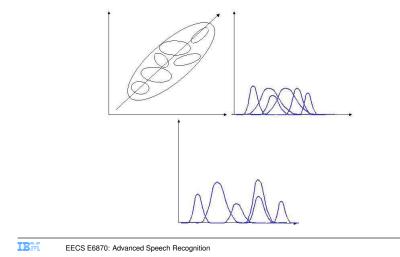
	Outline of Today's Lecture
EECS E6870 - Speech Recognition Lecture 11 Stanley F. Chen, Michael A. Picheny and Bhuvana Ramabhadran IBM T.J. Watson Research Center Yorktown Heights, NY, USA Columbia University stanchen@us.ibm.com, picheny@us.ibm.com, bhuvana@us.ibm.com	<ul> <li>Administrivia</li> <li>Linear Discriminant Analysis</li> <li>Maximum Mutual Information Training</li> <li>ROVER</li> <li>Consensus Decoding</li> </ul>
EECS E6870: Advanced Speech Recognition Administrivia	IBM       EECS E6870: Advanced Speech Recognition       1         Linear Discriminant Analysis
See http://www.ee.columbia.edu/ stanchen/fall09/e6870/readings/project_f09.html for suggested readings and presentation guidelines for final project.	A way to achieve robustness is to extract features that emphasize sound discriminability and ignore irrelevant sources of information. LDA tries to achieve this via a linear transform of the feature data. If the main sources of class variation lie along the coordinate axes there is no need to do anything even if assuming a diagonal covariance matrix (as in most HMM models):

## **Principle Component Analysis-Motivation**

If the main sources of class variation lie along the main source of variation we may want to rotate the coordinate axis (if using diagonal covariances):



## **Eigenvectors and Eigenvalues**

A key concept in feature selection are the eigenvalues and eigenvectors of a matrix.

The eigenvalues and eigenvectors of a matrix are defined by the following matrix equation:

 $\mathbf{A}\mathbf{x} = \lambda\mathbf{x}$ 

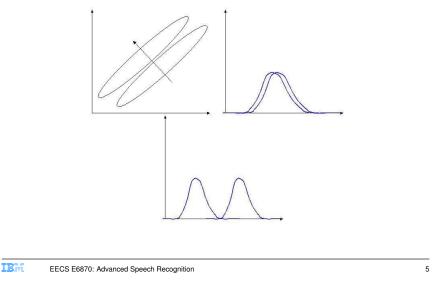
For a given matrix A the eigenvectors are defined as those vectors x for which the above statement is true. Each eigenvector has an associated eigenvalue,  $\lambda$ . To solve this equation, we can rewrite it as

$$(\mathbf{A} - \lambda \mathbf{I})\mathbf{x} = 0$$

If xis non-zero, the only way this equation can be solved is if the determinant of the matrix  $({\bf A}-\lambda {\bf I})$  is zero. The determinant of

## **Linear Discriminant Analysis - Motivation**

If the main sources of class variation do NOT lie along the main source of variation we need to find the best directions:



this matrix is a polynomial (called the *characteristic polynomial*)  $p(\lambda)$ . The roots of this polynomial will be the eigenvalues of **A**. For example, let us say

$$\mathbf{A} = \left[ \begin{array}{cc} 2 & -4 \\ -1 & -1 \end{array} \right]$$

In such a case,

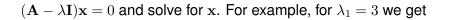
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$$p(\lambda) = \begin{vmatrix} 2-\lambda & -4 \\ -1 & -1-\lambda \end{vmatrix}$$
$$= (2-\lambda)(-1-\lambda) - (-4)(-1)$$
$$= \lambda^2 - \lambda - 6$$
$$= (\lambda - 3)(\lambda + 2)$$

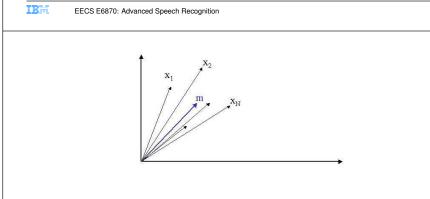
Therefore,  $\lambda_1 = 3$  and  $\lambda_2 = -2$  are the eigenvalues of A.

To find the eigenvectors, we simply plug in the eigenvalues into



[2-3]	-4 ]	$\begin{bmatrix} x_1 \end{bmatrix}$	$\begin{bmatrix} 0 \end{bmatrix}$
$\begin{bmatrix} -1 \end{bmatrix}$	$\begin{bmatrix} -4 \\ -1 - 3 \end{bmatrix}$	$\begin{bmatrix} x_2 \end{bmatrix} =$	

Solving this, we find that  $x_1 = -4x_2$ , so all the eigenvector corresponding to  $\lambda_1 = 3$  is a multiple of  $[-4 \ 1]^T$ . Similarly, we find that the eigenvector corresponding to  $\lambda_1 = -2$  is a multiple of  $[1 \ 1]^T$ .



Now, let  ${\bf e}$  be a unit vector in an arbitrary direction. In such a case, we can express a vector  ${\bf x}$  as

 $\mathbf{x} = \mathbf{m} + a\mathbf{e}$ 

For the vectors  $\mathbf{x}_k$  we can find a set of  $a_k$ s that minimizes the

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## **Principle Component Analysis-Derivation**

First consider the problem of best representing a set of vectors  $\mathbf{x}_1, \mathbf{x}_2, \ldots, \mathbf{x}_n$  by a single vector  $\mathbf{x}_0$ . More specifically let us try to minimize the sum of the squared distances from  $\mathbf{x}_0$ 

$$J_0(\mathbf{x}_0) = \sum_{k=1}^N |\mathbf{x}_k - \mathbf{x}_0|^2$$

It is easy to show that the sample mean, m, minimizes  $J_0$ , where m is given by

$$\mathbf{m} = \mathbf{x}_0 = \frac{1}{N} \sum_{k=1}^{N} \mathbf{x}_k$$

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mean square error:

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$$J_1(a_1, a_2, \dots, a_N, \mathbf{e}) = \sum_{k=1}^N |\mathbf{x}_k - (\mathbf{m} + a_k \mathbf{e})|^2$$

If we differentiate the above with respect to  $a_k$  we get

$$a_k = \mathbf{e}^T (\mathbf{x}_k - \mathbf{m})$$

i.e. we project  $\mathbf{x}_k$  onto the line in the direction of  $\mathbf{e}$  that passes through the sample mean  $\mathbf{m}$ . How do we find the best direction  $\mathbf{e}$ ? If we substitute the above solution for  $a_k$  into the formula for the overall mean square error we get after some manipulation:

$$J_1(\mathbf{e}) = -\mathbf{e}^T \mathbf{S} \mathbf{e} + \sum_{k=1}^N |\mathbf{x}_k - \mathbf{m}|^2$$

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where S is called the *Scatter* matrix and is given by:

$$\mathbf{S} = \sum_{k=1}^{N} (\mathbf{x}_k - \mathbf{m}) (\mathbf{x}_k - \mathbf{m})^T$$

Notice the scatter matrix just looks like N times the sample covariance matrix of the data. To minimize  $J_1$  we want to maximize  $e^T Se$  subject to the constraint that  $|e| = e^T e = 1$ . Using Lagrange multipliers we write

$$u = \mathbf{e}^T \mathbf{S} \mathbf{e} - \lambda \mathbf{e}^T \mathbf{e}$$

. Differentiating u w.r.t e and setting to zero we get:

$$2\mathbf{S}\mathbf{e} - 2\lambda\mathbf{e} = 0$$

or

 $\mathbf{Se} = \lambda \mathbf{e}$ 

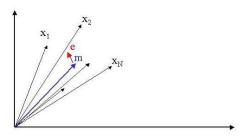
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In this case, we can write the mean square error as

$$J_d = \sum_{k=1}^N |(\mathbf{m} + \sum_{i=1}^d a_{ki} \mathbf{e}_i) - \mathbf{x}_k|^2$$

and it is not hard to show that  $J_d$  is minimized when the vectors  $e_1, e_2, \ldots, e_d$  correspond to the *d* largest eigenvectors of the scatter matrix S.

So to maximize  $e^T S e$  we want to select the eigenvector of S corresponding to the largest eigenvalue of S.



If we now want to find the best d directions, the problem is now to express  ${\bf x}$  as

$$\mathbf{x} = \mathbf{m} + \sum_{i=1}^d a_i \mathbf{e}_i$$

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## **Linear Discriminant Analysis - Derivation**

Let us say we have vectors corresponding to c classes of data. We can define a set of scatter matrices as above as

$$\mathbf{S}_i = \sum_{x \in \mathcal{D}_i} (\mathbf{x} - \mathbf{m}_i) (\mathbf{x} - \mathbf{m}_i)^T$$

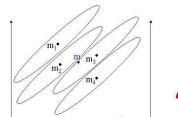
where  $\mathbf{m}_i$  is the mean of class *i*. In this case we can define the within-class scatter (essentially the average scatter across the classes relative to the mean of each class) as just:

$$\mathbf{S}_W = \sum_{i=1}^c \mathbf{S}_i$$

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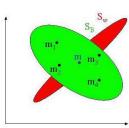




Another useful scatter matrix is the between class scatter matrix, defined as

$$\mathbf{S}_B = \sum_{i=1}^{c} (\mathbf{m}_i - \mathbf{m}) (\mathbf{m}_i - \mathbf{m})^T$$

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A reasonable measure of discriminability is the ratio of the volumes represented by the scatter matrices. Since the determinant of a matrix is a measure of the corresponding volume, we can use the ratio of determinants as a measure:

$$J = \frac{|\mathbf{S}_B|}{|\mathbf{S}_W|}$$

So we want to find a set of directions that maximize this expression. In the new space, we can write the above expression We would like to determine a set of projection directions V such that the classes c are maximally discriminable in the new coordinate space given by

 $\tilde{\mathbf{x}} = \mathbf{V}\mathbf{x}$ 

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as:

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$$\tilde{\mathbf{S}}_B = \sum_{i=1}^c (\tilde{\mathbf{m}}_i - \tilde{\mathbf{m}}) (\tilde{\mathbf{m}}_i - \tilde{\mathbf{m}})^T$$
$$= \sum_{i=1}^c \mathbf{V} (\mathbf{m}_i - \mathbf{m}) (\mathbf{m}_i - \mathbf{m})^T \mathbf{V}^T$$
$$= \mathbf{V} \mathbf{S}_B \mathbf{V}^T$$

and similarly for  $S_W$  so the discriminability measure becomes

$$J(\mathbf{V}) = \frac{|\mathbf{V}\mathbf{S}_B\mathbf{V}^T|}{|\mathbf{V}\mathbf{S}_W\mathbf{V}^T|}$$

With a little bit of manipulation similar to that in PCA, it turns out that the solution are the eigenvectors of the matrix

$$\mathbf{S}_W^{-1}\mathbf{S}_B$$

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which can be generated by most common mathematical packages.	Linear Discriminant Analysis in Speech Recognition
	The most successful uses of LDA in speech recognition are achieved in an interesting fashion.
	<ul> <li>Speech recognition training data are aligned against the underlying words using the Viterbi alignment algorithm described in Lecture 4.</li> </ul>
	<ul> <li>Using this alignment, each cepstral vector is tagged with a different phone or sub-phone. For English this typically results in a set of 156 (52x3) classes.</li> </ul>
	• For each time $t$ the cepstral vector $\mathbf{x}_t$ is spliced together with $N/2$ vectors on the left and right to form a "supervector" of $N$ cepstral vectors. ( $N$ is typically 5-9 frames.) Call this "supervector" $\mathbf{y}_t$ .
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	Training via Maximum Mutual Information
	The Fundamental Equation of Speech Recognition states that
<b>/</b>	p(S O) = p(O S)p(S)/P(O)
• The LDA procedure is applied to the supervectors $\mathbf{y}_t$ .	where <i>S</i> is the sentence and <i>O</i> are our observations. We model $p(O S)$ using Hidden Markov Models (HMMs). The HMMs themselves have a set of parameters $\theta$ that are estimated from a set of training data, so it is convenient to write this dependence explicitly: $p_{\theta}(O S)$ .
<ul> <li>The top M directions (usually 40-60) are chosen and the supervectors y<sub>t</sub> are projected into this lower dimensional space.</li> </ul>	We estimate the parameters $\theta$ to maximize the likelihood of the training data. Although this seems to make some intuitive sense, is this what we are after?
<ul> <li>The recognition system is retrained on these lower dimensional vectors.</li> <li>Performance improvements of 10%-15% are typical.</li> </ul>	Not really! (Why?). So then, why is ML estimation a good thing?

## **Maximum Likelihood Estimation Redux**

ML estimation results in a function that allows us to estimate parameters of the desired distribution from observed samples of the distribution. For example, in the Gaussian case:

$$\hat{\mu}_{\mathsf{MLE}} = \frac{1}{n} \sum_{k=1}^{n} \mathbf{x}_{k}$$
$$\hat{\Sigma}_{\mathsf{MLE}} = \frac{1}{n} \sum_{k=1}^{n} (\mathbf{x}_{k} - \hat{\mu}_{\mathsf{MLE}}) (\mathbf{x}_{k} - \hat{\mu}_{\mathsf{MLE}})^{T}$$

Since  $\mu$  and  $\Sigma$  themselves are computed from the random variables  $\mathbf{x}_k$  we can consider them to be random variables as well.

More generally we can consider the estimate of the parameters  $\theta$  as a random variable. The function that computes this estimate is called an *estimator*.

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This means the ML estimate on the average will produce the closest estimate to the true parameters of the system.

If we assume that the system has its best performance when the parameters match the true parameters, then the ML estimate will, on average, perform as good as or better than any other estimator.

Any estimator, maximum likelihood or other, since it is a random variable, has a mean and a variance. It can be shown that if

- The sample is actually drawn from the assumed family of distributions
- The family of distributions is well-behaved
- The sample is large enough

then, the maximum likelihood estimator has a Gaussian distribution with the following good properties:

- The mean converges to the true mean of the parameters (consistent)
- The variance has a particular form and is just a function of the true mean of the parameters and the samples (Fisher information)
- No other consistent estimator has a lower variance

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## Main Problem with Maximum Likelihood Estimation

The true distribution of speech is (probably) not generated by an HMM, at least not of the type we are currently using. (How might we demonstrate this?)

Therefore, the optimality of the ML estimate is not guaranteed and the parameters estimated may not result in the lowest error rates.

A reasonable criterion is rather than maximizing the likelihood of the data given the model, we try to maximize the a posteriori probability of the model given the data (Why?):

$$\theta_{\mathsf{MAP}} = \operatorname*{arg\,max}_{\theta} p_{\theta}(S|O)$$

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#### **MMI Estimation**

Let's look at the previous equation in more detail. It is more convenient to look at the problem as maximizing the logarithm of the a posteriori probability across all the sentences:

$$\begin{aligned} \theta_{\mathsf{MMI}} &= \arg \max_{\theta} \sum_{i} \log p_{\theta}(S_{i} | \mathbf{O}_{i}) \\ &= \arg \max_{\theta} \sum_{i} \log \frac{p_{\theta}(\mathbf{O}_{i} | S_{i}) p(S_{i})}{p_{\theta}(\mathbf{O}_{i})} \\ &= \arg \max_{\theta} \sum_{i} \log \frac{p_{\theta}(\mathbf{O}_{i} | S_{i}) p(S_{i})}{\sum_{j} p_{\theta}(\mathbf{O}_{i} | S_{i}^{j}) p(S_{i}^{j})} \end{aligned}$$

where  $S_i^j$  refers to the *j*th possible sentence hypothesis given a set of acoustic observations  $O_i$ 

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#### **Comparison to ML Estimation**

In ordinary ML estimation, the objective is to find  $\theta$ :

$$\theta_{\mathsf{ML}} = \operatorname*{arg\,max}_{\theta} \sum_{i} \log p_{\theta}(\mathbf{O}_{i}|S_{i})$$

Therefore, in ML estimation, for each i we only need to make computations over the correct sentence  $S_i$ . In MMI estimation, we need to worry about computing quanitities over all possibile sentence hypotheses - a much more computationally intense process.

Another advantage of ML over MMI is that there exists a relatively simple algorithm - the forward-backward, or Baum-Welch, algorithm, for iteratively estimating  $\theta$  that is guaranteed to converge. When originally formulated, MMI training had to be done by painful gradient search.

Why is this Called MMI Estimation?

There is a quantity in information theory called the Mutual Information. It is defined as:

$$E\left[\log\frac{p(X,Y)}{p(X)p(Y)}\right]$$

Since  $p(S_i)$  does not depend on  $\theta$ , the term can be dropped from the previous set of equations, in which case the estimation formula looks like the expression for mutual information, above.

When originally derived by Brown[1], the formulation was actually in terms of mutual information, hence the name. However, it is easier to quickly motivate in terms of maximizing the a posteriori probability of the answers.

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#### **MMI Training Algorithm**

A big breakthrough in the MMI area occured when it was shown that a forward-backward-like algorithm existed for MMI training [2]. The derivation is complex but the resulting esitmation formulas are surprisingly simple. We will just give the results for the estimation of the means in a Gaussian HMM framework.

The MMI objective function is

$$\sum_{i} \log \frac{p_{\theta}(\mathbf{O}_{i}|S_{i})p(S_{i})}{\sum_{j} p_{\theta}(\mathbf{O}_{i}|S_{i}^{j})p(S_{i}^{j})}$$

We can view this as comprising two terms, the numerator, and the denominator. We can increase the objective function in two ways:

- Increase the contribution from the numerator term
- Decrease the contribution from the denominator term

Basic idea:

- Collect estimation counts from both the numerator and denominator terms
- Increase the objective function by subtracting the denominator counts from the numerator counts.

More specifically, let:

$$\begin{array}{lll} \theta_{mk}^{num} & = & \displaystyle \sum_{i,t} \mathbf{O}_i(t) \gamma_{mki}^{num}(t) \\ \\ \theta_{mk}^{den} & = & \displaystyle \sum_{i,t} \mathbf{O}_i(t) \gamma_{mki}^{den}(t) \end{array}$$

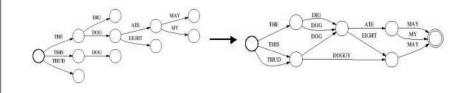
where  $\gamma_{mki}^{num}(t)$  are the counts for state k, mixture component m, computed from running the forward-backward algorithm on the "correct" sentence  $S_i$  and  $\gamma_{mki}^{den}(t)$  are the counts computed across all the sentence hypotheses corresponding to  $S_i$  The MMI

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## **Computing the Denominator Counts**

The major component of the MMI calculation is the computation of the denominator counts. Theoretically, we must compute counts for every possible sentence hypotheis. How can we reduce the amount of computation?

1. From the previous lectures, realize that the set of sentence hypotheses are just captured by a large HMM for the entire sentence:



estimate for  $\mu_{mk}$  is:

$$\mu_{km} = \frac{\theta_{mk}^{num} - \theta_{mk}^{den} + D_{mk}\mu'_{mk}}{\gamma_{mk}^{num} - \gamma_{mk}^{den} + D_{mk}}$$

The factor  $D_{mk}$  is chose large enough to avoid problems with negative count differences. Notice that ignoring the denominator counts results in the normal mean estimate. A similar expression exists for variance estimation.

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Counts can be collected on this HMM the same way counts are collected on the HMM representing the sentence corresponding to the correct path.

2. Use a ML decoder to generate a "reasonable" number of sentence hypotheses and then use FST operations such as determinization and minimization to compactify this into an HMM graph (*lattice*).

3. Do not regenerate the lattice after every MMI iteration.

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## **Other Computational Issues**

Because we ignore correlation, the likelihood of the data tends to be dominated by a very small number of lattice paths (Why?). To increase the number of confusable paths, the likelihoods are scaled with an exponential constant:

$$\sum_{i} \log \frac{p_{\theta}(\mathbf{O}_{i}|S_{i})^{\kappa} p(S_{i})^{\kappa}}{\sum_{j} p_{\theta}(\mathbf{O}_{i}|S_{i}^{j})^{\kappa} p(S_{i}^{j})^{\kappa}}$$

For similar reasons, a weaker language model (unigram) is used to generate the denominator lattice. This also simplifies denominator lattice generation.

Variations and Embellishments

## Results

MMIE	%WER		
Iteration	eval97sub	eval98	
0 (MLE)	44.4	45.6	
1	42,4	43.7	
1 (3xCHE)	42.0	43.5	
2	41.8	42.9	
2 (3xCHE)	41.9	42.7	

Table 6: Word error rates when using h5train00 training with and without CHE data weighting (3xCHE).

Adaptation	% WER eval98		
a so antos concercios y	MLE	MMIE	
None	44.6	42.5	
MLLR	42.1	39.9	

Table 8: Effect of MLLR on MLE and MMIE trained models.

Note that results hold up on a variety of other tasks as well.

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MPE

$$\sum_{i} \frac{\sum_{j} p_{\theta}(\mathbf{O}_{i}|S_{j})^{\kappa} p(S_{j})^{\kappa} A(S_{ref}, S_{j})}{\sum_{j} p_{\theta}(\mathbf{O}_{i}|S_{i}^{j})^{\kappa} p(S_{i}^{j})^{\kappa}}$$

- A(S<sub>ref</sub>, S<sub>j</sub> is a phone-frame accuracy function. A measures the number of correctly labeled frames in S
- Povey [3] showed this could be optimized in a way similar to that of MMI.
- Usually works somewhat better than MMI itself



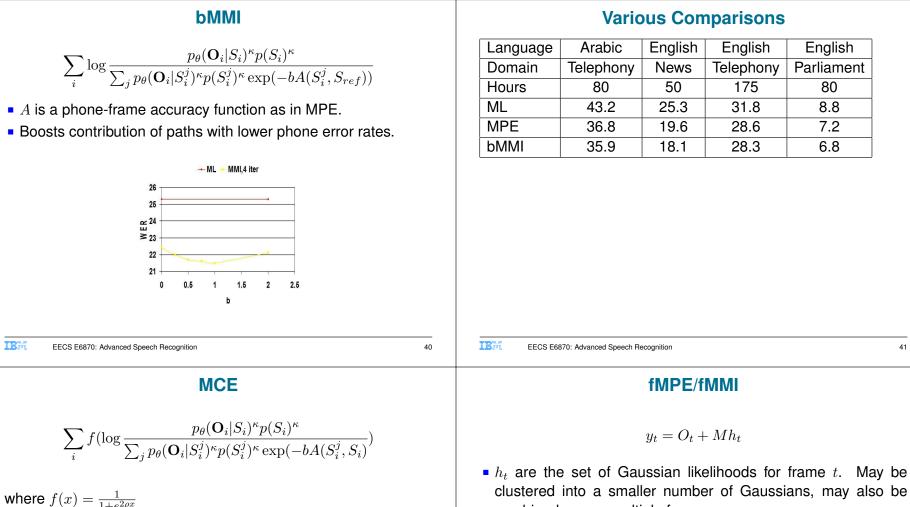
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- bMMI Boosted MMI
- MCE Minimum Classification Error

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fMPE/fMMI - feature-based MPE and MMI

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- The sum over competing models explicitly excludes the correct class (unlike the other variations)
- Approximates sentence error rate on training data
- Originally developed for grammar-based applications
- Comparable to MPE, never compared to bMMI

clustered into a smaller number of Gaussians, may also be combined across multiple frames.

- The training of M is exceedingly complex involving both the gradients of your favorite objective function with respect to Mas well as the model parameters  $\theta$  with multiple passes through the data.
- Rather amazingly gives significant gains both with and without MMI.

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80

8.8

7.2

6.8

## fMPE/fMMI Results

English BN 50 Hours, SI models

	RT03	DEV04f	RT04
ML	17.5	28.7	25.3
fBMMI	13.2	21.8	19.2
fbMMI+ bMMI	12.6	21.1	18.2

#### Arabic BN 1400 Hours, SAT Models

	DEV07	EVAL07	EVAL06
ML	17.1	19.6	24.9
fMPE	14.3	16.8	22.3
fMPE+ MPE	12.6	14.5	20.1

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## ROVER - Recognizer Output Voting Error Reduction[1]

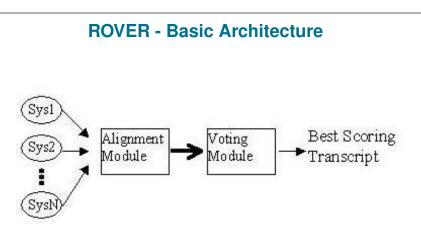
ROVER is a technique for combining recognizers together to improve recognition accuracy. The concept came from the following set of observations about 11 years ago:

- Compare errors of recognizers from two different sites
- Error rate performance similar 44.9% vs 45.1%
- Out of 5919 total errors, 738 are errors for only recognizer A and 755 for only recognizer B
- Suggests that some sort of voting process across recognizers might reduce the overall error rate

## References

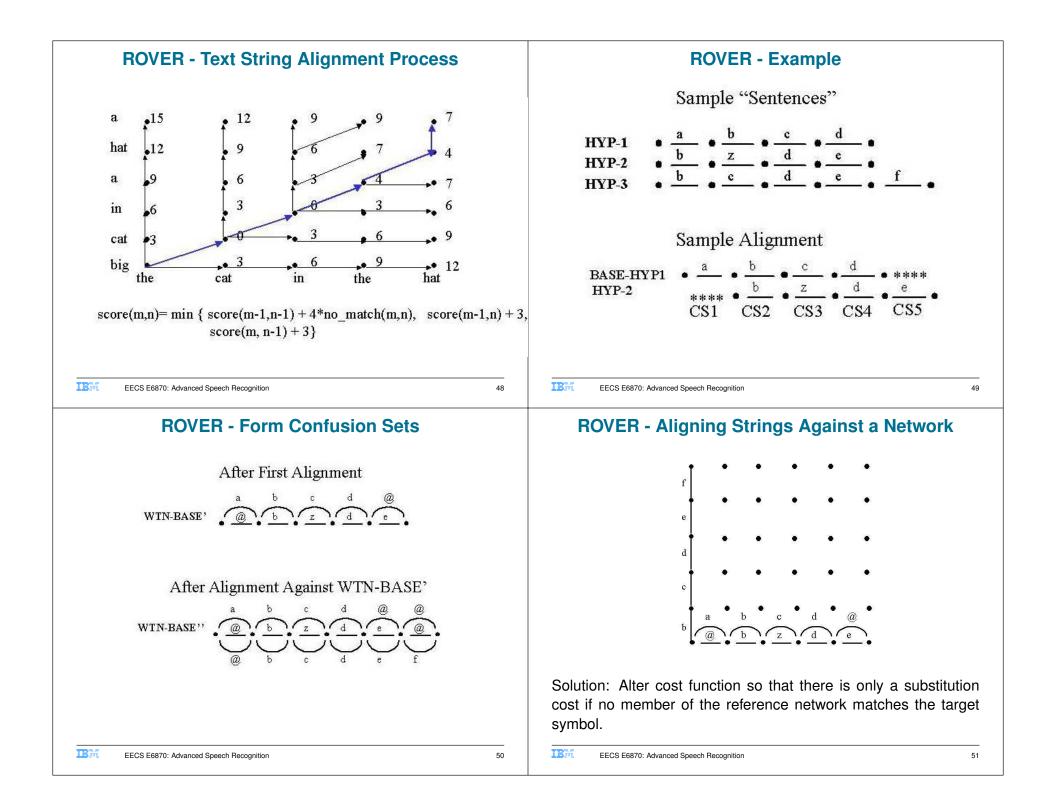
- [1] P. Brown (1987) "The Acoustic Modeling Problem in Automatic Speech Recognition", PhD Thesis, Dept. of Computer Science, Carnegie-Mellon University.
- [2] P.S. Gopalakrishnan, D. Kanevsky, A. Nadas, D. Nahamoo (1991) " An Inequality for Rational Functions with Applications to Some Statistical Modeling Problems", IEEE Trans. on Acoustics, Speech and Signal Processing, 37(1) 107-113, January 1991
- [3] D. Povey and P. Woodland (2002) "Minimum Phone Error and i-smoothing for improved discriminative training", Proc. ICASSP vol. 1 pp 105-108.

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- Systems may come from multiple sites
- Can be a single site with different processing schemes

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<b>ROVER - Aligning Networks Against Networks</b>	ROVER - Vote
No so much a ROVER issue but will be important for confusion networks. Problem: How to score relative probabilities and deletions? Solution: $cost\_subst(s_1,s_2) = (1 - p_1(winner(s_2)) + 1 - p_2(winner(s_1))/2$	<ul> <li>Main Idea: for each confusion set, take word with highest frequency</li> <li><u>SYS1 SYS2 SYS3 SYS4 SYS5 ROVER</u> 44.9 45.1 48.7 48.9 50.2 39.7</li> <li>Improvement very impressive - as large as any significant algorithm advance.</li> </ul>
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ROVER - Example	ROVER - As a Function of Number of Systems [2]
bbn1.ctm       there's       a       lot       of       @       like       societies       @       ruin       engineers       and       lakes         cmu-isl1.ctm       there's       the       labs       @       like       societies       @       ruin       engineers       and       lakes         was       @       alive       @       like       societies       @       true       of       engineers       and       like         sril.ctm       there's       a       lot       of       @       like       societies       @       true       engineers       and       like         sril.ctm       there's       a       lot       of       @       like       society's       for       women       engineers       and       like	Figure 1: 1998 Broadcast News word error rates in function of

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#### **References ROVER - Types of Systems to Combine** ML and MMI [1] J. Fiscus (1997) "A Post-Processing System to Yield Varying amount of acoustic context in pronunciation models Reduced Error Rates", IEEE Workshop on Automatic Speech (Triphone, Quinphone) Recognition and Understanding, Santa Barbara, CA Different lexicons [2] H. Schwenk and J.L. Gauvain (2000) "Combining Multiple Speech Recognizers using Voting and Language Model Different signal processing schemes (MFCC, PLP) Information" ICSLP 2000, Beijing II pp. 915-918 Anything else you can think of! Rover provides an excellent way to achieve cross-site collaboration and synergy in a relatively painless fashion. IBM IBM EECS E6870: Advanced Speech Recognition 56 EECS E6870: Advanced Speech Recognition 57 **Consensus Decoding[1] - Introduction Consensus Decoding - Motivation** Problem TABLE I: Example illustrating the difference between sentence and word error measures. Standard SR evaluation procedure is word-based Standard hypothesis scoring functions are sentence-based

#### Goal

Explicitly minimize word error metric:

$$\widehat{W} = \underset{W}{\operatorname{arg\,min}} E_{P(R|A)}[WE(W,R)] = \underset{W}{\operatorname{arg\,min}} \sum_{R} P(R|A) WE(W,R)$$

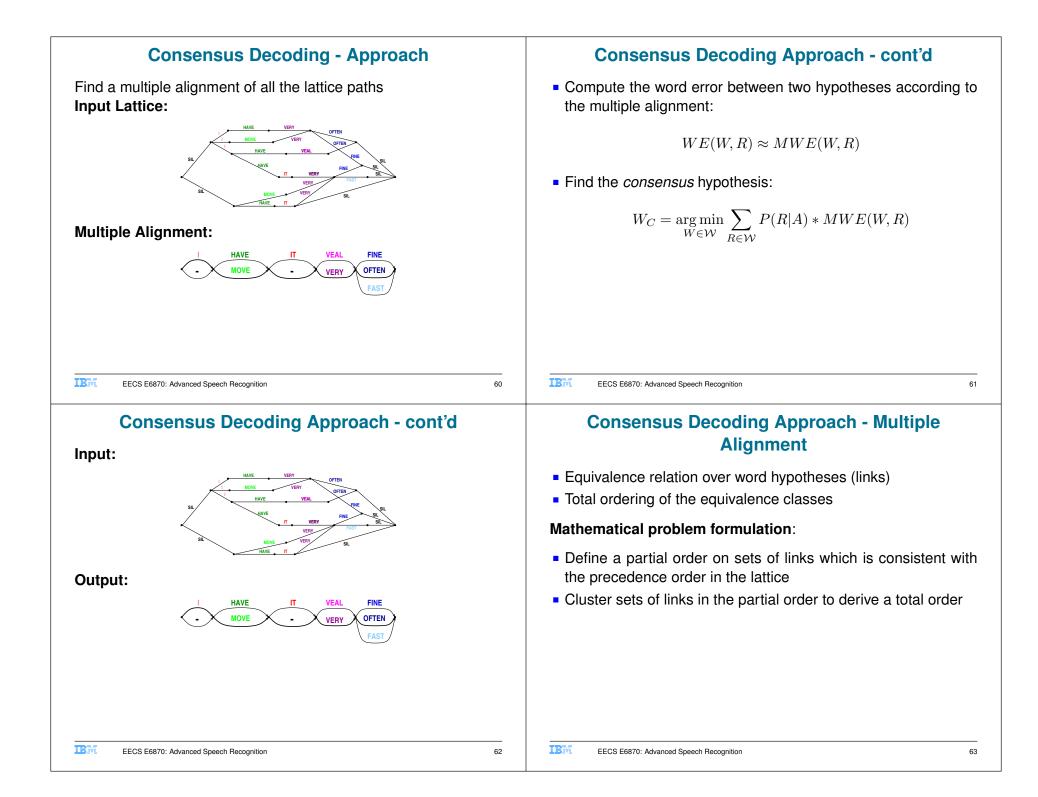
• For each candidate word, sum the word posteriors and pick the word with the highest posterior probability.

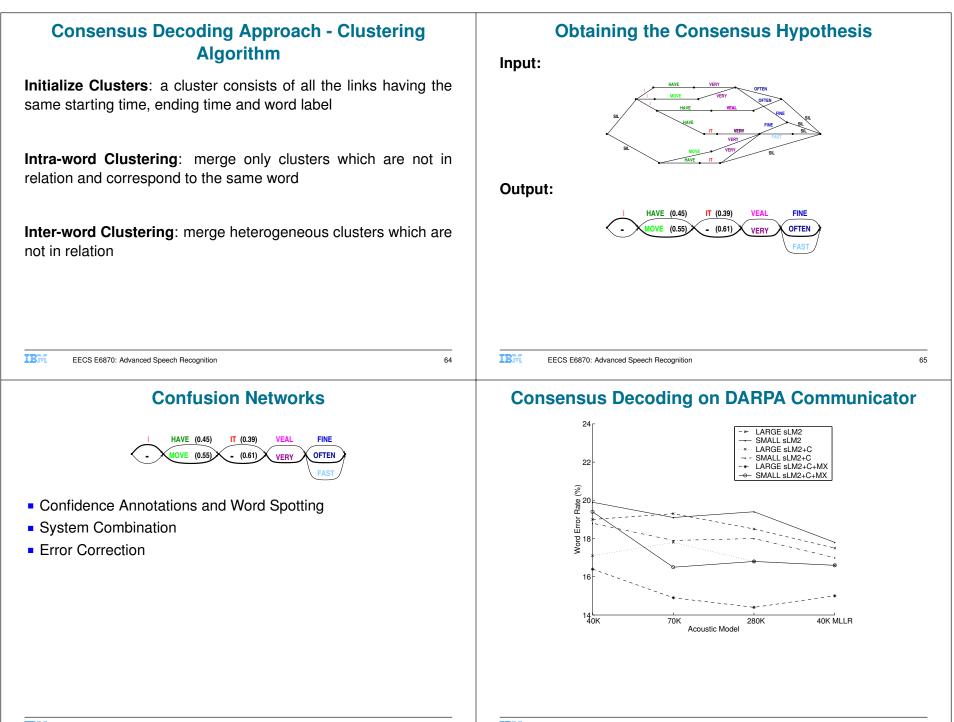
	Hypothe	sis(H)					
$w_1$	$w_2$	$w_3$	P(H A)	$P(w_1 A)$	$P(w_2 A)$	$P(w_3 A)$	E[correct]
I	DØ	INSIDE	0.16	0-34	0-29	0.16	0.79
I	DÖ	FINE	0.13	0-34	0.29	0.28	0.91
BY	DOING	FINE	0-11	0-45	0-49	0.28	1.22
BY	DÖING	WELL	0.11	0-45	0-49	0.11	1.05
BY	DÖING	SIGHT	0-10	0-45	0-49	0.10	1.04
BY	DÖING	BYE	0.07	0-45	0-49	0.07	1-01
BY	DÖING	THOUGHT	0.05	0-45	0-49	0.07	0.99
I	DOING	FINE	0-04	0-34	0-49	0.28	1.11
I	DON'T	BUY	0.01	0-34	0-01	0.01	0.36
BY	DÖING	FUN	0.01	0-45	0-49	0.01	0.95

Original work was done off N-best lists

Lattices much more compact and have lower oracle error rates

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## **Consensus Decoding on Broadcast News**

		Word Error Rate (%)						
	Avg	F0	F1	F2	F3	F4	F5	FX
C-	16.5	8.3	18.6	27.9	26.2	10.7	22.4	23.7
C+	16.0	8.5	18.1	26.1	25.8	10.5	18.8	22.5
		Word Error Rate (%)						
	Avg	F0	F1	F2	F3	F4	F5	FX
C-	14.0	8.6	15.8	19.4	15.3	16.0	5.7	44.8
C+	13.6	8.5	15.7	18.6	14.6	15.3	5.7	41.1

## **Consensus Decoding on Voice Mail**

	Word Error Rate (%)		
System	Baseline	Consensus	
S-VM1	30.2	28.8	
S-VM2	33.7	31.2	
S-VM3	42.4	41.6	
ROVER	29.2	28.5	

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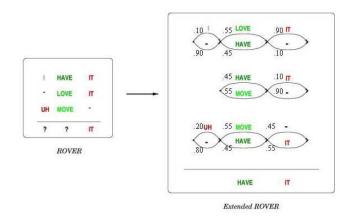
#### System Combination Using Confusion Networks

If we have multiple systems, we can combine the concept of ROVER with confusion networks as follows:

- Use the same process as ROVER to align confusion networks
- Take the overall confusion network and add the posterior probabilities for each word.
- For each confusion set, pick the word with the highest summed posteriors.

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## System Combination Using Confusion Networks



# (b) System Combination

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<section-header><section-header><section-header><section-header></section-header></section-header></section-header></section-header>	References [1] L. Mangu, E. Brill and A. Stolcke (2000) "Finding consensus in speech recognition: word error minimization and other applications of confusion networks", Computer Speech and Language 14(4), 373-400.
<ul> <li>ECS E6870: Advanced Speech Recognition</li> <li>COURSE FEEDBACK</li> <li>Was this lecture mostly clear or unclear? What was the muddiest topic?</li> <li>Other feedback (pace, content, atmosphere)?</li> </ul>	TEM EECS E6870: Advanced Speech Recognition 73
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