

Using Speech Models for Separation

Dan Ellis

Comprising the work of Michael Mandel and Ron Weiss

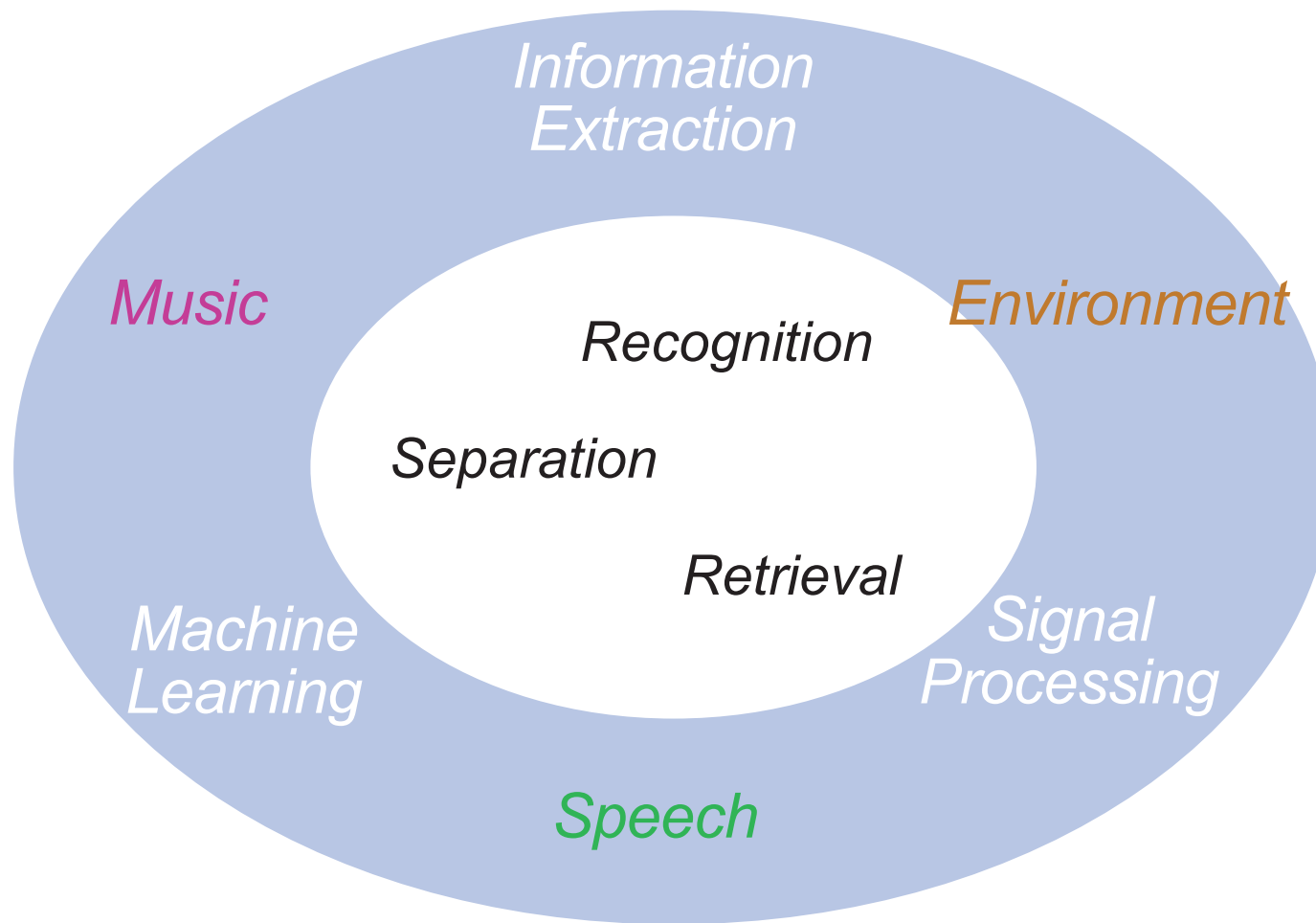
Laboratory for Recognition and Organization of Speech and Audio
Dept. Electrical Eng., Columbia Univ., NY USA

dpwe@ee.columbia.edu

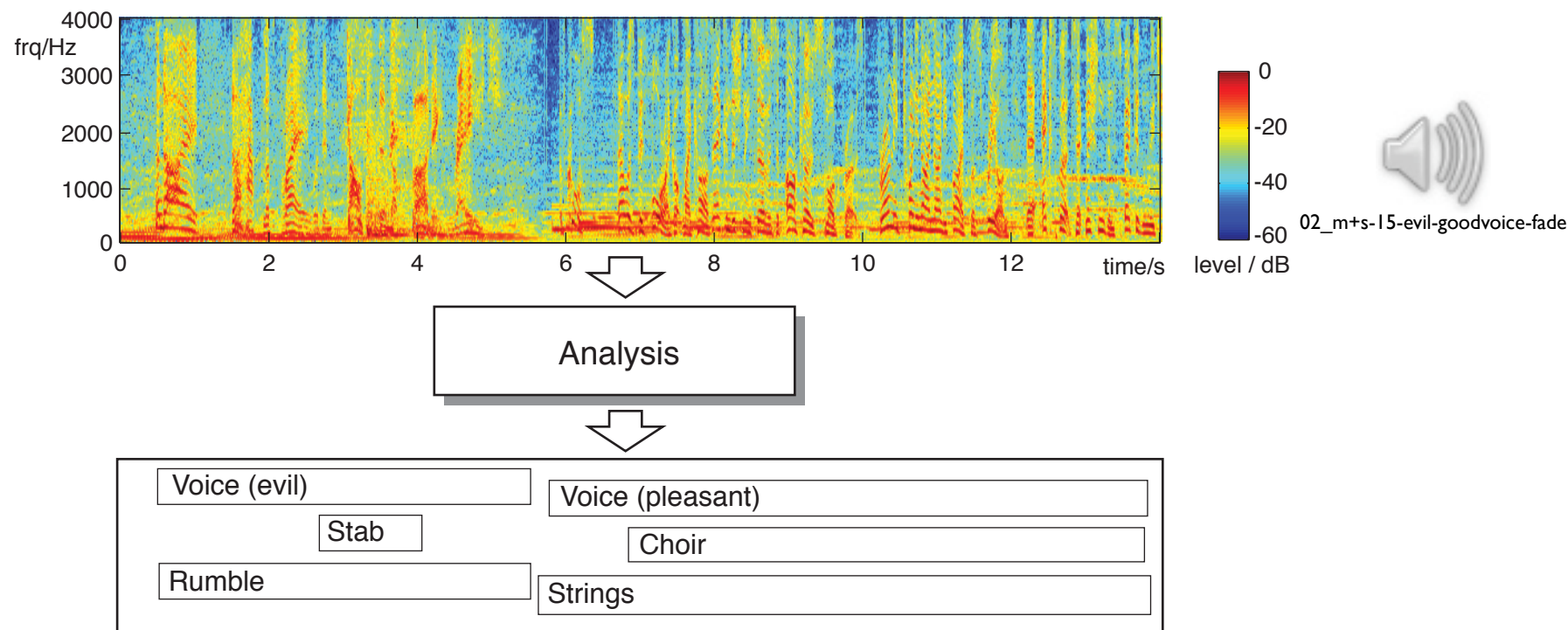
<http://labrosa.ee.columbia.edu/>

1. Source Models and Scene Analysis
2. Eigenvoice Speaker Models
3. Spatial Parameter Models in Reverb
4. Combining Source + Spatial

LabROSA Overview



I. Source Models and Scene Analysis

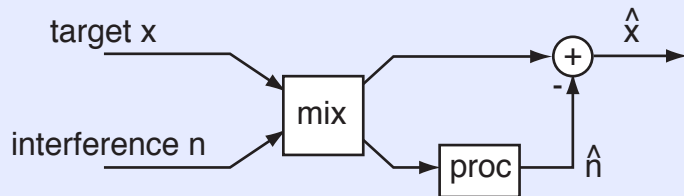


- Sounds rarely occur in **isolation**
 - .. so analyzing mixtures (“scenes”) is a problem
 - .. for humans and machines

Approaches to Separation

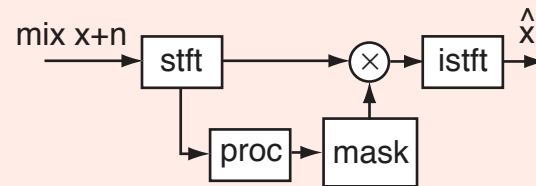
ICA

- Multi-channel
- Fixed filtering
- Perfect separation – maybe!



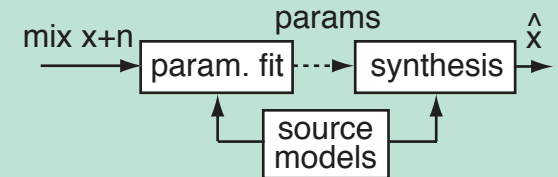
CASA

- Single-channel
- Time-var. filter
- Approximate separation



Model-based

- Any domain
- Param. search
- Synthetic output?



Separation vs. Inference

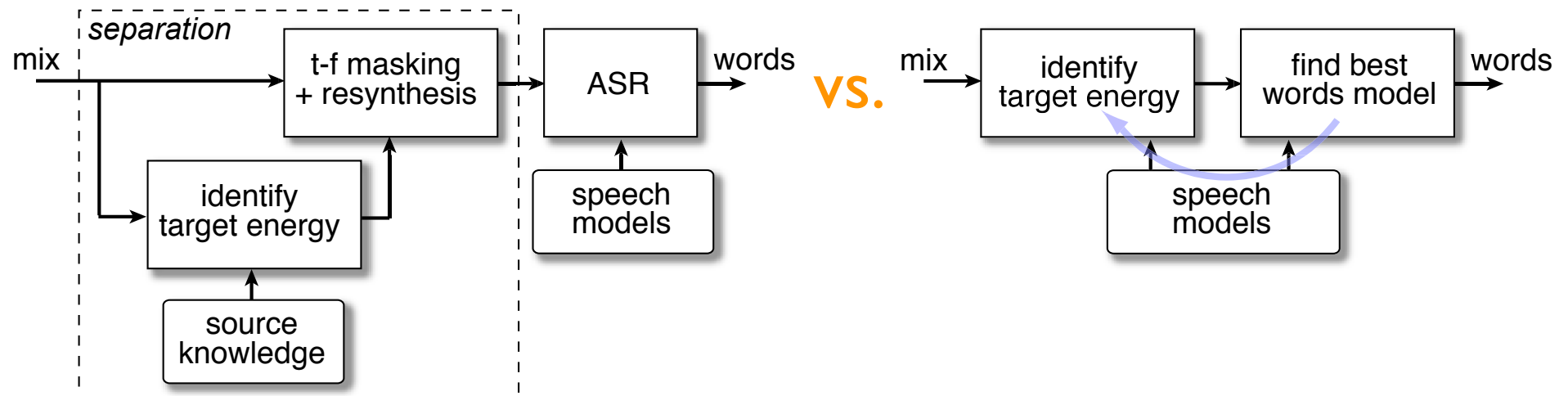
- **Ideal** separation is rarely possible
 - many situations where **overlaps** cannot be removed
- **Overlaps** → **Ambiguity**
 - scene analysis = find “**most reasonable**” explanation
- **Ambiguity can be expressed probabilistically**
 - i.e. posteriors of sources $\{S_i\}$ given observations X :
$$P(\{S_i\} | X) \propto \underbrace{P(X | \{S_i\})}_{\text{combination physics}} \prod_i \underbrace{P(S_i | M_i)}_{\text{source models}}$$
 - search over all source signal sets $\{S_i\}$??
- **Better source models** → **better inference**

2. Speech Separation Using Models

- **Cooke & Lee's Speech Separation Challenge**
 - pairs of short, grammatically-constrained utterances:
<command:4><color:4><preposition:4><letter:25><number:10><adverb:4>
e.g. "bin white by R 8 again"
 - task: report letter + number for "white"
 - (special session at Interspeech '06)



- **Separation or Description?**



Codebook Models

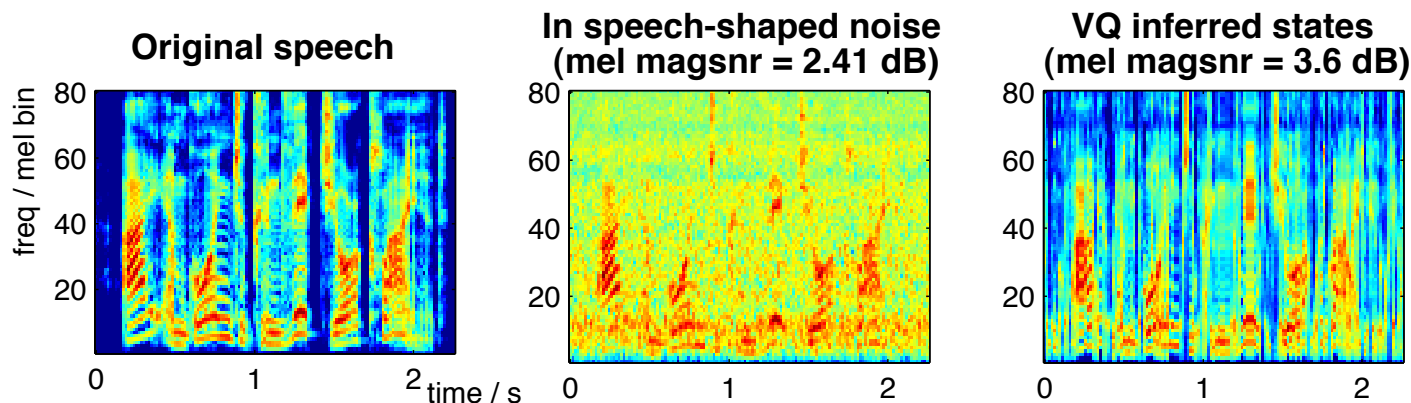
Roweis '01, '03
Kristjansson '04, '06

- Given **models** for sources, find “**best**” (most likely) states for spectra:

$$p(\mathbf{x}|i_1, i_2) = \mathcal{N}(\mathbf{x}; \mathbf{c}_{i_1} + \mathbf{c}_{i_2}, \Sigma) \quad \text{combination model}$$

$$\{i_1(t), i_2(t)\} = \operatorname{argmax}_{i_1, i_2} p(\mathbf{x}(t)|i_1, i_2) \quad \text{inference of source state}$$

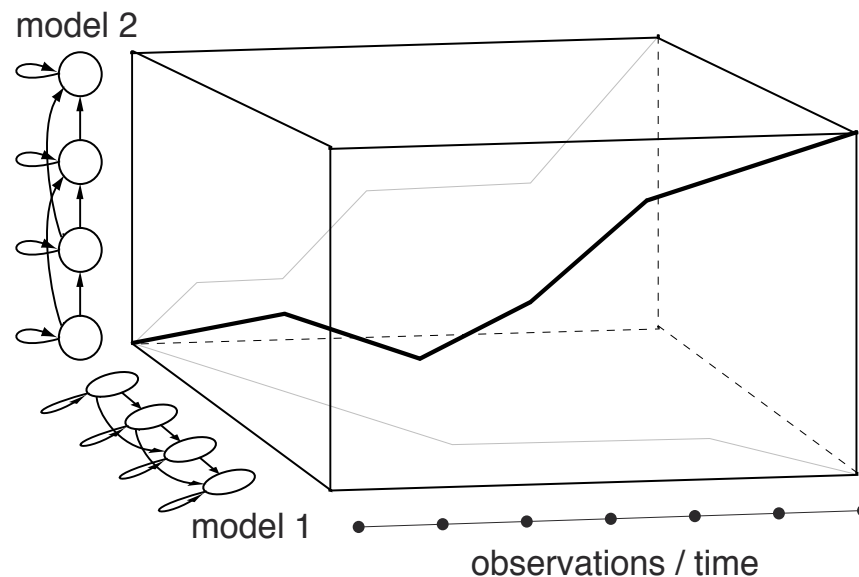
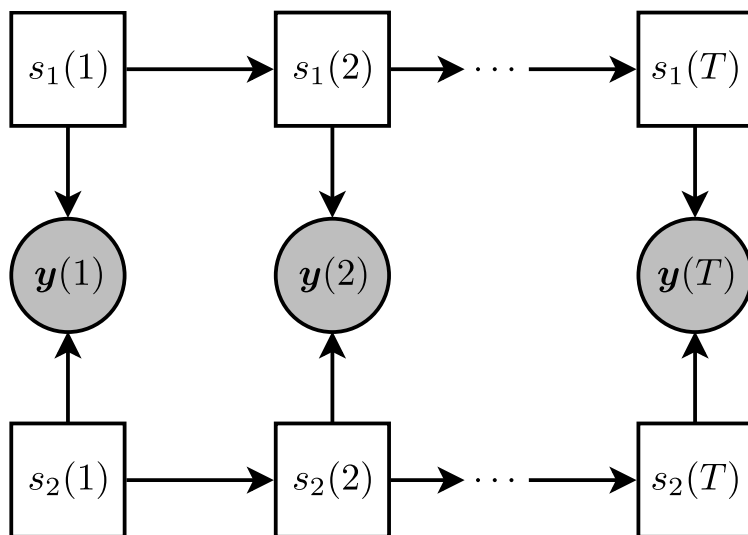
- can include **sequential** constraints...
- E.g. stationary noise:



Speech Recognition Models

Varga & Moore '90

- Speech recognizers contain speech models
 - ASR is just $\operatorname{argmax} P(W | X)$
- Recognize mixtures with **Factorial HMM**
 - i.e. two state sequences, one model for each voice
 - exploit **sequence constraints**, speaker differences



Speech Factorial Separation

Kristjansson, Hershey et al. '06

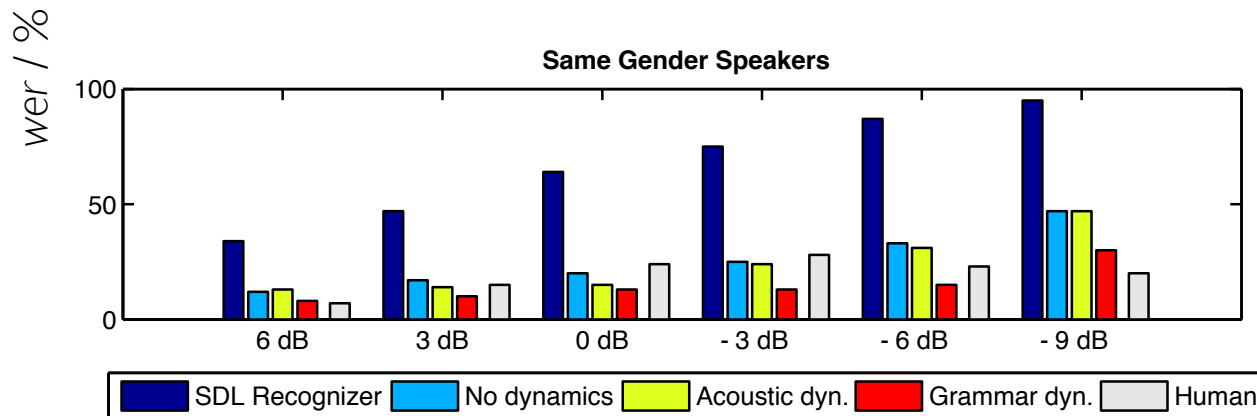
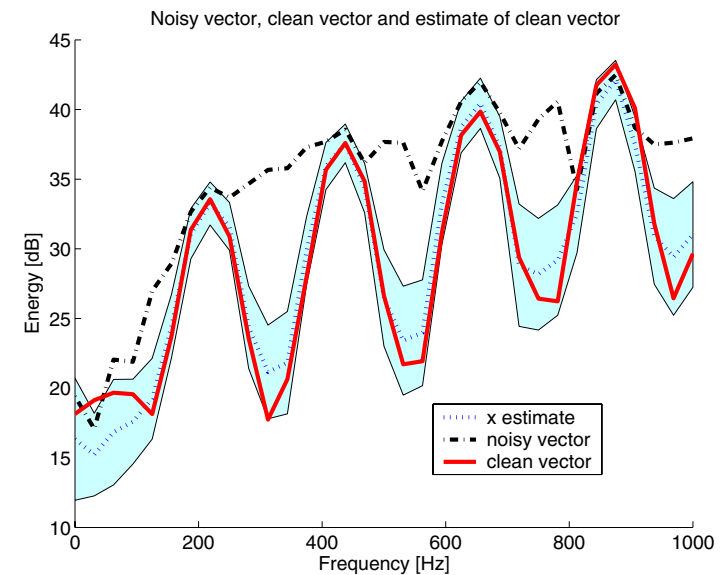
- IBM's 2006 **Iroquois** speech separation system

Key features:

- detailed state combinations
- large speech recognizer
- exploits grammar constraints
- 34 **per-speaker models**

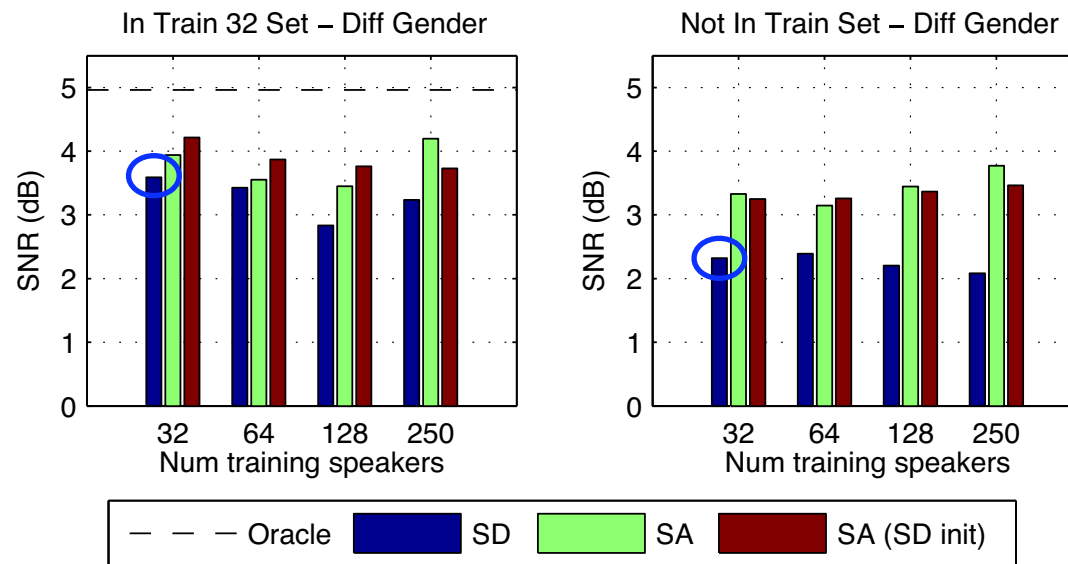
- **“Superhuman”** performance

- ... in some conditions



Adapting Source Models

- **Power** of model-based separation depends on **detail of model**
- Speech separation relies on **prior knowledge** of every speaker?



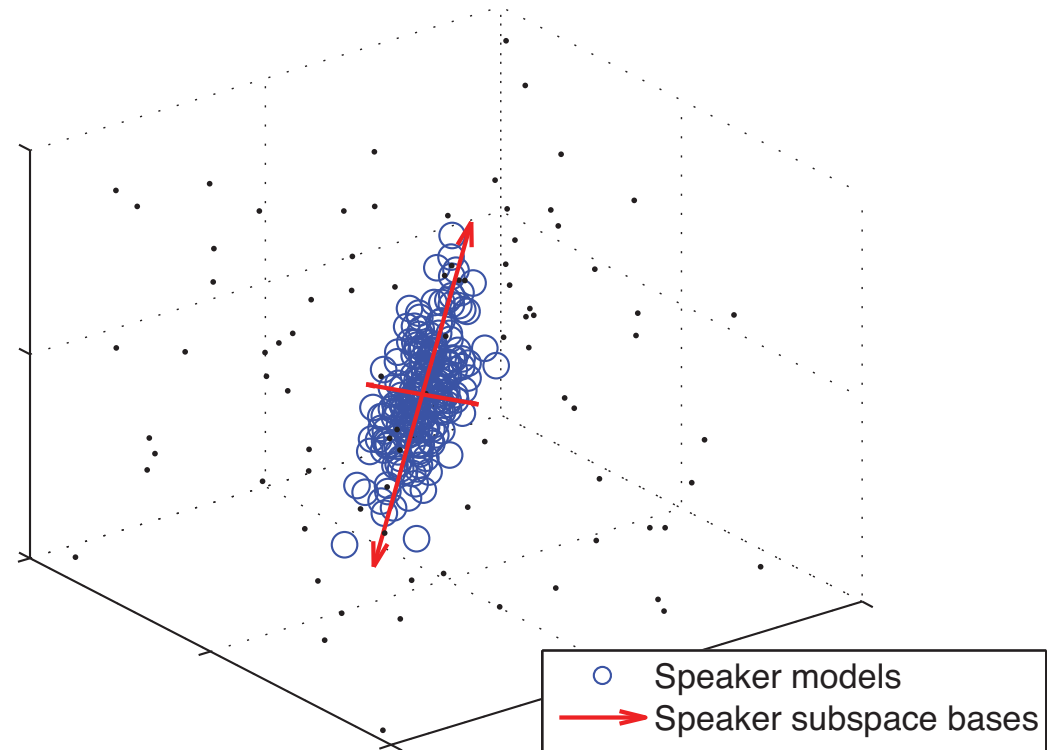
- Can this be **practical**?

Eigenvoices

Kuhn et al. '98, '00
Weiss & Ellis '07, '08, '09

- Idea: Find
model parameter space

- generalize without
losing detail?



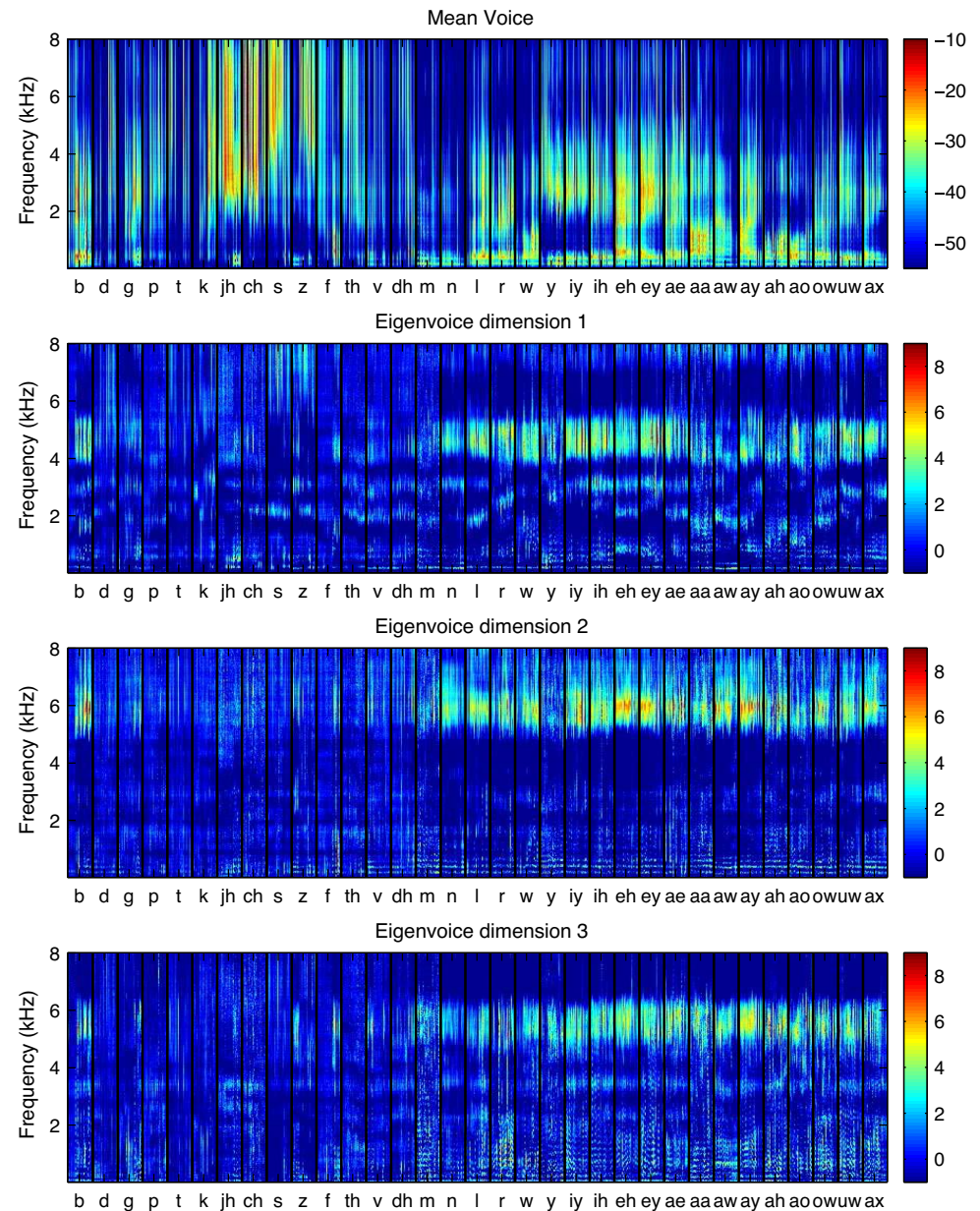
- Eigenvoice model:

$$\mu = \bar{\mu} + U \mathbf{w} + B \mathbf{h}$$

adapted model	mean voice	eigenvoice bases	weights	channel bases	channel weights
------------------	---------------	---------------------	---------	------------------	--------------------

Eigenvoice Bases

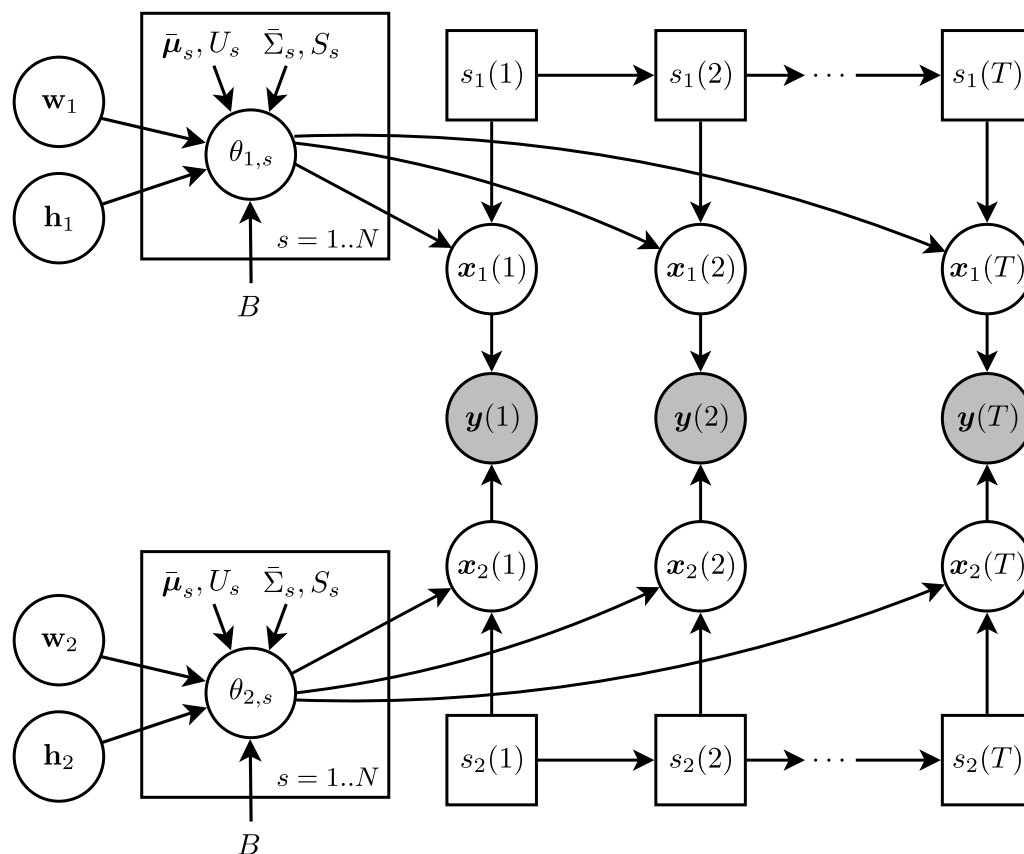
- Mean model
 - 280 states x 320 bins = 89,600 dimensions
- Eigencomponents shift formants/ coloration
 - additional components for channel



Speaker-Adapted Separation

Weiss & Ellis '08

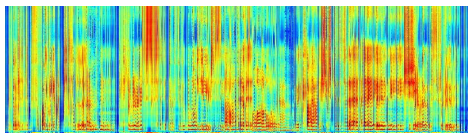
- Factorial HMM analysis
with tuning of source model parameters
= **eigenvoice speaker adaptation**



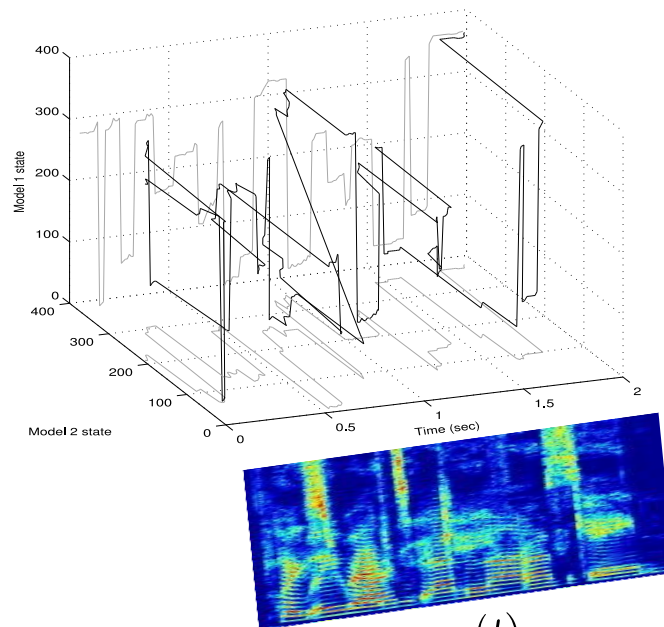
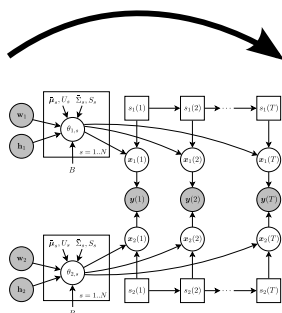
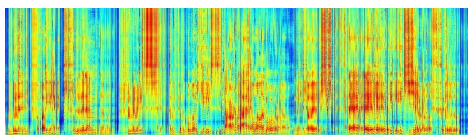
Speaker-Adapted Separation

Find Viterbi path

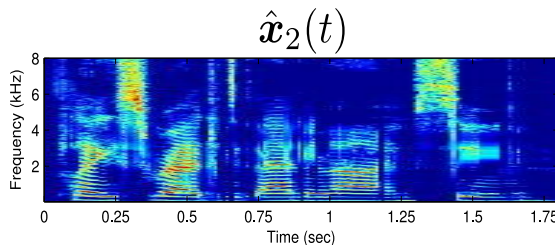
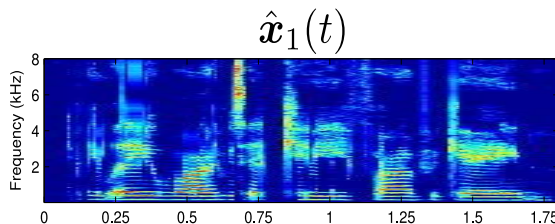
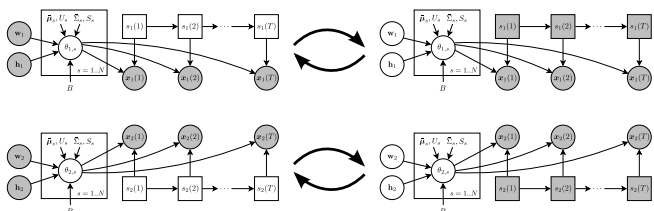
$$\mu_1 = U\mathbf{w}_1 + \bar{\mu}$$



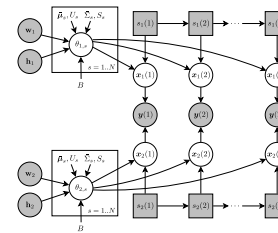
$$\mu_2 = U\mathbf{w}_2 + \bar{\mu}$$



Update model parameters using EM algorithm from Kuhn et al., (2000)

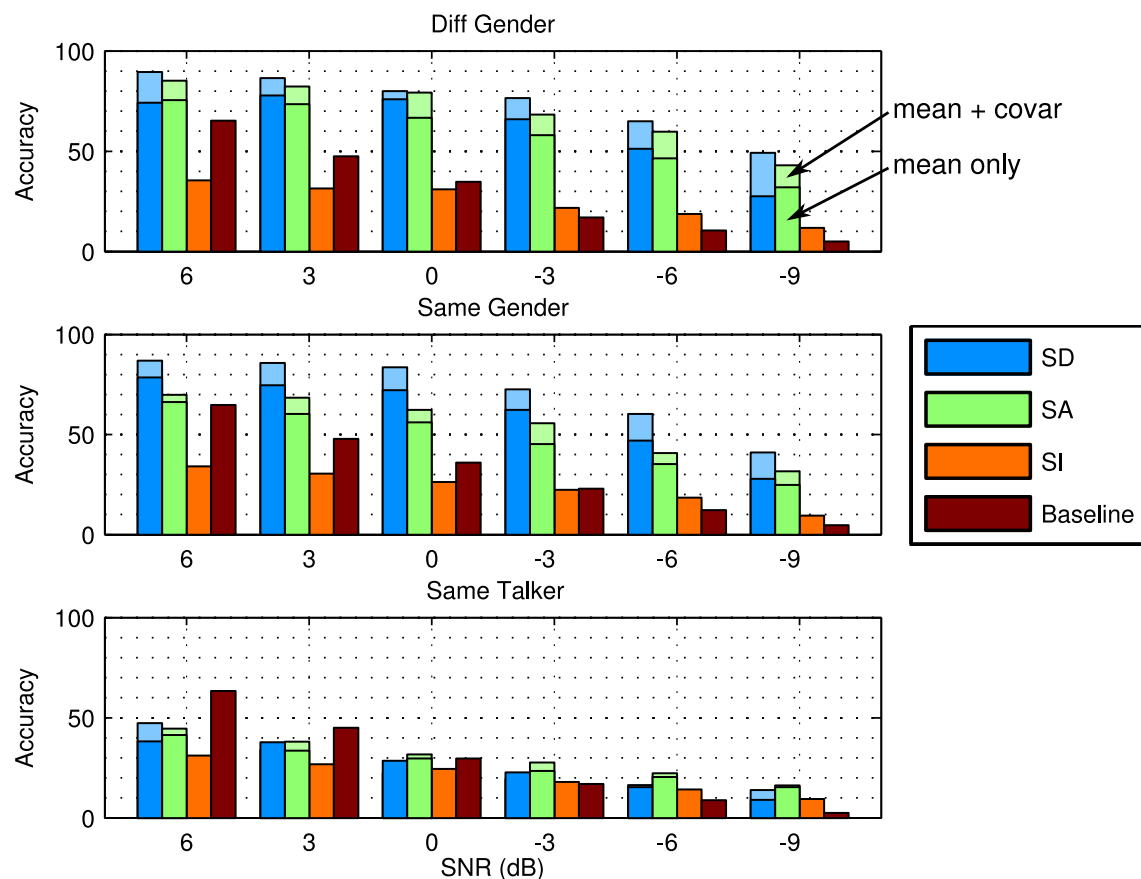


Estimate source signals



Speaker-Adapted Separation

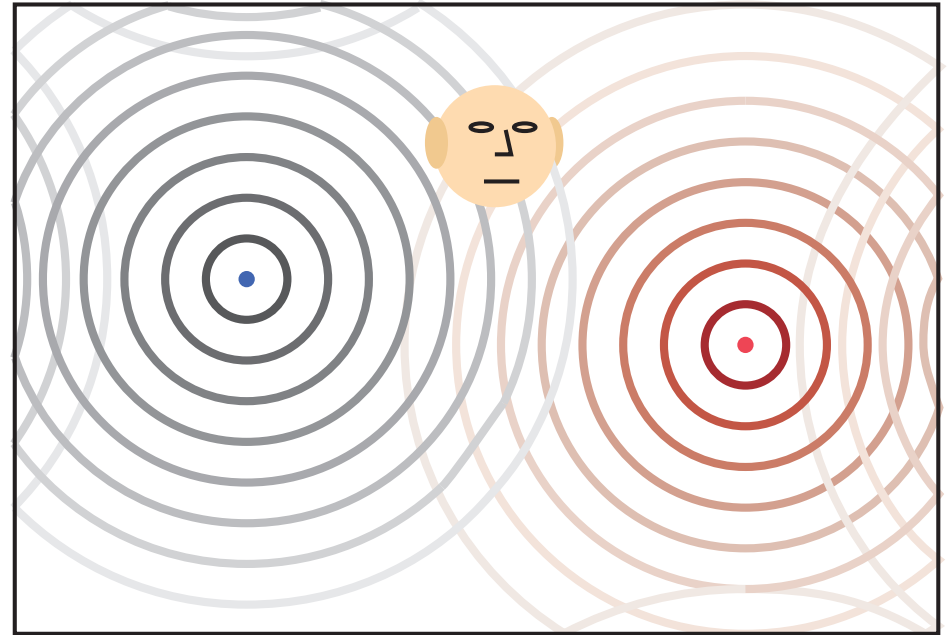
- Eigenvoices for Speech Separation task
 - speaker adapted (SA) performs midway between speaker-dependent (SD) & speaker-indep (SI)



3. Spatial Models & Reverb

Mandel & Ellis '07

- 2 or 3 sources in reverberation
 - assume just 2 'ears'

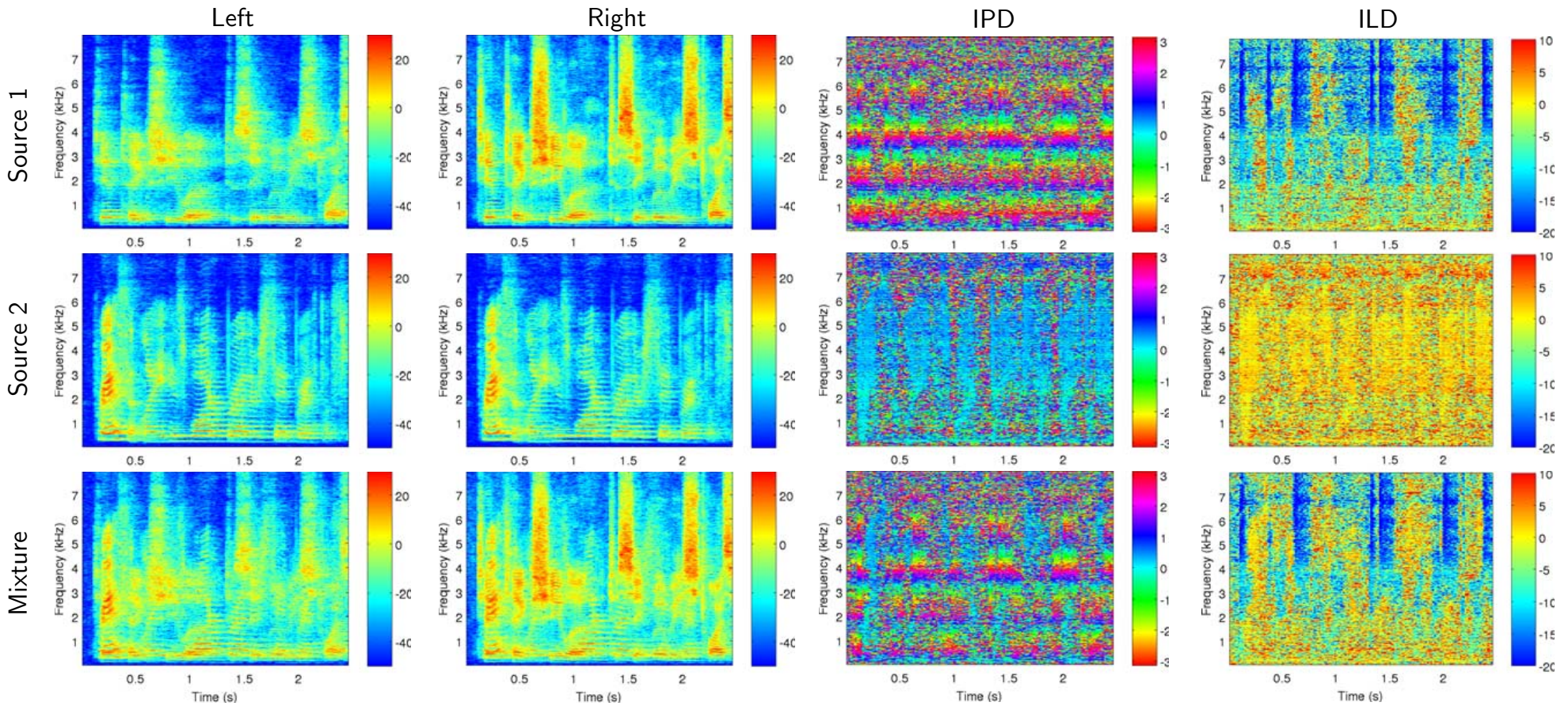


- Model interaural spectrum of each source as stationary level and time differences:

$$\frac{L(\omega, t)}{R(\omega, t)} = a(\omega) e^{j\omega\tau} N(\omega, t)$$

ILD and IPD

- Sources at 0° and 75° in reverb

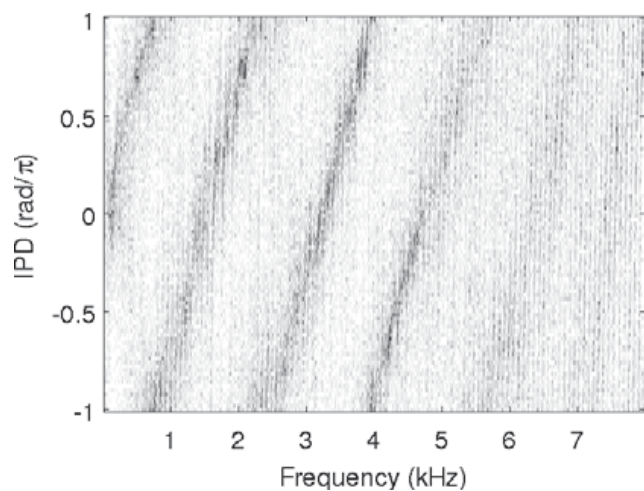


IPD, ILD Distributions

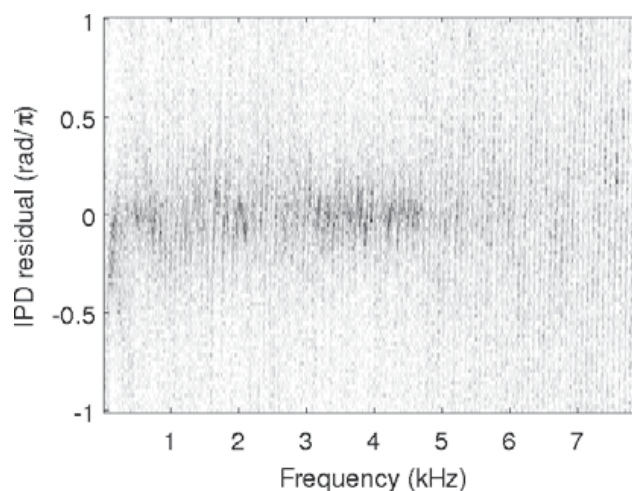
- Source at 75° in reverberation



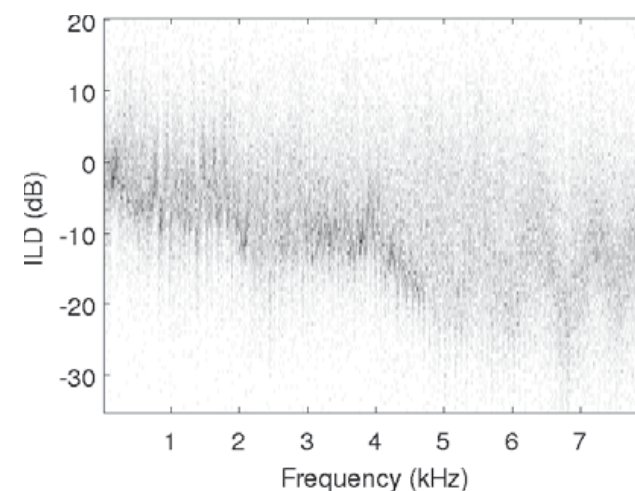
IPD



IPD residual



ILD



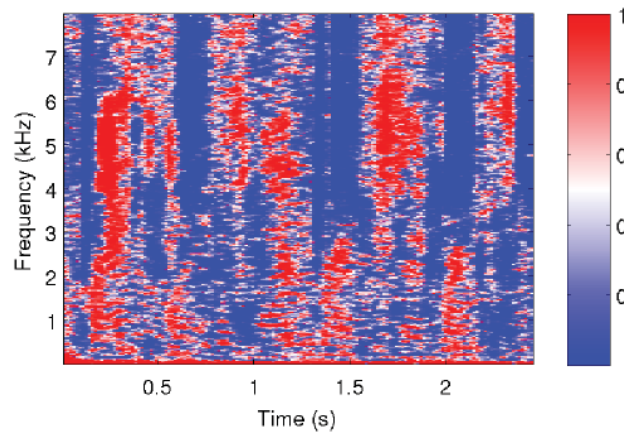
- IPD residual offsets phase by constant ωT
- IPD can be fit by single Gaussian
- ILD needs frequency-dependence

Model-Based EM Source Separation and Localization (MESSL)

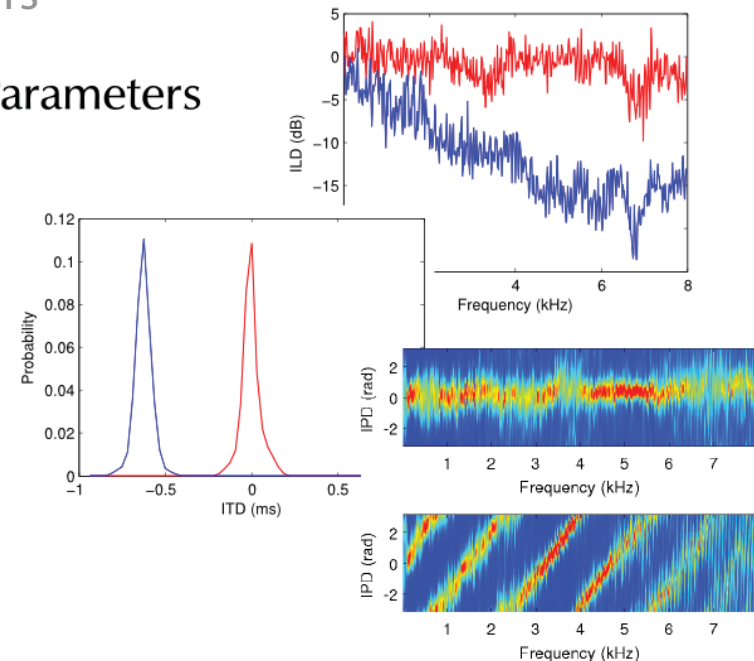
Mandel & Ellis '09

Re-estimate
source parameters

Masks



Parameters

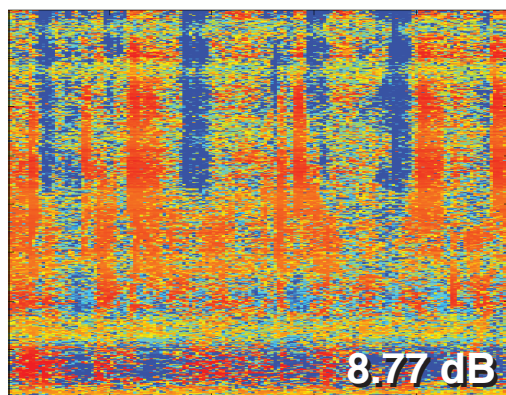


Assign spectrogram points
to sources

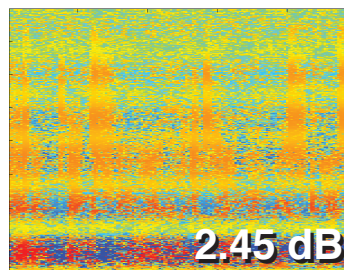
- can model more sources than sensors
- flexible initialization

MESSL Results

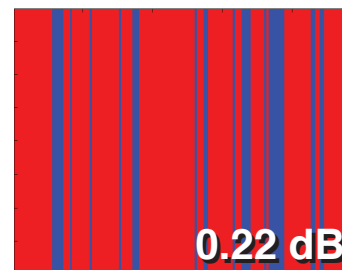
- **Modeling uncertainty** improves results
 - tradeoff between constraints & **noisiness**



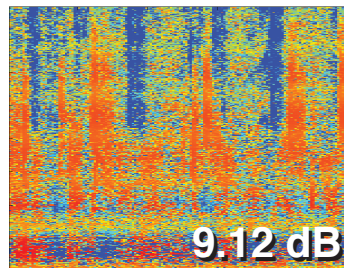
EM+ILD



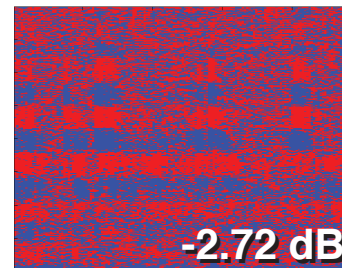
EM-ILD (only IPD)



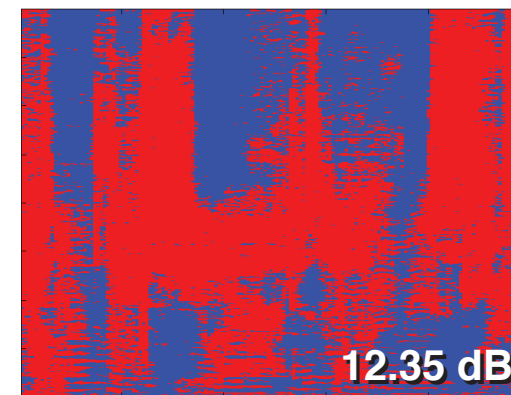
PHAT-histogram



EM+1ILD (tied means)



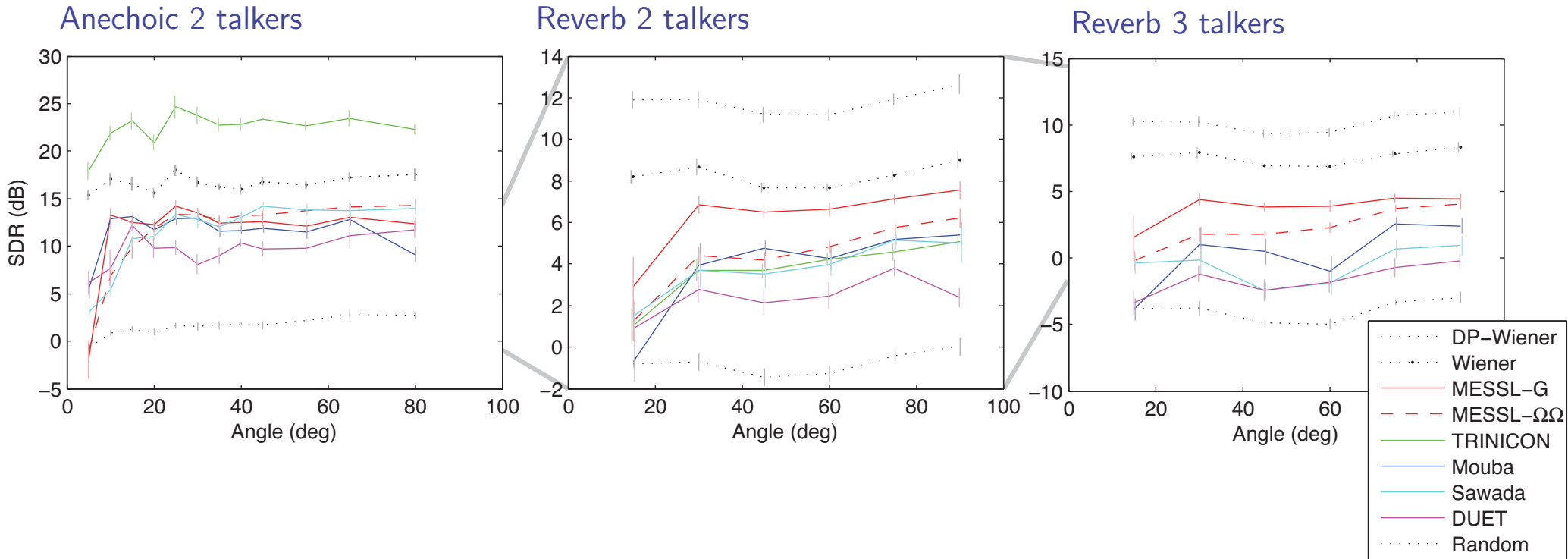
DUET



Ground Truth

MESSL Results

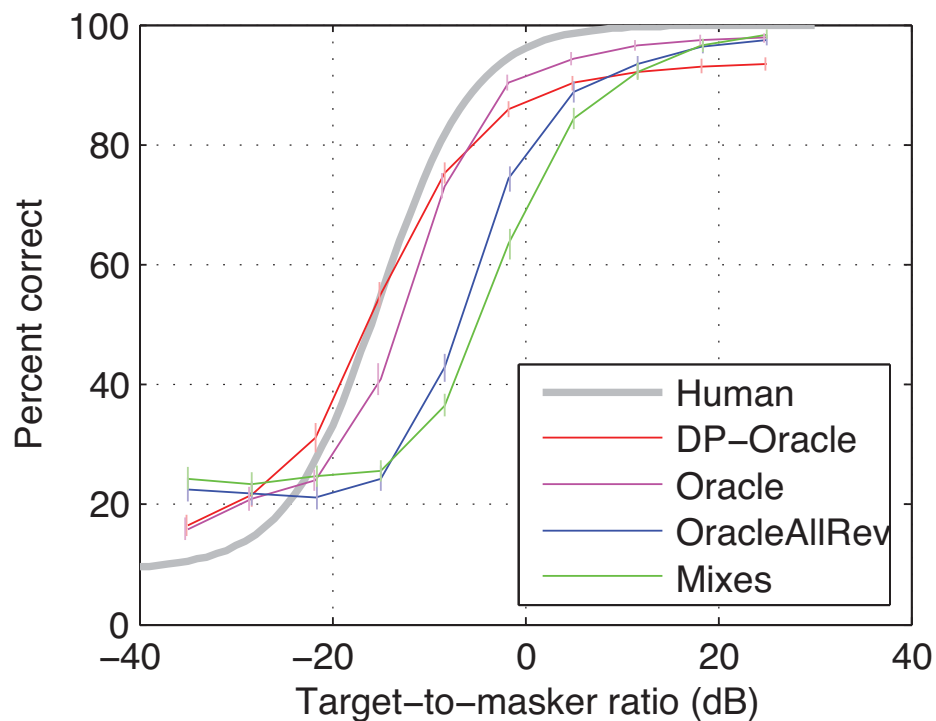
- Signal-to-Distortion Ratio (SDR)



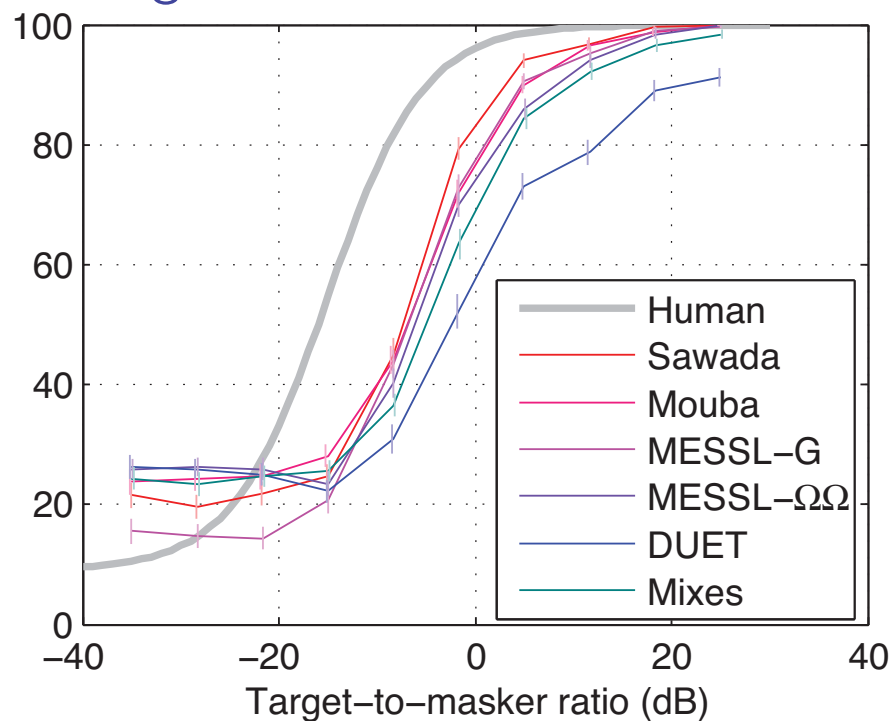
MESSL Results

- Speech recognizer (Digits)

Ground truth masks



Algorithmic masks



4. Combining Spatial + Speech Models

Weiss, Mandel & Ellis '08

- **Interaural** parameters give
 $ILD_i(\omega), ITD_i, \Pr(X(t, \omega) = S_i(t, \omega))$
- **Speech source model** can give
 $\Pr(S_i(t, \omega) \text{ is speech signal})$
- Can combine into one big **EM framework**...

E-step

$$p(u|\Theta^{(n)}) = p(x, u|\Theta^{(n)})/p(x|\Theta^{(n)})$$



M-step

$$\Theta^{(n+1)} = \operatorname{argmax}_{\Theta} E_{p(u|\Theta^{(n)})} p(x, u|\Theta)$$

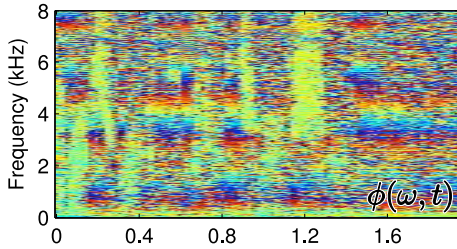
u is: $\Pr(\text{cell from source } i)$
phoneme sequence

Θ is: interaural params
speaker params

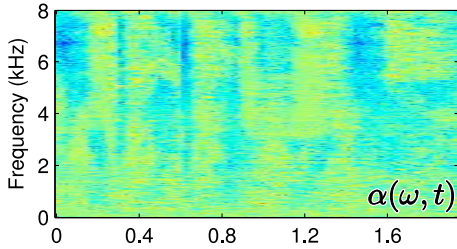
MESSL-SP (Source Prior)

Observations

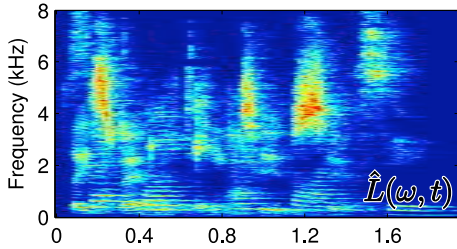
Mixture – IPD



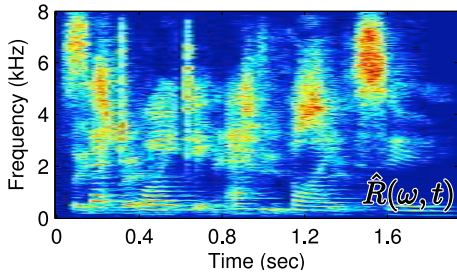
Mixture – ILD



Mixture – left channel

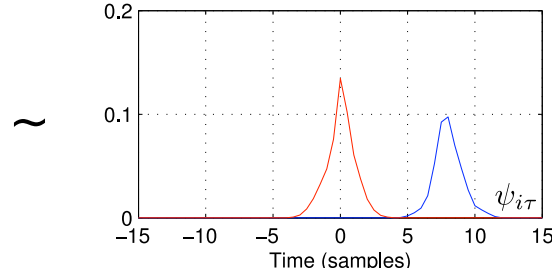


Mixture – right channel

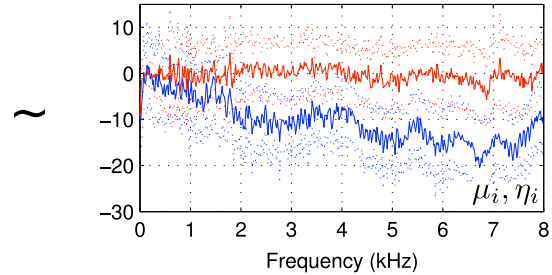


Parameters

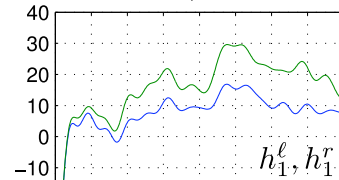
Per-source ITD



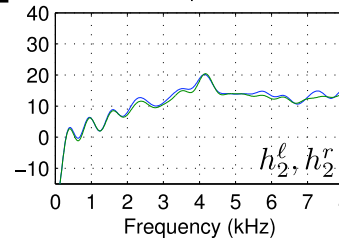
Per-source ILD



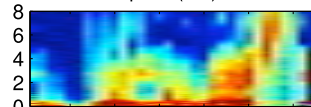
SP channel response – source 1



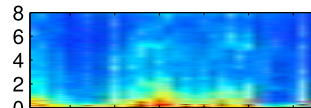
SP channel response – source 2



Source prior (SP) means



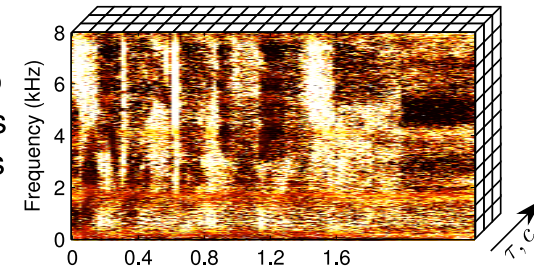
SP covars



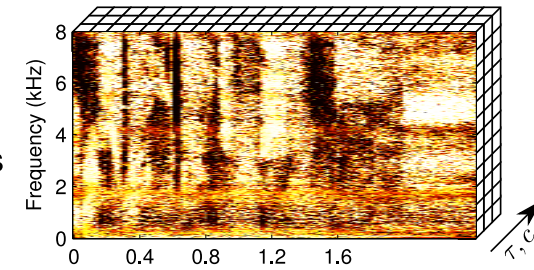
Posteriors

Each point in spectrogram is explained by a source, delay, and mixture component

Source 1 mask

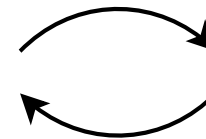


Source 2 mask



E-step

Use parameters to compute posteriors of hidden variables



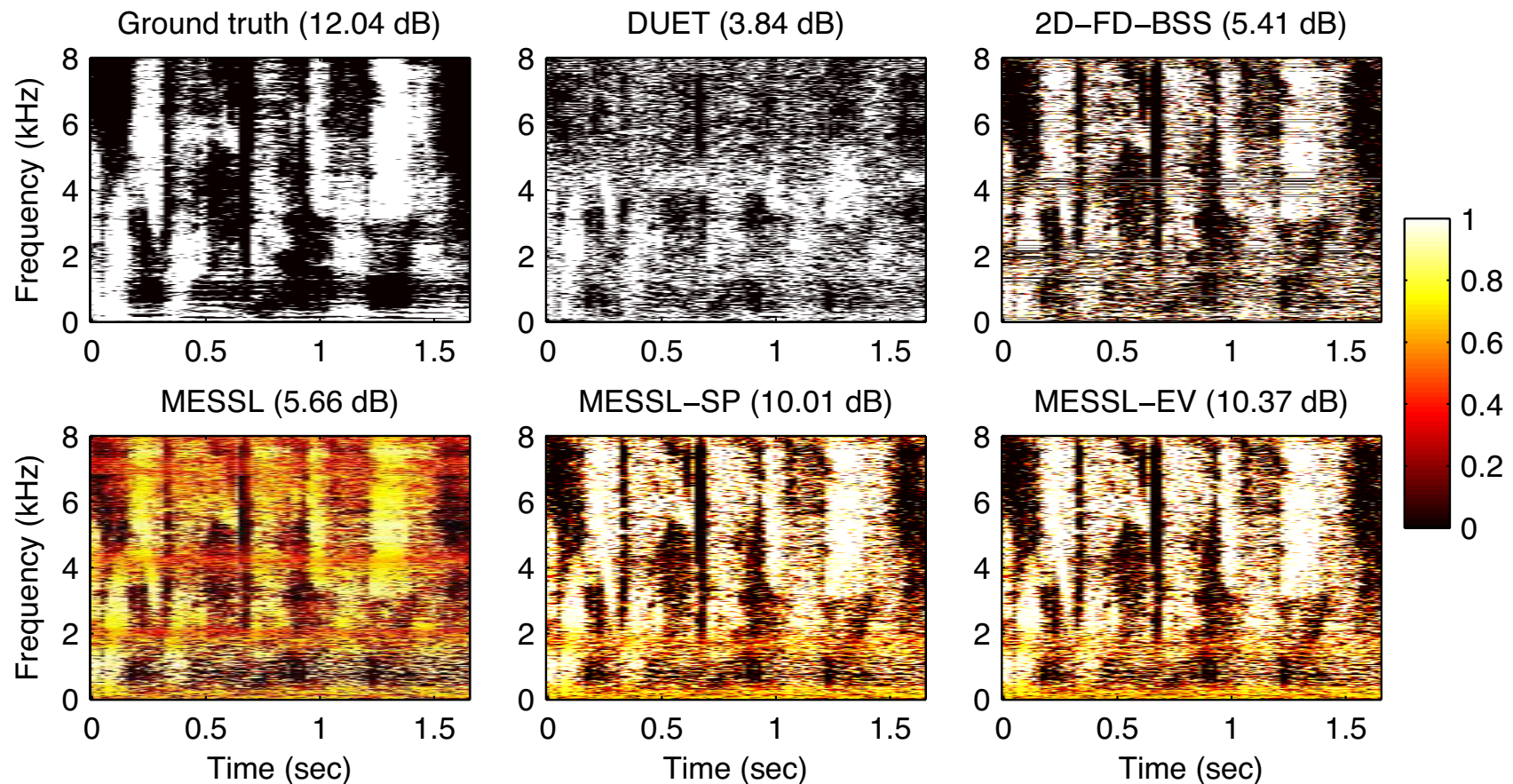
M-step

Use posteriors to update parameters

Separate sources by multiplying mixture by different masks

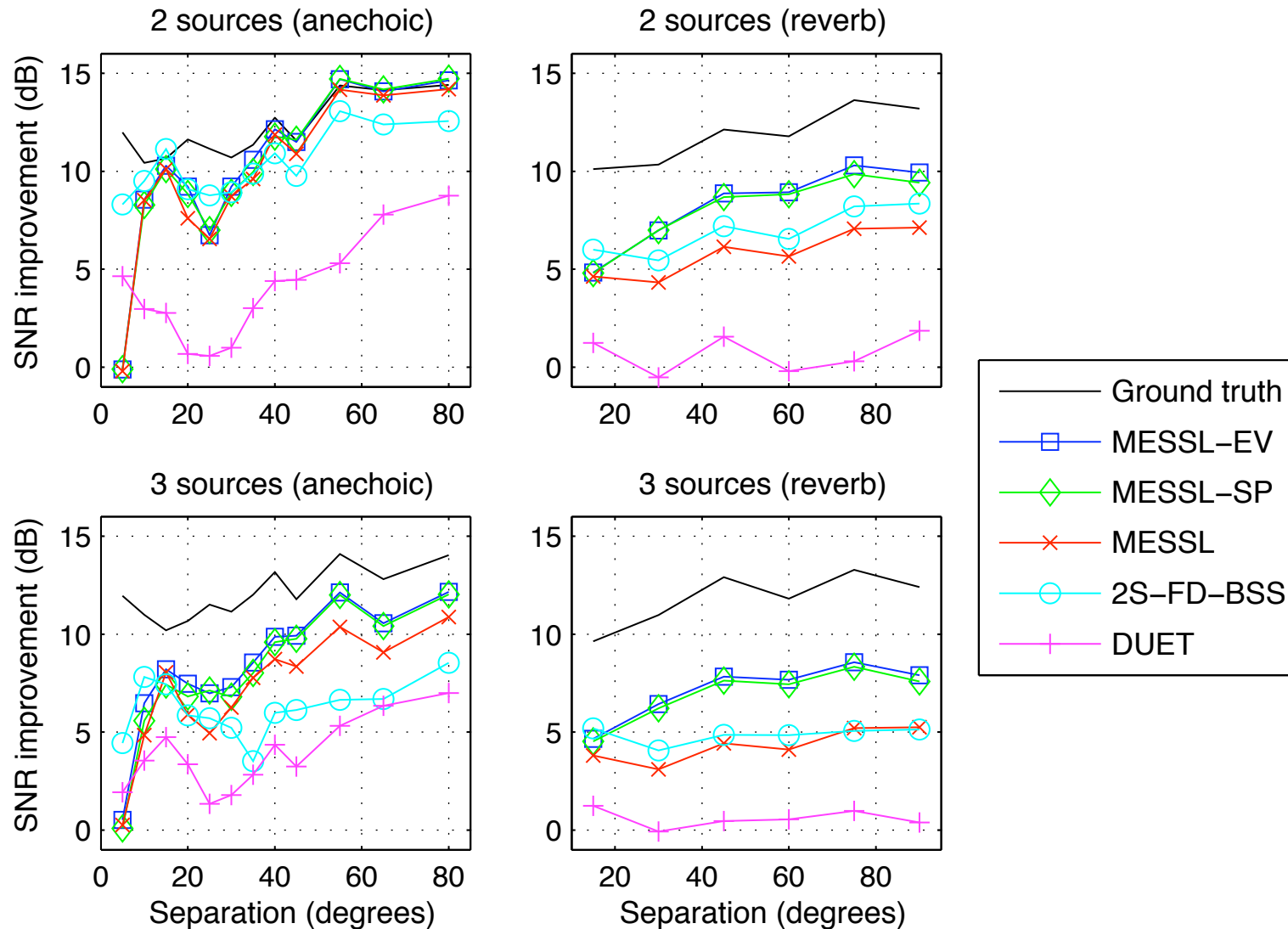
MESSL-SP Results

- Source models function as **priors**
- **Interaural** parameter spatial separation
 - source model prior **improves** spatial estimate



MESSL-SP Results

- SNR improvement vs. source angle separation



Future Work

- **Better parametric speaker models**
 - limitations of eigenvoices
 - varying style
- **Understanding reverb & ASR**
 - early echoes
 - what spoils ASR?
- **Models of other sources**
 - eigeninstruments?

Summary & Conclusions

- **Source models** provide the constraints to make **scene analysis** possible
- **Eigenvoices** (model subspace) can be used to provide detailed models that generalize
- Spatial parameters can identify more sources than models in reverb (**MESSL**)
- Can **combine** source + spatial models