

Some projects in Real-World Sound Analysis

Dan Ellis

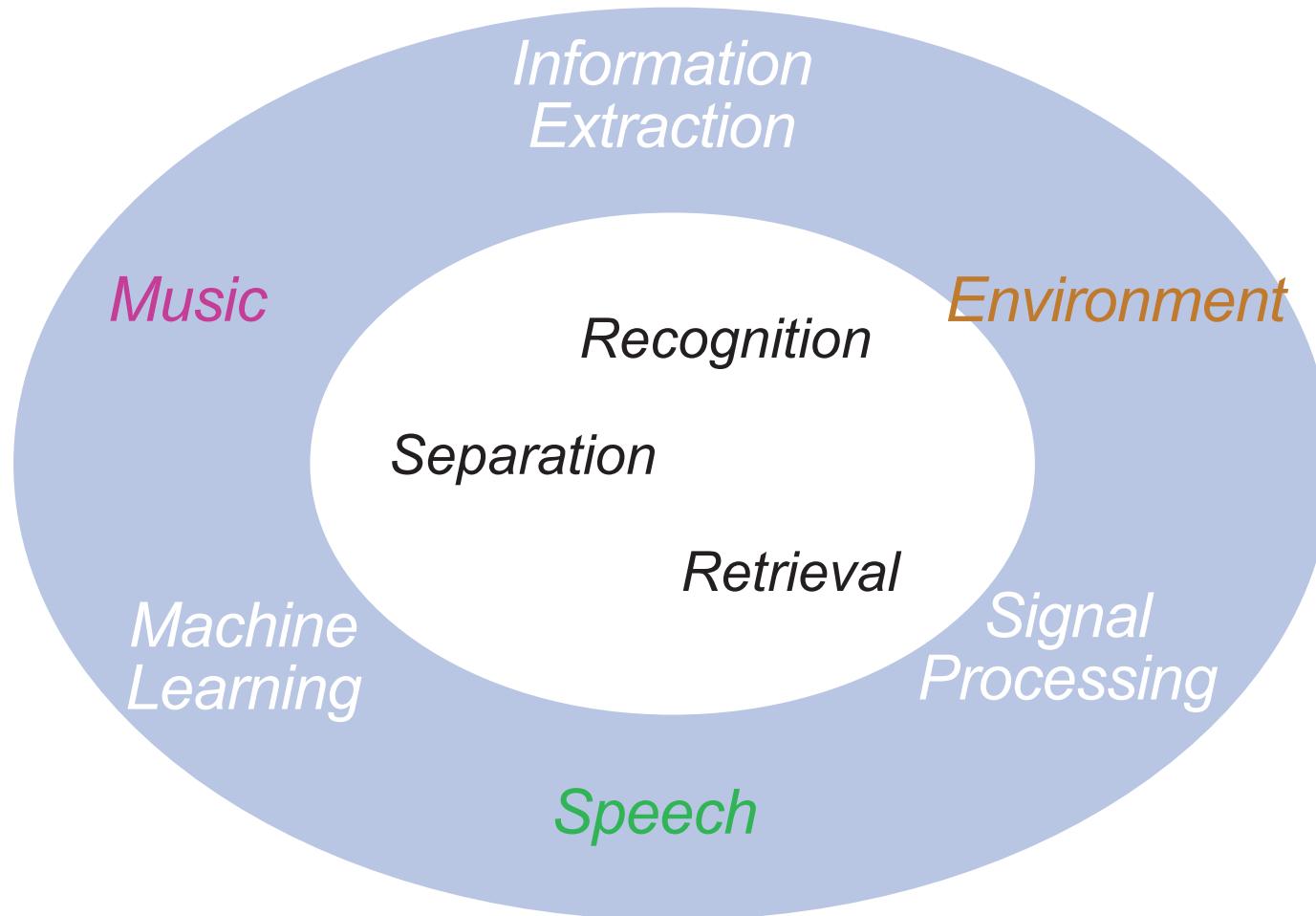
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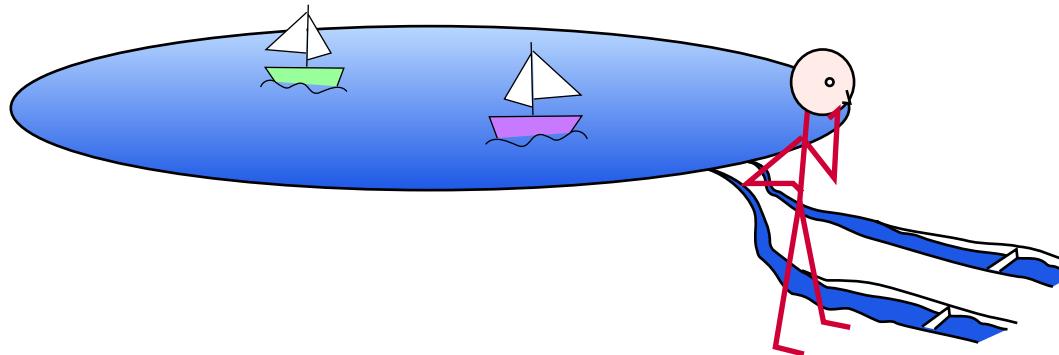
<http://labrosa.ee.columbia.edu/>

1. Real-World Sound
2. Speech Separation
3. Soundtrack Classification
4. Music Audio Analysis

LabROSA Overview



I. Real-World Sound



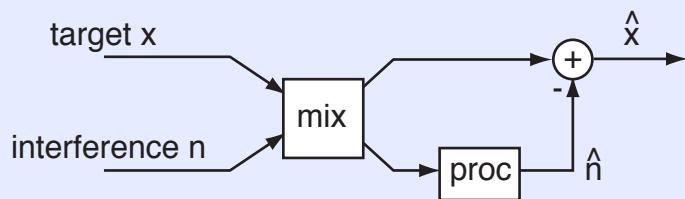
“Imagine two narrow channels dug up from the edge of a lake, with handkerchiefs stretched across each one. Looking only at the motion of the handkerchiefs, you are to answer questions such as: How many boats are there on the lake and where are they?” (after Bregman’90)

- Received waveform is a mixture
 - 2 sensors, N sources - underconstrained
- Use prior knowledge (**models**) to constrain

Approaches to Separation

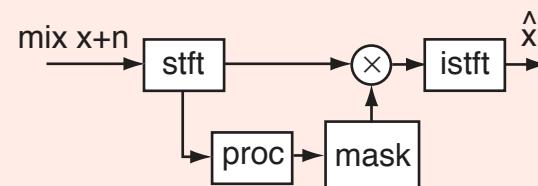
ICA

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



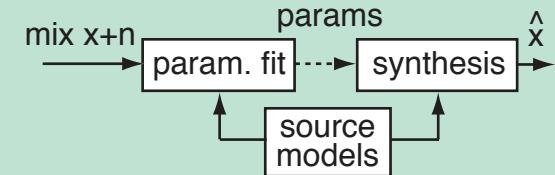
CASA

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



Model-based

- Any domain
- Param. search
- Synthetic output?



2. Speech Separation

Roweis '01, '03

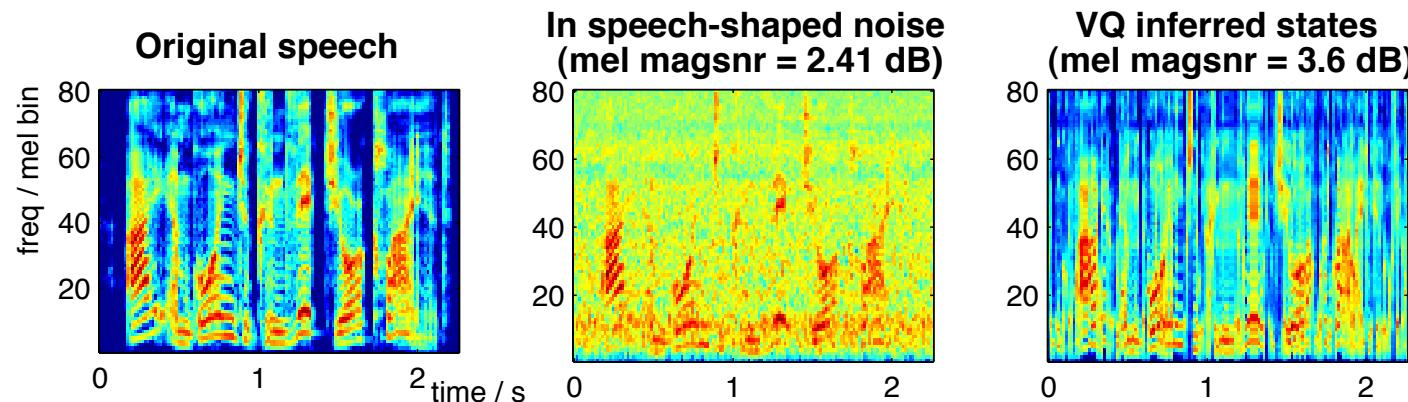
Kristjannson '04, '06

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

$$p(\mathbf{x}|i_1, i_2) = \mathcal{N}(\mathbf{x}; \mathbf{c}_{i1} + \mathbf{c}_{i2}, \Sigma) \quad \begin{matrix} \text{combination} \\ \text{model} \end{matrix}$$

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

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

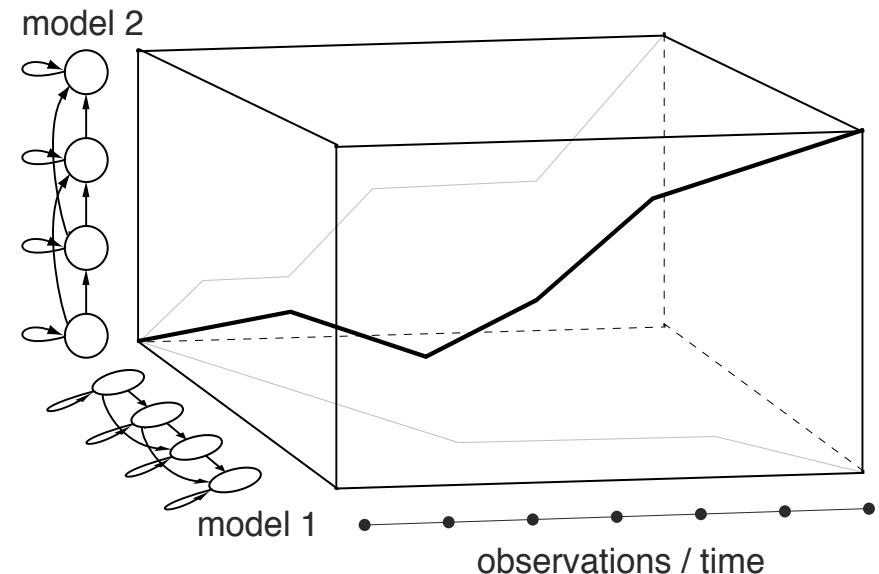
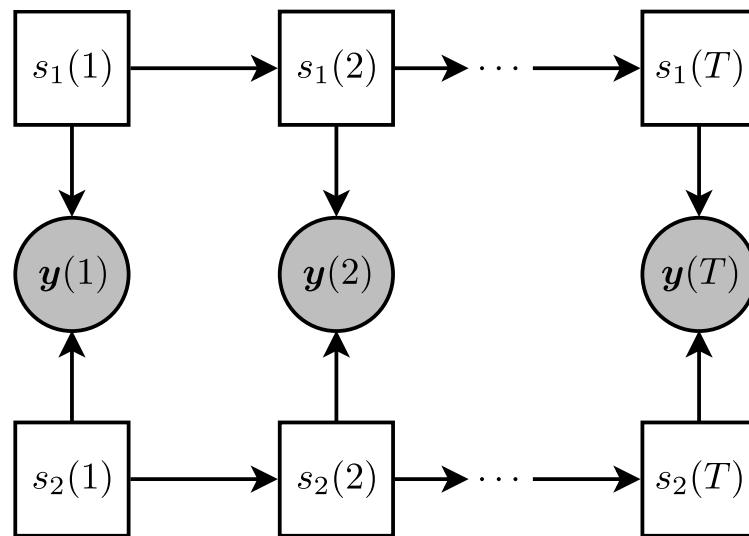


Speech Mixture Recognition

Varga & Moore '90

Kristjansson, Hershey et al. '06

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



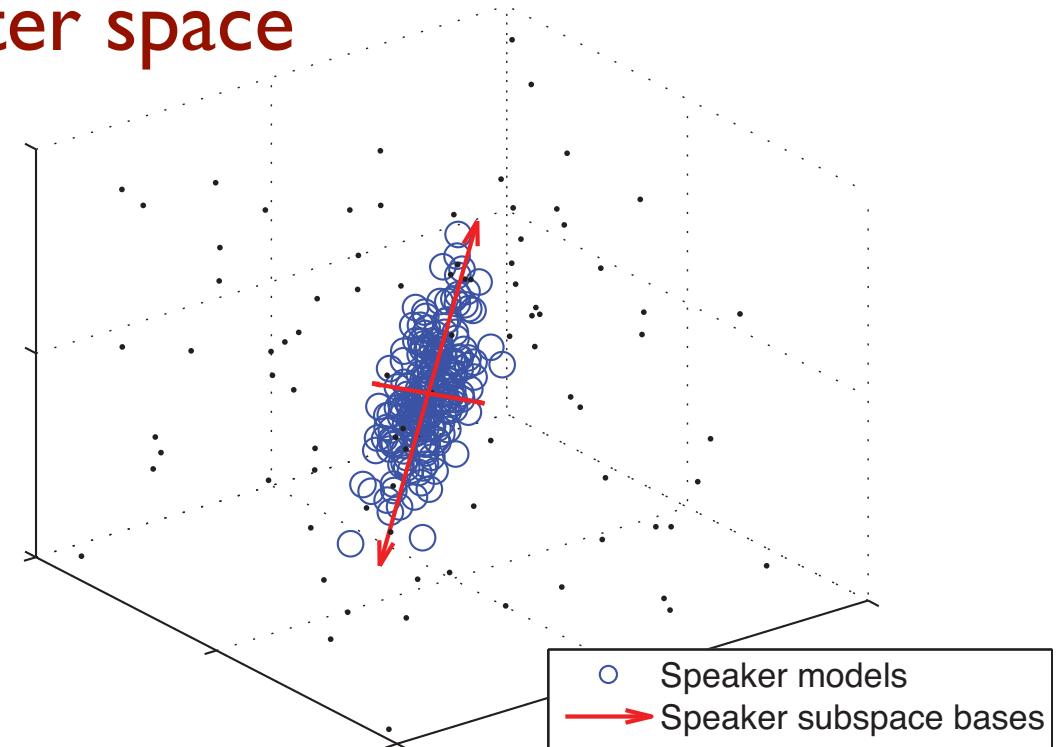
- Speaker-specific models...

Eigenvoices

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

- Idea: Find speaker model parameter space

- generalize without losing detail?



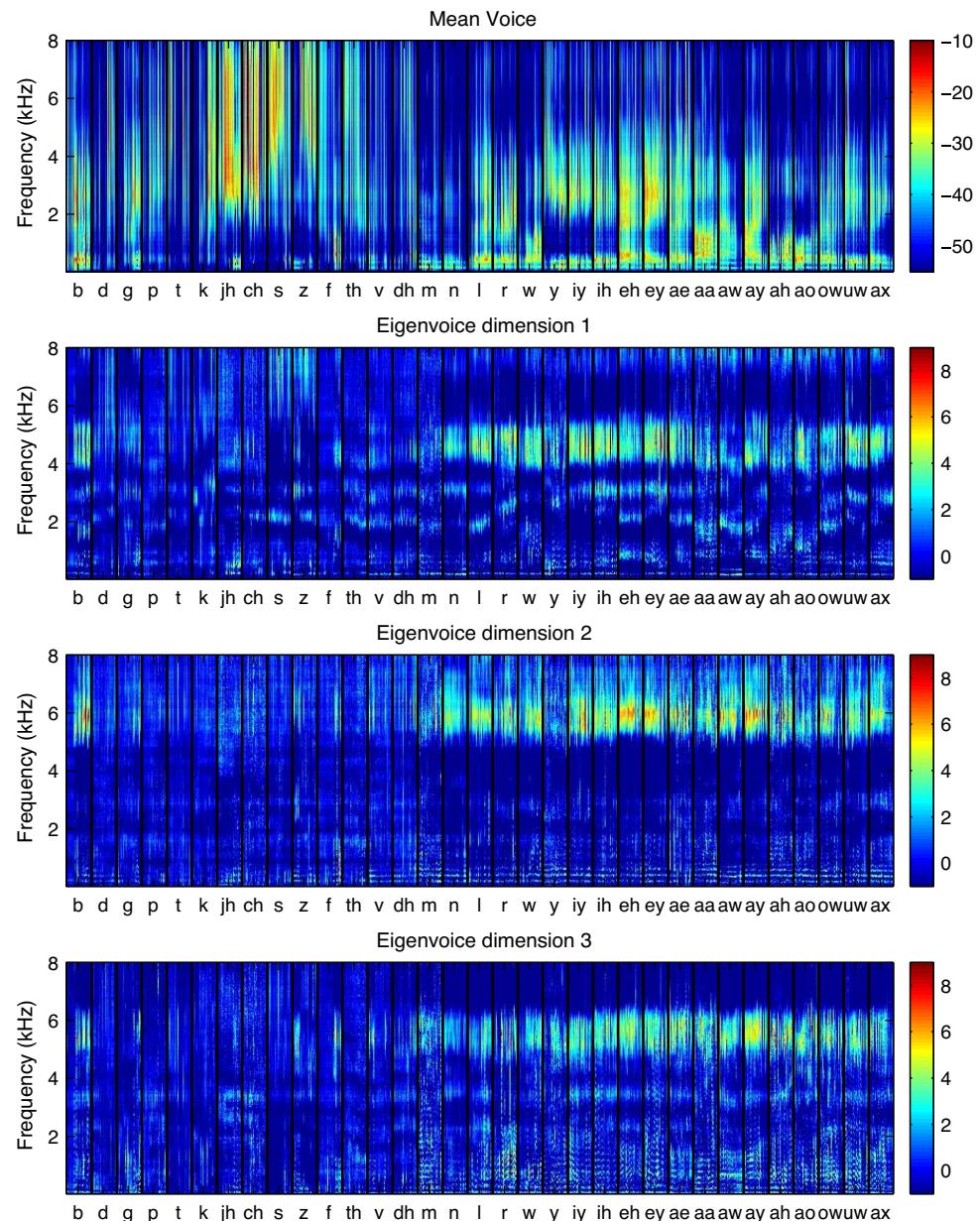
- Eigenvoice model:

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

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

Eigenvoice Bases

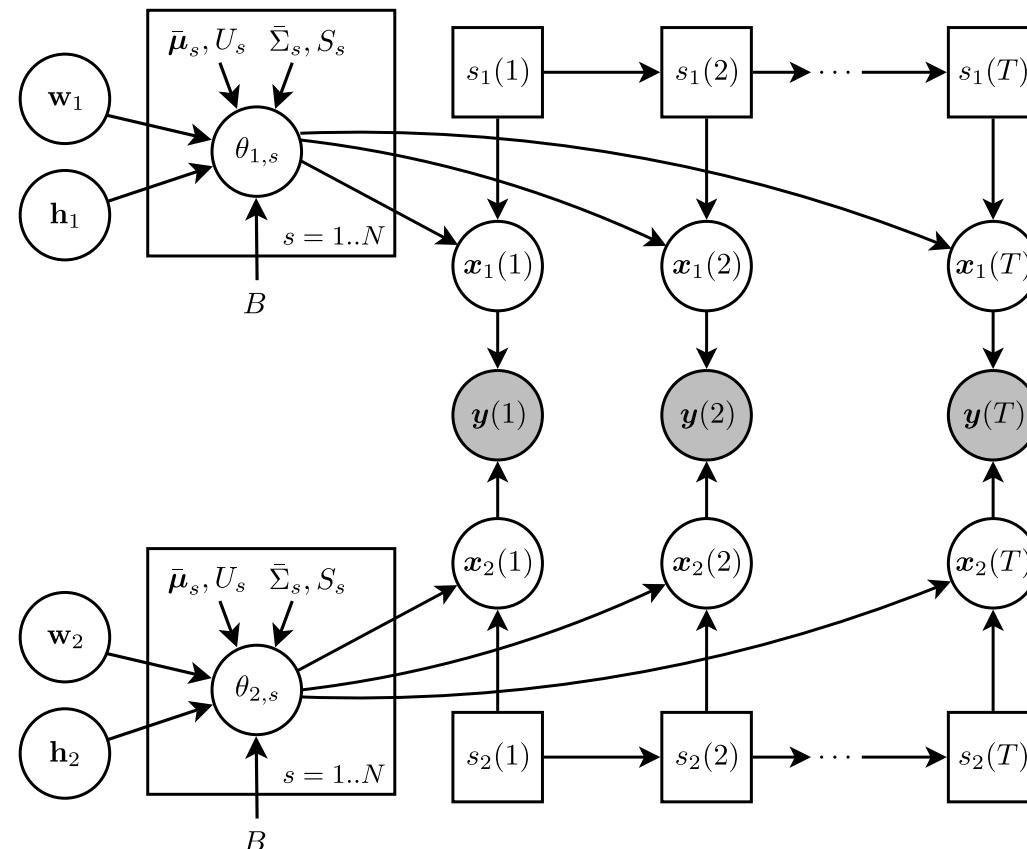
- Mean model
 - 280 states × 320 bins
= 89,600 dimensions
- Eigencomponents shift formants/
coloration
 - additional components for channel



Speaker-Adapted Separation

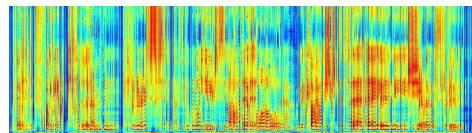
Weiss & Ellis '08

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

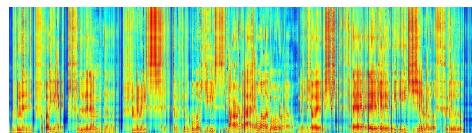


Speaker-Adapted Separation

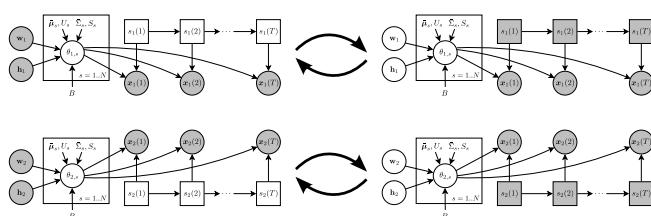
$$\mu_1 = U\mathbf{w}_1 + \bar{\mu}$$



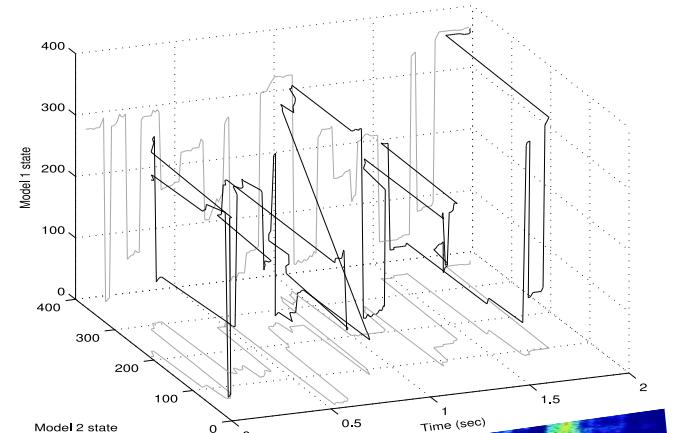
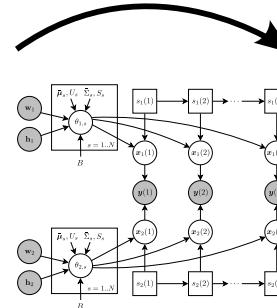
$$\mu_2 = U\mathbf{w}_2 + \bar{\mu}$$



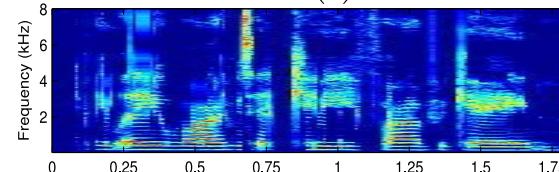
Update model parameters using EM algorithm from Kuhn et al., (2000)



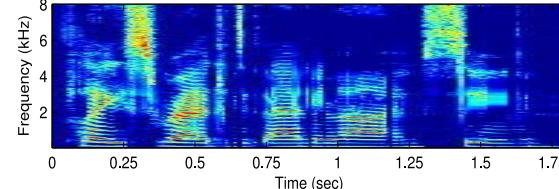
Find Viterbi path



$$\hat{x}_1(t)$$

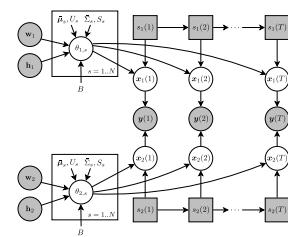


$$\hat{x}_2(t)$$



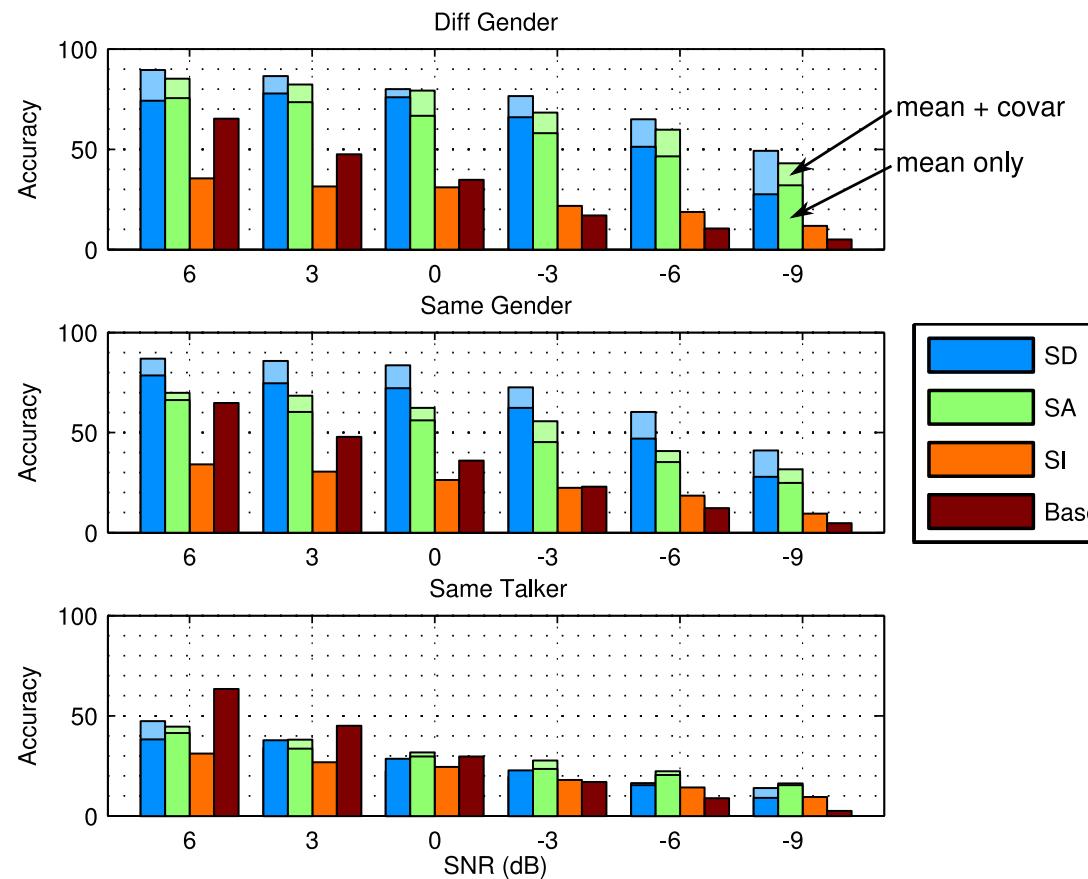
$$y(t)$$

Estimate source signals



Speaker-Adapted Separation

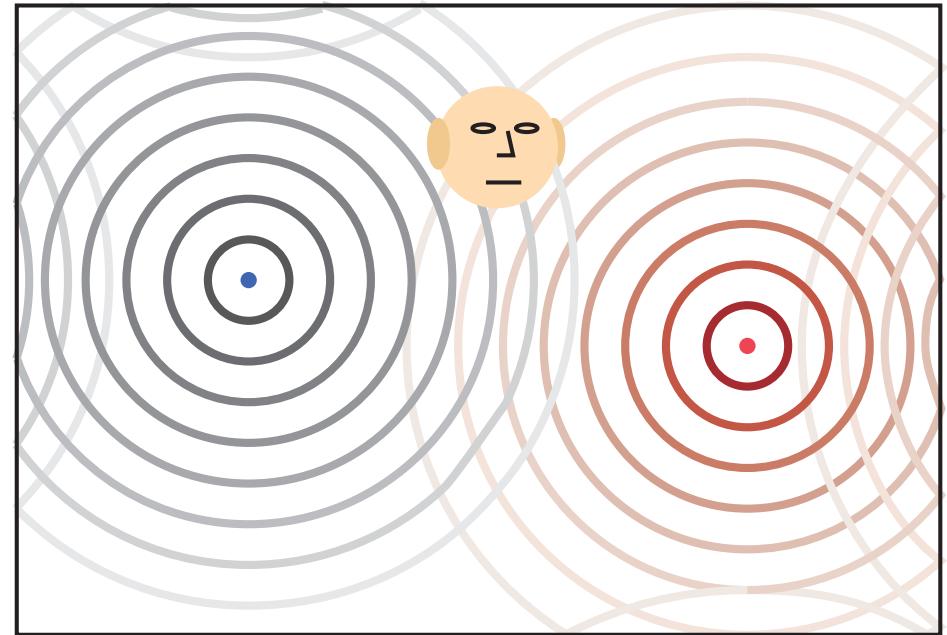
- **Eigenvoices for Speech Separation task**
 - speaker adapted (SA) performs midway between speaker-dependent (SD) & speaker-indep (SI)



Spatial Separation

Mandel & Ellis '07

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



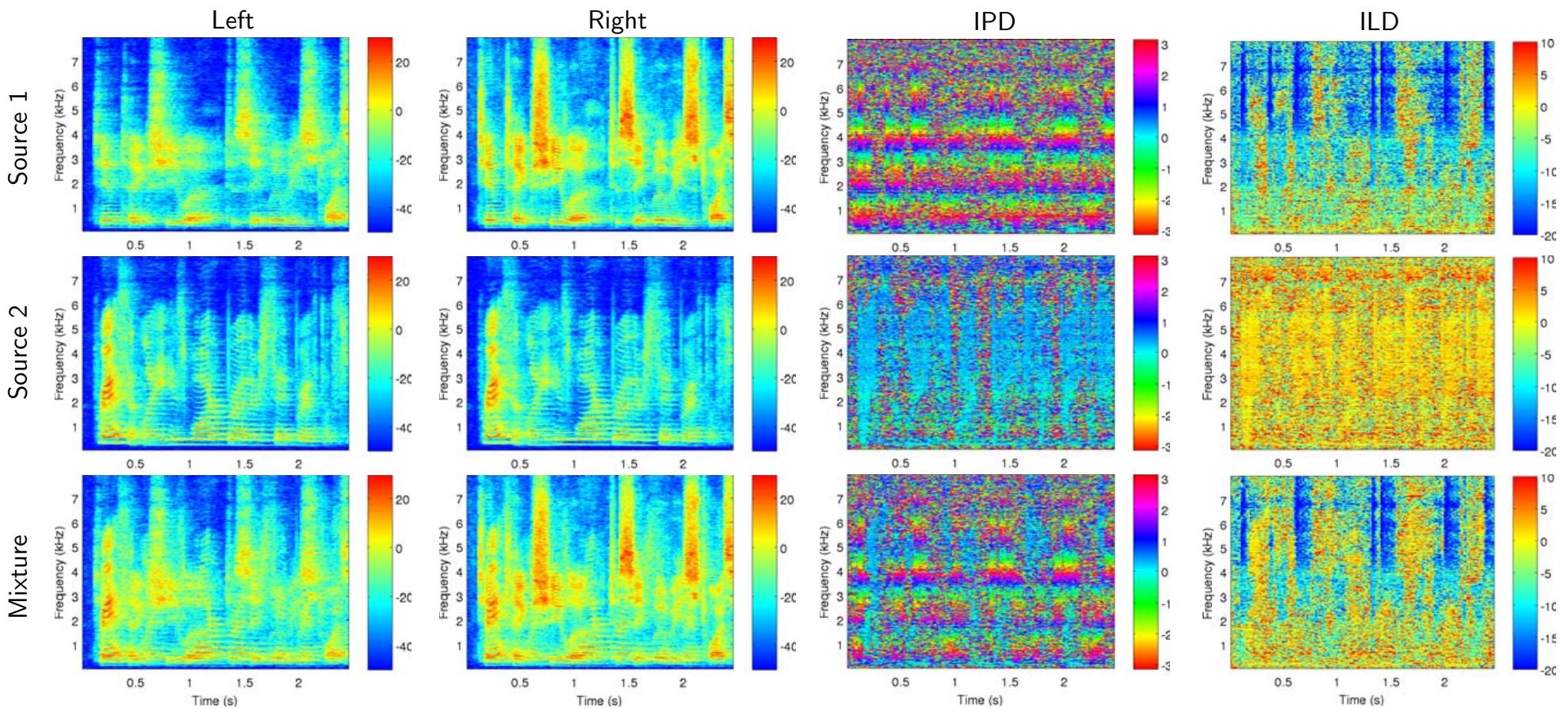
- Model interaural spectrum of each source as stationary level and time differences:

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

ILD and IPD

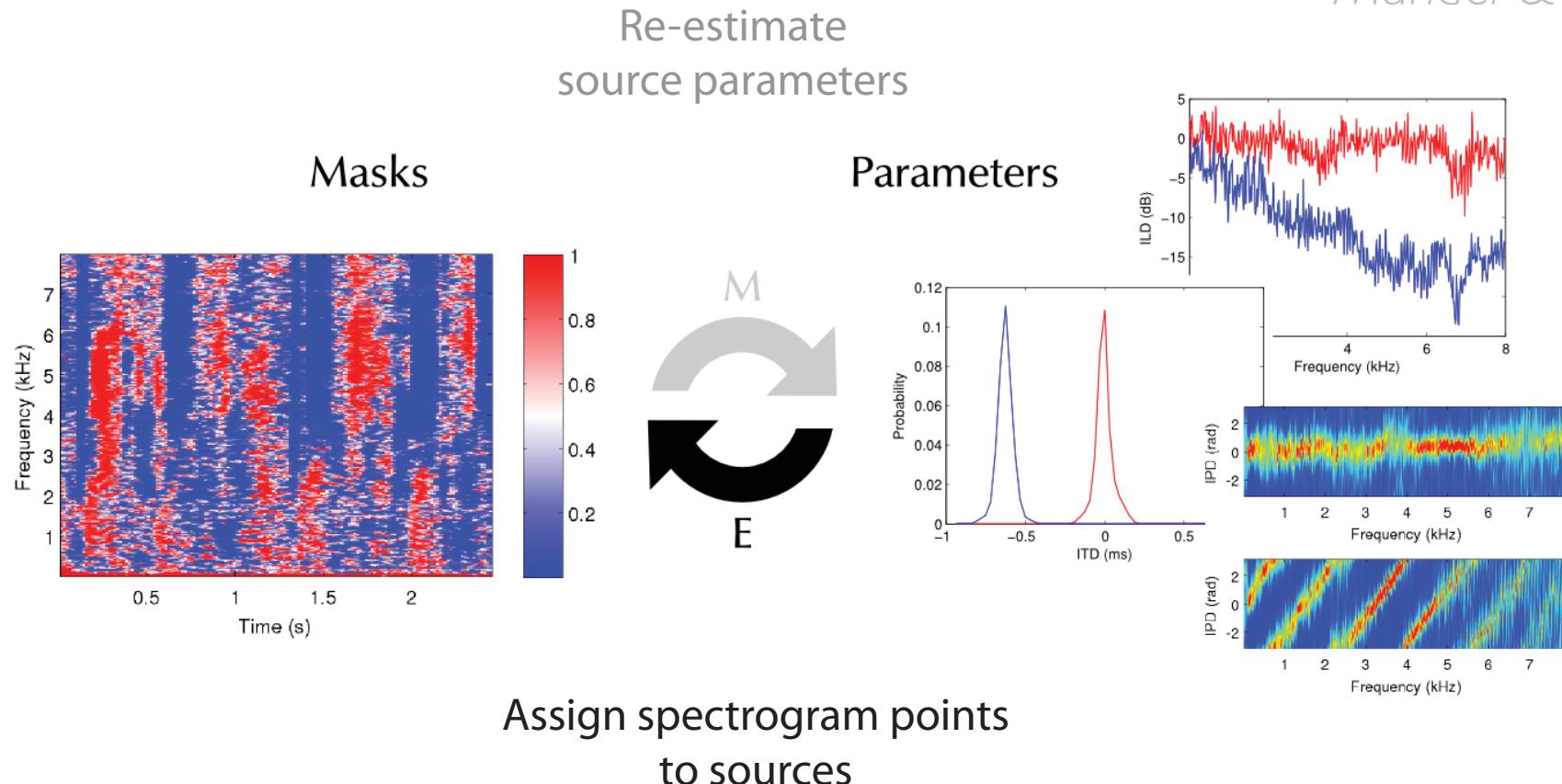


- Sources at 0° and 75° in reverb



Model-Based EM Source Separation and Localization (MESSL)

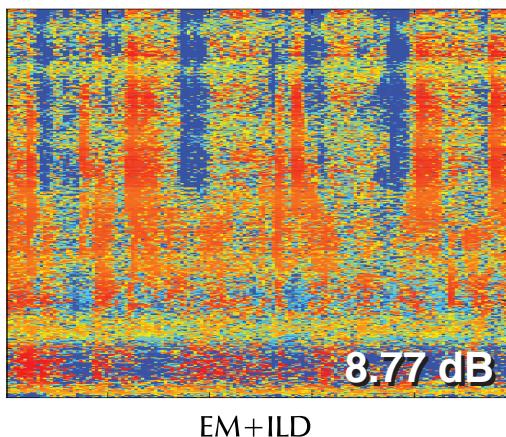
Mandel & Ellis '09



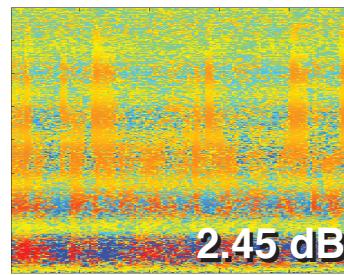
- can model more sources than sensors
- flexible initialization

MESSL Results

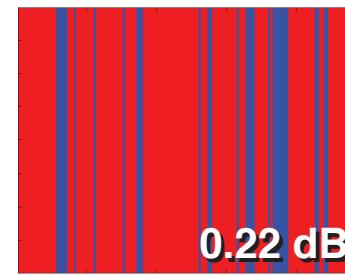
- Modeling uncertainty improves results
 - tradeoff between constraints & noisiness



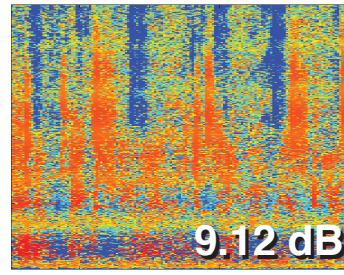
EM+ILD



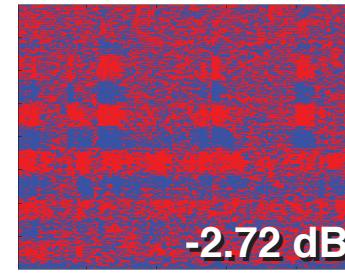
EM-ILD (only IPD)



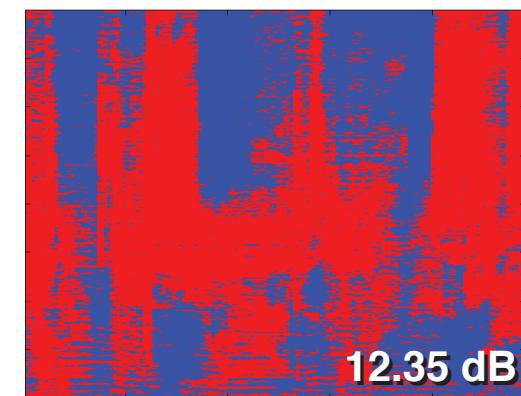
PHAT-histogram



EM+1ILD (tied means)



DUET

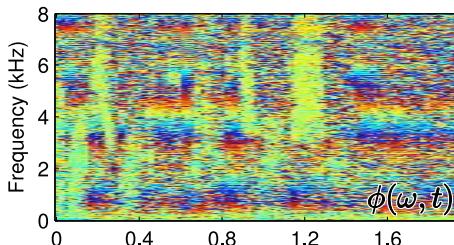


Ground Truth

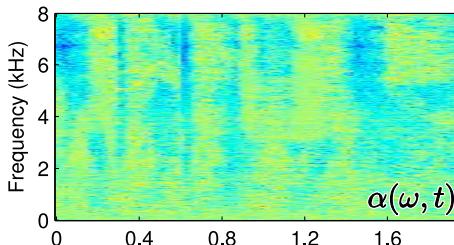
MESSL-SP (Source Prior)

Observations

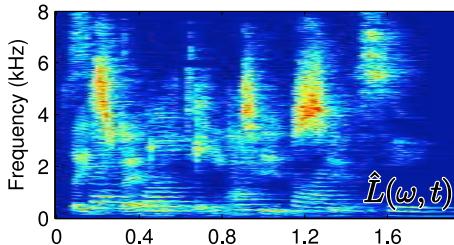
Mixture – IPD



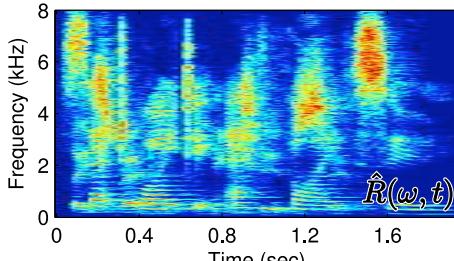
Mixture – ILD



Mixture – left channel

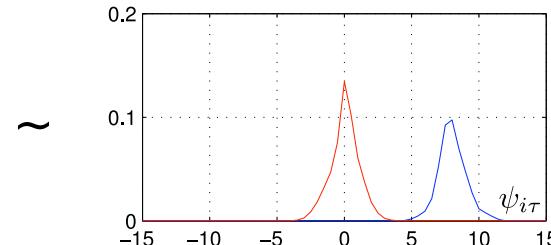


Mixture – right channel

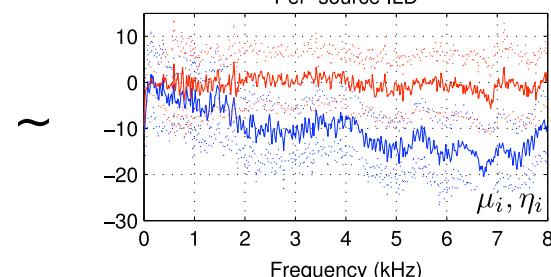


Parameters

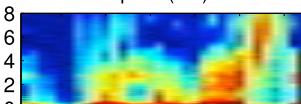
Per-source ITD



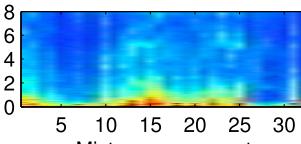
Per-source ILD



Source prior (SP) means



SP covars

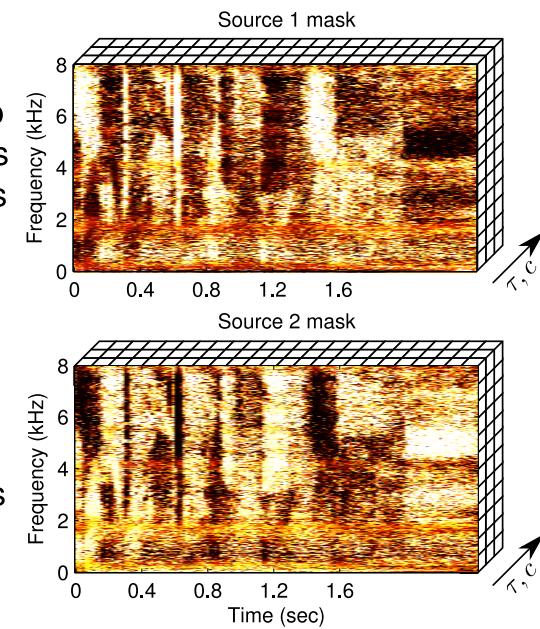


Mixture component



Postriors

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



E-step

Use parameters to compute posteriors of hidden variables

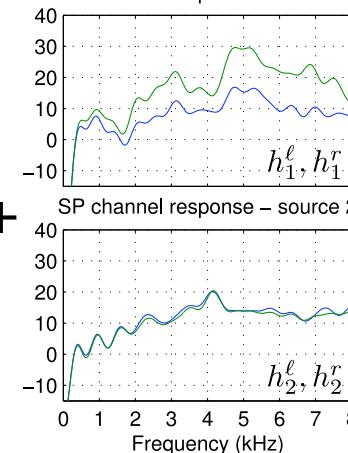


M-step

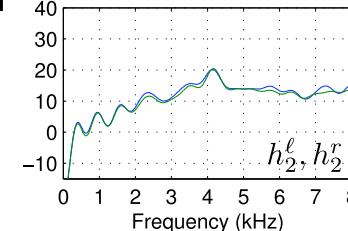
Use posteriors to update parameters

Separate sources by multiplying mixture by different masks

SP channel response – source 1

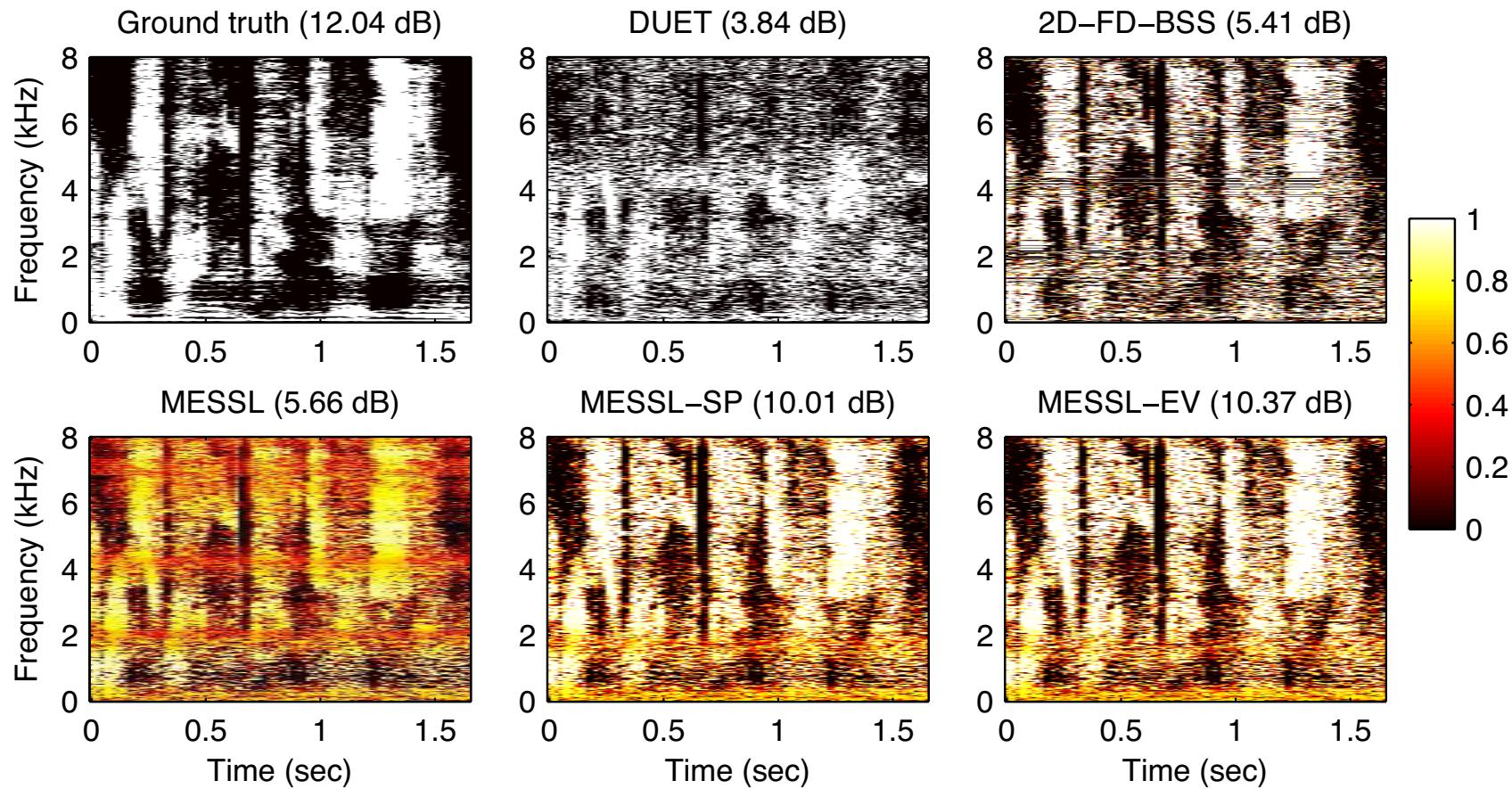


SP channel response – source 2



MESSL-SP Results

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



3. Soundtrack Classification

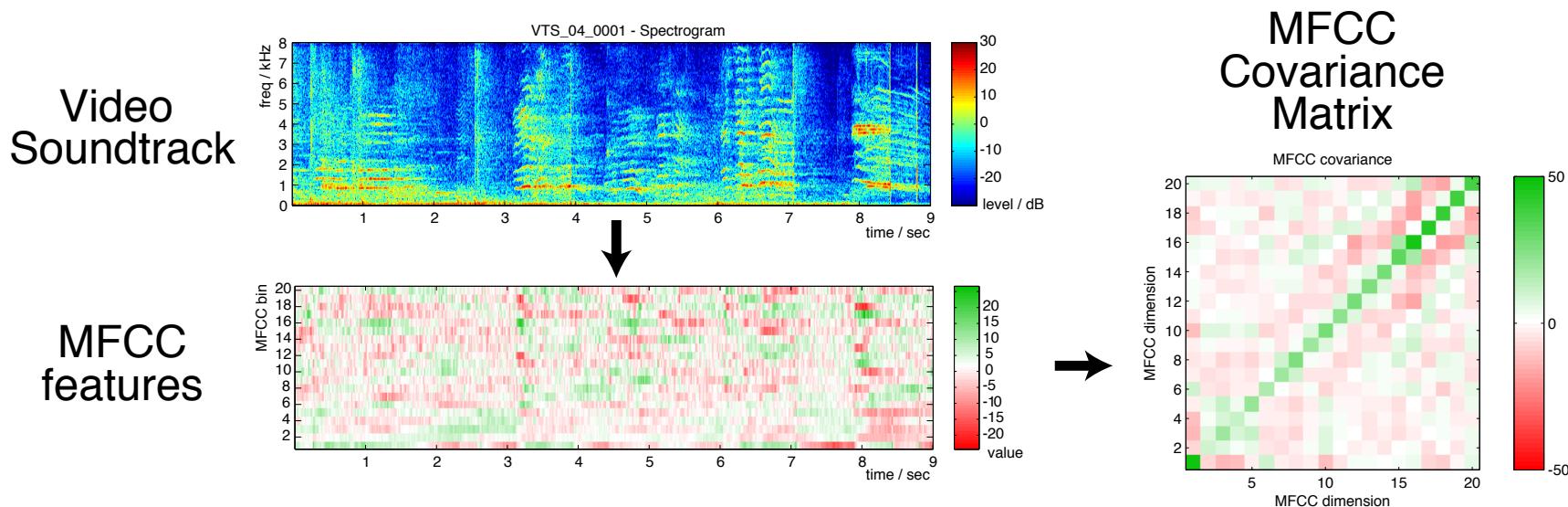
- Short video clips as the **evolution of snapshots**
 - 10-100 sec, one location, no editing
 - browsing?



- Need information for **indexing...**
 - video + audio
 - foreground + background

MFCC Covariance Representation

- Each clip/segment → **fixed-size statistics**
 - similar to speaker ID and music genre classification
- Full **Covariance matrix of MFCCs**
 - maps the kinds of **spectral shapes** present

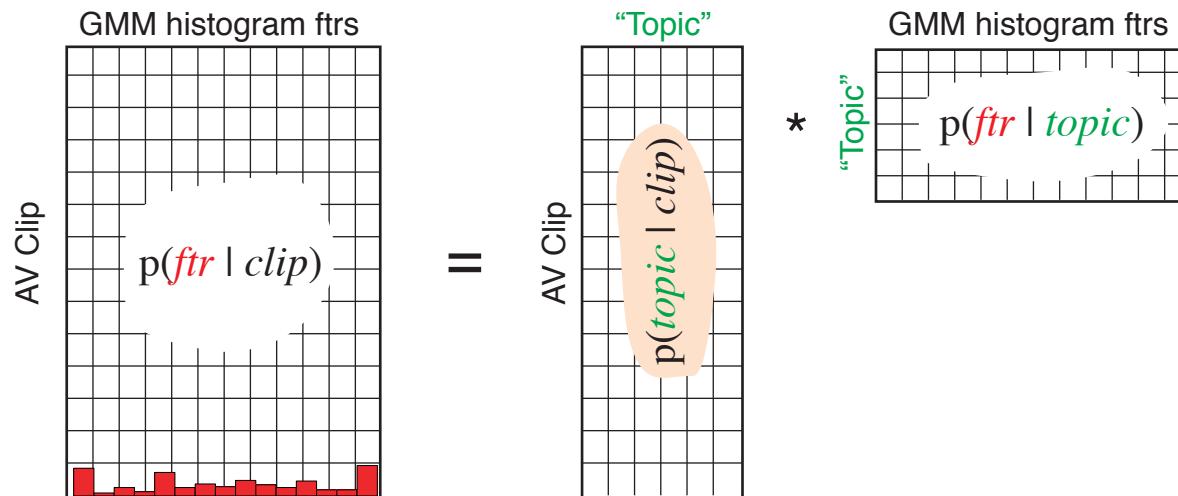


- Clip-to-clip **distances** for SVM classifier
 - by KL or 2nd Gaussian model

Latent Semantic Analysis (LSA)

Hofmann '99

- Probabilistic LSA (**pLSA**) models each histogram as a mixture of several ‘**topics**’
 - .. each clip may have several things going on
- Topic sets optimized through **EM**
 - $p(ftr \mid clip) = \sum_{topics} p(ftr \mid topic) p(topic \mid clip)$

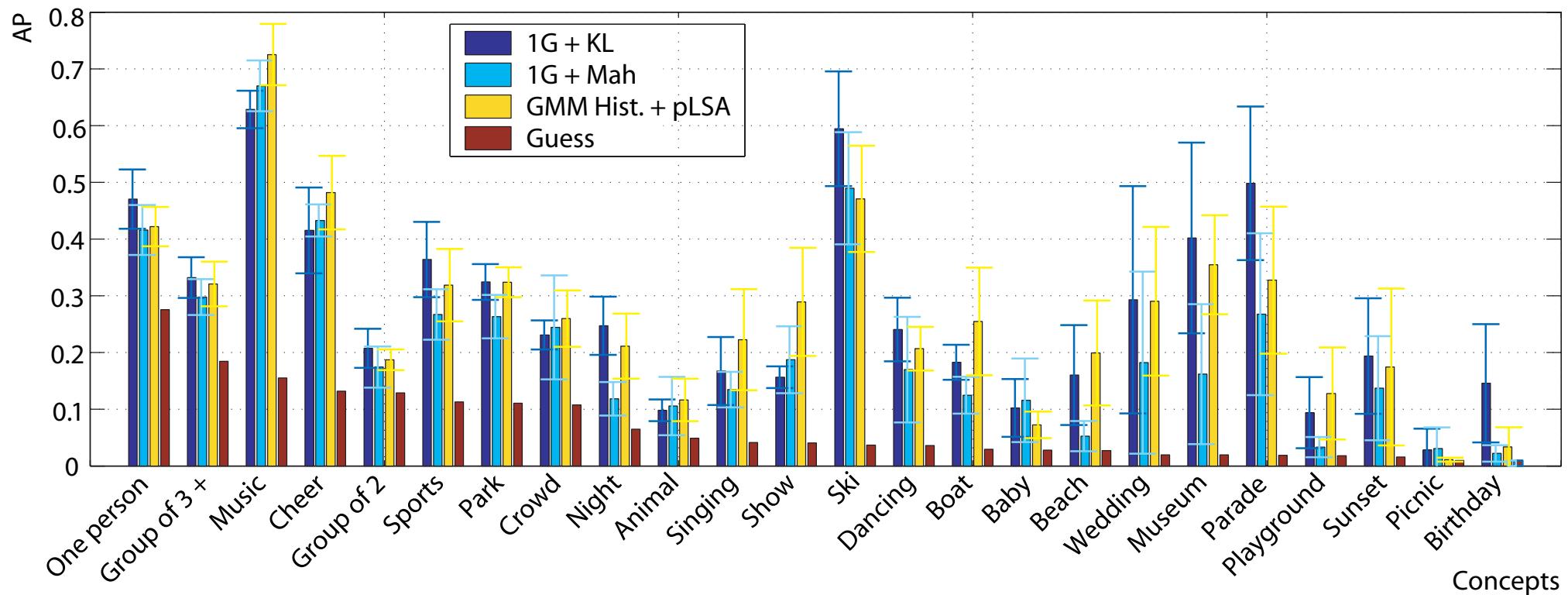


- use $p(topic \mid clip)$ as per-clip features

Classification Results

Chang, Ellis et al. '07
Lee & Ellis '10

- Wide range:

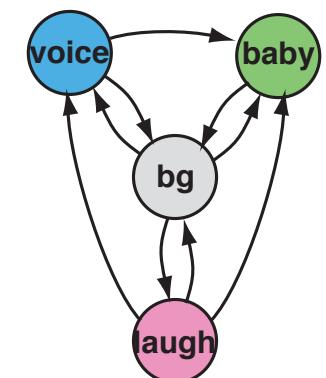
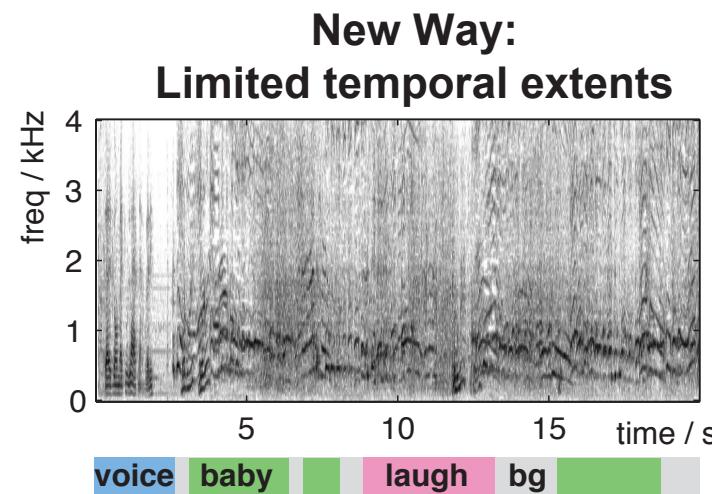
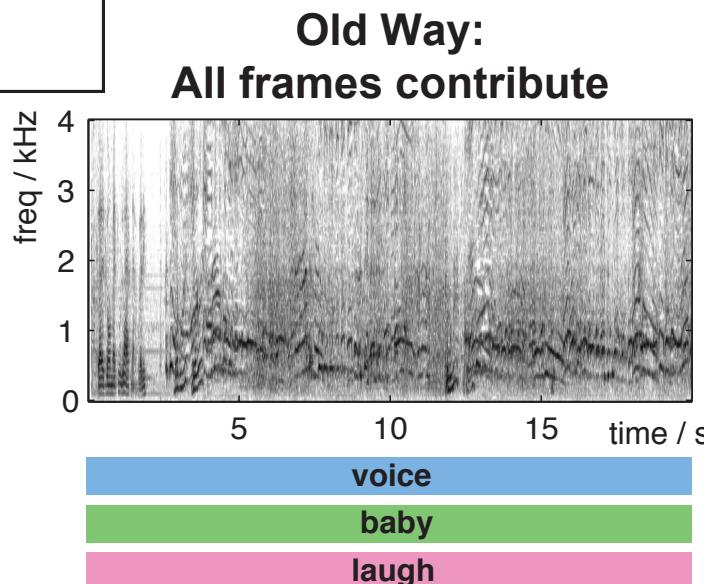


- audio (music, ski) vs. non-audio (group, night)
- large AP uncertainty on infrequent classes

Temporal Refinement

- **Global** vs. **local** class models
 - tell-tale acoustics may be ‘washed out’ in statistics
 - try iterative **realignment** of HMMs:

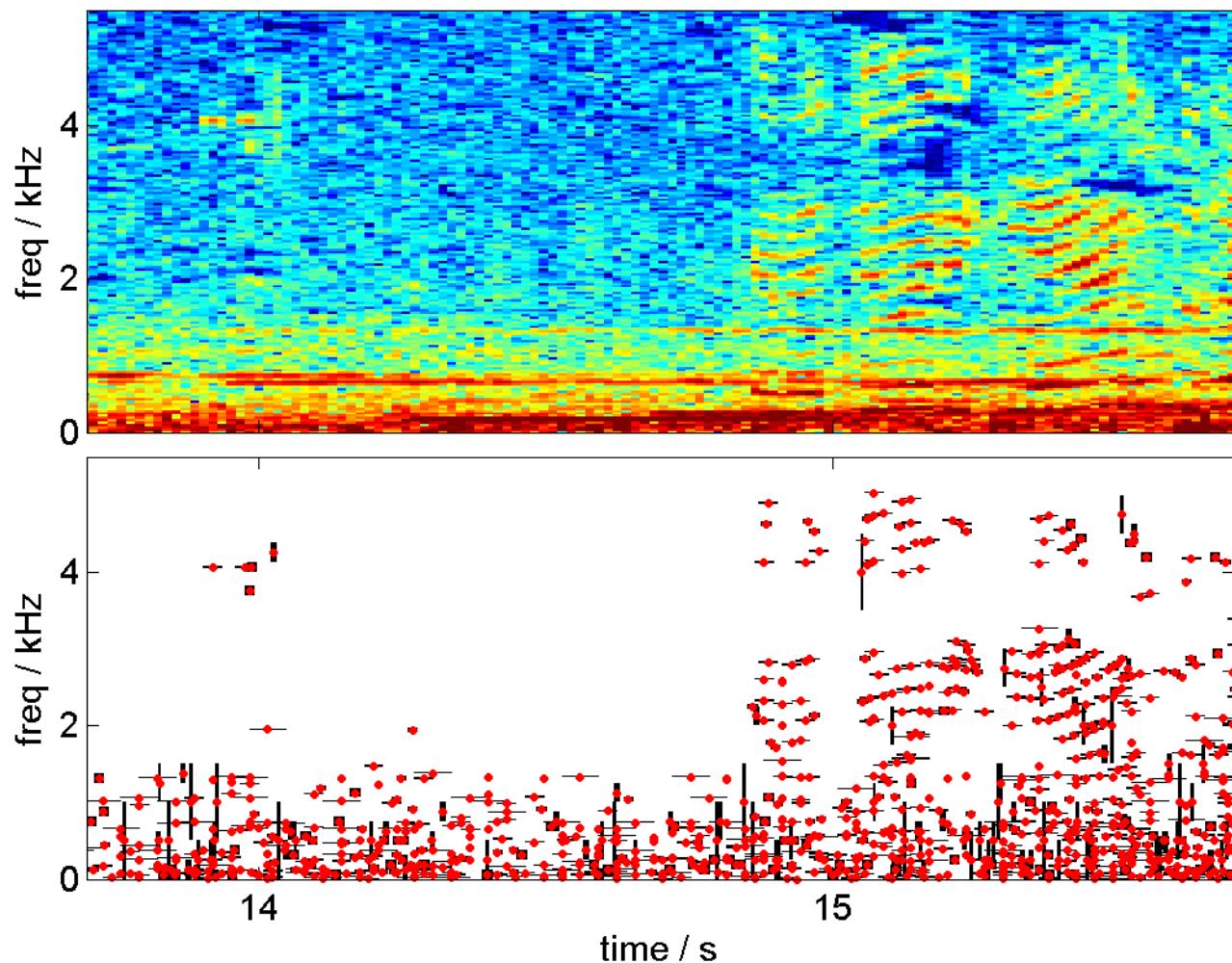
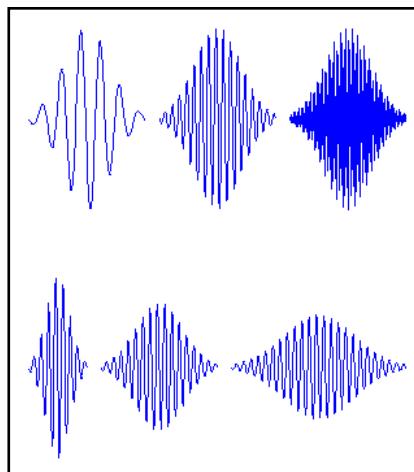
YT baby 002:
voice
baby
laugh



- “background” (bg) model shared by all clips
- Multiple-Instance Learning

Landmark-based Features

- Describe audio with biggest Gabor elements
 - extract with Matching Pursuit (MP)

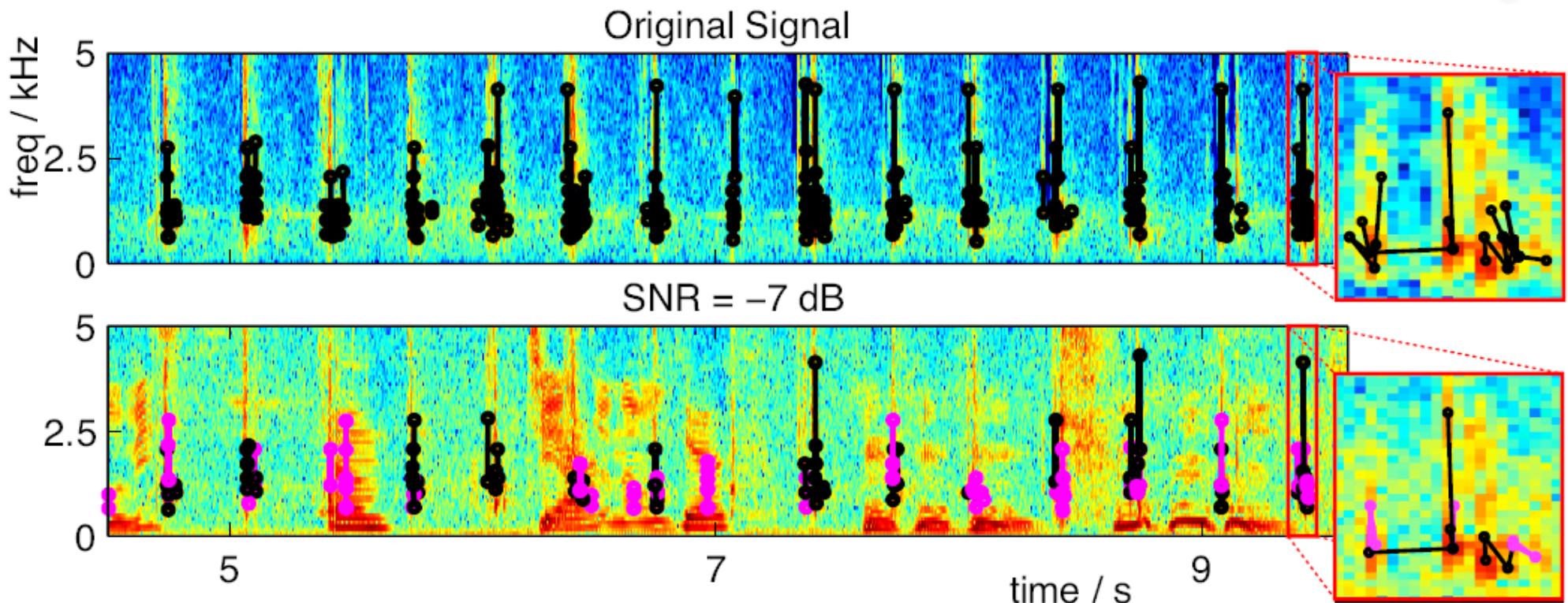


- neighbor pairs as “features”?

Landmark models for similar events

Cotton & Ellis '09

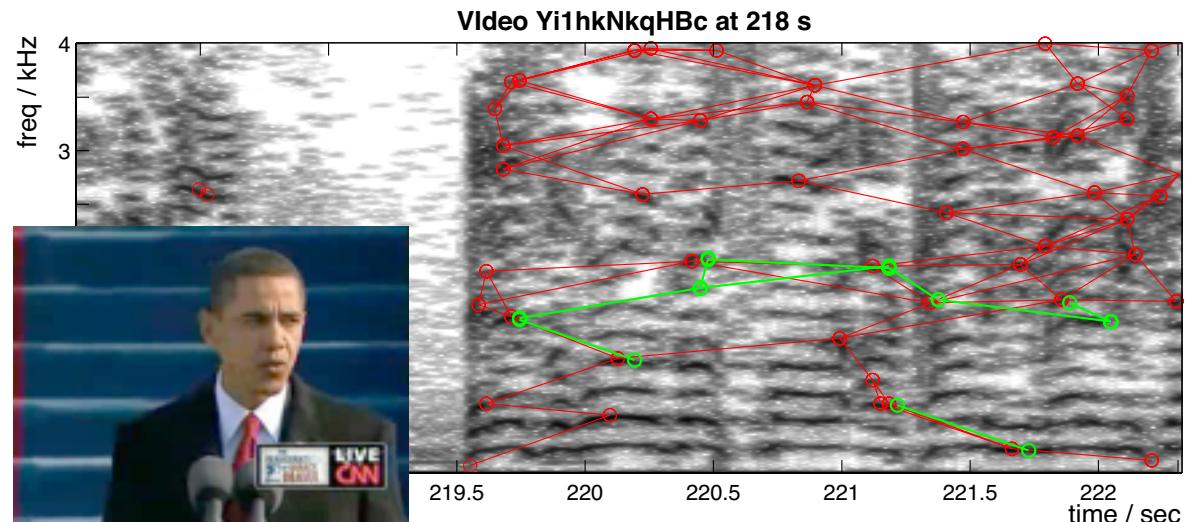
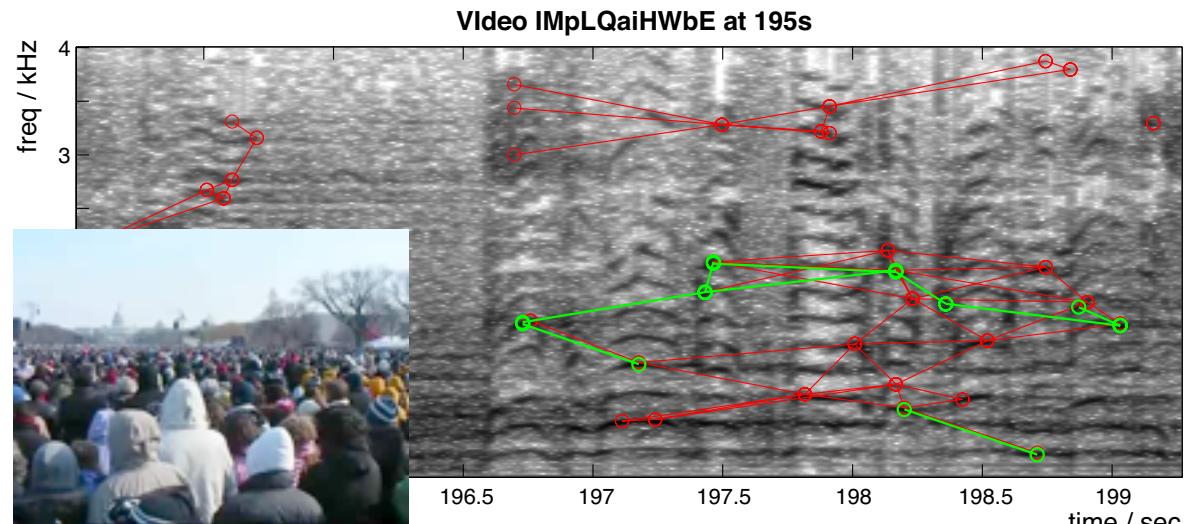
- Build index of Gabor neighbor pairs
 - recognize repeated events with similar pairs



Matching Videos via Fingerprints

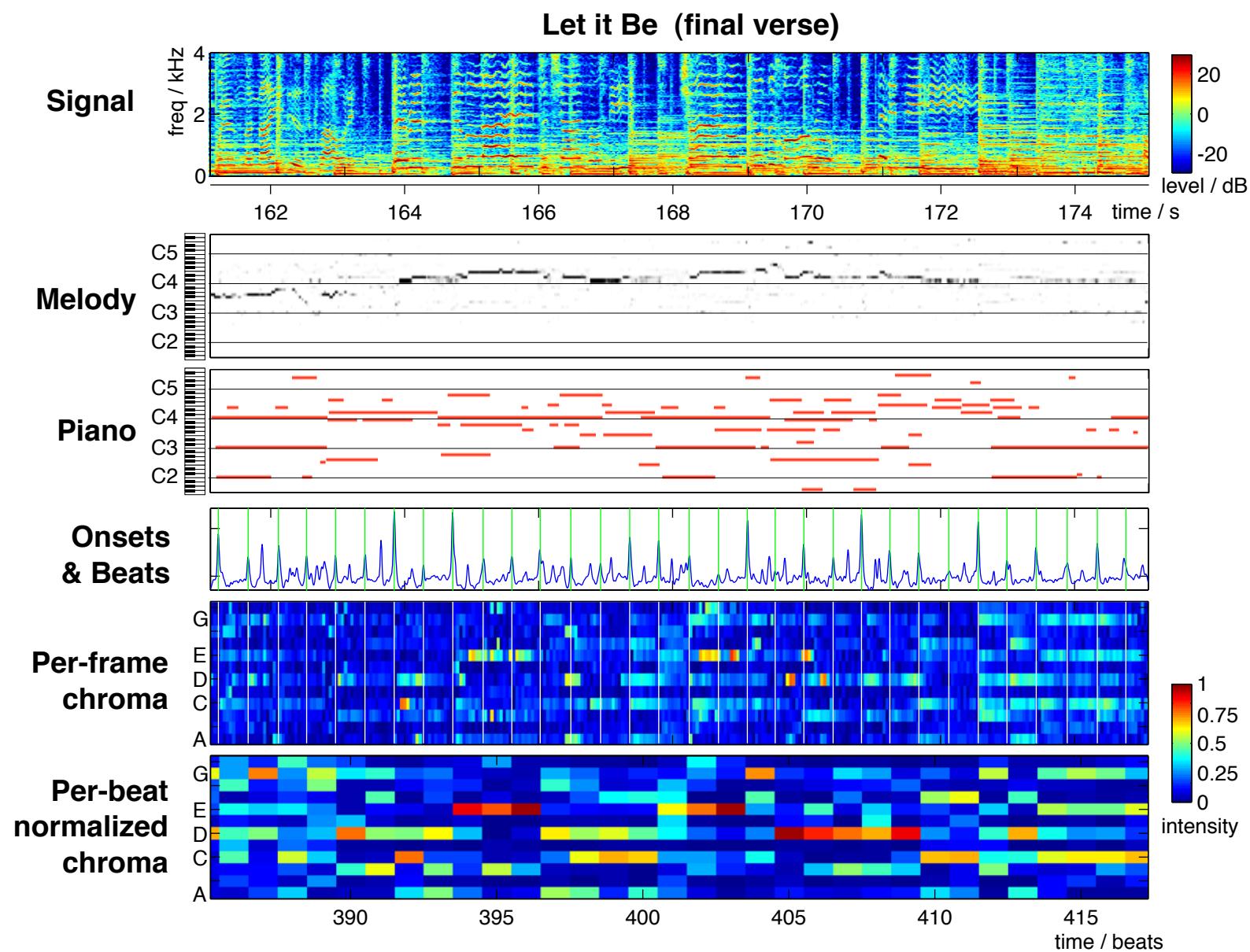
Cotton & Ellis '10?

- Landmark pairs are a noise-robust **fingerprint**
- Use to match distinct videos with same sound ambience



4. Music Audio Analysis

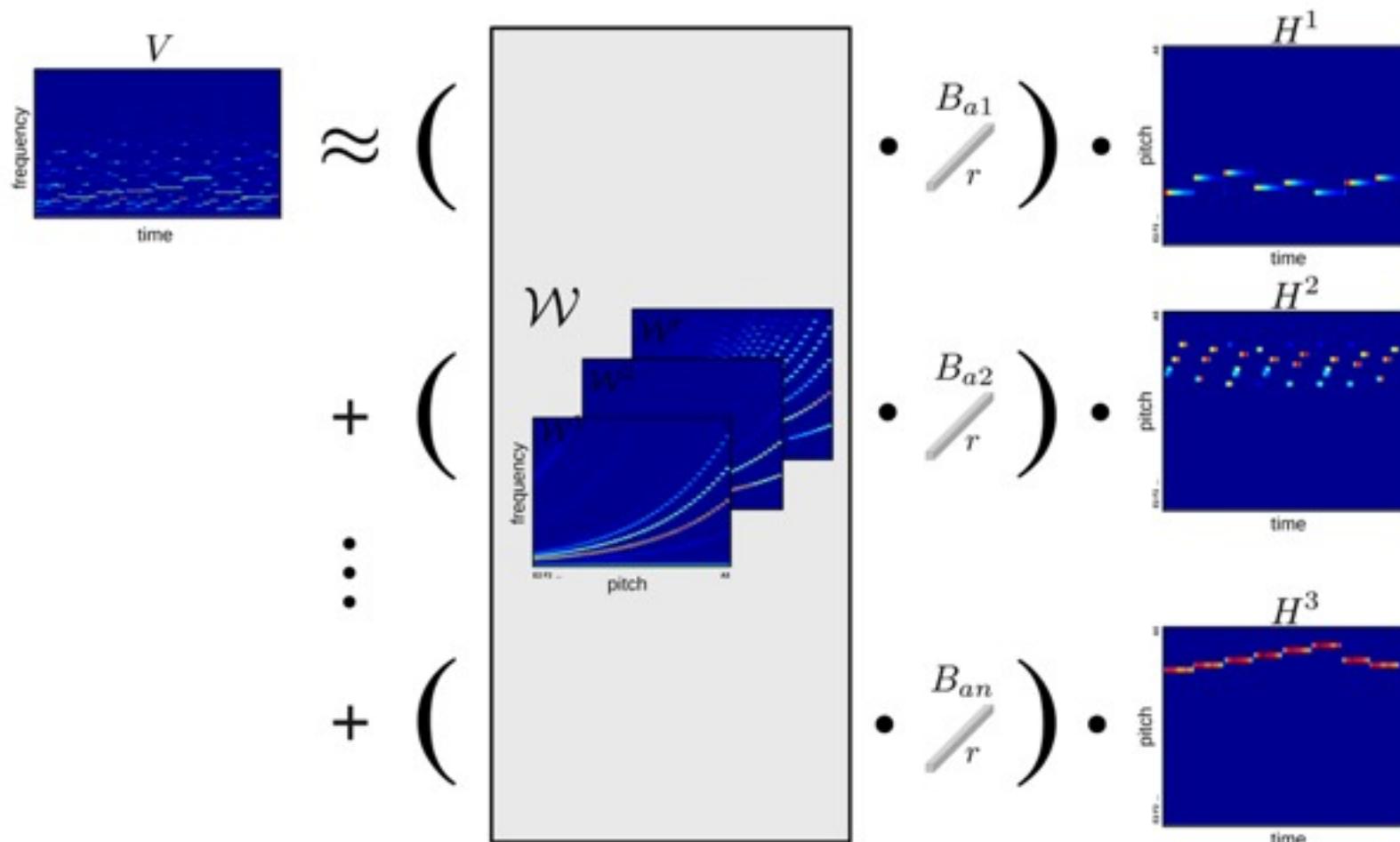
- Interested in all levels from notes to genres



Polyphonic Transcription

Grindlay & Ellis '09

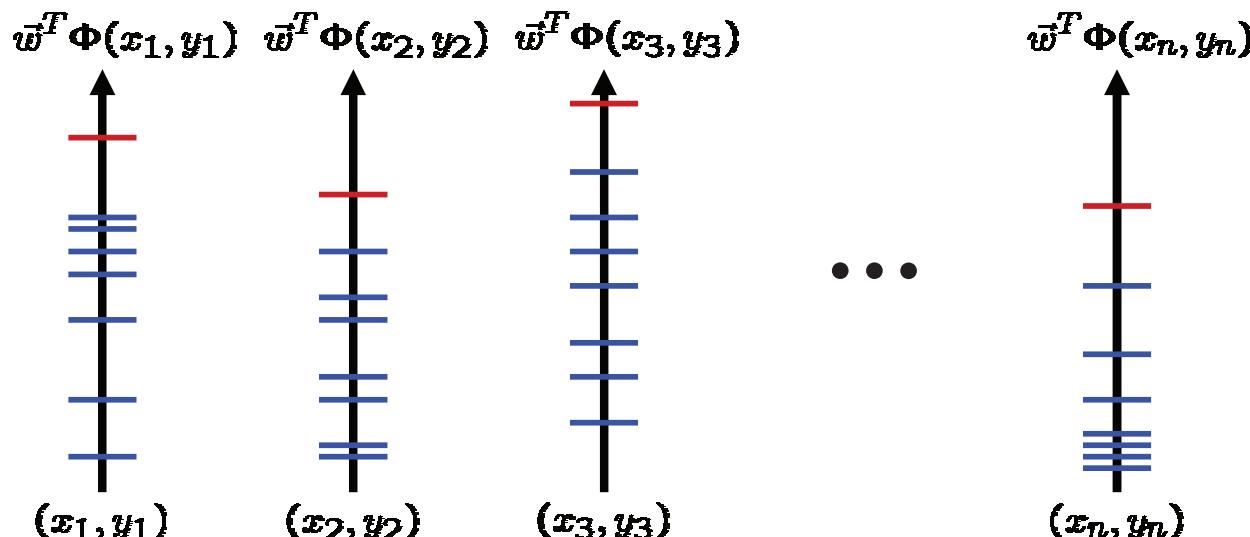
- Apply the Eigenvoice idea to music
 - eigeninstruments?
 - Subspace NMF



Chord Recognition

Structural Support Vector Machine

- Joint features $\Phi(x, y)$ describe match between x and y
- Learn weights \vec{w} so that $\vec{w}^T \Phi(x, y)$ is max for correct y



Tsochantaridis et al '05
Weller, Ellis, Jebara '09

- Much better results

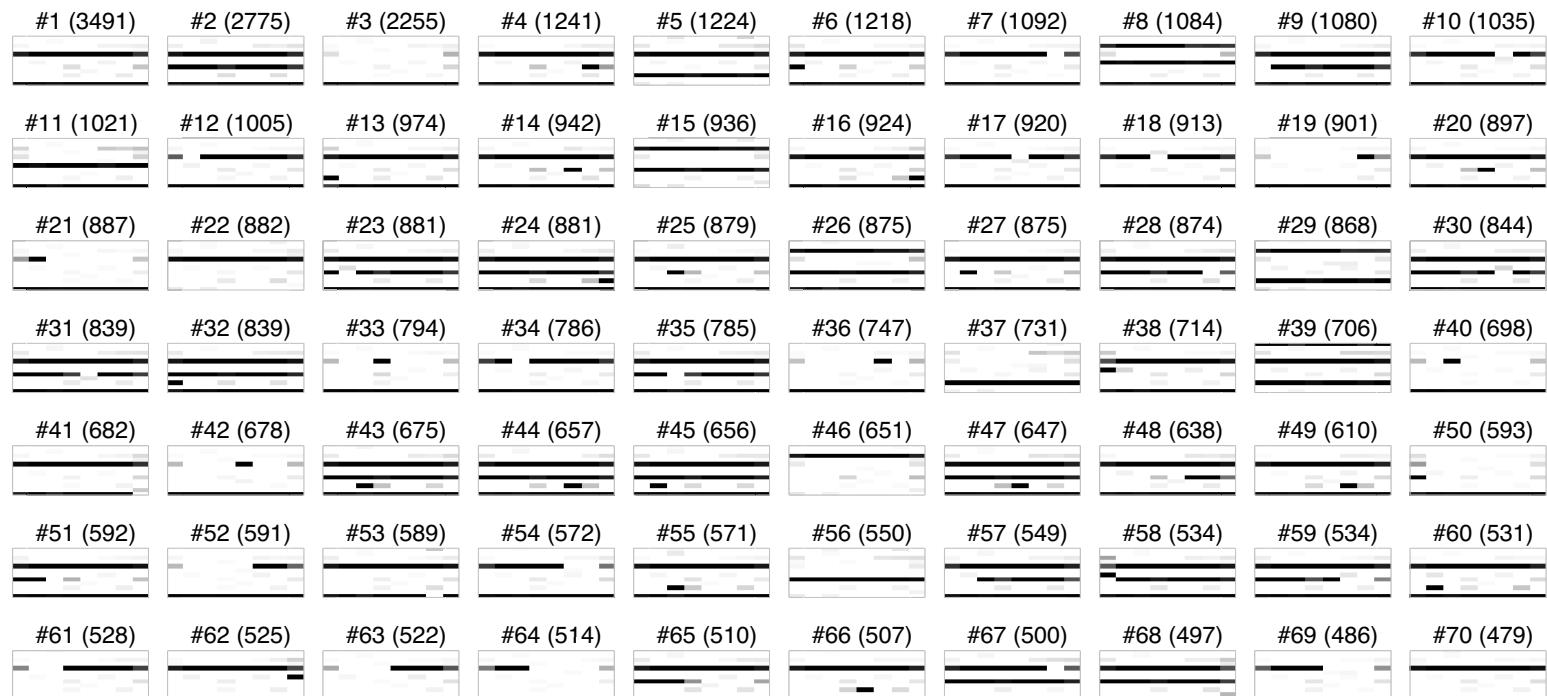
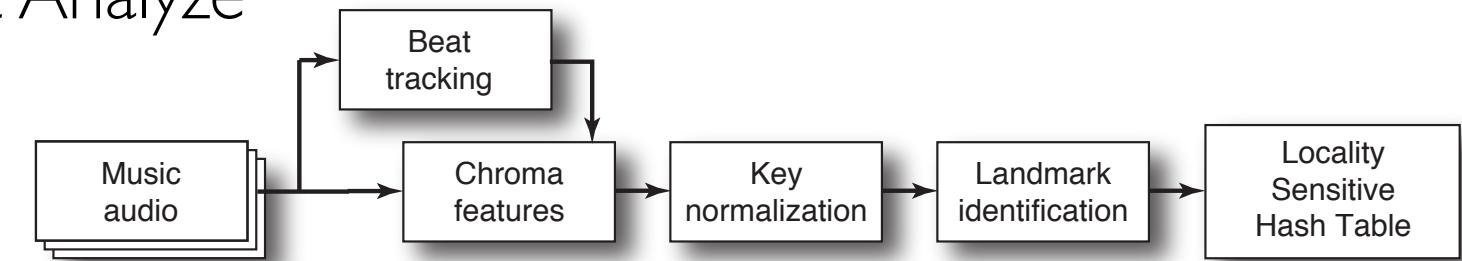
	RUSUSL	WEJ1	WEJ2	WEJ3	WEJ4
Wt. Ave. Overlap Score	0.701	0.704	0.723	0.723	0.742
Wt. Ave. Overlap Score (Merged maj/min)	0.760	0.743	0.762	0.760	0.777

Melodic-Harmonic Mining

- 100,000 tracks from morecowbell.dj

- as Echo Nest Analyze

- Frequent clusters of 12×8 binarized event-chroma



Summary

- LabROSA : getting information from sound
- Speech
 - monaural separation using eigenvoices
 - binaural + reverb using MESSL
- Environmental
 - classification of consumer video
 - landmark-based events and matching
- Music
 - transcription of notes, chords, ...
 - large corpus mining