## Statistical NLP Spring 2009



Berkeley

## Lecture 10: Acoustic Models

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## The Noisy Channel Model



- Search through space of all possible sentences.
- Pick the one that is most probable given the waveform.


## Speech Recognition Architecture



## Digitizing Speech



Thanks to Bryan Pellom for this slide!

## Frame Extraction

- A frame (25 ms wide) extracted every 10 ms


$$
\begin{array}{lll}
\mathbf{a}_{1} & \mathbf{a}_{2} & \mathbf{a}_{3}
\end{array}
$$

Figure from Simon Arnfield

## Mel Freq. Cepstral Coefficients

- Do FFT to get spectral information
- Like the spectrogram/spectrum we saw earlier
- Apply Mel scaling
- Linear below 1 kHz , log above, equal samples above and below 1 kHz
- Models human ear; more sensitivity in lower freqs
- Plus Discrete Cosine Transformation


## Final Feature Vector

- 39 (real) features per 10 ms frame:
- 12 MFCC features
- 12 Delta MFCC features
- 12 Delta-Delta MFCC features
- 1 (log) frame energy
- 1 Delta (log) frame energy
- 1 Delta-Delta (log frame energy)
- So each frame is represented by a 39D vector


## HMMs for Continuous Observations?

- Before: discrete, finite set of observations
- Now: spectral feature vectors are real-valued!
- Solution 1: discretization
- Solution 2: continuous emissions models
- Gaussians
- Multivariate Gaussians
- Mixtures of Multivariate Gaussians
- A state is progressively:
- Context independent subphone (~3 per phone)
- Context dependent phone (=triphones)
- State tying of CD phone


## Vector Quantization

- Idea: discretization
- Map MFCC vectors onto discrete symbols
- Compute probabilities just by counting
- This is called Vector Quantization or VQ
- Not used for ASR any more; too simple
- Useful to consider as a starting point

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## Gaussian Emissions

- VQ is insufficient for real ASR
- Instead: Assume the possible values of the observation vectors are normally distributed.
- Represent the observation likelihood function as a Gaussian with mean $\mu_{j}$ and variance $\sigma_{j}^{2}$

$$
f(x \mid \mu, \sigma)=\frac{1}{\sigma \sqrt{2 \pi}} \exp \left(-\frac{(x-\mu)^{2}}{2 \sigma^{2}}\right)
$$

## Gaussians for Acoustic Modeling

## A Gaussian is parameterized by a mean and

 a variance:

- $P(o \mid q):$



## Multivariate Gaussians

- Instead of a single mean $\mu$ and variance $\sigma$ :

$$
f(x \mid \mu, \sigma)=\frac{1}{\sigma \sqrt{2 \pi}} \exp \left(-\frac{(x-\mu)^{2}}{2 \sigma^{2}}\right)
$$

- Vector of means $\mu$ and covariance matrix $\Sigma$

$$
f(x \mid \mu, \Sigma)=\frac{1}{(2 \pi)^{n / 2}|\Sigma|^{1 / 2}} \exp \left(-\frac{1}{2}(x-\mu)^{T} \Sigma^{-1}(x-\mu)\right)
$$

- Usually assume diagonal covariance
- This isn't very true for FFT features, but is fine for MFCC features


## Gaussian Intuitions: Size of $\Sigma$



- $\mu=\left[\begin{array}{ll}0 & 0\end{array}\right]$
$\mu=\left[\begin{array}{ll}0 & 0\end{array}\right]$
$\mu=\left[\begin{array}{ll}0 & 0\end{array}\right]$
- $\Sigma=1$
$\Sigma=0.61$
$\Sigma=21$
- As $\Sigma$ becomes larger, Gaussian becomes more spread out; as $\Sigma$ becomes smaller, Gaussian more compressed


## Gaussians: Off-Diagonal



$$
\Sigma=\left[\begin{array}{ll}
1 & 0 \\
0 & 1
\end{array}\right] ; \quad \Sigma=\left[\begin{array}{cc}
1 & 0.5 \\
0.5 & 1
\end{array}\right] ; . \Sigma=\left[\begin{array}{cc}
1 & 0.8 \\
0.8 & 1
\end{array}\right]
$$

- As we increase the off diagonal entries, more correlation between value of $x$ and value of $y$


## But we're not there yet

- Single Gaussian may do a bad job of modeling distribution in any dimension:

- Solution: Mixtures of Gaussians


## Scatter Plots




## Mixtures of Gaussians

- M mixtures of Gaussians:

$$
\begin{gathered}
f\left(x \mid \mu_{j k}, \Sigma_{j k}\right)=\sum_{k=1}^{M} c_{j k} N\left(x, \mu_{j k}, \Sigma_{j k}\right) \\
b_{j}\left(o_{t}\right)=\sum_{k=1}^{M} c_{j k} N\left(o_{t}, \mu_{j k}, \Sigma_{j k}\right)
\end{gathered}
$$

- For diagonal covariance:

$$
b_{j}\left(o_{t}\right)=\sum_{k=1}^{M} \frac{c_{j k}}{2 \pi^{D / 2} \prod_{d=1}^{D} \sigma_{j k d}{ }^{2}} \exp \left(-\frac{1}{2} \sum_{d=1}^{D} \frac{\left(x_{j k d}-\mu_{j k d}\right)^{2}}{\sigma_{j k d}{ }^{2}}\right)
$$

## GMMs

- Summary: each state has a likelihood function parameterized by:
- M mixture weights
- M mean vectors of dimensionality D
- Either
- M covariance matrices of DxD
- Or often
- M diagonal covariance matrices of DxD
which is equivalent to
- M variance vectors of dimensionality D


## HMMs for Speech

Word Model

Observation
Sequence (spectral feature vectors)


## Phones Aren't Homogeneous



## Need to Use Subphones

Phone Model


## A Word with Subphones



## ASR Lexicon: Markov Models




## Markov Process with Bigrams



## Training Mixture Models

- Forced Alignment
- Computing the "Viterbi path" over the training data (where the transcription is known) is called "forced alignment"
- We know which word string to assign to each observation sequence.
- We just don't know the state sequence.
- So we constrain the path to go through the correct words (by using a special example-specific language model)
- And otherwise do normal Viterbi
- Result: state sequence!


## Modeling phonetic context



## "Need" with triphone models



## Implications of Cross-Word Triphones

- Possible triphones: $50 \times 50 \times 50=125,000$
- How many triphone types actually occur?
- 20K word WSJ Task (from Bryan Pellom)
- Word internal models: need 14,300 triphones
- Cross word models: need 54,400 triphones
- But in training data only 22,800 triphones occur!
- Need to generalize models.


## State Tying / Clustering

- [Young, Odell, Woodland 1994]
- How do we decide which triphones to cluster together?
- Use phonetic features (or 'broad phonetic classes')
- Stop
- Nasal
- Fricative
- Sibilant
- Vowel
- lateral


## State Tying

- Creating CD phones:
- Start with monophone, do EM training
- Clone Gaussians into triphones
- Build decision tree and cluster Gaussians
- Clone and train mixtures (GMMs


Tie states in each leaf node


## Standard subphone/mixture HMM

$\rightarrow \underset{\sim}{\text { statat }} \rightarrow \rightarrow$ End $\cdots \cdots$ Structure


| Model | Error rate |
| :--- | ---: |
| HMM Baseline | $\mathbf{2 5 . 1 \%}$ |

## Our Model

Fully
Connected



## Refinement of the /ih/-phone



## Refinement of the /ih/-phone



Refinement of the /ih/-phone


## HMM states per phone



Inference


- State sequence:
$d_{1}-d_{6}-d_{6}-d_{4}-a e_{5}-a e_{2}-a e_{3}-a e_{0}-d_{2}-d_{2}-d_{3}-d_{7}-d_{5}$
- Phone sequence:
$d-d-d-d-a e-a e-a e-a e-d-d-d-d-d$
- Transcription

Variational
d - ae - d
Viterbi

