

The USTC Systems for The NIST-2005 Speaker Recognition Evaluation



Beiqian Dai, Yanlu Xie, Xi zhou,
Zhiqiang Yao, Jixu Chen, Minghui Liu



Introduction

**Participant
Task:**

		Test Segment Condition			
		10 sec 2-chan	1 conv 2-chan	1 conv summed- chan	1 conv aux mic
Training Condition	10 seconds 2-channel	<input type="radio"/>	<input type="radio"/>		
	1 conversation 2-channel	<input type="radio"/>	<input type="radio"/>		
	3 conversation 2-channel	<input type="radio"/>	<input type="radio"/>		
	8 conversation 2-channel	<input type="radio"/>	<input type="radio"/>		
	3 conversation summed- channel		<input type="radio"/>	<input type="radio"/>	



USTC SSIP Lab.

One Speaker System





Main Modules

- FrontEnd Processing
- Universal Background Model Training
- Speaker Model Adaptation
- LLR Score Computation
- Fusion
- Making Decision





FrontEnd Processing

- FrontEnd Processing for MFCC
- FrontEnd Processing for Pitch
- FrontEnd Processing with Wavelet





FrontEnd Processing for MFCC

- Band-limited (300Hz – 3400Hz)
- MFCC+Delta(16+16) with the 0th removed
- RASTA
- CMS
- Remove Silence
- Kurtosis Normalization





Silence Removal

- Energy based threshold to remove long period silence
- Predictive Segment
 - H_0 : current frame is a new segment first frame
 - H_1 : current frame is belong to previous segment
 - $|X_t - \text{Seed}_{t-1}| < |X_t - O|$, choose H_0 ,
 - Else, choose H_1
- Energy & Duration based threshold to remove silence segment

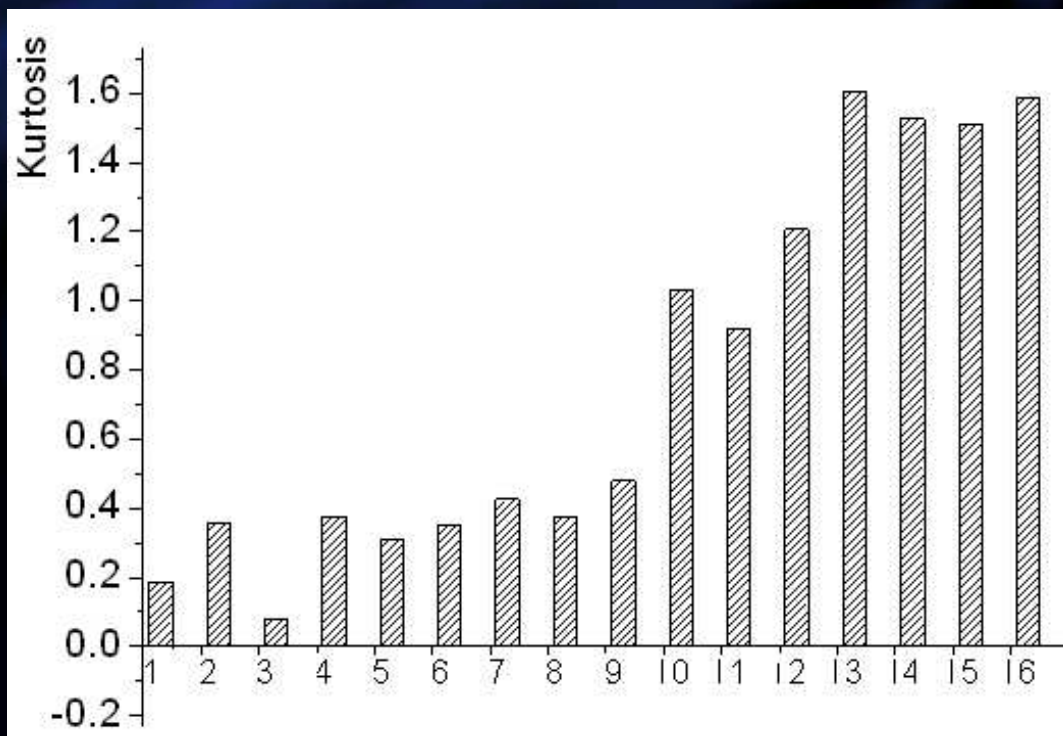




Kurtosis Normalization

The kurtosis of a random variable x is defined as

$$K(x) = \frac{E(x^4)}{E(x^2)^2} - 3$$



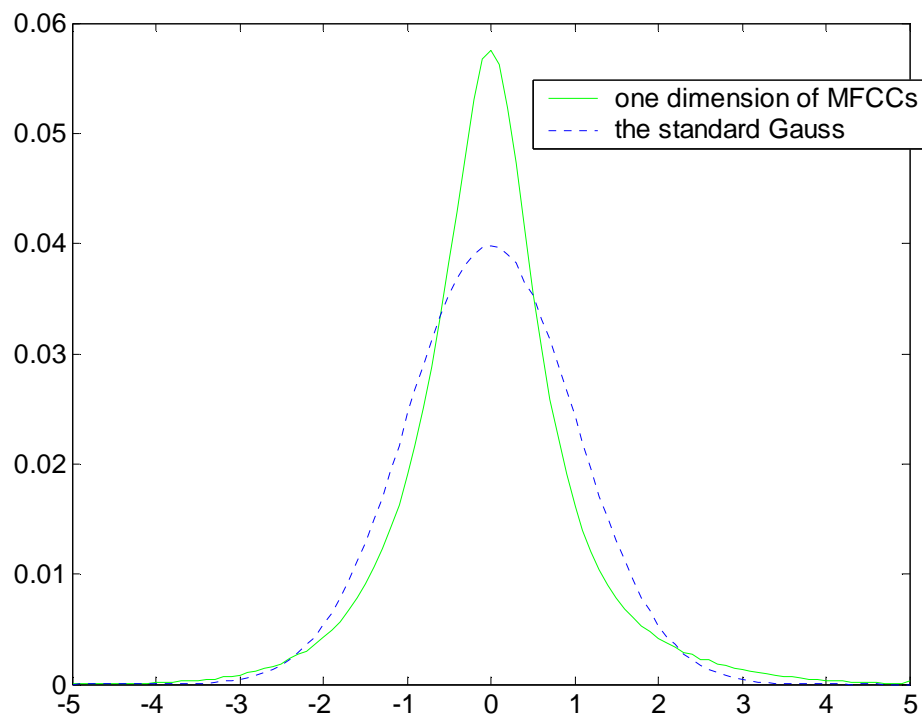
If a random variable has a kurtosis less than zero, it is termed platykurtic i.e. sub-Gauss. If it has kurtosis greater than zero, it is termed leptokurtic i.e. super-Gauss. Speech signals are generally leptokurtic, so are speech cepstral parameters.





Kurtosis Normalization

The comparison of pdfs
between MFCCs and the
standard normal.



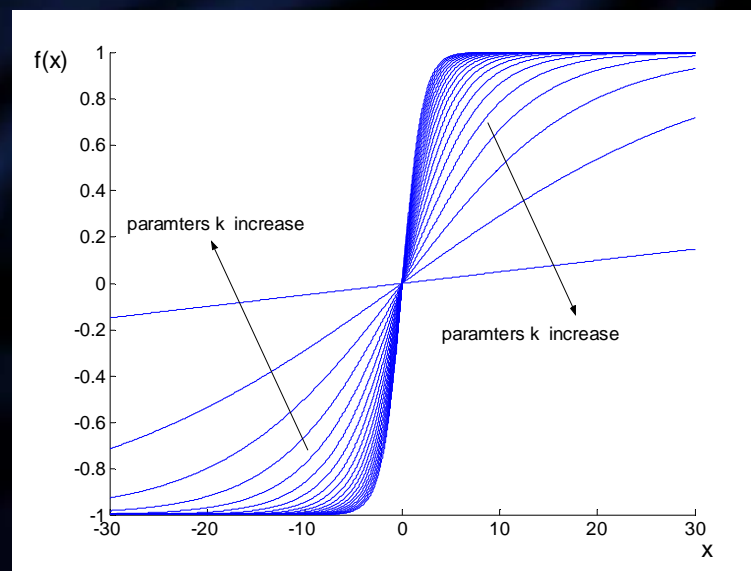


Kurtosis Normalization

the sigmoid functions

$$f(x) = \frac{a}{1 + \exp(-kx)} - b$$

where a and b are constant coefficients, $k > 0$. In order to keep the means of speech parameters invariable, coefficients a and b are chosen to be 2 and 1 respectively.



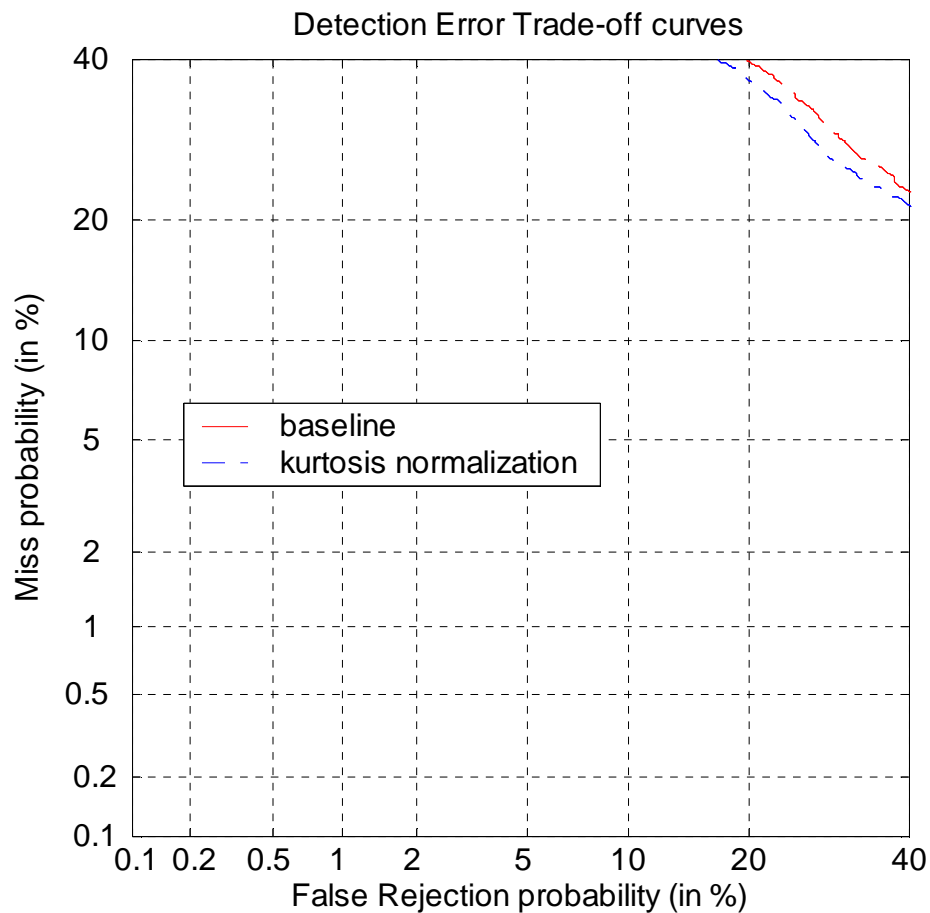
We have proved that the optimization the parameter k of the sigmoid functions can make the kurtosis be zero for speech parameters.





Kurtosis Normalization

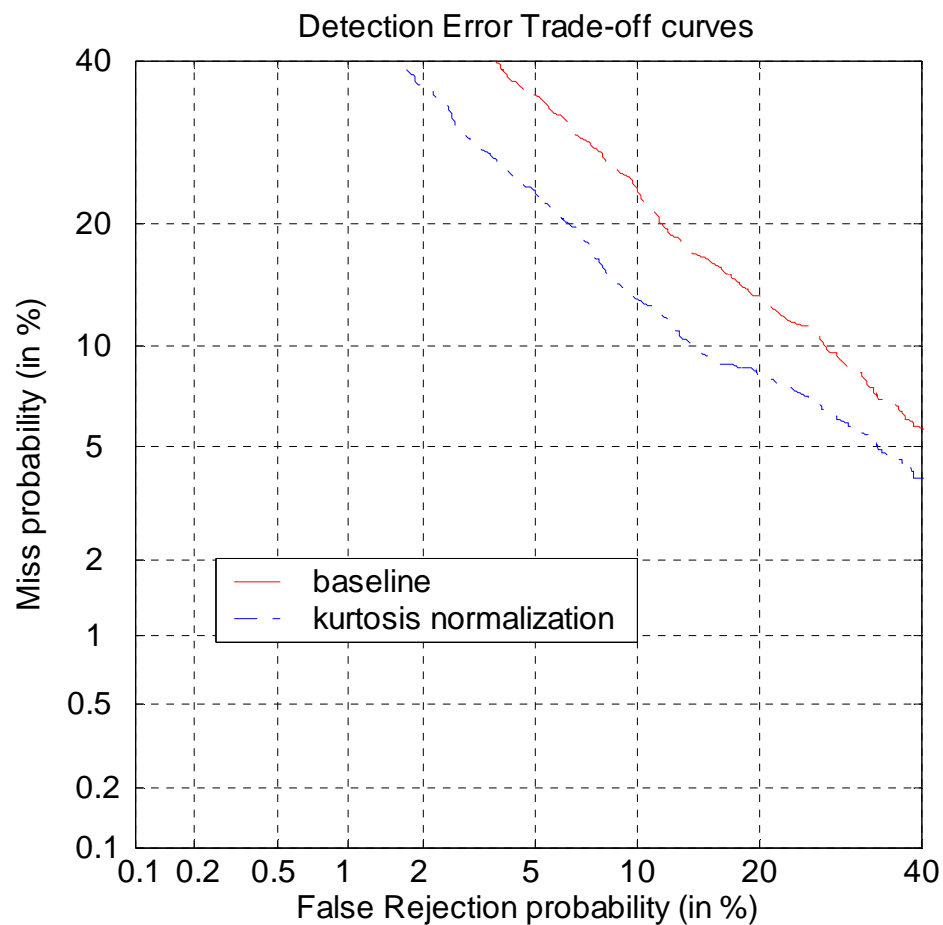
Experiment
on NIST'04
10seconds-
10seconds
male
database





Kurtosis Normalization

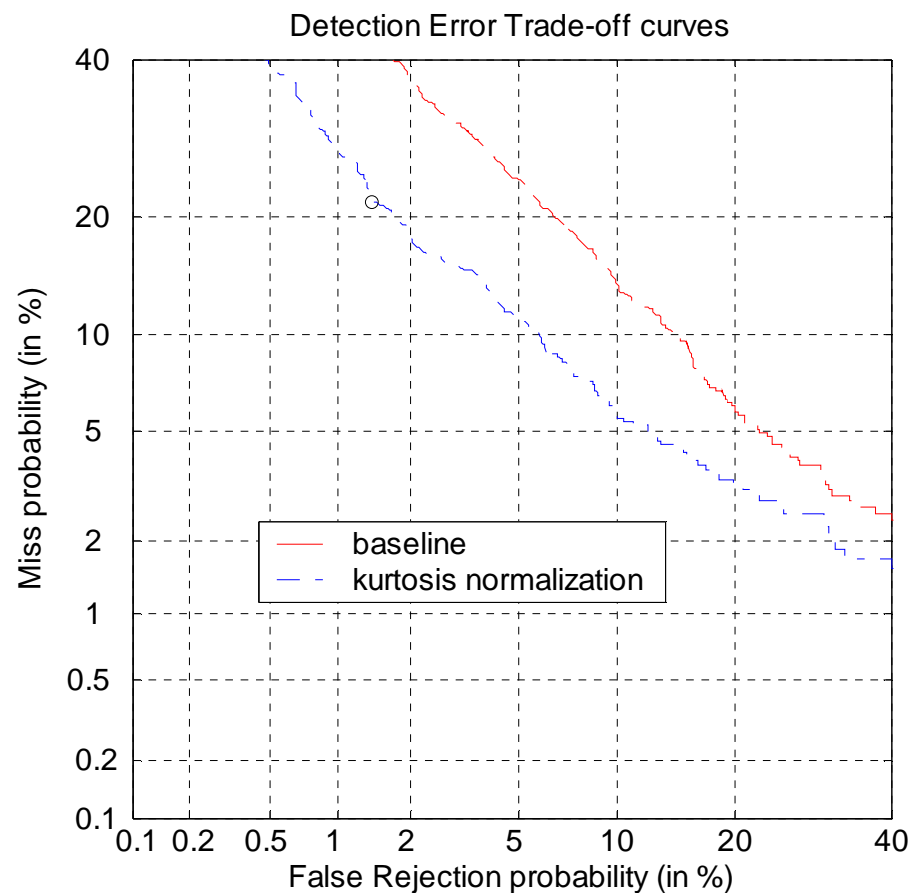
Experiment
on NIST'04
1conv-1conv
male
database





Kurtosis Normalization

Experiment
on NIST'04
8conv-1conv
male
database





Kurtosis Normalization

**Maybe the more speech is used,
the performance of the system is
improved further with kurtosis
normalization method.**





FrontEnd Processing for pitch

We firstly split pitch and energy contours into segment with 7 frames length. 4 parameters related to pitch were extracted:

- $\log(\text{mean_F0})$ averaged over a segment
- $\log(\text{max_F0})$ of a segment
- $\log(\text{min_F0})$ of a segment
- F0_slop of a segment

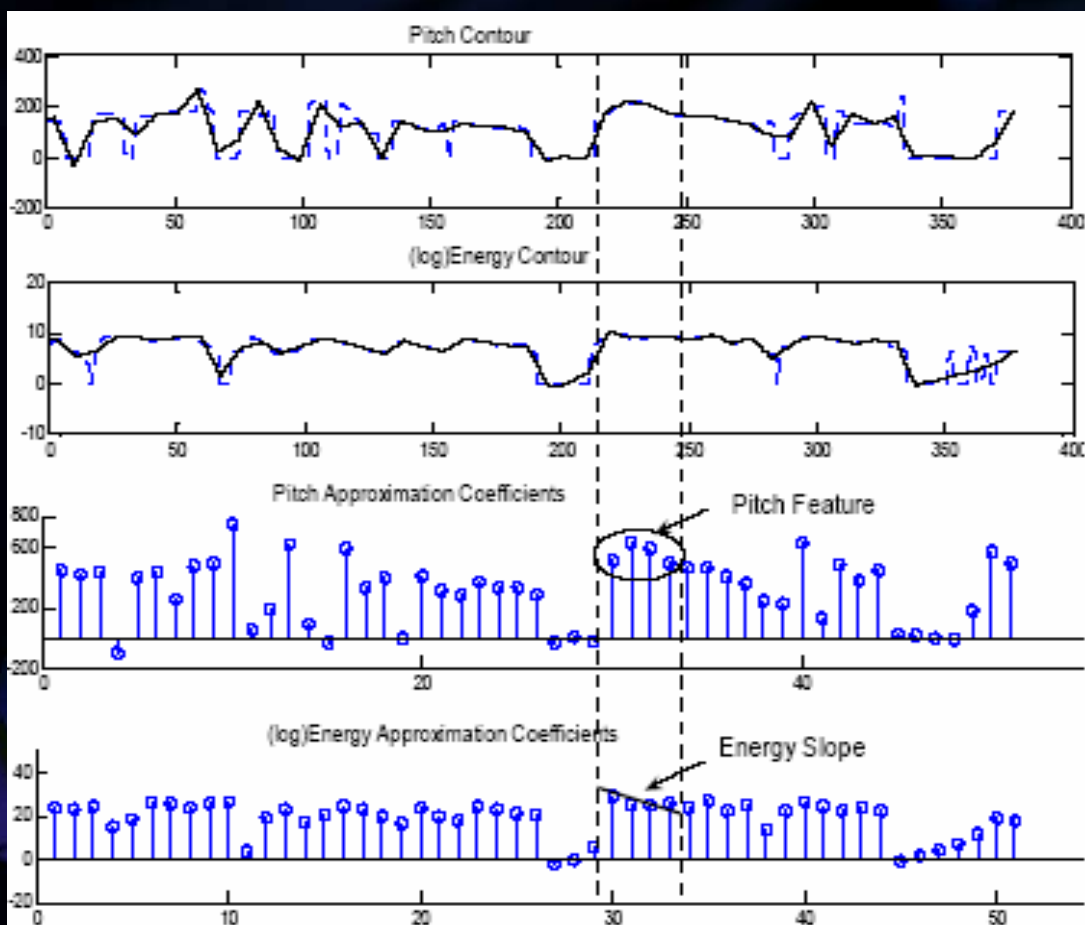
Another 4 parameters related to energy are extracted as above. Total 8 parameters of a segment comprise an 8-dimension vector.





FrontEnd Processing with wavelet

We made wavelet analysis of the f0 and energy contour. Subsequently, the prosodic features were extracted only from the 3rd level approximation coefficients



Prosodic Feature:

[cA1 cA2 cA3 cA4 ESlope]





Universal Background Model

- Model Type
 - GMM consist of 2048 mixtures (1conv)
 - GMM consist of 512 mixtures (10seconds)
 - UBM_F for female and UBM_M for male
- Training data
 - Selected from NIST'03&04 training and test data
- Training Algorithm
 - EM Algorithm





Speaker Model Adaptation

- Model Type
 - Same as UBM
- Training data
 - Training data in NIST'05
- Training algorithm
 - MAP from UBM_M or UBM_F





LLR Score Computation

- Log Likelihood Ratio

$$\Lambda(\mathbf{O}) = \frac{1}{T} \sum_{t=1}^T (\log p(\mathbf{O}_t | \lambda_{tar}) - \log p(\mathbf{O}_t | \lambda_{UBM}))$$

- TNORM
 - A speaker-specific T-norm selection
 - The closest set of P cohort models are used to Tnorm during run time where P is chosen to be 50.





Fusion

- The scores from the sub-systems are fused with a perceptron classifier. The number of input nodes of the perceptron is the same as the number of sub-systems applied. There is no hidden layers and only one output node.





Making Decision

- Threshold is tested with NIST'04 test utterances when the minimal DCF is reached.



USTC 2-sp System





Main Modules

- FrontEnd Processing
- Universal Background Model Training
- Segmentation
- Speaker Model Adaptation
- LLR Score Computation
- Making Decision





FrontEnd Processing

- Feature for 2-sp Segmentation
 - Band-limited(0Hz - 4000Hz)
 - MFCC(23) (without delta)





FrontEnd Processing

- Feature for Speaker Verification
 - Band-limited(300Hz - 3400Hz)
 - MFCC + Delta(16 + 16)
 - RASTA
 - CMS
 - Remove Silence
 - Kurtosis Normalization





Universal Background Model

- UBM-F training
- UBM-M training
- Gender Independent UBM training





Gender Dependent UBM training (UBM-F and UBM-M)

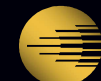
- Setting
 - 2048 x 1
- Training Data:
 - NIST'03&04 Dev Training Data (IDs are selected)
- Training Algorithm:
 - EM algorithm





Gender Independent UBM training

- Setting
 - 4096 x 1
- Training Algorithm
 - Merge from UBM-F and UBM-M





Unsupervised Speaker Segmentation

- Hierarchical agglomerative clustering
 - Divide the speech into 1sec segments as initial clusters.
 - Merge two clusters which have minimum pair distance.
 - Until obtain three clusters (speaker 1, speaker 2, overlap of two speakers)





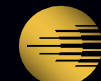
Pair-wise Distance Computing

- Likelihood Ratio Score for Segment

$$L(x:\theta_x) = \prod_{j=1}^r \sum_{k=1}^K g_k(x) N_k(v_j)$$

- Likelihood Ratio

$$\lambda_L = \frac{L(z:\theta_z)}{L(x:\theta_x)L(y:\theta_y)}$$





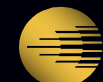
Pair-wise Distance Computing

- Transition Probability

$$f(n) \equiv \Pr[S_{i+n} = S_i] = \frac{1 + (2p-1)^n}{2}$$

- Duration time bias

$$\lambda_D = \frac{\prod_i^c f(n_i)}{\prod_i^c (1 - f(n_i))}$$





Pair-wise Distance Computing

$$d(x, y) = -\log(\lambda_L) - \alpha \log(\lambda_D)$$

$$\alpha = 4$$





Speaker Model Adaptation

- Setting
 - Same as UBM
- Training data
 - 3 of the 9 Clusters are selected
 - Select most similar 3 clusters from 9 clusters.
- Training algorithm
 - MAP from UBM

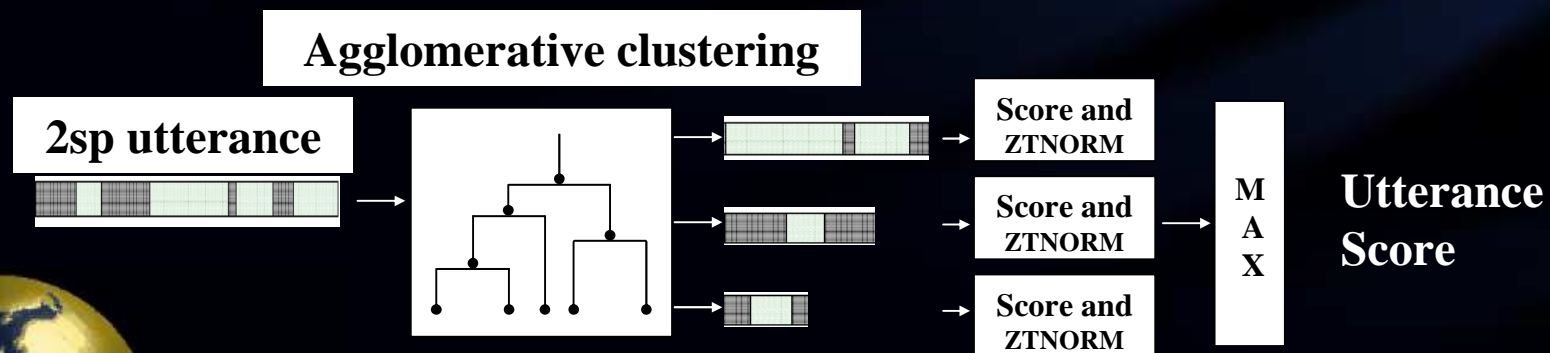




LR Score Computation

- Likelihood Ratio Score

$$\Lambda(\mathbf{O}) = \frac{1}{T} \sum_{t=1}^T (\log p(\mathbf{O}_t | \lambda_{tar}) - \log p(\mathbf{O}_t | \lambda_{UBM}))$$





Making Decision

- Threshold Selecting
 - NIST04 2-spk Evaluation Test Segments
 - Minimal DCF

