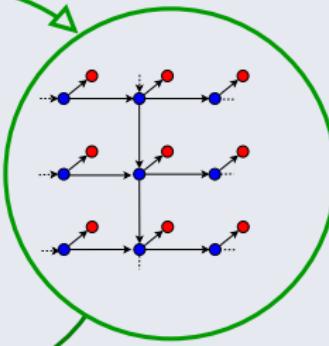
## Iterative Turbo Decoding Scheme

Horizontal modified FBA

$$\mathcal{H}\mathbf{u}(m, k) = (\mathcal{H}\alpha(m, k) \oslash \pi) \circ \mathcal{H}\beta(m, k)$$



Vertical modified FBA



$$\mathcal{H}\mathbf{u}(m, k) = (\mathcal{V}\alpha(m, k) \oslash \pi) \circ \mathcal{V}\beta(m, k)$$

- Based on modified forward-backward algorithm (FBA) along time and frequency direction
- Extrinsic information exchange between both directions

## Polar model of $p(\mathbf{X}(m, k) | Z(m, k))$

### Average *a-posteriori* SNR

$$\varphi(m, k) := \frac{1}{D} \mathbf{X}^H(m, k) \boldsymbol{\Phi}_{\mathbf{NN}}^{-1}(k) \mathbf{X}(m, k)$$

where  $\boldsymbol{\Phi}_{\mathbf{NN}} = \mathbb{E}[\mathbf{NN}^H]$

- Modeled by scaled chi-squared distribution

### Frequency and unit-norm normalize observation vector

$$\begin{aligned}\tilde{Y}_j(m, k) &:= |X_j(m, k)| \exp \left\{ i \frac{\arg[X_j(m, k) X_1^*(m, k)]}{2(k-1)f_s d_{\max}(K c_v)^{-1}} \right\} \\ \mathbf{Y}(m, k) &:= \tilde{\mathbf{Y}}(m, k) / \|\tilde{\mathbf{Y}}(m, k)\|\end{aligned}$$

- Modeled by complex Watson distribution

### Assumption

$$p(\mathbf{X}(m, k) | Z(m, k)) = p(\mathbf{Y}(m, k) | Z(m, k)) \cdot p(\varphi(m, k) | Z(m, k))$$

# Expectation Maximization algorithm for BSS

## Parameters

- Set of unknown parameters  $\Theta = \{\mathbf{W}_1, \dots, \mathbf{W}_I\}$
- Parameters of 2D-HMM are pretrained with speech mixtures

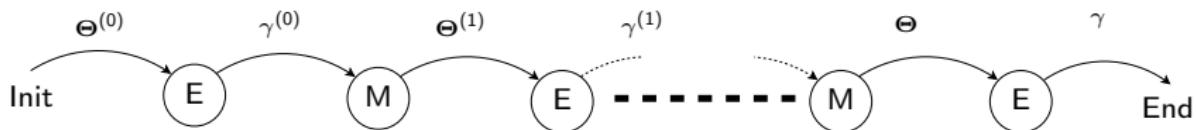
## E-step: Estimate posterior speech activity probabilities

- Use proposed turbo decoding scheme to estimate posterior speech activity probabilities  $\gamma_i^{(\nu)}(m, k) := P(Z(m, k) | \mathbf{X}(1..M, 1..K); \Theta^{(\nu)})$

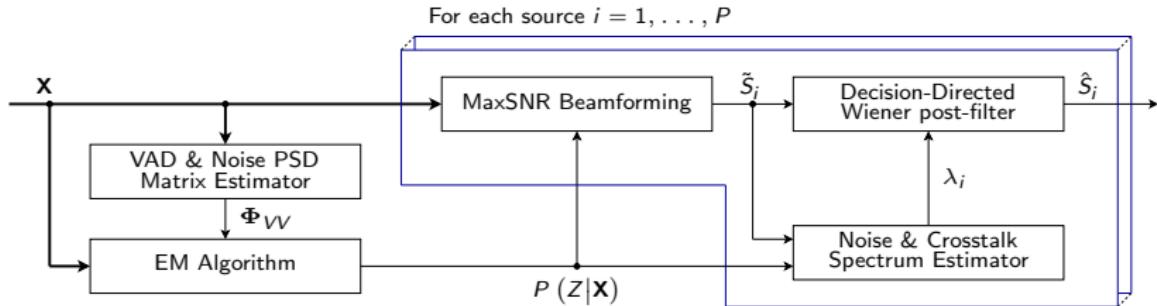
## M-step: Update unknown parameter

- Principal eigenvector of source dependent matrix

$$\Phi_{\mathbf{Y}\mathbf{Y}, i}^{(\nu)} := \frac{\sum_{m=1}^M \sum_{k=1}^K \gamma_i^{(\nu)}(m, k) \mathbf{Y}(m, k) \mathbf{Y}^H(m, k)}{\sum_{m=1}^M \sum_{k=1}^K \gamma_i^{(\nu)}(m, k)}$$



# System overview



## Properties

- Spatial filtering with generalized eigenvector (MaxSNR)-beamforming
- Spectral filtering with Wiener post-filter
- Adaptation control with source activity probability  $P(Z|X)$

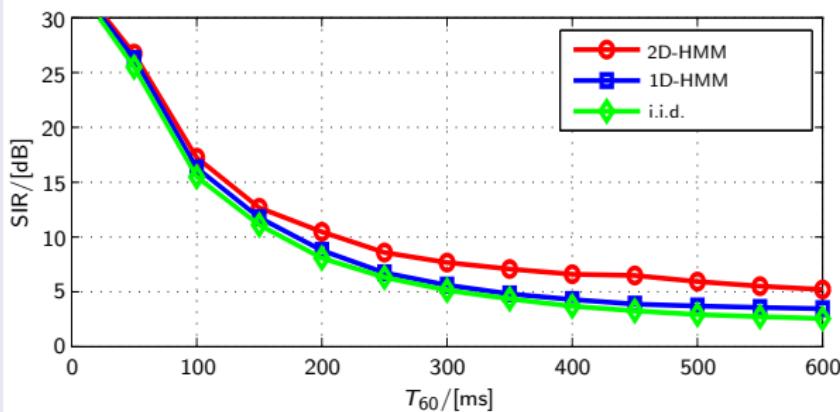
# Evaluation

## Setup

- Four sensor array arranged at the vertices of regular tetrahedron with lateral length of 2cm (No spatial aliasing!)
- 3 sources randomly positioned around the microphone
- Noise recordings of the fan noise of a video projector at  $-10\text{ dB}$
- White noise at  $-20\text{ dB}$
- Image method for reverberant room simulation

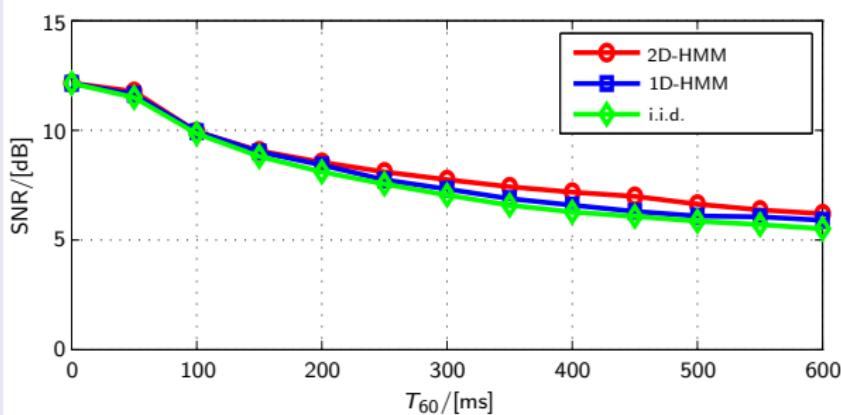
# Evaluation

## Gain in signal-to-interference-ratio (SIR)



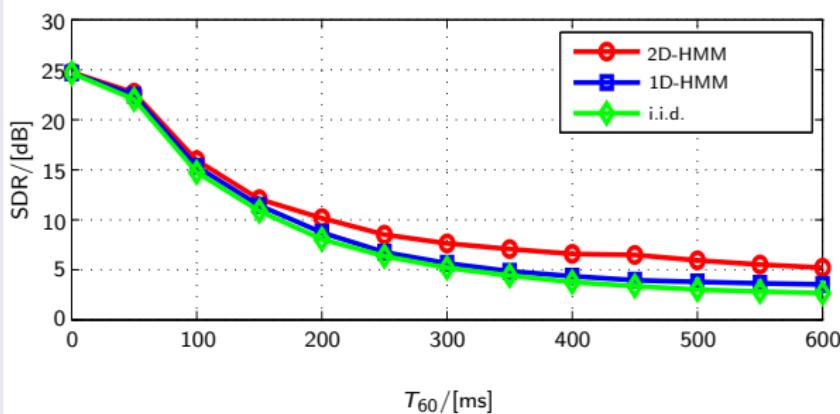
# Evaluation

## Gain in signal-to-noise-ratio (SNR)



# Evaluation

## Gain in signal-to-distortion-ratio (SDR)



# Summary and outlook

## Summary

- Exploiting correlations of adjacent TF-slots for noisy BSS
  - ▶ 2D-HMM to capture temporal and spectral correlations
  - ▶ Iterative decoding scheme with modified FBA algorithm using extrinsic information exchange
- Improved performance in all cases and w.r.t. all measures
  - ▶ Advantages are evident especially in highly reverberant recording conditions

## Outlook

- Exploitation of harmonic structures of speech
- Low latency block online implementation



**Thank you for your attention!**

**Questions ?**

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