

Sound, Mixtures, and Learning

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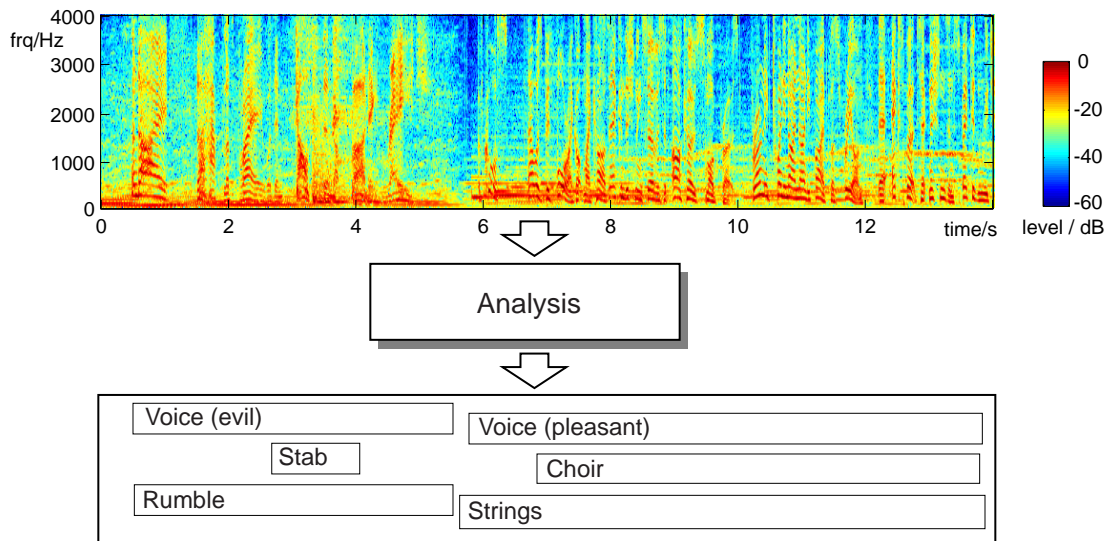
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Outline

- 1 Human sound organization
- 2 Computational Auditory Scene Analysis
- 3 Speech models and knowledge
- 4 Sound mixture recognition
- 5 Learning opportunities

1

Human sound organization

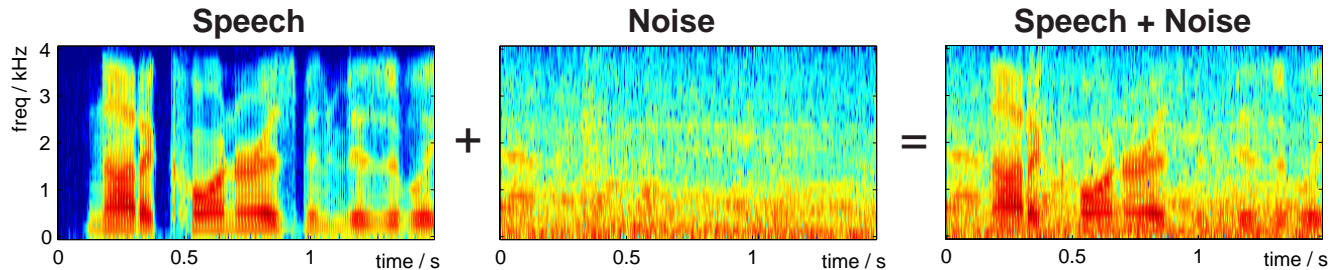


- **Analyzing and describing complex sounds:**
 - continuous sound mixture → distinct events
- **Hearing is *ecologically* grounded**
 - reflects 'natural scene' properties
 - subjective *not* canonical (ambiguity)
 - *mixture* analysis as primary goal



Sound mixtures

- **Sound ‘scene’ is almost always a mixture**
 - always stuff going on
 - sound is ‘transparent’ - but big energy range



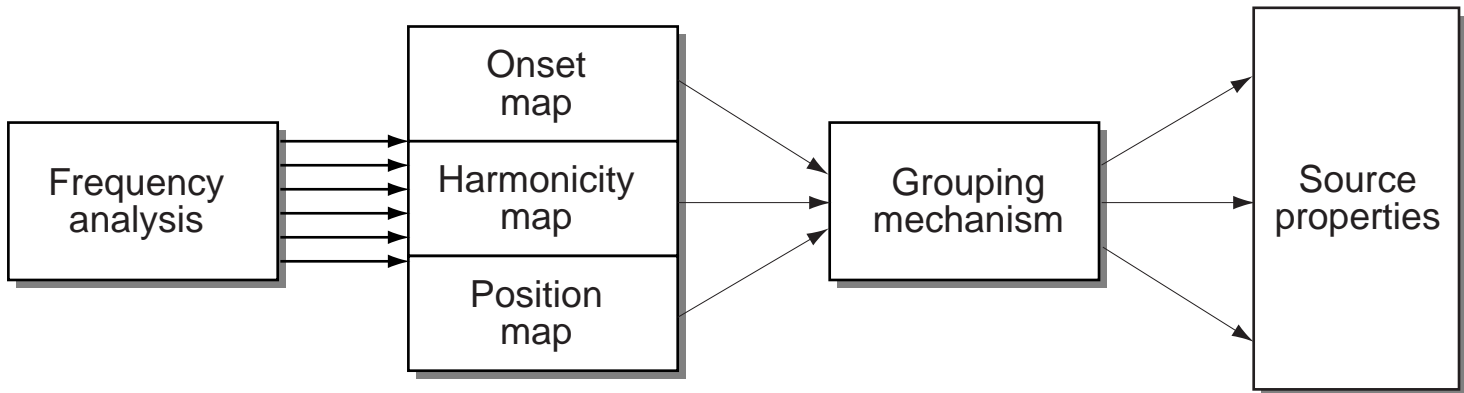
- **Need information related to our ‘world model’**
 - i.e. separate objects
 - a wolf howling in a blizzard is the same as a wolf howling in a rainstorm
 - whole-signal statistics won't do this
- **‘Separateness’ is similar to independence**
 - objects/sounds that change in isolation
 - but: depends on the situation e.g. passing car vs. mechanic's diagnosis



Auditory scene analysis

(Bregman 1990)

- **How do people analyze sound mixtures?**
 - break mixture into small *elements* (in time-freq)
 - elements are *grouped* in to sources using *cues*
 - sources have aggregate *attributes*
- **Grouping 'rules' (Darwin, Carlyon, ...):**
 - cues: common onset/offset/modulation, harmonicity, spatial location, ...

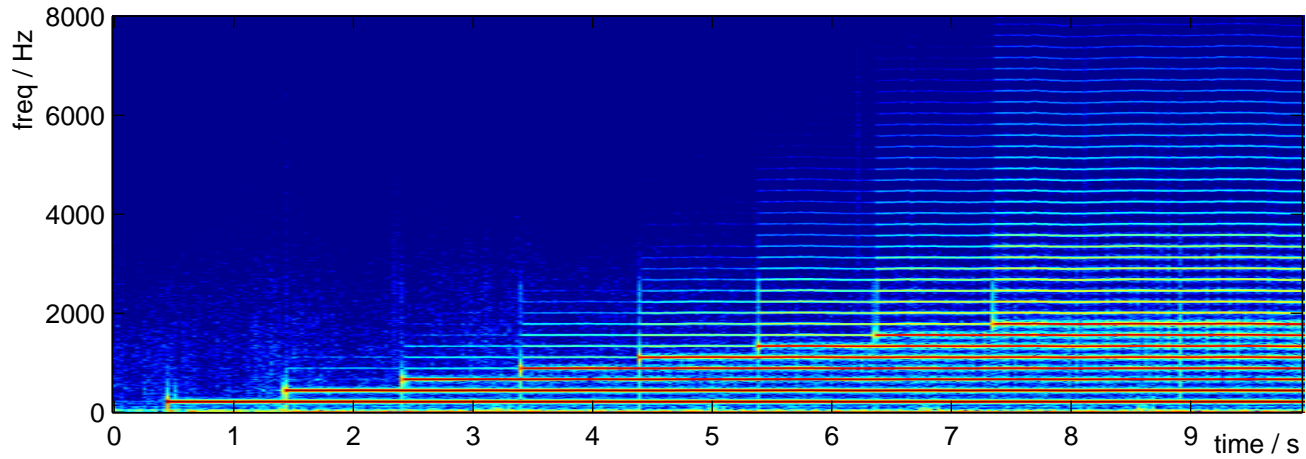


(after Darwin, 1996)



Cues to simultaneous grouping

- **Elements + attributes**

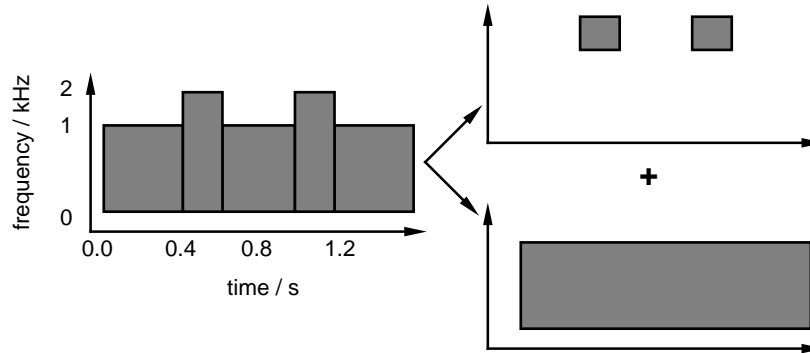


- **Common onset**
 - simultaneous energy has common source
- **Periodicity**
 - energy in different bands with same cycle
- **Other cues**
 - spatial (ITD/IID), familiarity, ...



The effect of context

- **Context can create an ‘expectation’:**
i.e. a bias towards a particular interpretation
- **e.g. Bregman’s “old-plus-new” principle:**
A change in a signal will be interpreted as an *added* source whenever possible



- a different division of the same energy depending on what preceded it

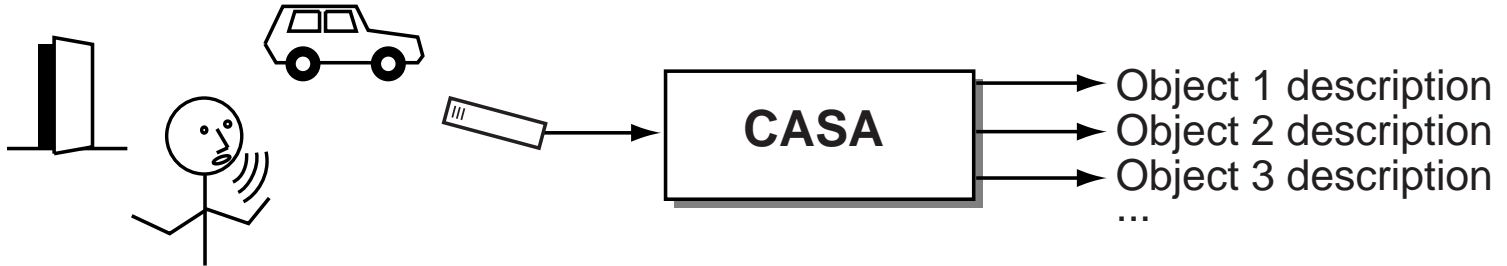


Outline

- 1 Human sound organization
- 2 **Computational Auditory Scene Analysis**
 - sound source separation
 - bottom-up models
 - top-down constraints
- 3 Speech models and knowledge
- 4 Sound mixture recognition
- 5 Learning opportunities



2 Computational Auditory Scene Analysis (CASA)

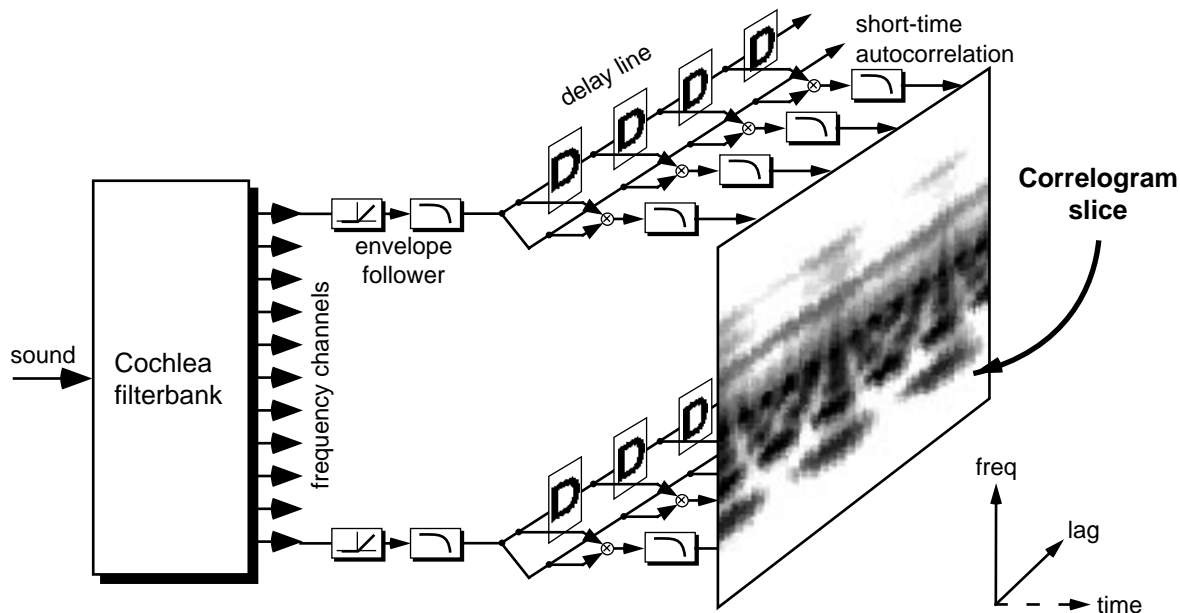


- **Goal: Automatic sound organization ;**
Systems to ‘pick out’ sounds in a mixture
 - ... like people do
- **E.g. voice against a noisy background**
 - to improve speech recognition
- **Approach:**
 - psychoacoustics describes grouping ‘rules’
 - ... just implement them?



CASA front-end processing

- **Correlogram:**
Loosely based on known/possible physiology



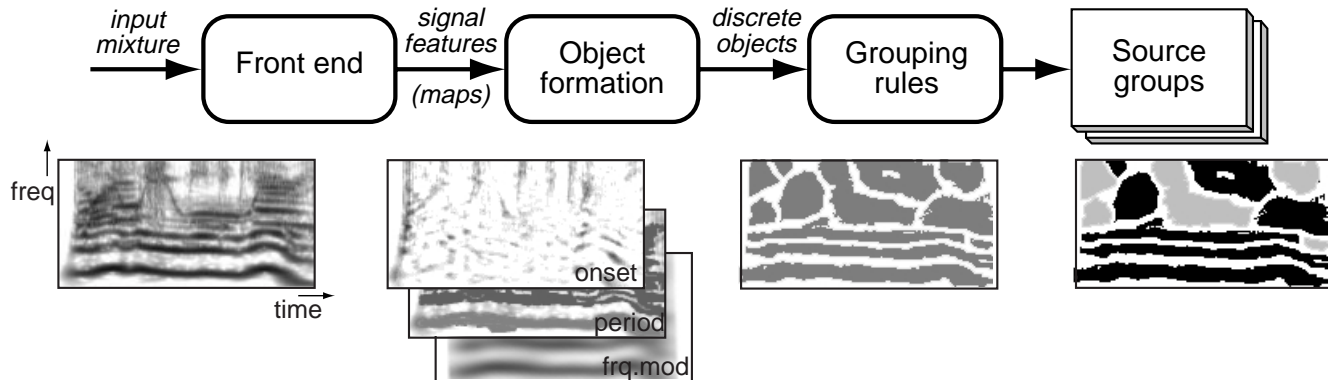
- linear filterbank cochlear approximation
- static nonlinearity
- zero-delay slice is like spectrogram
- periodicity from delay-and-multiply detectors



The Representational Approach

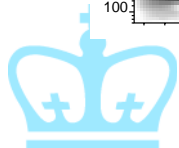
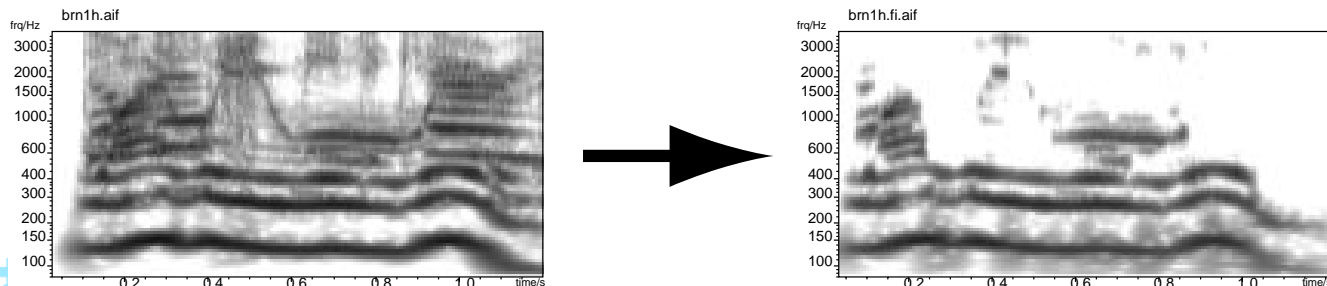
(Brown & Cooke 1993)

- Implement psychoacoustic theory

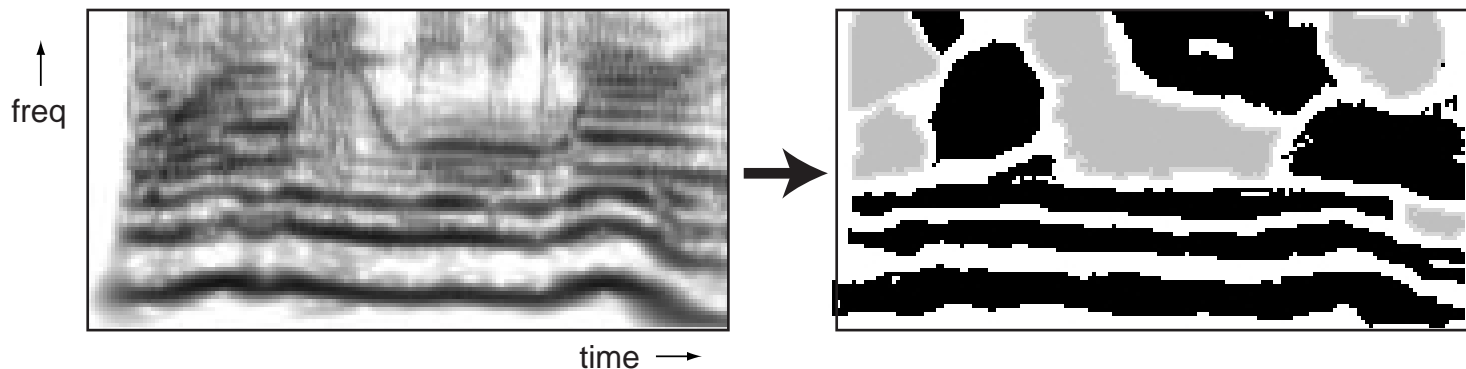


- 'bottom-up' processing
- uses common onset & periodicity cues

- Able to extract voiced speech:



Problems with 'bottom-up' CASA

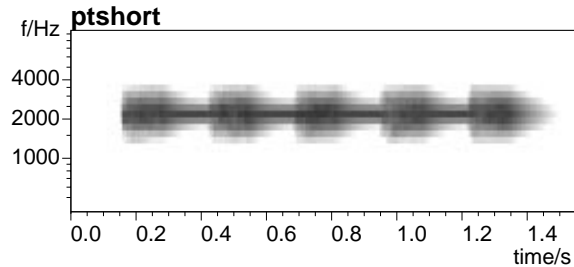


- **Circumscribing time-frequency elements**
 - need to have 'regions', but hard to find
- **Periodicity is the primary cue**
 - how to handle aperiodic energy?
- **Resynthesis via masked filtering**
 - cannot separate within a single t-f element
- **Bottom-up leaves no ambiguity or context**
 - how to model illusions?

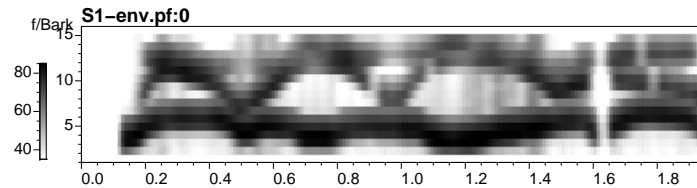


Restoration in sound perception

- Auditory 'illusions' = hearing what's not there
- The continuity illusion



- SWS



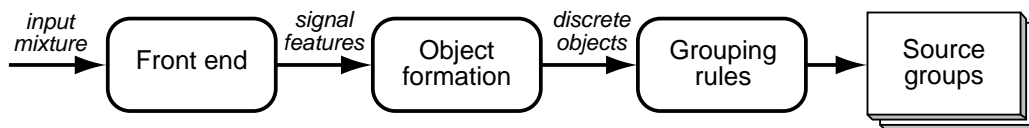
- duplex perception



Adding top-down constraints

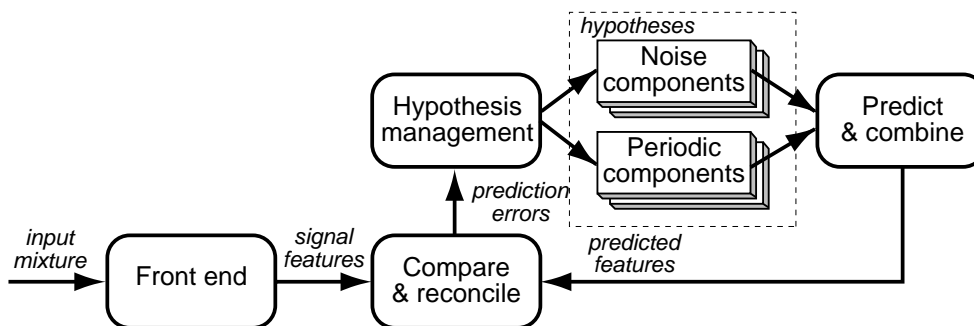
Perception is not *direct*
but a *search for plausible hypotheses*

- **Data-driven (bottom-up)...**



- objects irresistibly appear

vs. Prediction-driven (top-down)



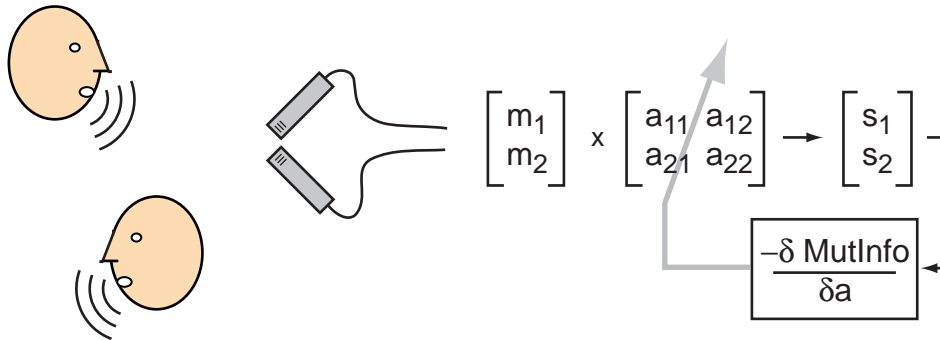
- match observations with parameters of a world-model
- need world-model constraints...



Aside: Optimal techniques (ICA, ABF)

(Bell & Sejnowski etc.)

- **General idea:**
Drive a parameterized separation algorithm to maximize independence of outputs



- **Attractions:**
 - mathematically rigorous, minimal assumptions
- **Problems:**
 - limitations of separation algorithm (N x N)
 - essentially bottom-up



Outline

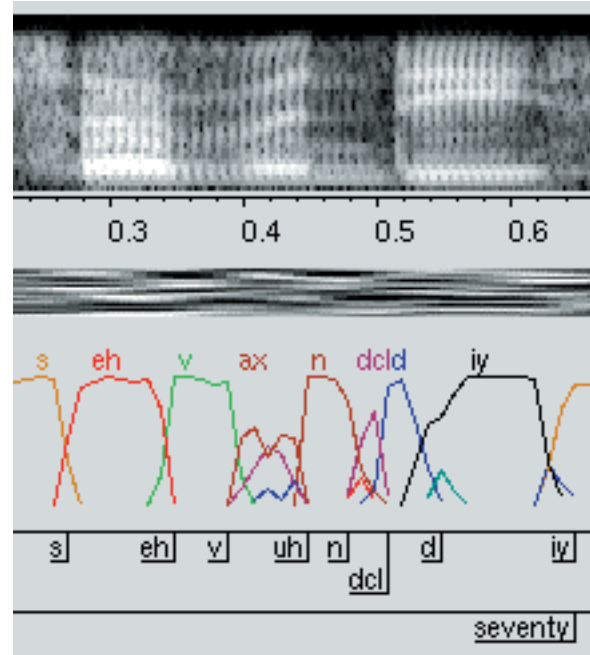
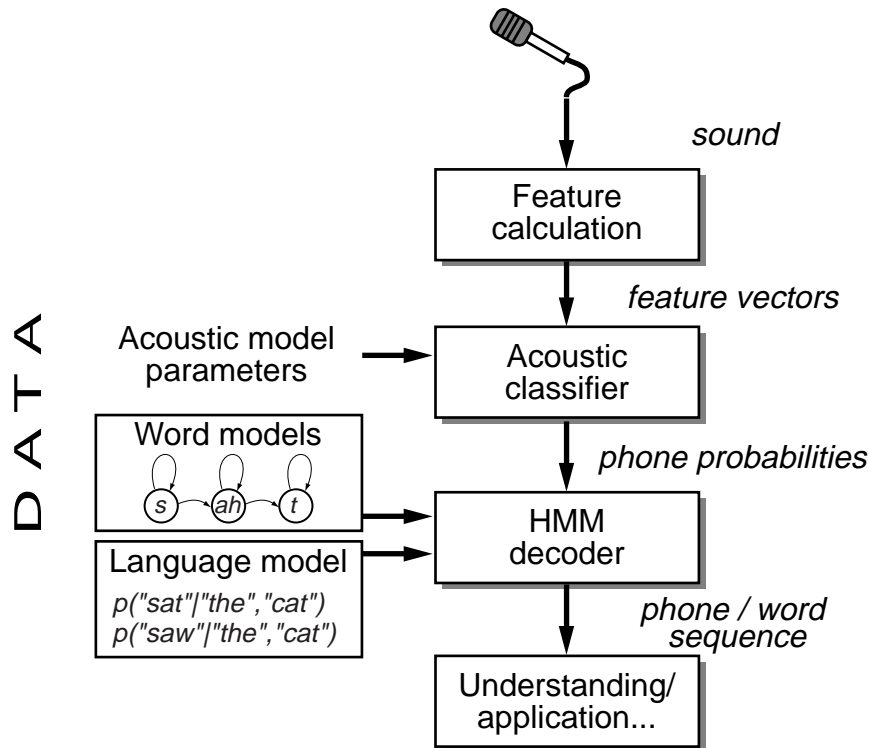
- 1 Human sound organization
- 2 Computational Auditory Scene Analysis
- 3 Speech models and knowledge**
 - automatic speech recognition
 - subword states
 - cepstral coefficients
- 4 Sound mixture recognition
- 5 Learning opportunities



3

Speech models & knowledge

- Standard speech recognition

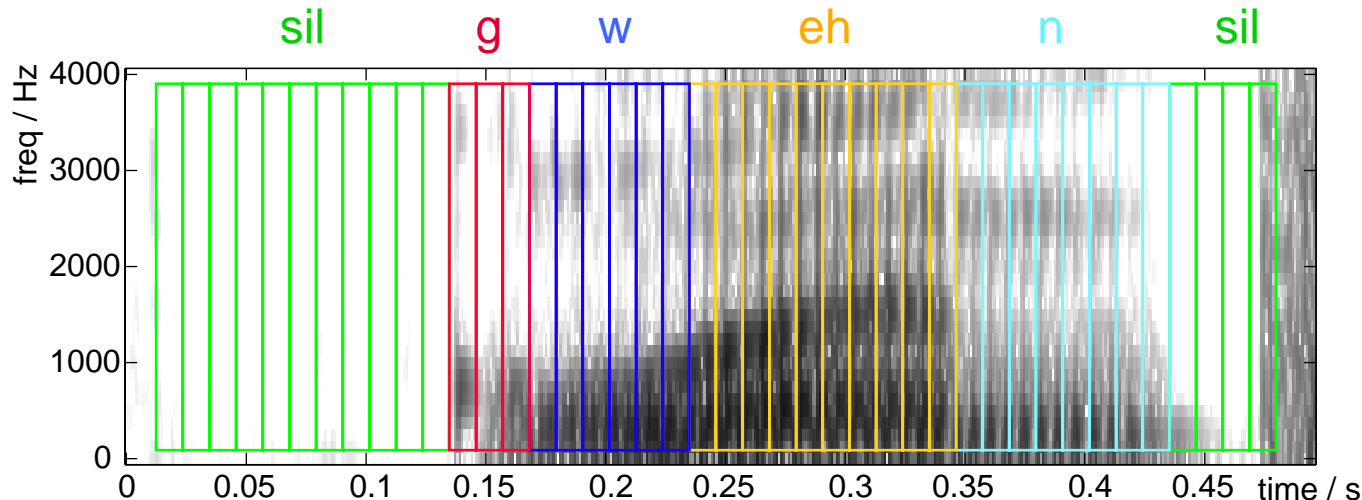


- **'State of the art' word-error rates (WERs):**
 - 2% (dictation) - 30% (telephone conversations)



Speech units

- **Speech is highly variable**
 - simple templates won't do
 - spectral variation (voice quality)
 - *time-warp* problems
- **Match short segments (states), allow repeats**
 - model with pseudo-stationary slices of ~ 10 ms



- **Speech models are distributions $p(X|q)$**

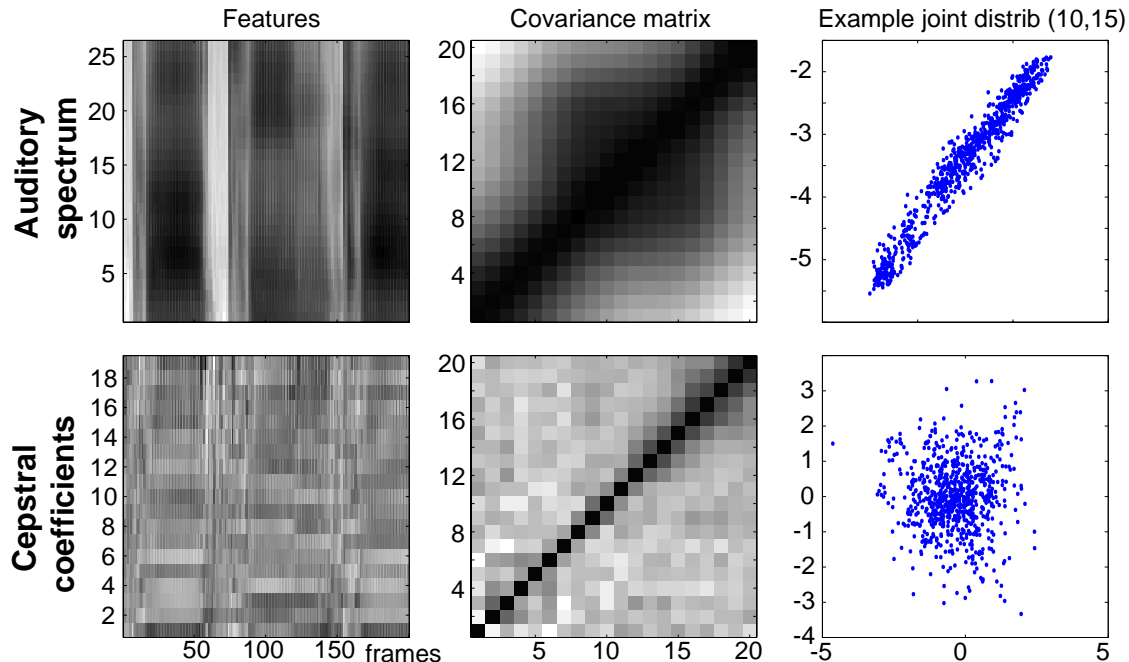


Speech features: Cepstra

- **Idea: Decorrelate & summarize spectral slices:**

$$X_m[l] = IDFT\{\log|S[mH, k]|\}$$

- easier to model:



- C_0 'normalizes out' average log energy

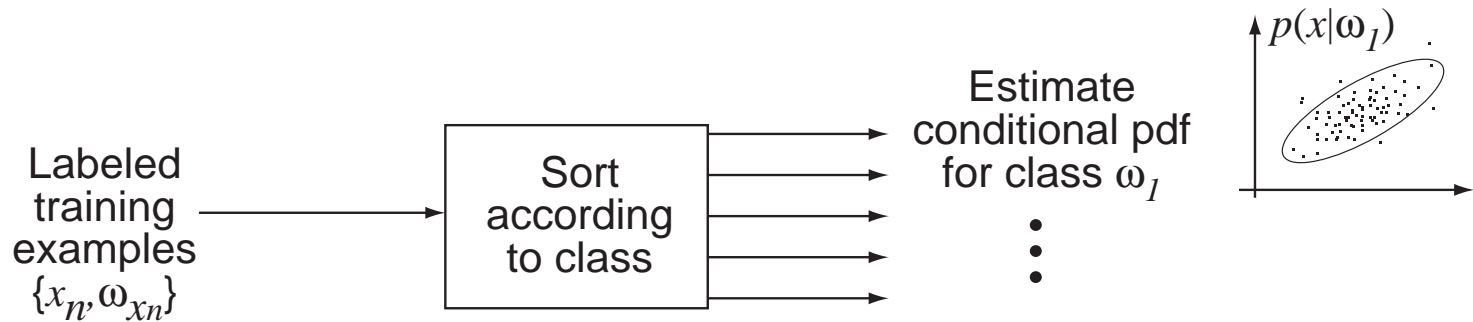
- **Decorrelated pdfs fit diagonal Gaussians**

- DCT is close to PCA for log spectra



Acoustic model training

- **Goal: describe $p(X|q)$ with e.g. GMMs**

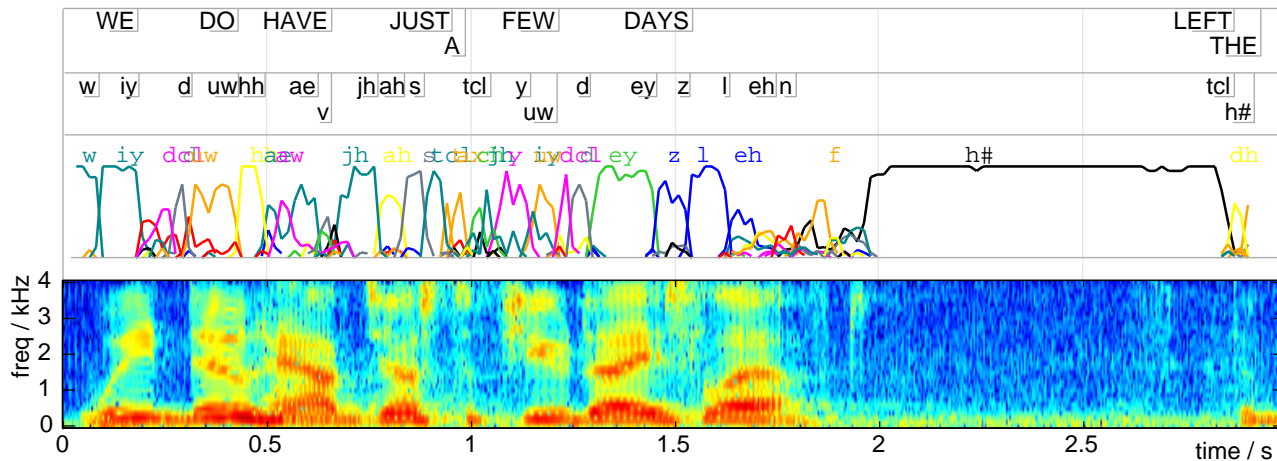


- **Training data labels from:**
 - manual phonetic annotation
 - 'best path' from earlier classifier (Viterbi)
 - EM: joint estimation of labels & pdfs



HMM decoding

- **Feature vectors cannot be reliably classified into phonemes**



- **Use top-down constraints to get good results**
 - allowable phonemes
 - dictionary of known words
 - grammar of possible sentences
- **Decoder searches all possible state sequences**
 - at least notionally; pruning makes it possible



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- 1 Human sound organization
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- 4 Sound mixture recognition**
 - feature invariance
 - mixtures including
 - general mixtures
- 5 Learning opportunities



4

Sound mixture recognition

- **Biggest problem in speech recognition is background noise interference**
- **Feature invariance approach**
 - use features that reflect only speech
 - e.g. normalization, mean subtraction
 - but: non-static noise?
- **Or: more complex models of the signal**
 - HMM decomposition
 - missing-data recognition
- **Generalize to other, multiple sounds**



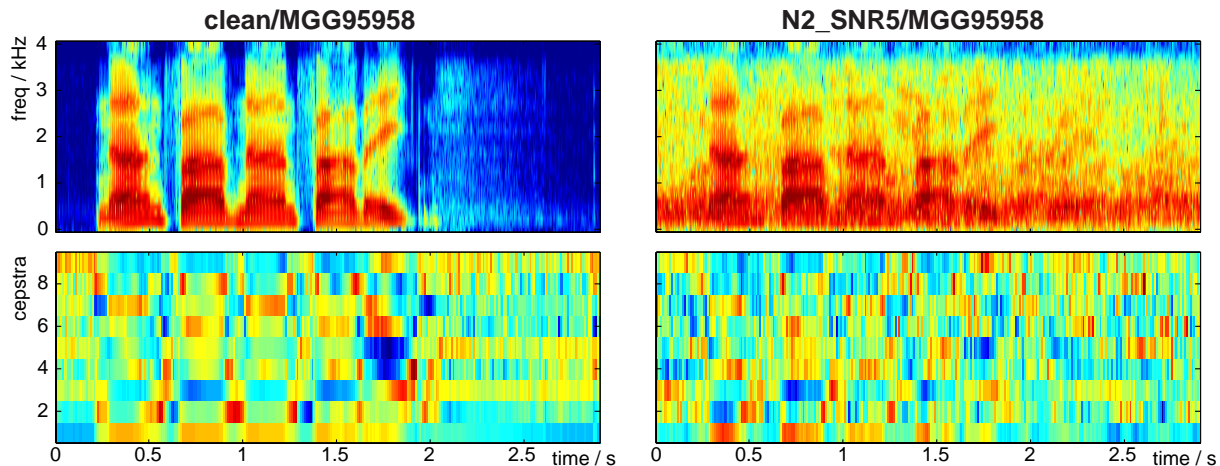
Feature normalization

- **Idea:** feature *variations*, not absolute level
- **Hence:** calculate average level & subtract it:
$$X[k] = S[k] - \text{mean}\{S[k]\}$$
- **Factors out fixed channel frequency response:**

$$s[n] = h[n] * e[n]$$

$$\log|S[k]| = \log|H[k]| + \log|E[k]|$$

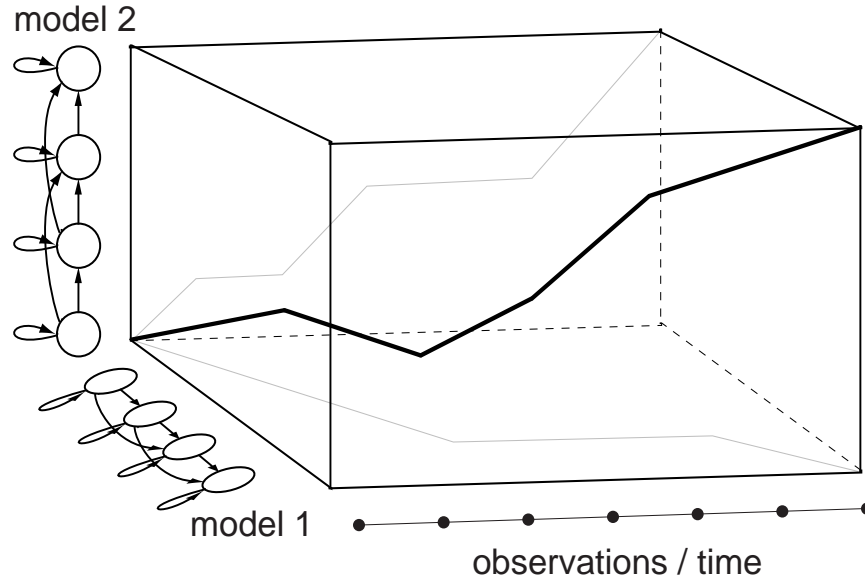
- **Normalize variance to handle added noise?**



HMM decomposition

(e.g. Varga & Moore 1991, Roweis 2000)

- **Total signal model has independent state sequences for 2+ component sources**



- **New combined state space** $q' = \{q_1 q_2\}$
 - new observation pdfs for each combination

$$p(X^i | q_1^i, q_2^i)$$



Problems with HMM decomposition

- $O(q_k)^N$ is exponentially large...
- **Normalization no longer holds!**
 - each source has a different gain
→ model at various SNRs?
 - models typically don't use overall energy C_0
 - each source has a different *channel* $H[k]$
- **Modeling every possible sub-state combination is inefficient, inelegant and impractical**



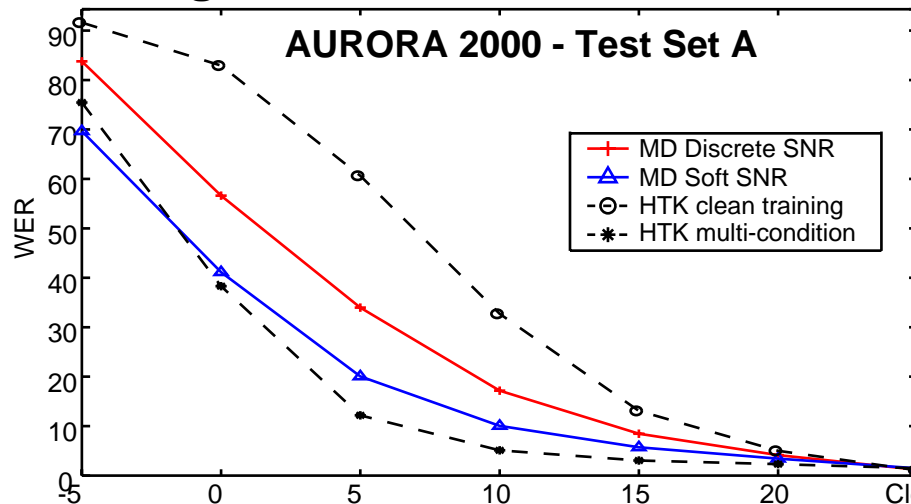
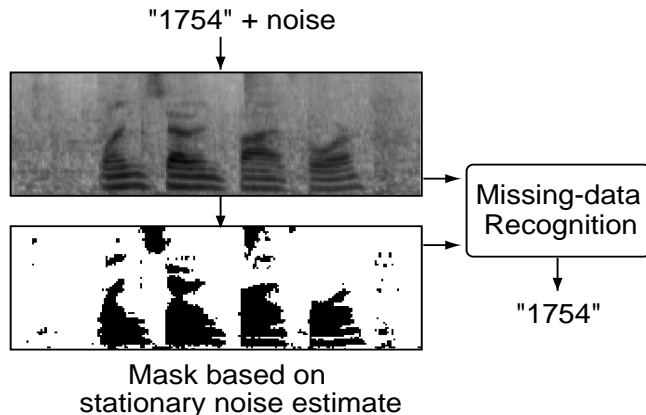
Missing data recognition

(Cooke, Green, Barker @ Sheffield)

- **Energy overlaps in time-freq. hide features**
 - some observations are effectively *missing*
- **Use missing feature theory...**
 - integrate over missing data x_m under model M

$$p(x|M) = \int p(x_p|x_m, M)p(x_m|M)dx_m$$

- **Effective in speech recognition**



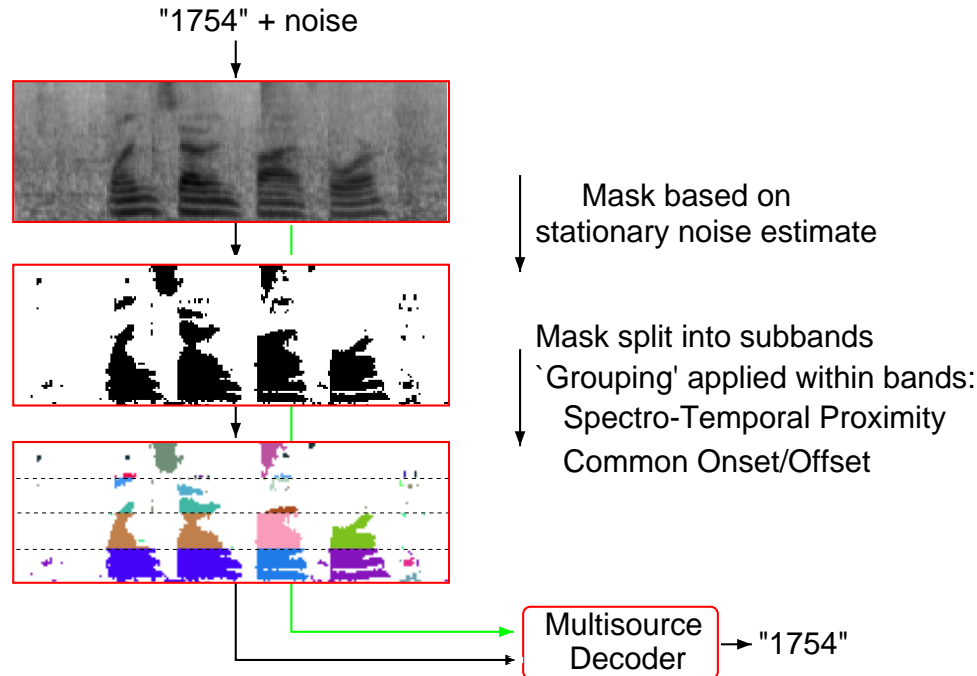
- **Problem: finding the missing data mask**



Maximum-likelihood data mask

(Jon Barker @ Sheffield)

- **Search of sound-fragment interpretations**



- **Decoder searches over data mask K :**

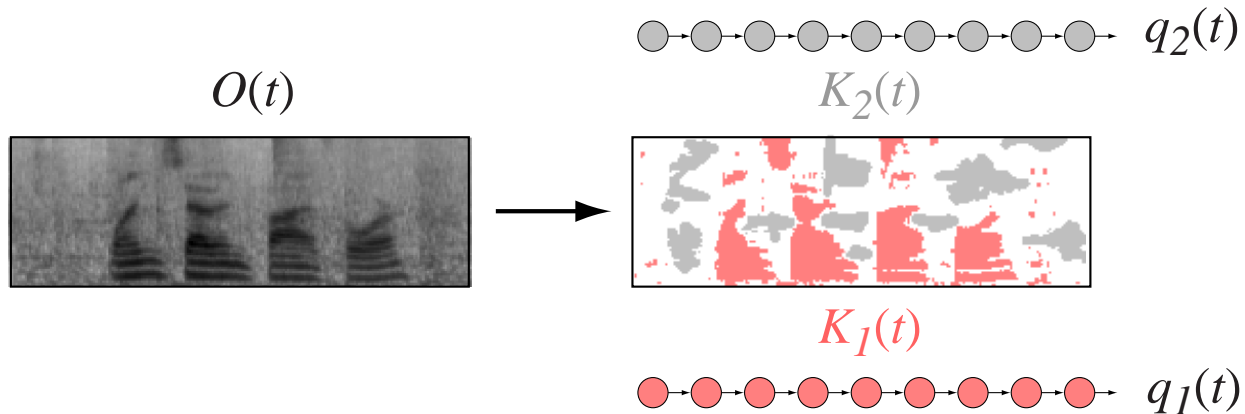
$$p(M, K|x) \propto p(x|K, M)p(K|M)p(M)$$

- how to estimate $p(K)$



Multi-source decoding

- Search for more than one source



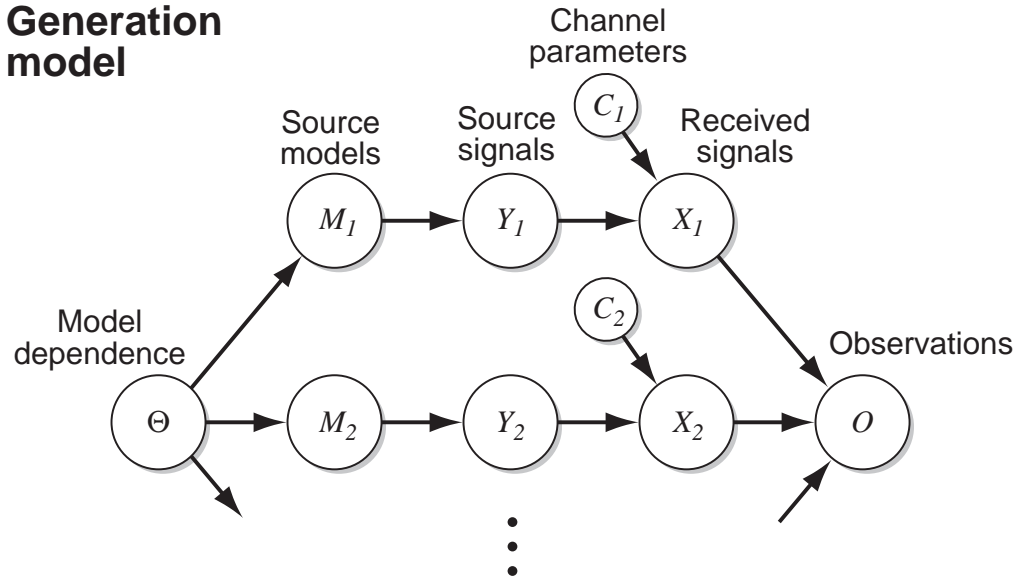
- Mutually-dependent data masks
- Use CASA processing to propose masks
 - locally coherent regions
 - $p(K|q)$
- Theoretical vs. practical limits



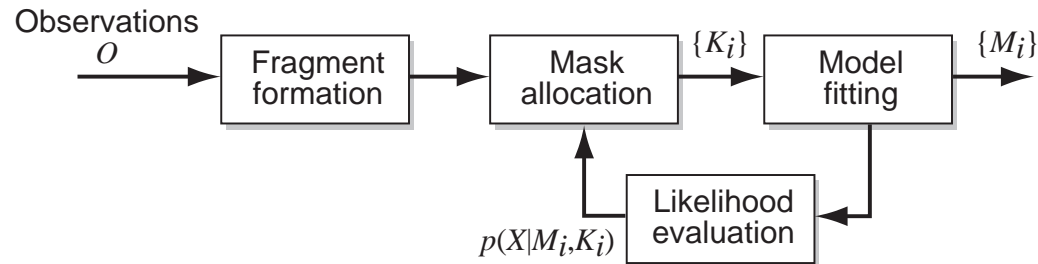
General sound mixtures

- Search for generative explanation:

Generation model



Analysis structure



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- 1 Human sound organization
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- 4 Sound mixture recognition
- 5 Opportunities for learning**
 - learnable aspects of modeling
 - tractable decoding
 - some examples



5

Opportunities for learning

- **Per model feature distributions** $P(Y|M)$
 - e.g. analyzing isolated sound databases
- **Channel modifications** $P(X|Y)$
 - e.g. by comparing multi-mic recordings
- **Signal combinations** $P(O|\{X_i\})$
 - determined by acoustics
- **Patterns of model combinations** $P(\{M_i\})$
 - loose dependence between sources
- **Search for most likely explanations**
$$P(\{M_i\}|O) \propto P(O|\{X_i\})P(\{X_i\}|\{M_i\})P(\{M_i\})$$
- **Short-term structure: repeating events**



Source models

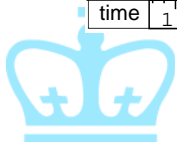
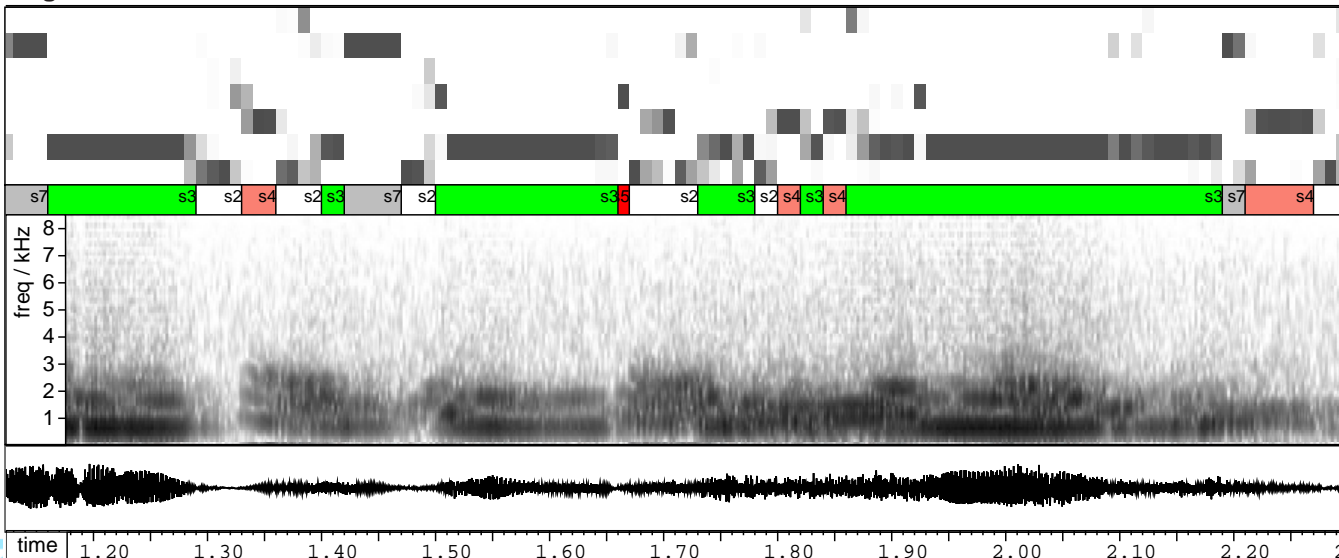
- **The speech recognition lesson:
Use the data as much as possible**
 - what can we do with unlimited data feeds?
- **Data sources**
 - clean data corpora
 - identify near-clean segments in real sound
- **Model types**
 - templates
 - parametric/constraint models
 - HMMs



What are the HMM states?

- No sub-units defined for nonspeech sounds
- Final states depend on EM initialization
 - labels
 - clusters
 - transition matrix
- Have ideas of what we'd like to get
 - investigate features/initialization to get there

dogBarks2



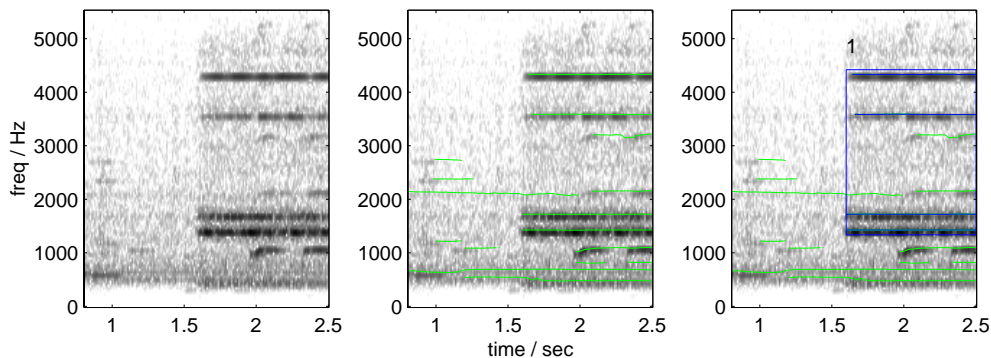
Tractable decoding

- **Speech decoder notionally searches all states**
- **Parametric models give infinite space**
 - need closed-form partial explanations
 - examine residual, iterate, converge
- **Need general cues to get started**
 - return to Auditory Scene Analysis:
 - onsets
 - harmonic patterns
 - then parametric fitting
- **Need multiple hypothesis search, pruning, efficiency tricks**
- **Learning?**
Parameters for new source events
 - e.g. from artificial (hence labeled) mixtures

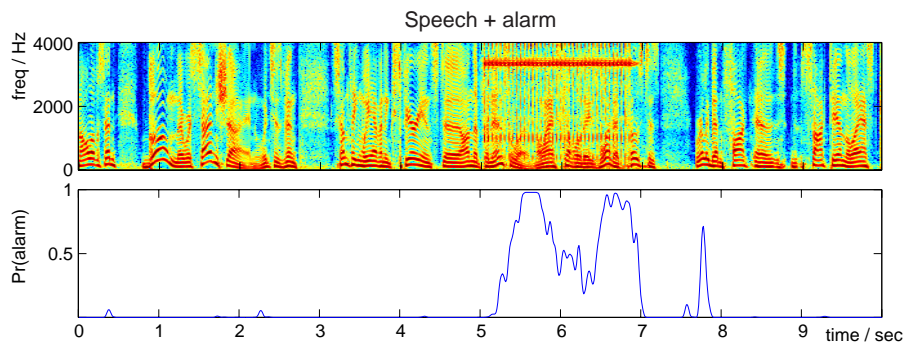


Example: Alarm sound detection

- Alarm sounds have particular structure
 - people 'know them when they hear them'
- Isolate alarms in sound mixtures



- sinusoid peaks have invariant properties

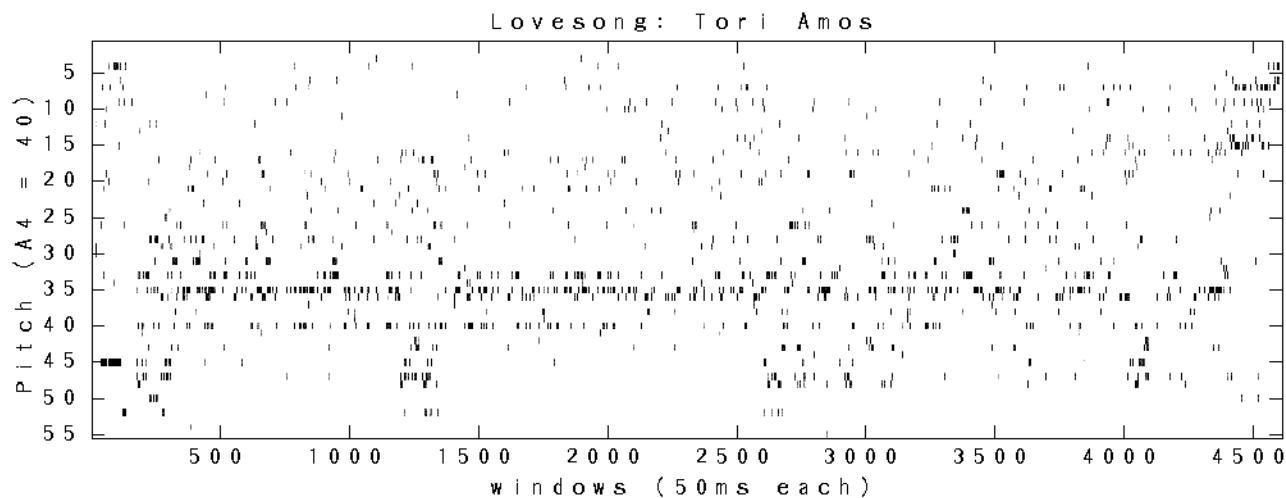


- Learn model parameters from examples



Example: Music transcription (e.g. Masataka Goto)

- **High-quality training material:**
Synthesizer sample kits
- **Ground truth available:**
Musical scores
- **Find ML explanations for scores**
 - guide by multiple pitch tracking (hyp. search)



- **Applications in similarity matching**



Summary

- **Sound contains lots of information**
... but it's always mixed up
- **Psychologists describe ASA**
... but bottom-up computer models don't work
- **Speech recognition works for isolated speech**
... by exploiting top-down, context constraints
- **Speech in mixtures via multiple-source models**
... practical combinatorics are the main problem
- **Generalize this idea for all sounds**
... need models of 'all sounds'
... plus models of channel modification
... plus ways to propose segmentations
... plus missing-data recognition



Further reading

- [BarkCE00] J. Barker, M.P. Cooke & D. Ellis (2000). “Decoding speech in the presence of other sound sources,” *Proc. ICSLP-2000*, Beijing.
<ftp://ftp.icsi.berkeley.edu/pub/speech/papers/icslp00-msd.pdf>
- [Breg90] A.S. Bregman (1990). *Auditory Scene Analysis: the perceptual organization of sound*, MIT Press.
- [Chev00] A. de Cheveigné (2000). “The Auditory System as a Separation Machine,” *Proc. Intl. Symposium on Hearing*.
<http://www.ircam.fr/pcm/cheveign/sh/ps/ATReats98.pdf>
- [CookeE01] M. Cooke, D. Ellis (2001). “The auditory organization of speech and other sources in listeners and computational models,” *Speech Communication* (accepted for publication).
<http://www.ee.columbia.edu/~dpwe/pubs/tcfkas.pdf>
- [Ellis99] D.P.W. Ellis (1999). “Using knowledge to organize sound: The prediction-driven approach to computational auditory scene analysis...,” *Speech Communications* 27.
<http://www.ee.columbia.edu/~dpwe/pubs/spcomcasa98.pdf>
- [Roweis00] S. Roweis (2000). “One microphone source separation.,” *Proc. NIPS 2000*.
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