

# Computer-Assisted Language Learning (CALL) Systems

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# OUTLINE

- Introduction (TK)
- Segmental Aspect & Speech Recognition Tech. (TK)
  - Pronunciation Structure Model (NM)
- Prosodic Aspect (NM)
- Speech Synthesis Tech. for CALL (NM)
- CALL Systems (TK)
- Database for CALL (NM)

# (Traditional) LL → CALL

- (Traditional) LL: magnetic audio tape
  - Single media, Sequential access



- Computer-Assisted LL

- Multi-media, Random access
  - Easier comparison of learner's speech and model speech
- **Speech technology can be incorporated**
  - Partly replace rater's or teacher's jobs



# Speech Technology for LL

- Automate assessment of proficiency
  - PhonePass → Versant
  - ETS-TOEFL
  - PSC (Putonghua Shuiping Ceshi)
- Assist LL
  - With light supervision...CALL classroom
  - Self-learning
    - Need to keep motivating...Edutainment
    - Need to avoid enhancement of errors

# Target Population of CALL

- **Non-native speakers**
  - Particular L1 (ex.) English LL for Japanese people
    - Still diverse in proficiency level, but L1 knowledge useful
  - Unlimited L1 (ex.) Japanese LL for people in the world
- Children (native) [Russel 1996]
- Handicapped (Hearing or Articulation-impaired) people [Bernstein 1977]
- Accented (dialect) people
  - Putonghua [Hu 2008]
  - Operators at Call Centers

# Target Skill of CALL

- Reading
- Writing
- Listening
- **Speaking-Pronunciation**
  - Phone, word
  - Sentence, paragraph
  - Segmental, prosodic
- Vocabulary, Grammar
- Pragmatic Dialog (Communication)
  - travel-shopping, business-negotiation

# Importance of Pronunciation Training

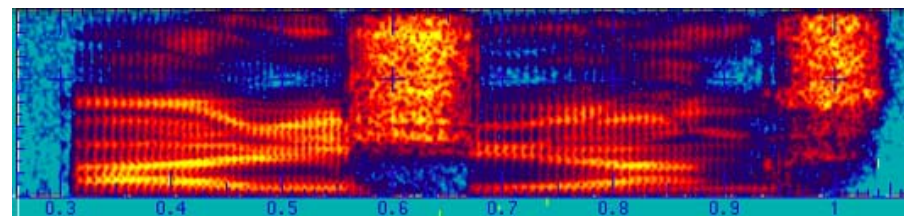
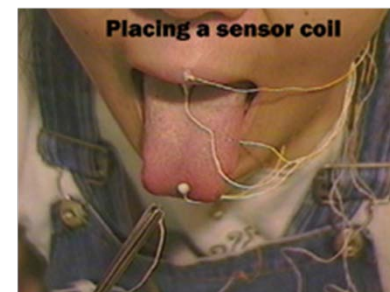
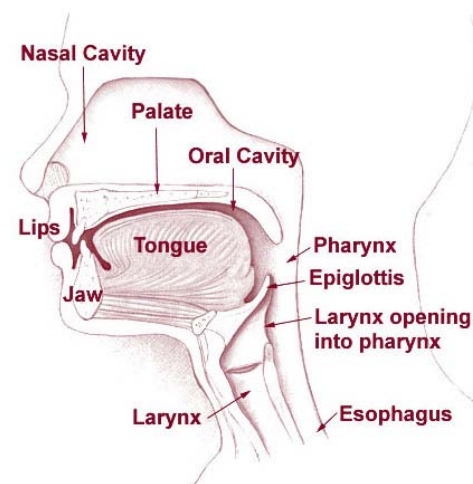
## [Bernstein 2003]

$$Comm = pron * lex * (1+syn+rhet+prag+soc)$$

- comm. = communication skills
  - pron. = pronunciation
  - lex. = lexical control and vocabulary
  - syn. = syntax
  - rhet. = rhetorical form
  - prag. = pragmatics
  - soc. = sociolinguistics
- Pronunciation skill affects entire communicative performance
  - Native-sounding pronunciation may not be needed, but acceptable (intelligible enough) pronunciation is desired for smooth communication

# Articulation → Speech

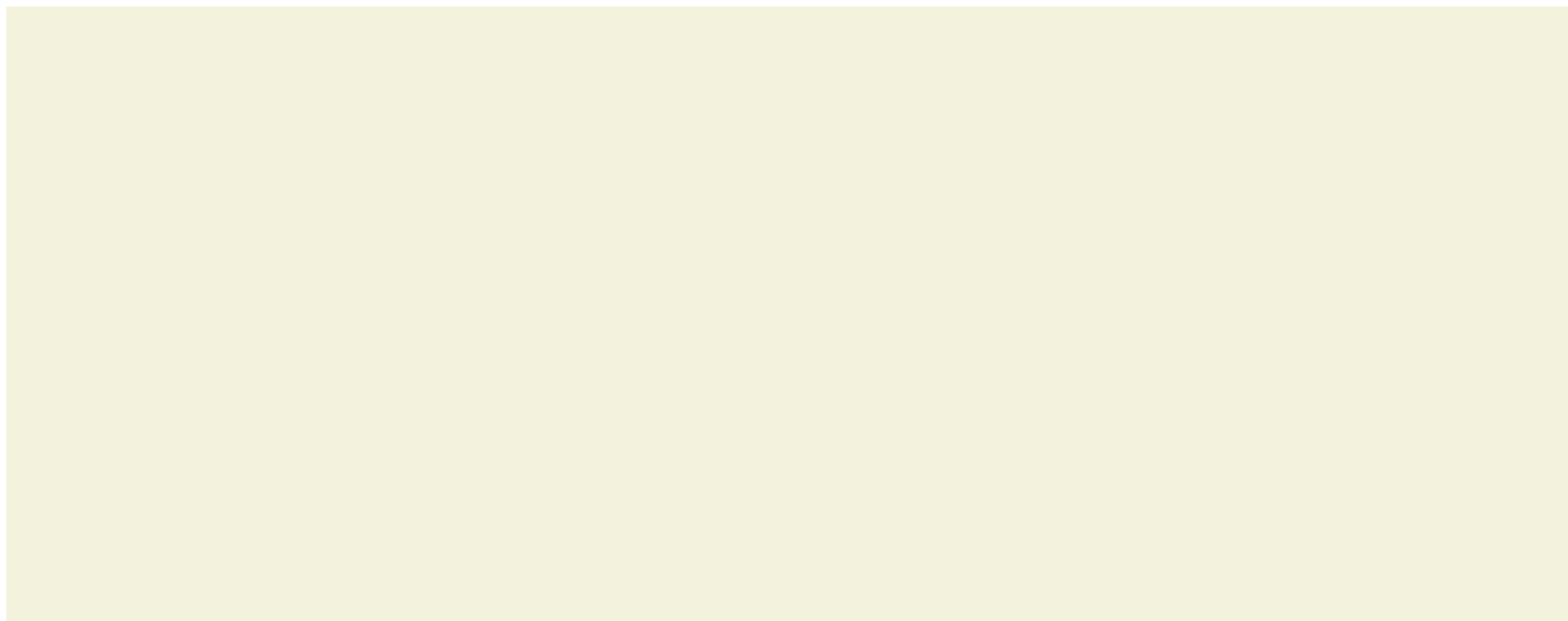
- Students must learn how to control articulators (vocal tract)
- But it is not easy to observe the movement of these organs
- Observation is feasible for acoustic aspect of speech





# Visual Presentation of Articulation

- Talking Head showing correct articulation [Massaro 2006]
- Acoustic-to-articulatory inversion to estimate the articulatory movements [Badin 2010]



# Segmental and Prosodic Aspects

- Segmental Pronunciation → Kawahara
  - Phonemes (Sub-words)
  - Features: spectrum envelop-based
- Prosody → Minematsu
  - Tones
  - Lexical accents
  - Intonation and rhythm patterns
  - Features: fundamental frequency, power, and duration

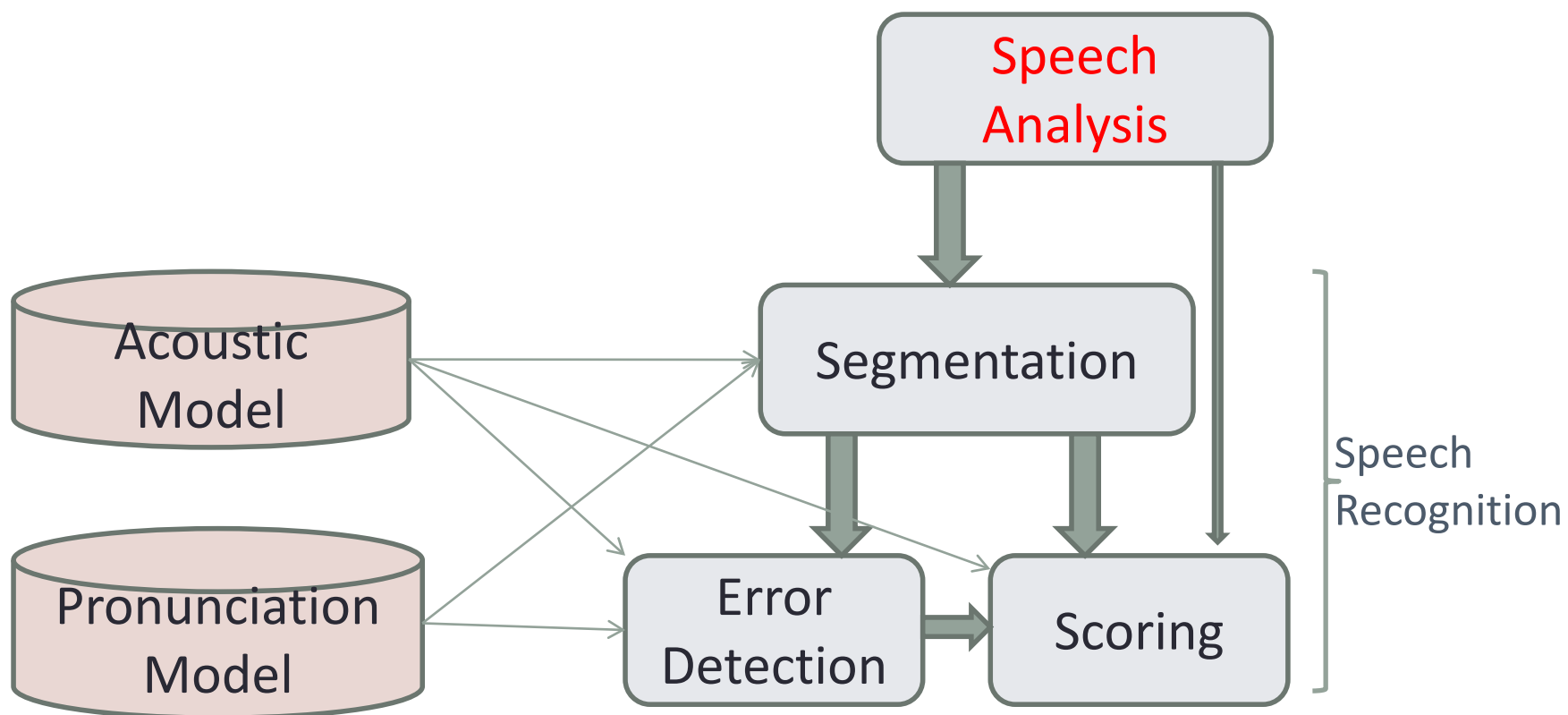
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# Speech Technology used in CALL

- Speech analysis
  - spectrum, pitch, power
  - Feature normalization required for objective comparison with model speaker
- (Constrained) speech recognition (ASR)
  - Speech segmentation-alignment
  - Error detection
  - Scoring
  - Need to model non-native speech and handle erroneous input
  - Not only segmental aspect, but also prosodic aspects
- Speech synthesis (Minematsu)

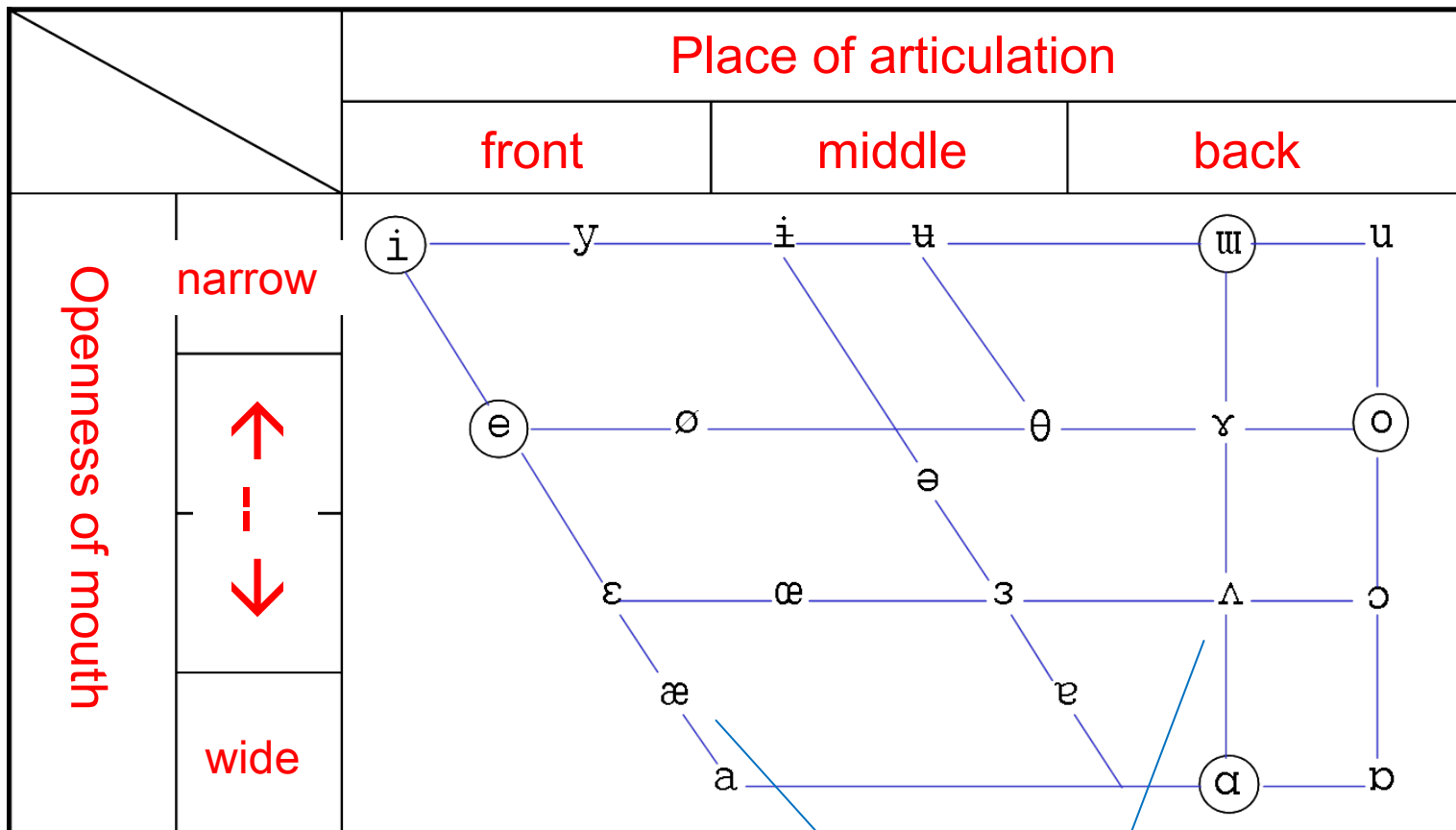
# Flowchart of Pronunciation Error Detection and Scoring



# Formant and Articulatory Features

- Potentially useful for effective diagnosis and feedback
  - Direct relationship with articulation
- Not easy to make reliable and robust estimation
  - Not used in ASR

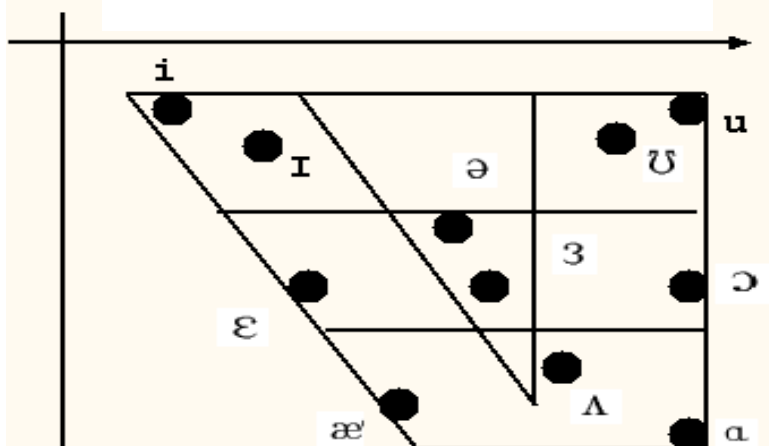
# Classification of Vowels



"bat" vs. "but"

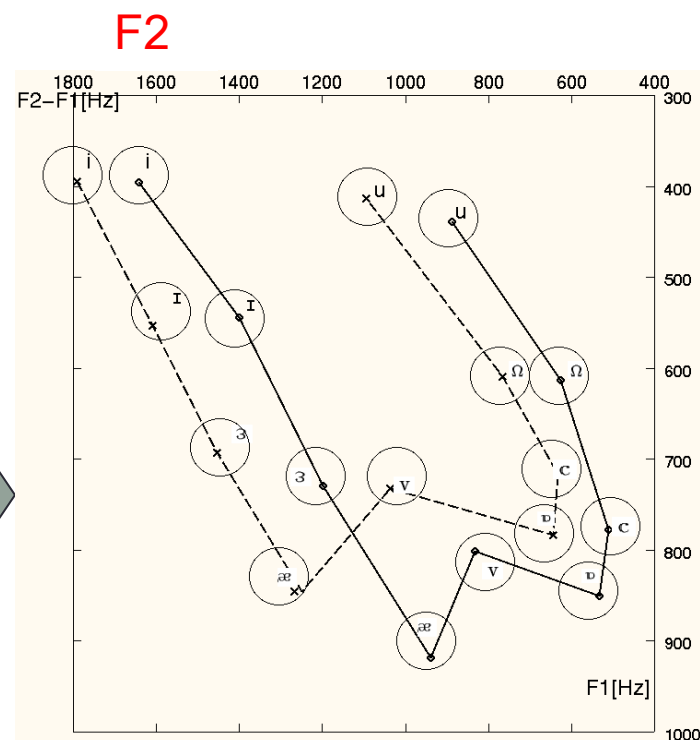
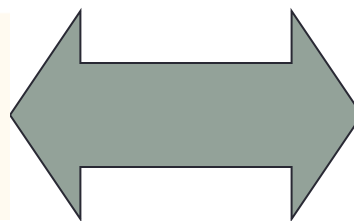
# Relationship between Articulation and Formants

Place of Articulation



Openness

Articulation Chart



Formant Chart **F1**



# Classification of Consonants (Japanese)

	Bilabial		Alveolar		Palatal		Glottal
	voiced	unvoiced	voiced	unvoiced	voiced	unvoiced	unvoiced
Fricative		f*)	z	s	ʒ	ʃ	h*)
Affricate			dz	ts	dʒ	tʃ	
Stop	b	p	d	t	g	k	
Semi-vowel	w		r**) )		j		
Nasal	m		n		ɲ		

“sea” vs. “she”

# MFCC: Mel-Frequency Cepstrum Coefficient

- Most widely-used spectral feature
    - Mel-bandwidth  $\leftarrow$  human perception
    - Cepstrum  $\rightarrow$  spectrum envelope
      - orthogonal & less correlated  $\rightarrow$  appropriate for statistical model
1. DFT(FFT)  $\rightarrow$  power spectrum
  2. Mel-conversion (Mel-band filter bank)
  3. Logarithm + Cosine Transform (IDFT)  $\rightarrow$  cepstrum
  4. Extract low quefreny (12) coefficients

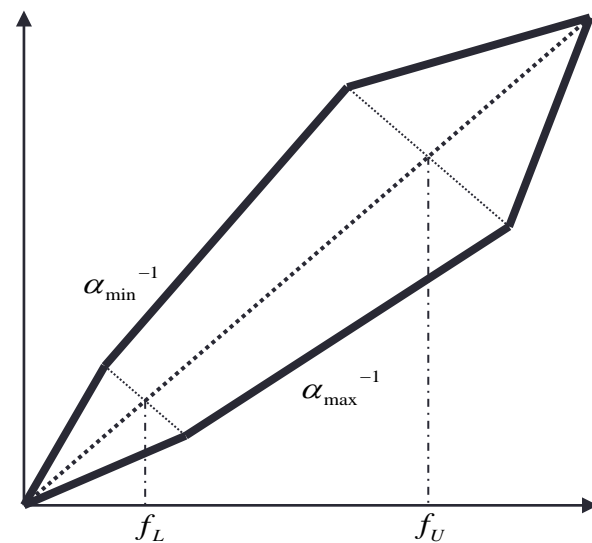
# Feature Normalization in Speech Analysis

- Feature normalization
  - for objective comparison with model speaker
  - for score calculation via speech recognition
  - against speakers (native/non-native)
  - against acoustic channels (database/users)
- Normalization methods for MFCC
  - Cepstrum Mean Normalization (CMN)
  - Cepstrum Variance Normalization (CVN)
  - Histogram Equalization

# Speaker Normalization in Speech Analysis

- Vocal-Tract Length Normalization (VTLN)

- Warping spectral dimension
- Based on acoustic model likelihood



- Pronunciation Structure (by Minematsu)

- Invariant-feature (F-divergence)

# Speech Recognition for CALL

- Tasks
  - Speech segmentation-alignment
  - Error detection
  - Scoring
- Challenges
  - Modeling non-native speech
  - Handling erroneous speech input
- Constraint
  - Target word or sentence is given

# ASR vs. CALL

$X$ : speech input,  $W$ : phone label  $\leftarrow$  word sequence (target)

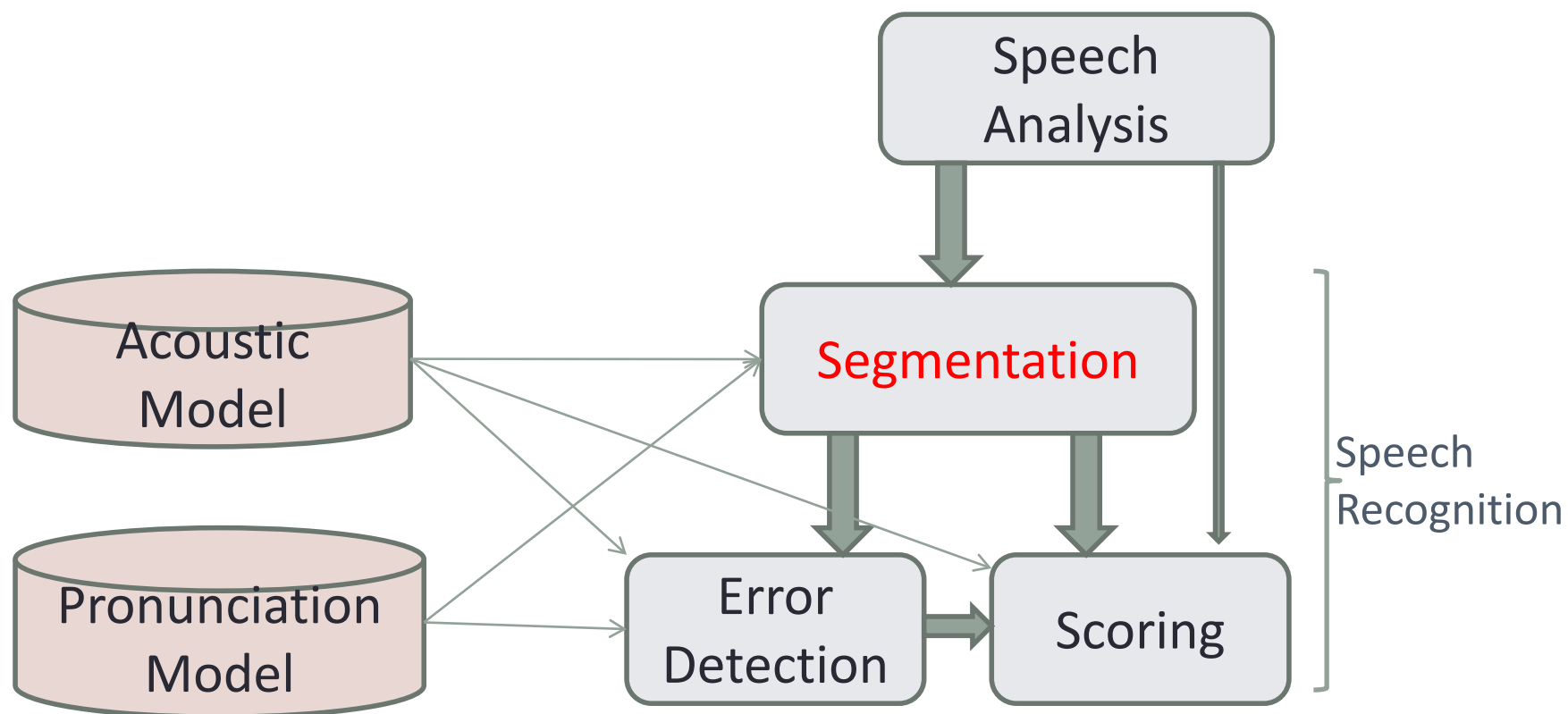
- ASR

- For given  $X$ , find  $W$  that maximizes  $p(W|X)$
- Solved by  $\max p(W)*p(X|W)$
- Each phone model  $p(x|w)$  is trained

- CALL

- $W$  (oracle) and  $X$  (not reliable) given,
- Segmentation: Viterbi forced alignment
- Error detection: find  $W'$  such that  $p(X|W') > p(X|W)$
- Scoring: evaluate  $p(X|W)$ ?? How to train the model??

# Flowchart of Pronunciation Error Detection and Scoring

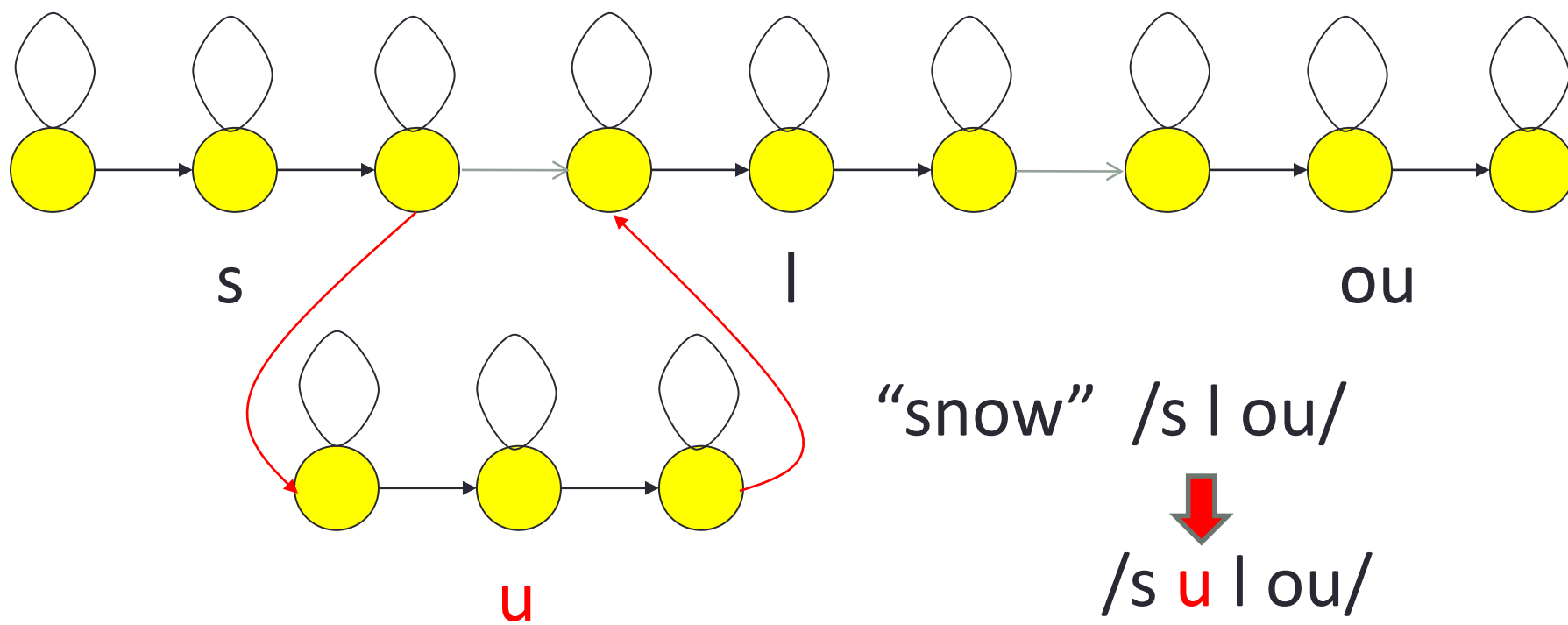


# Segmentation

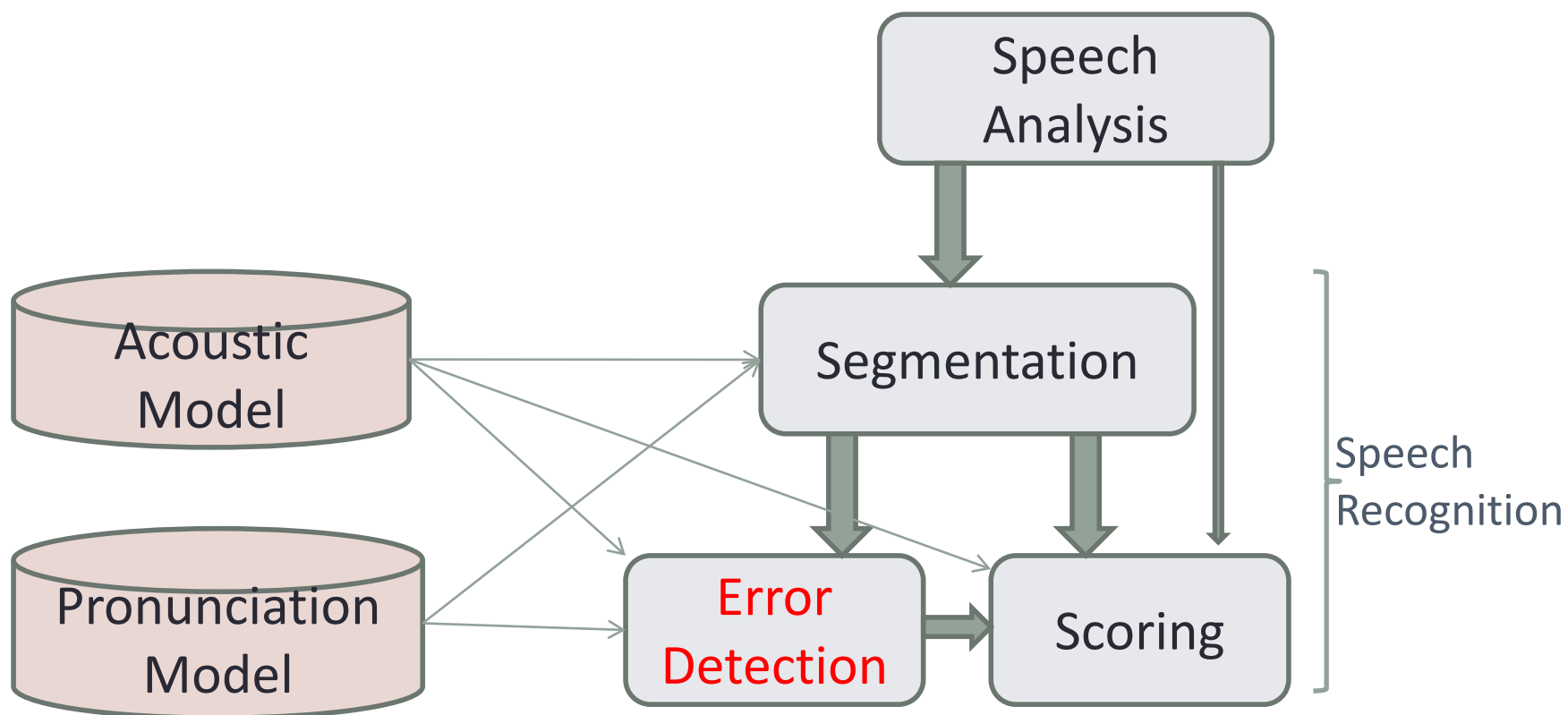
- Pre-process for scoring
- Viterbi forced alignment with HMM representing  $W$
- In fact, there may be pronunciation errors in  $X$ 
  - Insertion & deletion seriously affect alignment
  - Error prediction/detection may be necessary



# Segmentation



# Flowchart of Pronunciation Error Detection and Scoring

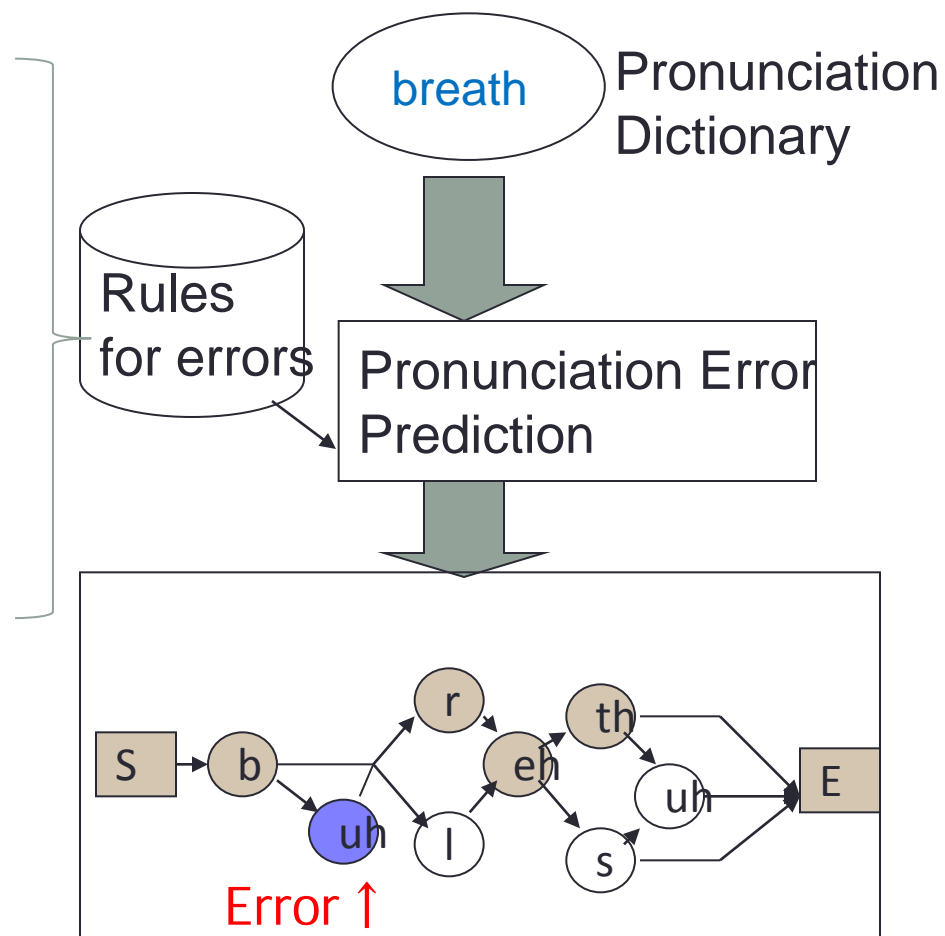


# Error Detection

- Find  $W'$  such that  $p(X|W') > p(X|W)$
- Compute scores  $p(x|w')$  for alternative phones  $w'$  for each segmented region  $x$
- When we take into account insertions and deletions, we need to generate a network of possible errors
- Error prediction can be done with prior knowledge, such as L1
  - Alternative phones  $w'$  can be taken from L1

# Error Prediction in Pronunciation Model

- No equivalent syllable in L1  
(ex.) *sea* → *she*
- No equivalent phoneme in L1  
(ex.) *l* → *r*, *v* → *b*
- Vowel insertions  
(ex.) *b-r* → *b-uh-r*



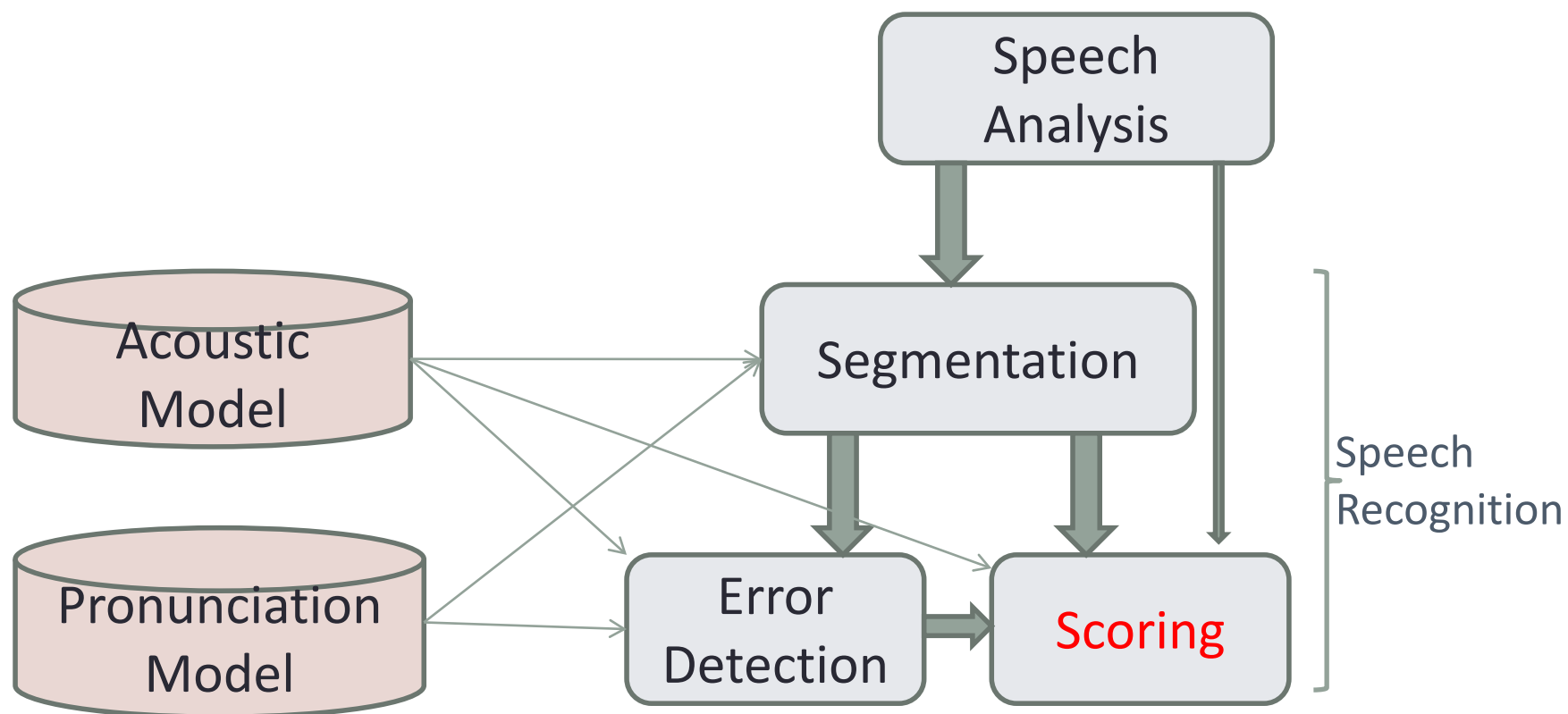
# Error Detection based on Classification Approach

- Not necessarily compute  $p(x|w')$ , but test if  $w'$  is more likely than  $w$
- Explicit classifier (verifier) learning
  - Incorporate many features
  - Focus on error detection
  - by assuming segmentation
- Linear Discriminant Analysis (LDA)
- Support Vector Machines (SVM)

# Other Issues in Error Detection

- Filter and prioritize many (possible) individual phone errors
- error miss >> false alarm
  - Not to discourage learners
- Feedback
  - How to correct errors

# Flowchart of Pronunciation Error Detection and Scoring



# Scoring: Standpoints

- Native-likeness

“How close to golden native speakers?”

→  $P(X|W, \lambda_G)$

- What is the “golden” model? British? American?...
- Impossible to free from L1 effect, speaker characteristic

- Intelligibility

“How distinguishable (less confusable) from other phones?”

→  $p(W|X)$

- Some pronunciation may not be recognized as anything
- Need to consider L1 phones as well → assume L1



# Scoring based on Native-Likeness

- How close to golden native speakers?
  - Defined by  $p(X|W, \lambda_G)$   $\lambda_G$ : golden model
  - Normalized by  $p(X|W, \lambda_N)$   $\lambda_N$ : non-native model
  - In summary, likelihood ratio

$$\frac{p(X | W, \lambda_G)}{p(X | W, \lambda_N)} \approx \prod \frac{p(x | w, \lambda_G)}{p(x | w, \lambda_N)} \approx \prod \prod_t \frac{p(x_t | w, \lambda_G)}{p(x_t | w, \lambda_N)}$$

Mean w.r.t. phones
Mean w.r.t. time-frame

$\Pi$ : geometric mean= arithmetic mean in logarithm

# Scoring based on Intelligibility

- How distinguishable (less confusable) from other phones?
  - Measured by  $p(W|X)$

$$\frac{p(X|W)}{\sum_{W'} p(X|W')} \approx \prod \frac{p(x|w)}{\sum_{w'} p(x|w')} \quad \text{posterior prob.}$$

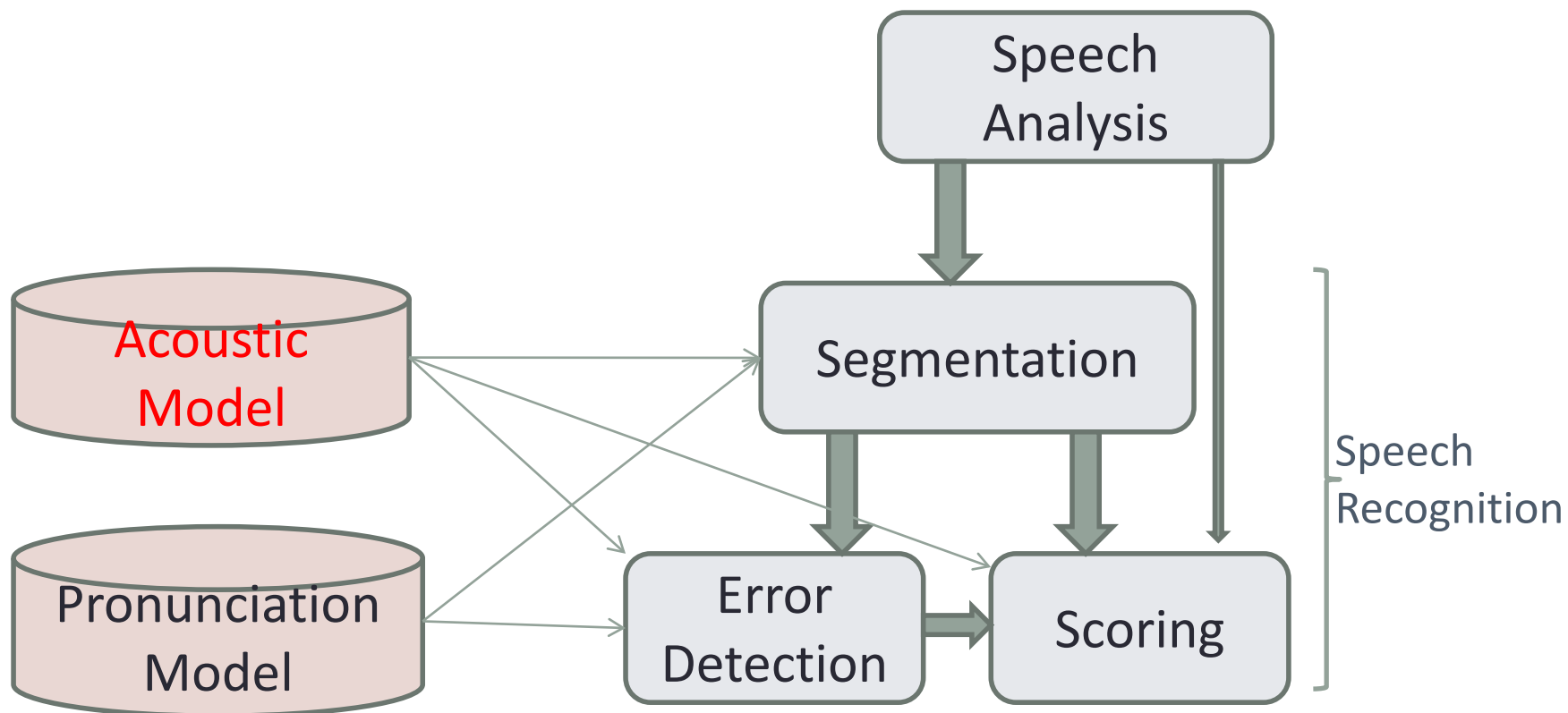
$$\approx \prod_t \frac{p(x_t|w)}{\max_{w'} p(x_t|w')} \quad \begin{array}{l} \text{forced alignment} \\ \text{Viterbi score} \end{array}$$

- Often called **GOP (Goodness Of Pronunciation)**
  - becomes 1 if best  $w'=w$
- Need to adapt to non-native speech
- Need to consider L1 phones


# Scoring to Assessment

- Other factors
  - Duration modeling & evaluation
  - Other prosodic aspects...accent, intonation
  - Speech rate
- Score mapping
  - Linear regression to fit to human rater's evaluation

# Flowchart of Pronunciation Error Detection and Scoring



# Acoustic Modeling: Native vs. Non-native

- Native speech
    - “Gold standard”, but does not match
  - Non-native speech
    - Matched, but error-prone
    - There is not large database available
- 
- Adaptation from native to non-native
  - Phone model of L1 is used for the same phone (in the IPA inventory)

# Context-Independent Modeling

- Context-dependent (e.g. triphone) models are widely used in ASR
- Context-independent (monophone) model works well, even better, in CALL
  - Phonetic context is not reliable in non-native speech ← insertion of vowels
  - Better segmentation accuracy even in native speech

## Speaker Adaptation of Acoustic Model to Non-native Speech

- Pronunciation of adaptation data may not be correct
- Compare baseform label (automatic but error prone) and hand label (correct but costly)
- Phone accuracy: measured based on hand-label including errors

[Tsubota 2004]

Acoustic model (native model)	Phone accuracy
No adaptation	75.4
Hand label	81.0
Baseform label	<b>80.6</b>

Lexicon baseform label is sufficient

# Acoustic Model:

## Native model vs. Non-native model

- Non-native speech database (MEXT project)
  - 13129 utterances by 178 speakers
  - Pronunciation errors are not annotated (too costly)
  - Dictionary label vs. automatic label with ASR
    - Both are error prone

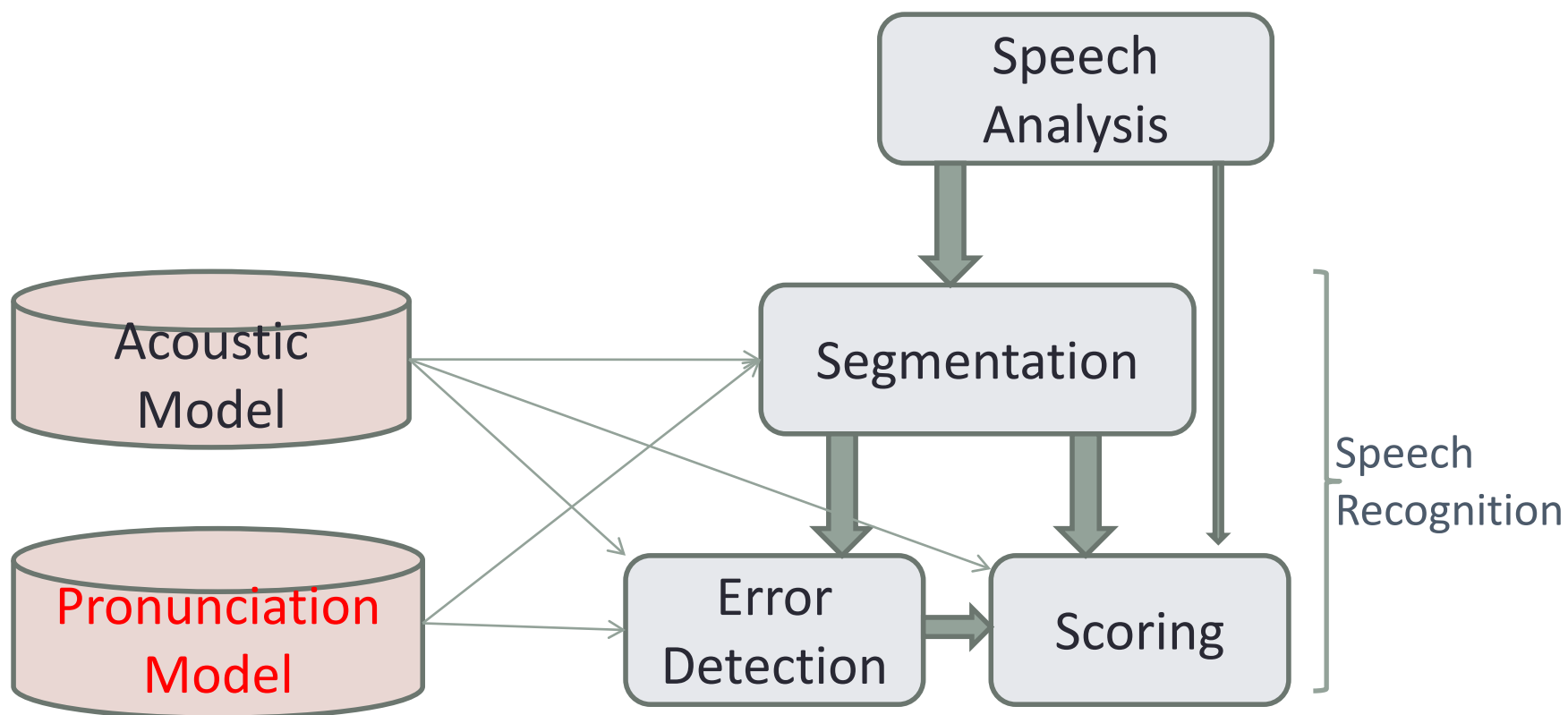
[Tsubota 2004]

Acoustic model	baseline	speaker adapt
Native English model	75.4	80.6
Non-native model (baseform)	<b>78.0</b>	81.8
Non-native model (ASR)	77.1	81.5

- Non-native model is more effective, even with dictionary label
- The superiority is reduced with speaker adaptation



# Flowchart of Pronunciation Error Detection and Scoring

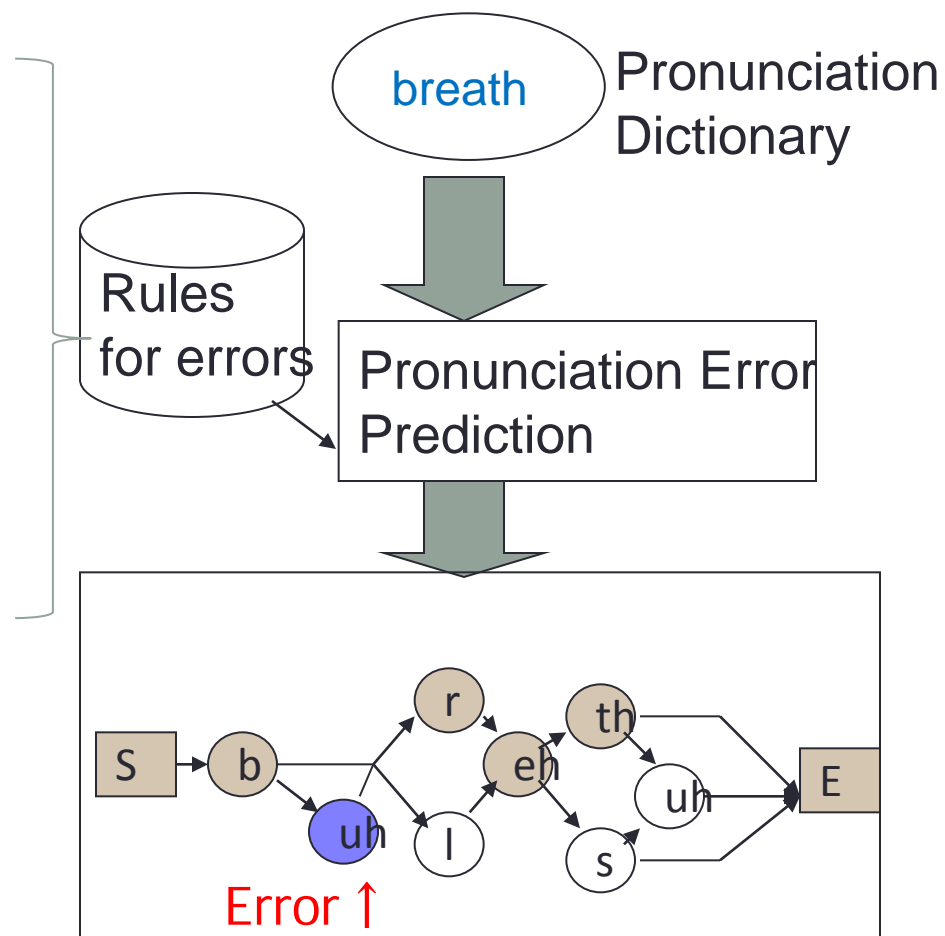


# Pronunciation Model

- Standard baseform → possible errors
- Constraint of L1 is effective
- Linguistic knowledge
  - /v/ → /b/, /θ/ → /s/
  - Substitution with similar phone of L1
  - Insertion of vowels
- For GOP score computation, simple phone loop model (=no pronunciation model) is used

# Error Prediction in Pronunciation Model

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- No equivalent phoneme in L1  
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- Vowel insertions  
(ex.) *b-r* → *b-uh-r*




# Pronunciation Model Training

- Hand-craft phonological rules
  - Expert knowledge needed
  - Too many rules cause false alarms, degrading recognition performance
  - Tradeoff between coverage and perplexity
- Machine learning from annotated data
  - Statistical learning of rewriting rules [Meng 2011]
  - Decision tree to find critical rule set [Wang 2009]

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## Another approach for segmental assessment

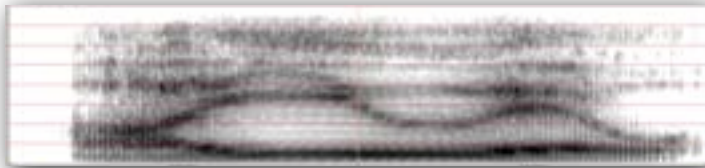
- Pronunciation training is not impersonation training [Minematsu'07].
    - Impersonation = trying to speak exactly like a target speaker
      - Not needed for pronunciation training.
      - Students are not myna birds!!
- 
- Likelihood scores are impersonation scores, not pron. scores.
    - $P(o|p)$  = similarity bet. a student's  $p$  and the mean speaker's  $p$  in training data.
    - Inadequate if a student is a child and HMMs are trained from adult teachers.
  - Posterior probability (GOP) is a score with normalization.
 
$$P(p|o) = \frac{P(o|p)P(p)}{\sum_q P(o|q)P(q)} \approx \frac{P(o|p)}{\max_q P(o|q)}$$

← forced alignment  
← cont. phoneme recognition

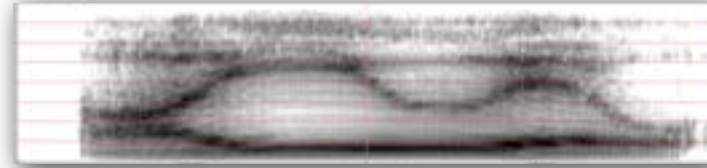
    - But alignment and recognition fails due to mismatch bet. students and teachers.
      - Then, speaker-adapted HMMs are often used or native children's data are collected.
      - So, posterior probability is a score of impersonation, again?

## Another approach for segmental assessment

- The essential problem lies in the use of spectrum envelopes.
  - SE carries information both of linguistic content and speaker identity.



Tall speaker



Short speaker

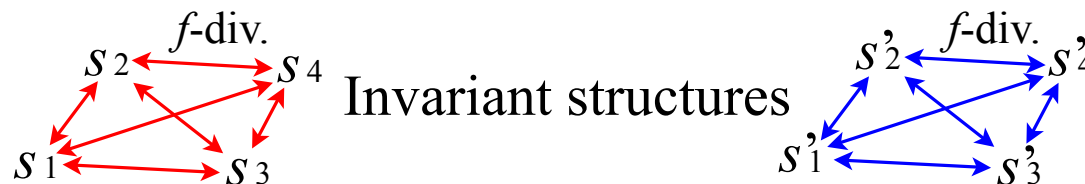
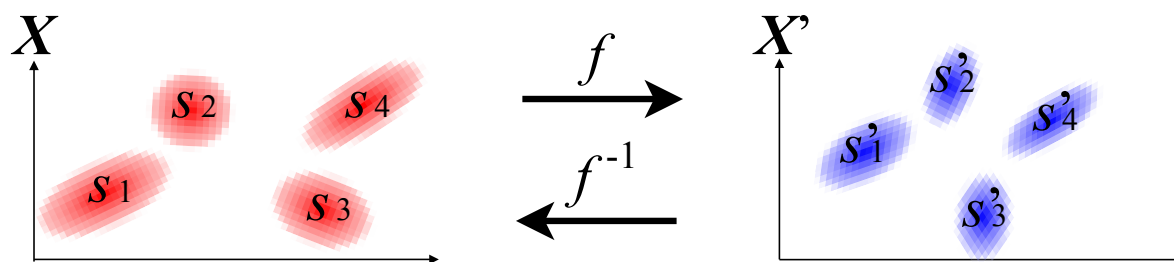
- But students imitate only the linguistic content!
  - Speaker information in the teacher's utterance is ignored by students.
    - What does "Hcopy" copy from utterances? What do students copy from utterances?
  - How to make a machine ignore the speaker component in an utterance?
- What is the commonly observed speech pattern?
  - Among linguistically identical but acoustically different utterances.
  - This pattern is the target of students' imitation but what is that?

# Another approach for segmental assessment

- Speaker difference is often modeled as feature space transformation.
  - The question is “what are transform-invariant patterns or features?”
  - $f$ -divergence is invariant with any kind of invertible transform (sufficiency).
    - The invariant features have to be  $f$ -divergence (necessity). [Qiao+’10]

$$f_{div}(p_1, p_2) = \int p_2(x) g\left(\frac{p_1(x)}{p_2(x)}\right) dx \quad \text{KL-div, Bhattacharyya distance} \in f\text{-div.}$$

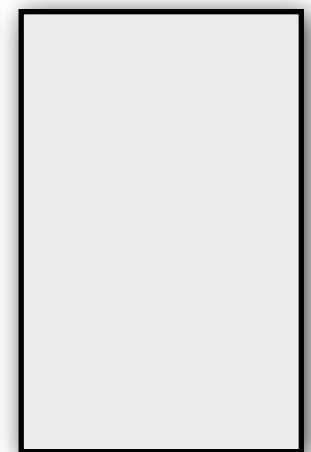
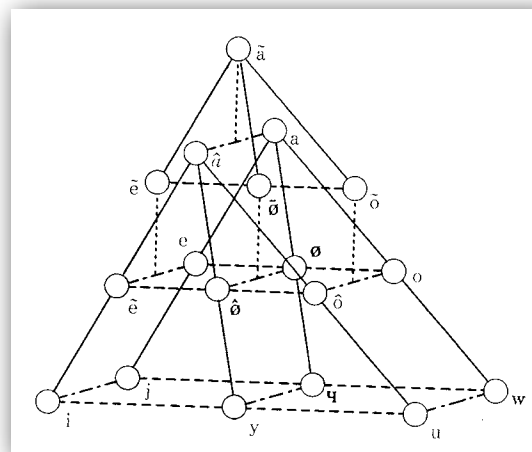
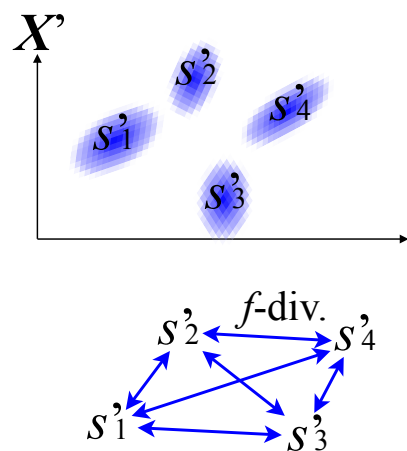
- From individual sounds to their sound system [Minematsu’04]
  - Each sound is dependent on speaker but their system is independent of speaker.
  - Any event has to be characterized as distribution not as point.





# Another approach for segmental assessment

- From individual sounds to their sound system [Minematsu+'06]
  - It should be focused on whether *the native sound system* is found in a student's utterances not whether *native sounds* are found there.
- From phonetics to (structural) phonology
  - Acoustic phonetics focus on acoustic features of individual phones.
  - Structural phonology focuses on features of their sound system.
  - Roman Jakobson (1896-1982)
    - The sound shape of language (1987)
    - *We have to put aside the accidental properties of individual sounds and substitute a general expression that is the common denominator of these variables.*



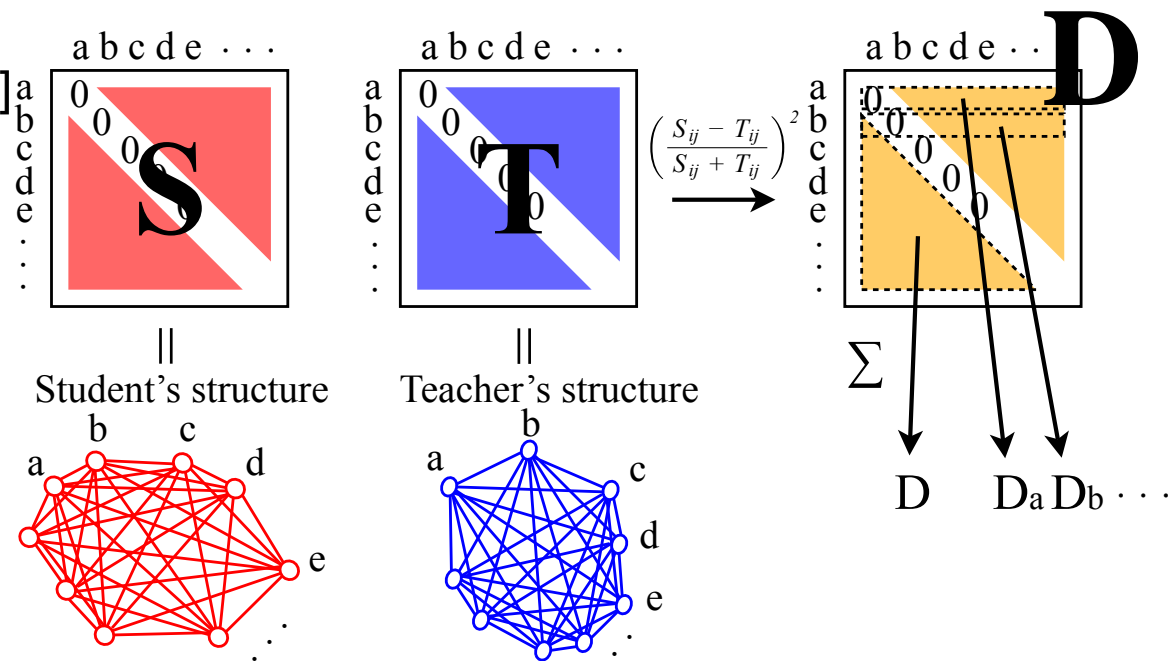
# Pronunciation structure

- Topological difference between a student and a teacher
  - Speaker-dependent phoneme HMMs are build.
    - Phoneme-based *f*-div. distance matrix is calculated from a student and a teacher.
    - *S* : matrix from a student, *T* : matrix from a teacher
  - $S - T = D$  : difference matrix between *S* and *T*

$$S_{ij}, T_{ij} = \sqrt{\text{Bhattacharyya distance bet. two phonemes}} \quad \text{BD} \in \text{f-div.}$$

$$D_{ij} = (S_{ij} - T_{ij})^2 \quad [\text{Minematsu+'06}]$$

$$D_{ij} = \left( \frac{S_{ij} - T_{ij}}{S_{ij} + T_{ij}} \right)^2 \quad [\text{Suzuki+'10}]$$



# Pronunciation structure

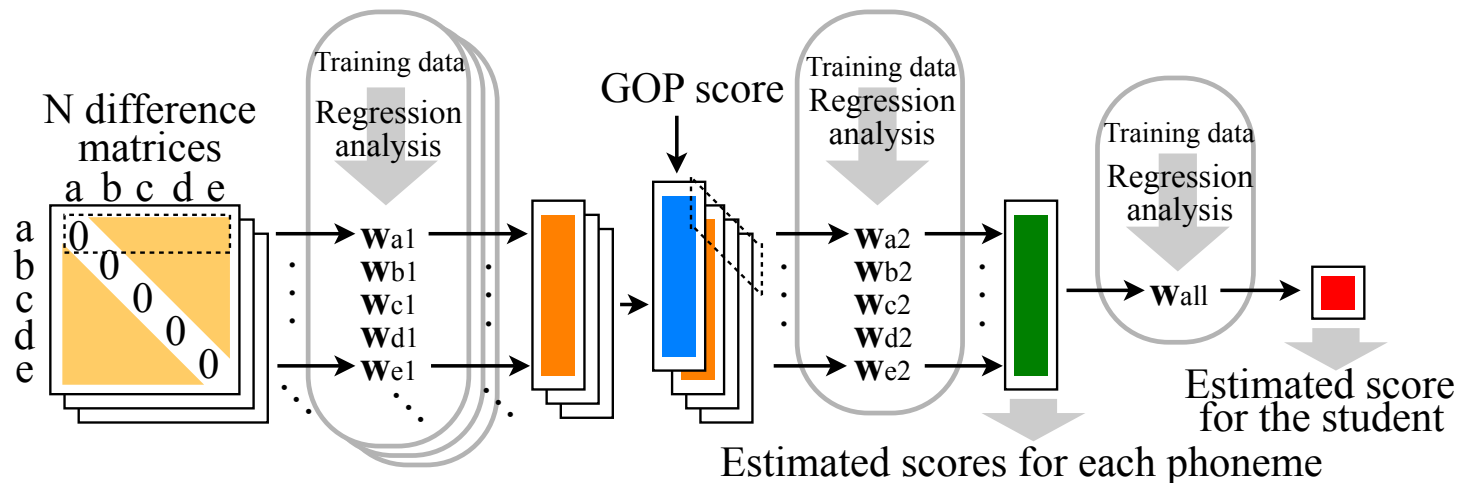
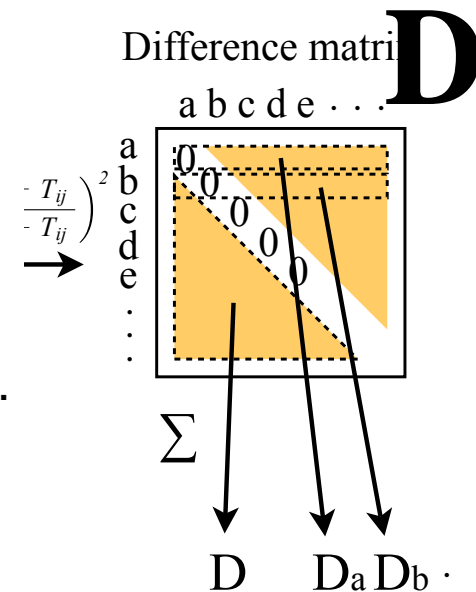
- Global Assessment Score calculated from D matrix

- $GAS = \sum_{i < j} D_{ij}$  [Minematsu+'06]

- Very effective when the target sounds are vowel-like sounds only.
- Not effective when all the phonemes are considered.

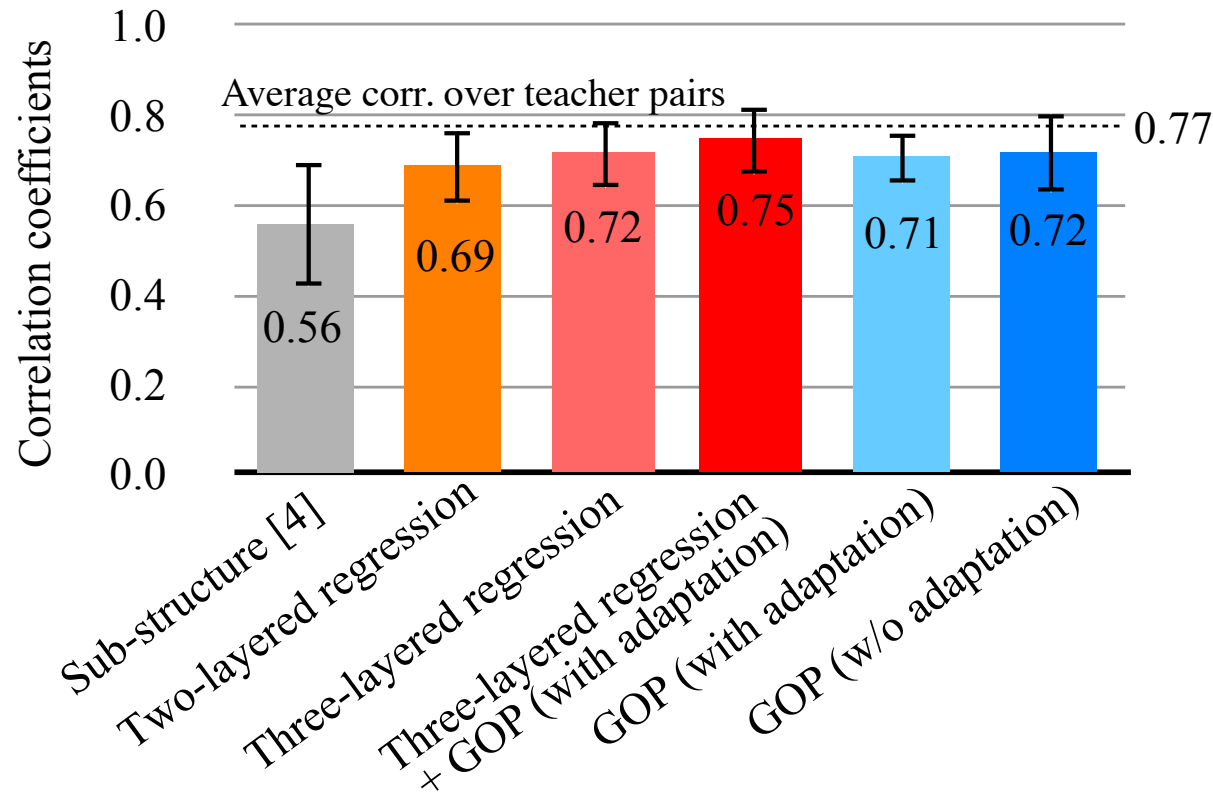
- $GAS = \text{weighted sum of } D_{ij}$  [Suzuki+'10]

- Can treat all kinds of phonemes well.
- Not simple linear regression but multilayer linear regression is applied.
- D matrices obtained from different teachers (features) can be used additionally.
- Phoneme-based GOP scores can be used additionally.



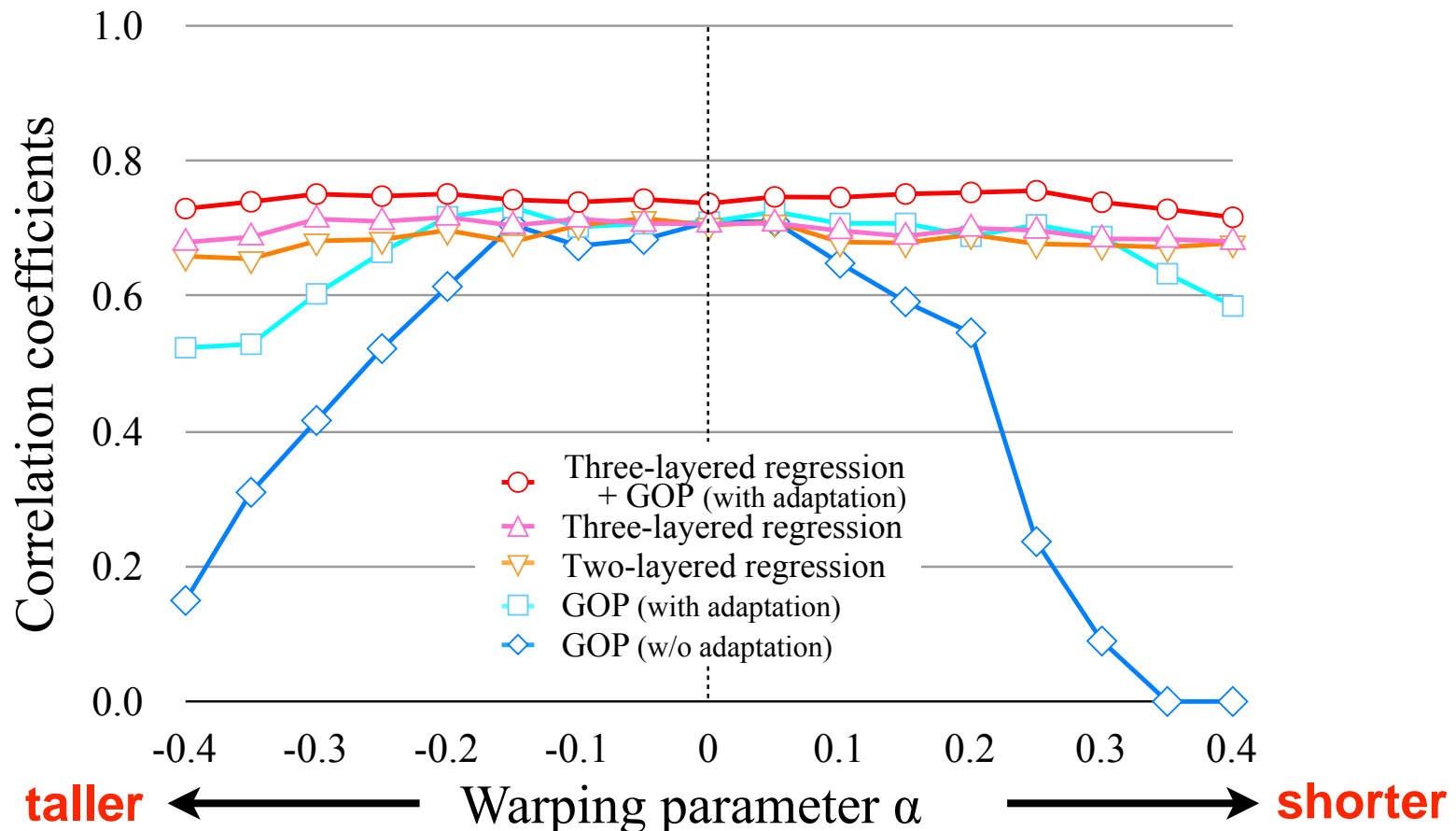
# Pronunciation structure

- Experiment using pronunciation structures [Suzuki+'10]
  - About 60 utterances per student (teacher) to train a spk-dependent HMM set.
  - Number of teachers used for the experiment
    - Two-layered regression : only 1 male teacher
    - Three-layers regression : only 1 male and 1 female teachers
  - Correlation between human teachers' scores and machine scores



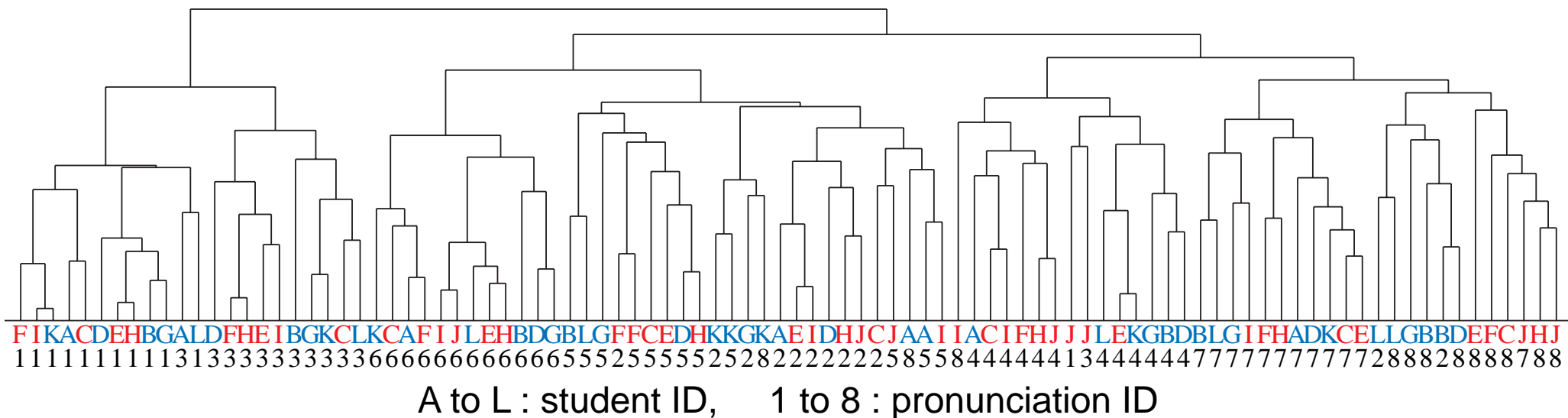
# Pronunciation structure

- Experiment using warped utterances [Suzuki+'10]
  - Simulated very tall students and very short students.
  - Only a single teacher is used in the two-layered regression.



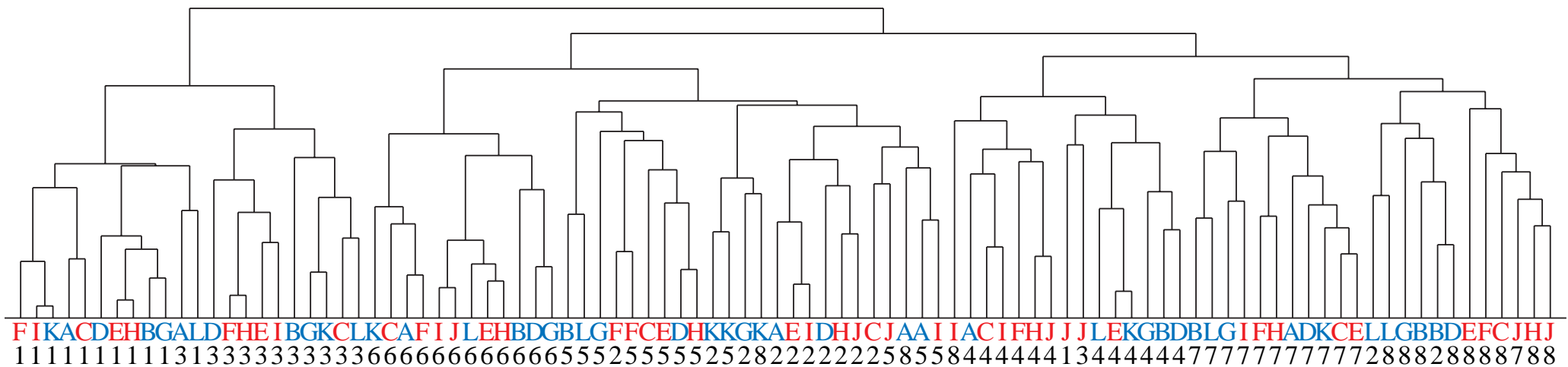
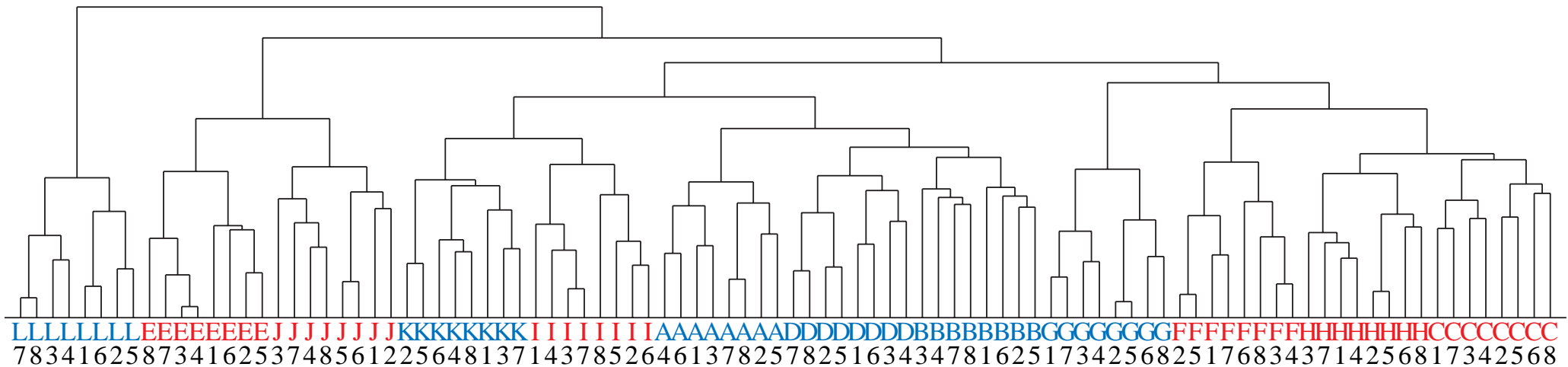
# Learner clustering based on their pron.

- Clustering “simulated” 96 students [Minematsu+'06,'07]
  - Only vowels are focused.
  - Speakers are 12 very good learners of American English (spk-A to spk-L).
  - They are asked to produce AE vowels and JE vowels, uttered in word context.
    - 7 differently accented vowel structures and a good and normal vowel structure.
    - 1-7 : Japanese accented structures, 8 : non-accented structure
    - ex)  $\alpha$ ,  $\Lambda$ ,  $\text{æ}$ ,  $\text{ə}$ ,  $\text{ɚ}$ ,  $\text{ɔ}$ ,  $\text{ɛ}$ ,  $\text{ɪ}$ ,  $\text{i}$ ,  $\text{ʊ}$ ,  $\text{u}$  (Red vowels are replaced by Japanized versions.)
  - 12 students x 8 pronunciations = 96 simulated students



# Learner clustering based on their pron.

- Acoustic clustering vs. structural clustering [Minematsu+'06,'07]



A to L : student ID, 1 to 8 : pronunciation ID

# OUTLINE

- Introduction (TK)
- Segmental Aspect & Speech Recognition Tech. (TK)
  - Pronunciation Structure Model (NM)
- **Prosodic Aspect** (NM)
- Speech Synthesis Tech. for CALL (NM)
- CALL System (TK)
- Database for CALL (NM)



# Basic prosodic features

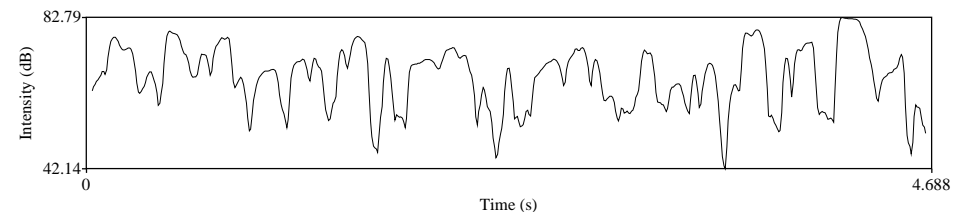
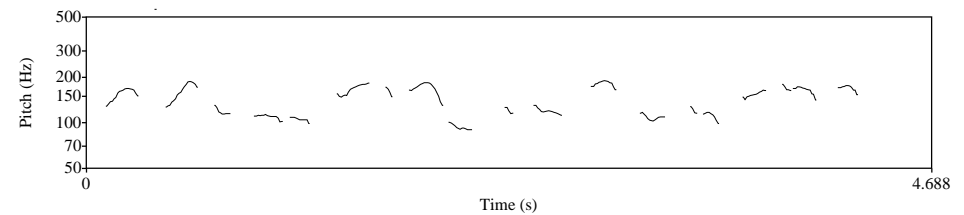
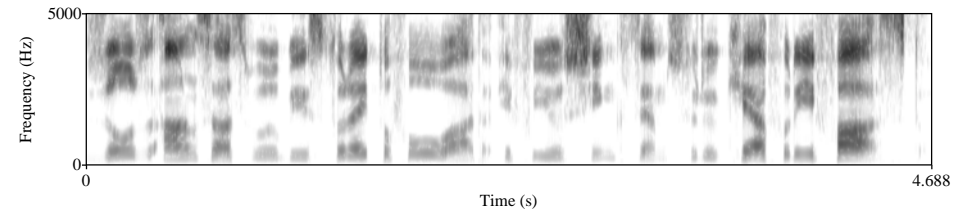
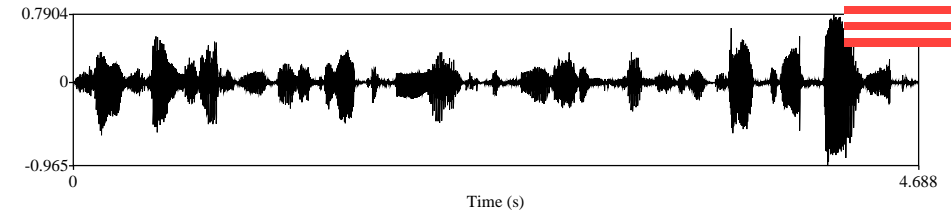
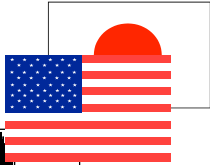
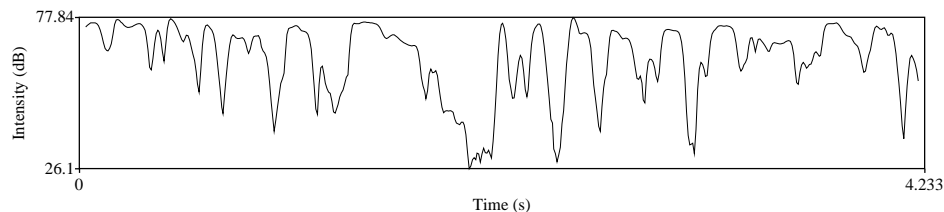
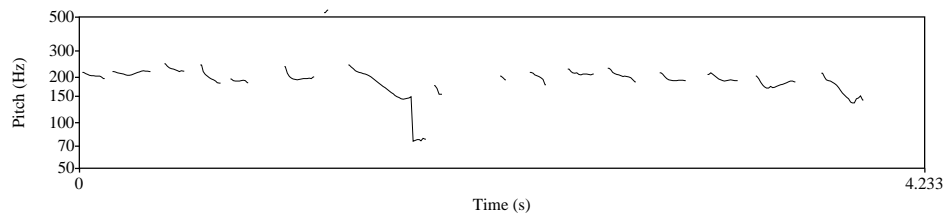
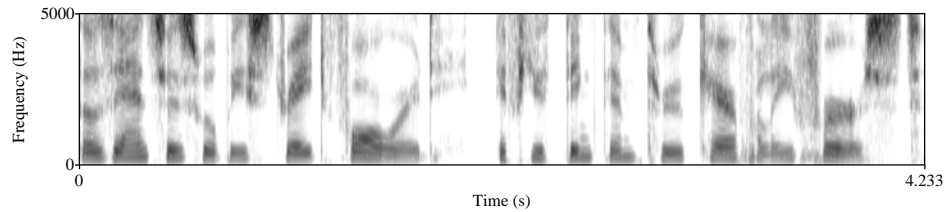
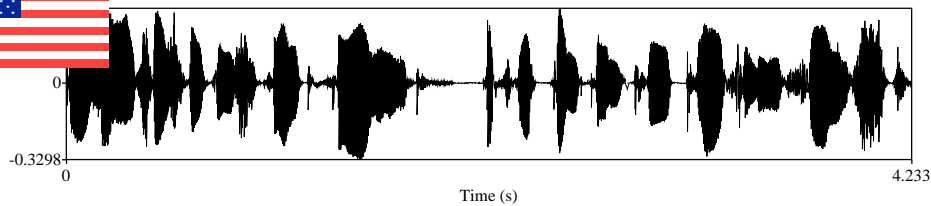
- Three basic psychological terms and their acoustic correlates

psychological	physical (acoustic)	related phenomena
pitch	fundamental frequency	intonation, word accent speaker identity
loudness	energy, intensity, sound pressure level	word accent (word stress)
duration*	duration*	rhythm
timbre	spectrum envelope	phoneme speaker identity

- It seems that two distinct terms are not prepared well for perceptual length and physical length of a sound.
- Foreign accent and prosodic features
  - Various types of prosodic deviation can be found in foreign accented speech depending on the native language of a learner and the target language.

# Basic prosodic features

- “Those answers will be straightforward if you think them through...”
- Results of acoustic analysis using Praat.
- <http://www.fon.hum.uva.nl/praat/>



# Prosodic assessment of pronunciation

- Use of various prosodic metrics to estimate prosodic quality
  - Duration-based metrics to predict “fluency” [Cucchiarini+'98,'02]
  - Model-based and non-model based prosodic metrics [Maier+'09][Huang+'10]
- Additional prosodic features used to estimate overall proficiency
  - Duration log-likelihood [Kim+'97], rate of speech [Franco+'00]
  - Linear combination of various scores to predict proficiency [Hirabayashi+'10]
- Word accent (word stress) generation assessment
  - Position [Minematsu+'97][Imoto+'99] and manner [Minematsu+'00]
- Rhythm assessment
  - Rhythm metrics [Ramus+'99,'02][Grabe+'99,'02]
- Intonation(+energy) pattern comparison bet. a student and a model
  - Word-based comparison [Suzuki+'08][Cheng+'11]
  - Multiple units for comparison [Yamashita+'05]
- Corrective feedback generation
  - Decision-tree based generation [Liao+'10], using a learner's voice [Hirose+'03]

## Duration-based metrics

- Duration-based metrics to predict “fluency” [Cucchiaroni+'98,'02]
  - 60 non-native learners of Dutch and 20 native speakers
  - Forced alignment using an ASR engine
  - 3 groups of raters, 3 raters per group (phonetician, therapist1, therapist2)
  - Fluency assessment was done for each material (sentence?).

	intrarater reliability			interrater reliability
	rater 1	rater 2	rater 3	
<b>ph</b>	.97	.94	.95	.96
<b>st1</b>	.94	.97	.96	.93
<b>st2</b>	.90	.76	.91	.90

**Table 1** Intrarater and interrater reliability coefficients (Cronbach's alpha) for the three rater groups, ph, st1, and st2.

## Duration-based metrics

- Duration-based metrics to predict “fluency” [Cucchiaroni+'98,'02]
  - Each duration-based metric is examined as its ability to predict “fluency”.

- ros = rate of speech: # segments / total duration of speech plus sentence-internal pauses
- ptr = phonation/time ratio: total duration of speech without pauses / total duration of speech plus sentence-internal pauses
- art = articulation rate : # segments / total duration of speech without pauses
- tdp = total duration of sentence-internal pauses: all silences longer than or equal to 0.2 sec
- alp = average length of pauses
- #p = # of silent pauses
- mlr = mean length of runs: average number of phones occurring between unfilled pauses of not less than 0.20 secs
- #fp = # filled pauses:  $\emptyset$ ,  $\emptyset m$
- #dy = # dysfluencies (repetitions, restarts, repairs)

	Phoneticians	Speech therapists 1	Speech therapists 2
ros	.93	.91	.90
ptr	.86	.88	.89
art	.88	.84	.81
#p	-.84	-.89	-.89
tdp	-.81	-.86	-.86
alp	-.65	-.62	-.65
mlr	.85	.86	.88
#fp	.34	.33	.38
#dy	.42	.48	.40

**Table 5** Correlations (corrected for attenuation) between the fluency ratings by the three rater groups and the quantitative measures.

# Speech rhythm metrics

- The three rhythm classes
  - Stress-timed languages: English, German, Dutch, Portuguese, etc
  - Syllable-timed languages: French, Italian, Spanish, Cantonese Chinese, etc
  - Mora-timed languages: Japanese, etc
  - X-timed = the *perceptual* interval between two consecutive Xes is constant
    - Stress isochrony, syllable isochrony, and mora isochrony
- Pairwise Variability Index (PVI) [Grabe+'99,'02]

- Raw PVI (rPVI) and normalized PVI (nPVI)

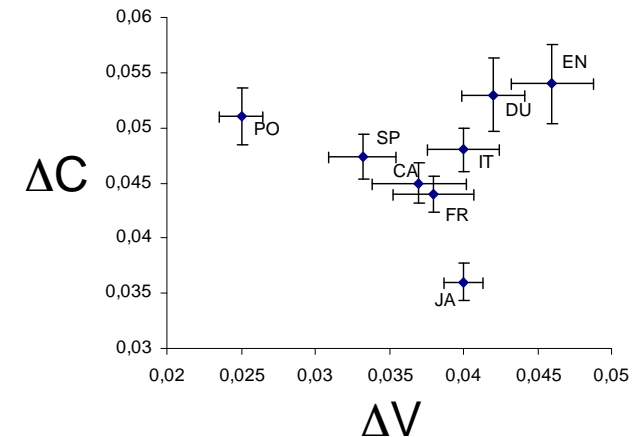
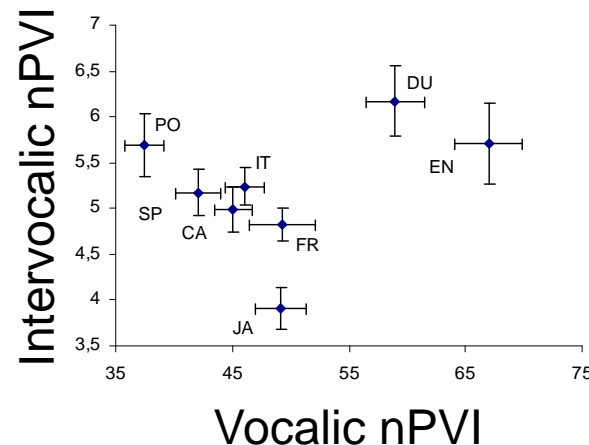
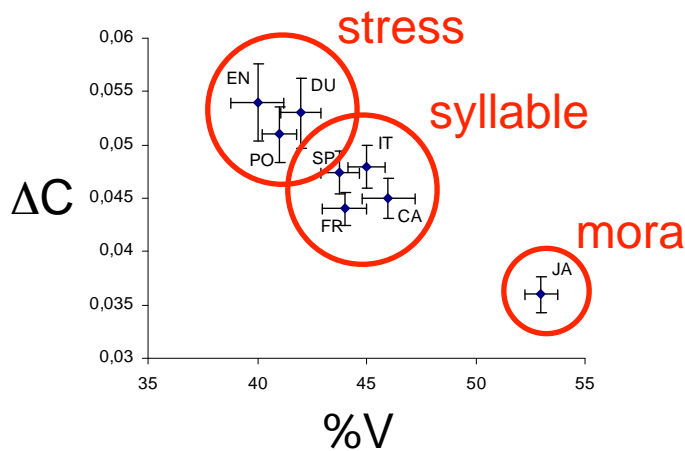
$$rPVI = \frac{100}{m-1} \sum_{k=1}^{m-1} |d_k - d_{k+1}|$$

$$nPVI = \frac{100}{m-1} \sum_{k=1}^{m-1} \frac{|d_k - d_{k+1}|}{(d_k + d_{k+1})/2}$$

- $d_k$  is the duration of the k-th interval. m is the number of intervals.
- “interval” is the vocalic interval or the consonantal interval.
- Used to classify input utterances as one of the three rhythm groups

# Speech rhythm metrics

- Combination of durational statistics of  $\Delta V$ ,  $\Delta C$  and %V [Ramus'99,'02]
  - $\Delta X$  : standard deviation of the duration of Vowel intervals or Consonant intervals within a sentence
    - X interval: interval of a X or a sequence of consecutive Xes
    - Intervocalic interval: interval of a consonant or a consonant sequence
  - %V : percentage of duration taken up by vowel intervals within a sentence
  - Used to cluster various languages in terms of their rhythmic structure.



# Various prosodic metrics

- Lang-independent feature set for prosody evaluation [Maier+'09]
  - Word-based 21 metrics + sentence-based 16 metrics
    - Related to F0, energy, and duration
    - 37 metrics x [max, min, mean, std] = 148 features
  - Text-independent 187 prosodic features
  - Support vector regression to predict prosodic quality

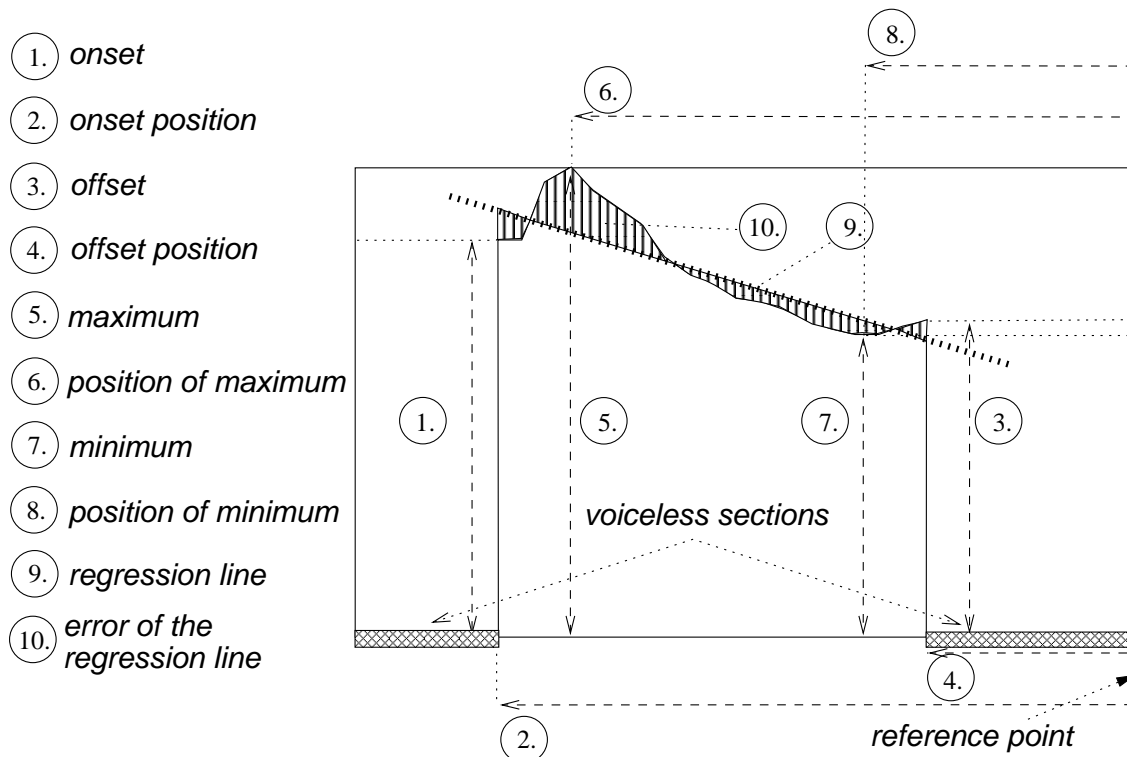


Table 1: Correlations between the automatic evaluation system and the human raters in comparison to the inter-rater correlation

language	inter-rater	word-based		text-independent	
		SVR	SVR (CFS)	SVR	SVR (CFS)
German	<b>0.88</b>	0.89	<b>0.92</b>	<b>0.88</b>	0.75
Japanese	<b>0.92</b>	-	-	0.76	<b>0.83</b>



# Linear combination of many features

- Prediction of overall naturalness [Hirabayashi+'10]
  - Many possible features are linearly combined to predict pron. scores
    - LL with native HMMs =  $LL_{\text{native}}$
    - LL with HMMs adapted into non-native =  $LL_{\text{non-native}}$
    - LL obtained with phone-loop grammar and native HMMs =  $LL_{\text{best-native}}$
    - LL ratio =  $LR = LL_{\text{native}} - LL_{\text{non-native}}$
    - Posterior probability =  $LR' = LL_{\text{native}} - LL_{\text{best-native}}$
    - Another LL ratio =  $LR_{\text{adapt}} = LL_{\text{best-native}} - LL_{\text{best-non-native}}$
    - Another LL ratio =  $LR_{\text{mother}} = LL_{\text{best-native}} - LL_{\text{best-mother-tongue}}$
    - Phoneme recognition rates (rates of correct, substitution and deletion)
    - Word recognition results (rates of correct, substitution and deletion)
    - Standard deviation of power and F0
    - Phoneme-based rate of speech

# Linear combination of many features

- Prediction of overall naturalness [Hirabayashi+'10]
- Results of linear prediction of pronunciation scores

Table 2: Correlation between acoustic measures and pronunciation score ("\*" denotes a text-independent measure)

Measure	1 sentence	5 sentences	10 sentences
$LL_{native}$	-0.466	-0.625	-0.669
$LL_{non-native}$	<b>-0.638</b>	<b>-0.771</b>	<b>-0.804</b>
<b>LR</b>	<b>0.800</b>	<b>0.859</b>	<b>0.880</b>
* $LL_{best}$	-0.473	-0.613	-0.660
* $LR_{mother}$	<b>0.719</b>	<b>0.804</b>	<b>0.811</b>
* $LR_{adap}$	<b>0.772</b>	<b>0.827</b>	<b>0.822</b>
$LR'$	0.214	0.273	0.349
Phoneme recog( <i>Sub.</i> )	-0.298	-0.567	-0.662
Phoneme recog( <i>Del.</i> )	0.056	0.116	0.220
Phoneme recog( <i>Cor.</i> )	0.299	0.461	0.483
Word recog(WSJ, <i>Cor.</i> )	0.102	0.163	0.261
Word recog(EURO, <i>Cor.</i> )	0.113	0.256	0.281
* $Power$	-0.066	-0.057	-0.002
* $Pitch(F_0)$	0.495	0.638	0.691
<b>Rate of speech</b>	<b>0.523</b>	<b>0.692</b>	<b>0.773</b>

Table 3: Correlation between combination of acoustic measures and learner's pronunciation score by human raters

Acoustic measures	Number of sentences for evaluation		1 sentence		5 sentences		10 sentences	
	CLOSED	S.OPEN	CLOSED	S.OPEN	CLOSED	S.OPEN	CLOSED	S.OPEN
$LL_{non-native}, LR, LR_{mother}, Power, Phoneme\ recog(Del.)$	0.851	0.804	0.910	0.851	0.927	0.864		
Word recog(EURO, <i>Cor.</i> ), $LR, Power, Word\ recog(WSJ, Cor.)$	0.815	0.770	0.902	0.866	0.929	0.884		
Word recog(EURO, <i>Cor.</i> ), $LR, Power$	0.814	0.771	0.893	0.858	0.918	0.887		
$LL_{best}, LR_{mother}, Power$	0.819	0.779	0.891	0.853	0.912	0.878		

# Word stress detection

- Modeling of (un)stressed syllables [Minematsu+'97][Imoto+'02]
  - HMM-based modeling of syllables (C..CVC..C)
    - Syllable structure dependent (V, C..CV, VC..C, and C..CVC..C)
    - Vowel type dependent (short vowels, long vowels, and diphthongs)
    - Vowel position dependent (head, tail and other in a word)

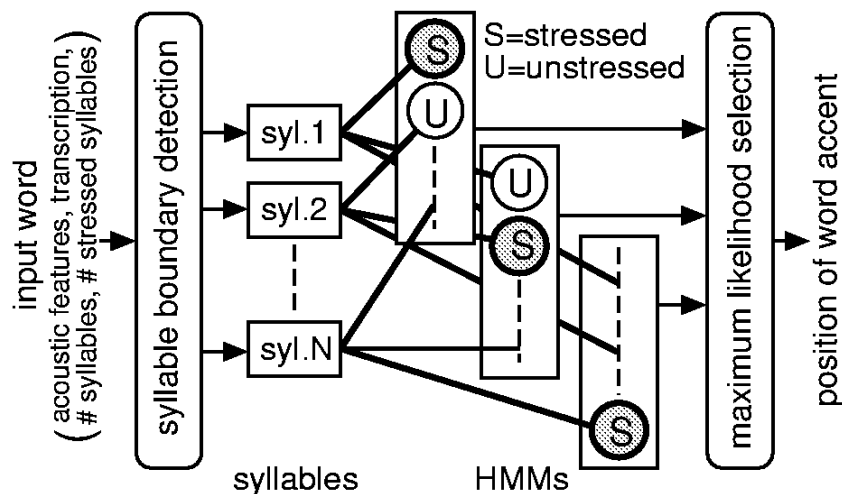


Figure 1: Accent detection using syllable boundaries

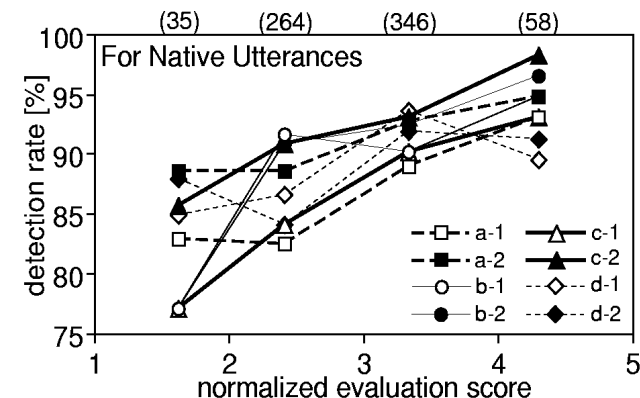


Figure 2: Detection rates for native speakers

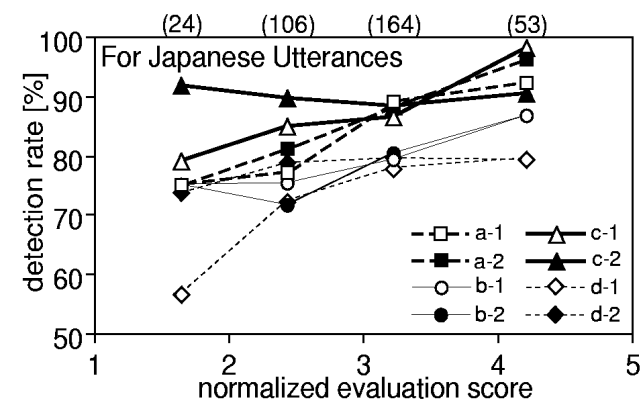


Figure 3: Detection rates for Japanese students

# Manner of word stress generation

- Estimation of pron. habit in word stress generation [Minematsu+'00]
  - Word accent in Japanese : pitch accent
    - Fundamental frequency (F0)
  - Word accent in English : stress accent
    - Four multiple factors of F0, duration, power, and vowel quality
    - Japanese tend to produce English word stress mainly by pitch change [Shibuya'96].
  - Stress / unstress identification using multiple weights
    - $P(o|M) = P(F_0|M)^{w_{F_0}} P(dur|M)^{w_d} P(pow|M)^{w_p} P(env|M)^{w_e}$
    - The optimal weights represent the pronunciation habit of individual students.
    - Larger  $w_{F_0}$  is observed in word stress generation by Japanese?

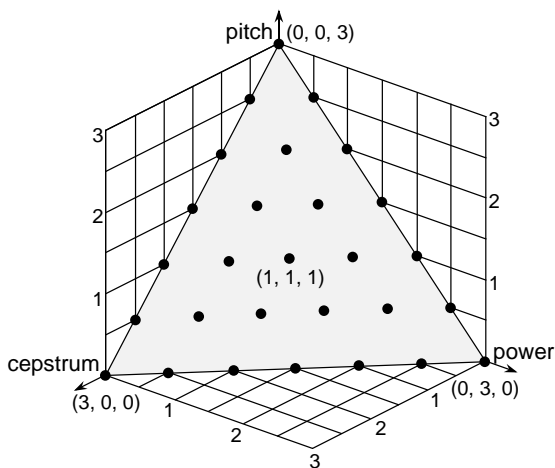
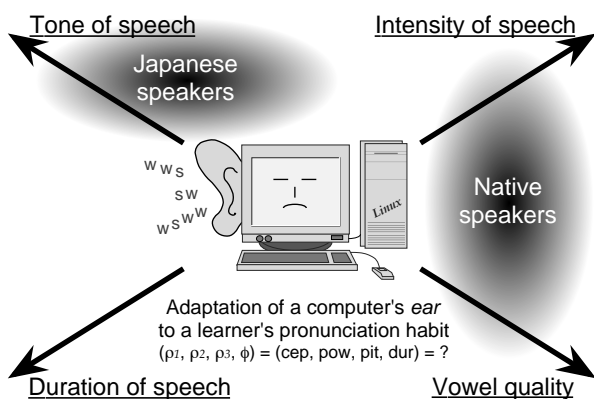
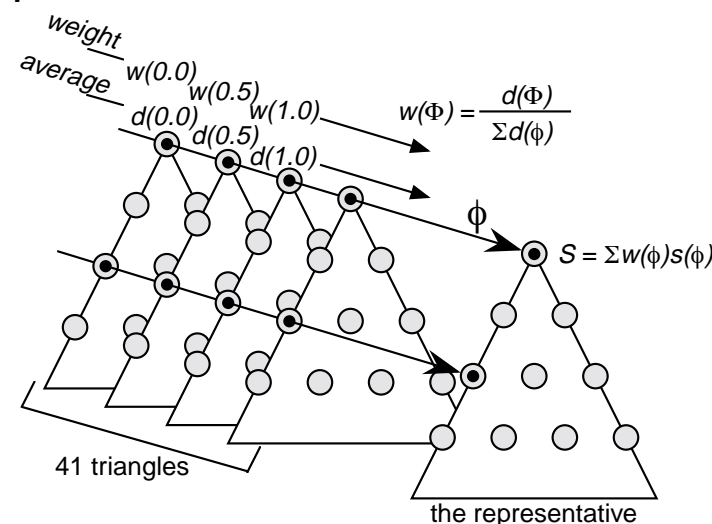
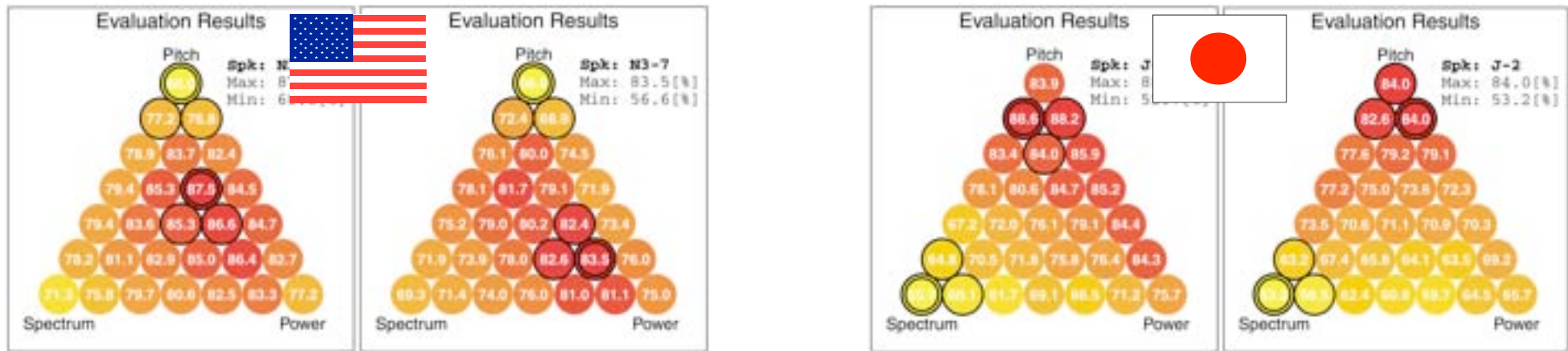


Figure 3: Distribution of the weight combination of  $\{\rho_s\}$

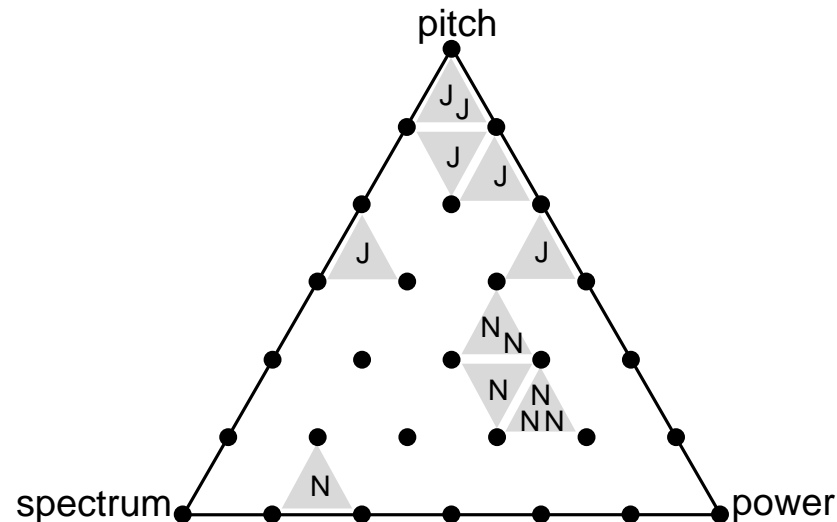


# Manner of word stress generation

- Results of pronunciation habit estimation [Minematsu+'00]
- Four examples of estimation results: two natives and two Japanese



- Locations of the optimal weights of 7 natives and 6 Japanese students



# Tone error detection using DT

- Use of a decision tree to detect tone errors [Liao+'10]
  - A decision tree tells us “why and how the input tone pattern is bad”.
    - This info. can be used as easy-to-understand feedback to students.
  - 1 to 5 point human rating scores are converted into binary rating
    - 1/2 = bad and 3/4/5 = good, which are used as labels for training decision trees.
  - A syllable is divided into three segments and F0 mean is obtained from each.
    - F0 differences bet. any segment pair are added to a feature vector.

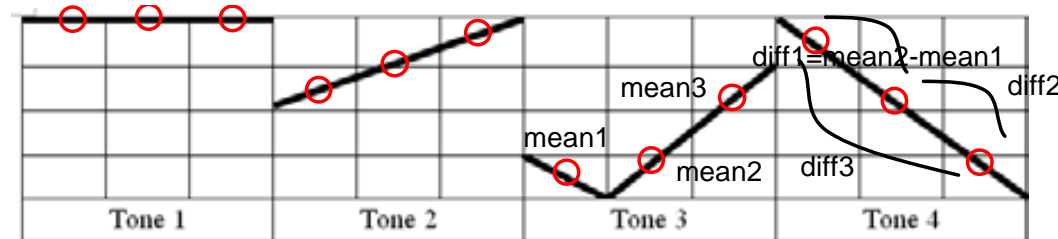
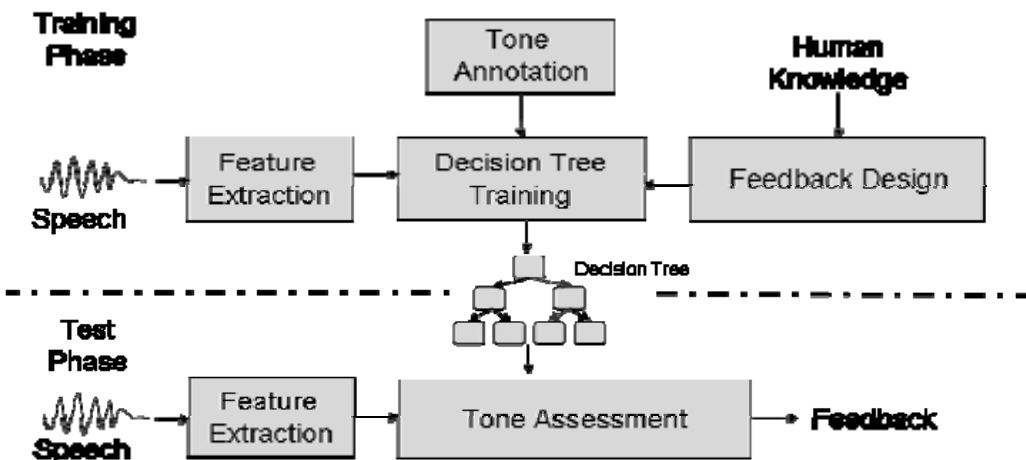
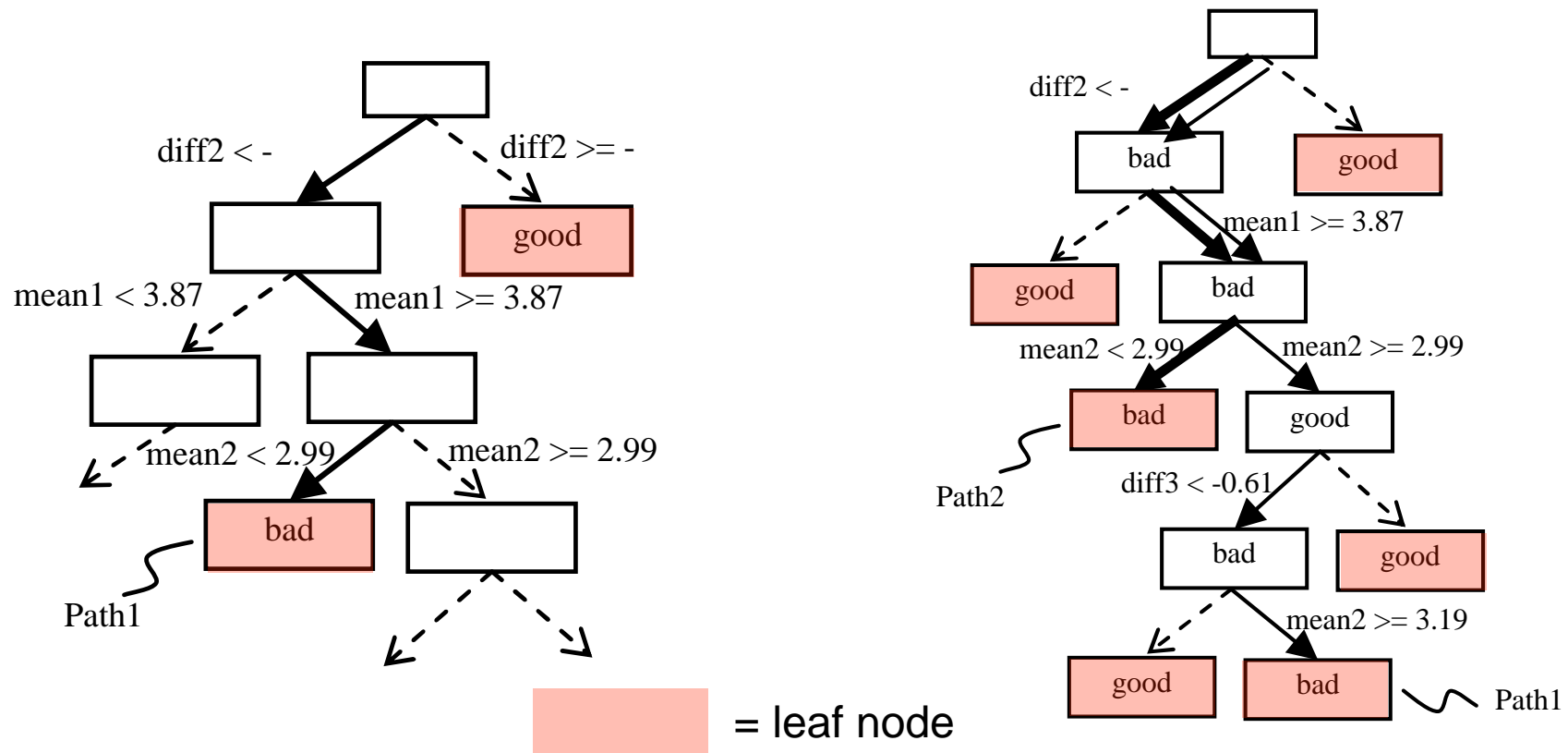


Figure 3: Illustration of feature vector extraction.

Figure 1: A decision tree based tone assessment system block diagram with training (upper) and testing (lower) subsystems.

# Tone error detection using DT

- Use of a decision tree to detect tone errors [Liao+'10]
  - Right-context-dependent models are adopted.
  - A set of questions prepared in terms of F0mean and F0diff.
  - Approx. 90% of correct binary judgment (good or bad) for testing data.
  - Potential use of traversed paths for feedback generation



# Utterance-based prosodic comparison

- Consideration of characteristics of Japanese English
  - Word-by-word pronunciation [Sugito'98]
  - Too many or too few peak-and-valleys in intonation [Shimizu'95]
- Prosodic comparison between utterances [Yamashita+'05]
  - Multiple units such as word, word boundary, prosodic phrase, and sentence
  - Each unit is determined by phoneme labels obtained from an HMM aligner.

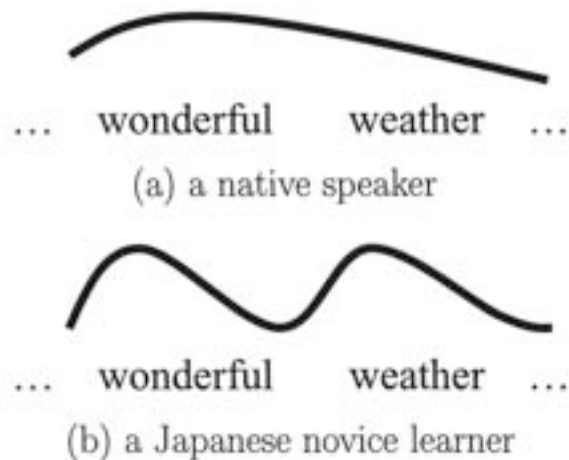
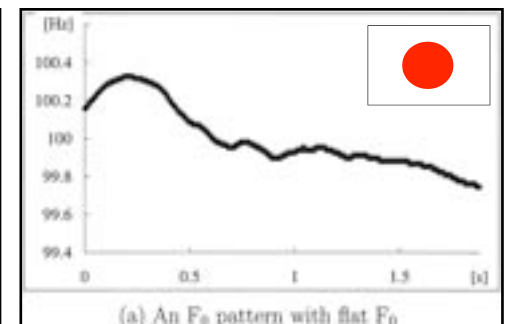
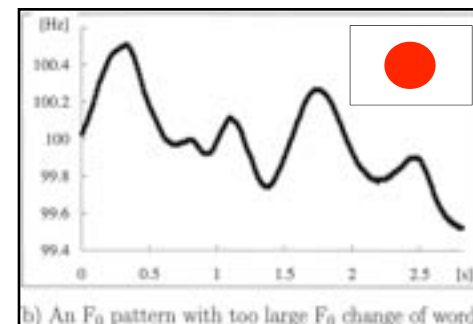
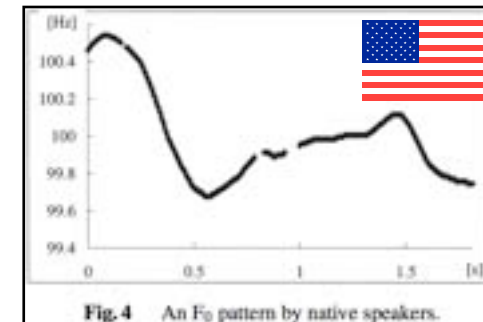


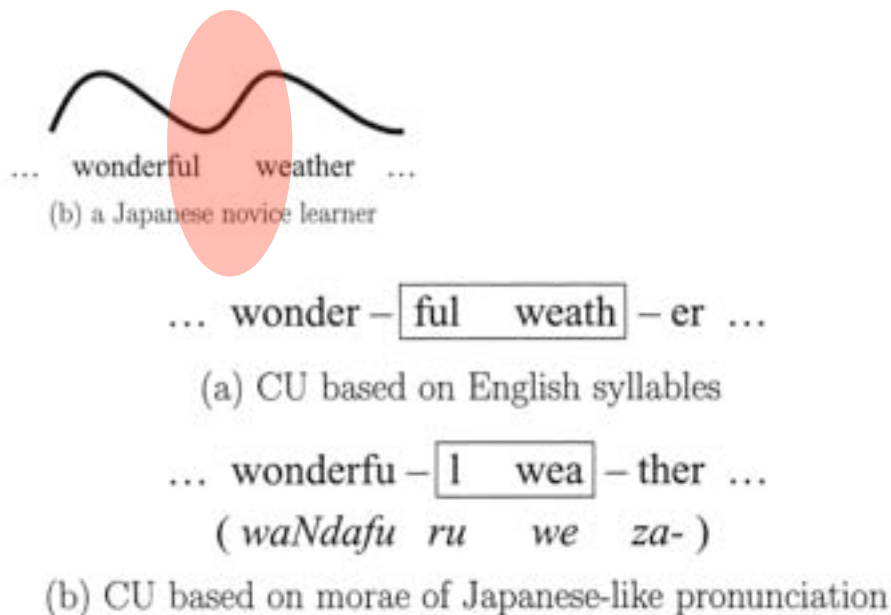
Fig. 1 Typical  $F_0$  patterns at a word boundary.



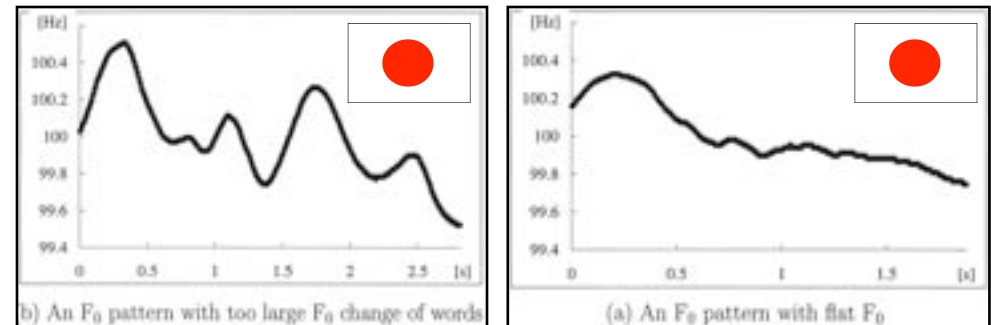


# Utterance-based prosodic comparison

- Prosodic comparison between utterances [Yamashita+'05]
  - F0 contour, power contour, total duration, word duration, pause duration
  - Deviation of an observed contour from its 1-st or 2-nd order approximation.
    - Very low deviation expects that the contour is flat.
  - Linear regression of these prosodic scores to predict human scores.
  - The correlation bet. machine and human is not high.



**Fig. 2** Definition of the comparison unit.



**Table 4** The correlation between the teachers' score and automatic scoring.

measure set	closed	open
set-0 (baseline)	0.40	0.41
set-I (proposed)	0.69	0.51

# Prosodic comparison with DTW

- Prosodic assessment with word importance factors [Suzuki+'08]
  - Word segmentation is done by forced alignment using an ASR engine.
  - Word-based prosodic comparison between a student and a teacher
    - Ratio of word-based durations, DTW of stress patterns (log energy contour) and DTW of intonation patterns (F0 + log energy contour)
  - Word class importance factor is introduced to improve the performance.
    - A sentence score is obtained as linear combination of the word-based scores.
    - Different words should have different contributions to the final prosodic assessment.
    - DTs are trained so that linear regression errors should be minimized.
      - Leaf-node-dependent linear regressions are used.

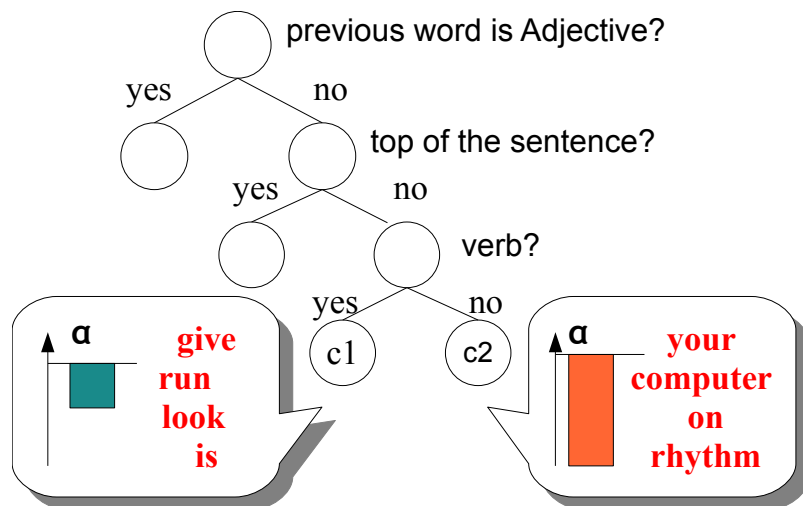
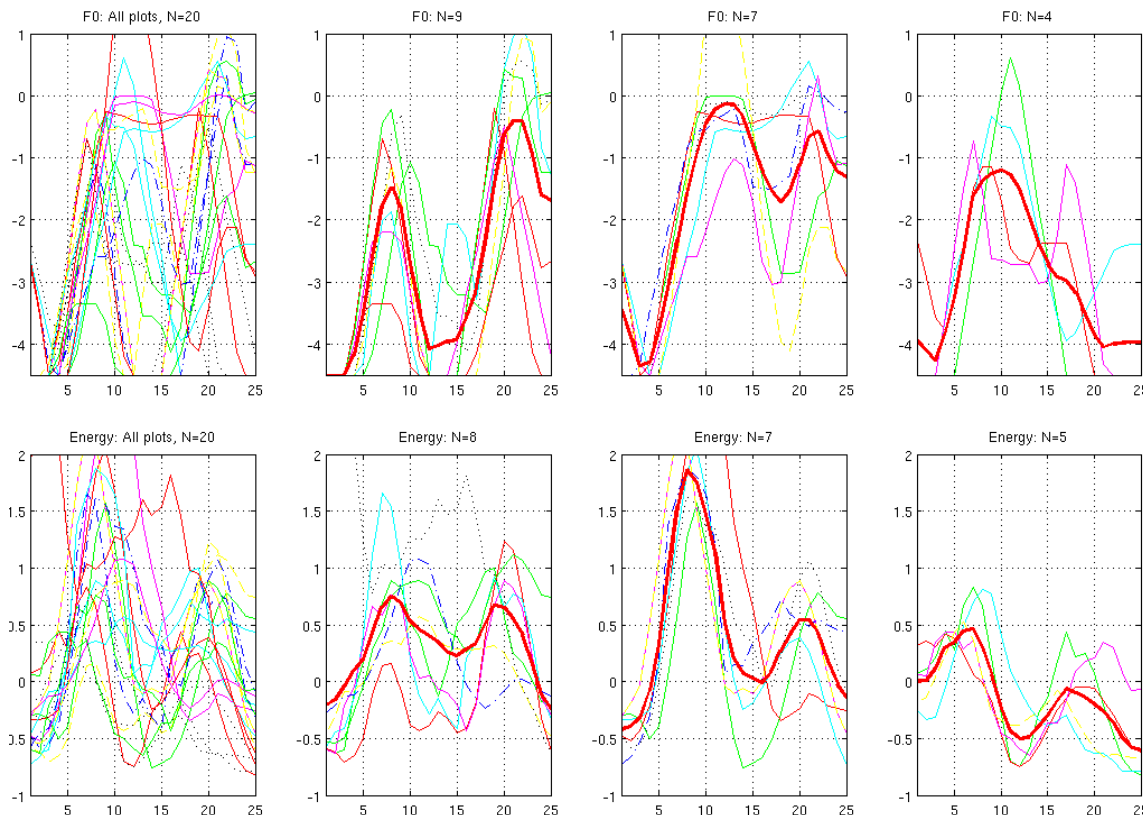


Table 8: Results of intonation evaluation using integration of both scores

	Intonation only	Both scores
Closed	0.59	0.64
Open	0.45	0.48

# Prosodic comparison w/o DTW

- Word-based modeling of F0 and energy contours [Cheng'11]
  - 25-point resampling of prosodic patterns for each word
  - Each word has three templates for each of F0 and energy contours.
  - Euclidean distance is used to quantify a difference between a student pattern and a model pattern.



Clustering results of F0 contours and energy contours of word "strategy" using 20 utterances

# Prosodic comparison w/o DTW

- Phoneme duration and inter-word silence duration [Cheng'11]
  - Phoneme duration likelihood, similar to [Franco+'00]
    - $\log\_seg\_prob = \frac{1}{N-2} \sum_{i=2}^{N-1} \log(Pr(D_i^{seg}))$
  - Inter-word silence duration likelihood
    - $\log\_sil\_prob = \frac{1}{M} \sum_{i=1}^M \log(Pr(D_i^{sil}))$
- Linear regression of F0, energy, and duration scores to predict human scores

Features	Correlation
F0	0.67
Energy	0.67
F0 + Energy	0.73
<i>iw_log_seg_prob</i>	0.54
<i>log_seg_prob</i>	0.76
Linear regression	0.80

Table 2: Correlations using different features.

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# Text-to-speech technology

- Two main streams of TTS technology
  - Unit-selection-based generation of waveforms
    - Selection and concatenation of waveform templates
  - HMM-based generation of waveforms
    - Cepstrum-vocoder based generation
  - Comparison of the two frameworks
    - The former tends to be higher in naturalness.
    - The latter is higher in flexible control.
- Use of TTS technology for CALL [Handley+'05][Black'07]
  - As model pronunciation
    - Use of TTS in pronunciation training
    - Required naturalness is extremely high.
  - As reading machine
    - Use of TTS in dictation practice, shadowing practice, etc
    - Required naturalness is high.
  - As dialogue partner in a dialogue-based CALL system
    - Required naturalness is not so high.



Oxford-Hachette French Dictionary

# Some demos of high-quality TTS

- “Globalvoice English” produced by HOYA service corp., Japan
- <http://voicetext.jp>
- Used in dictation practice and shadowing practice in college English classes

The screenshot displays a Windows desktop environment. On the left, a Windows Internet Explorer browser window is open, showing a webpage titled "Xi'an Information". The page features a photograph of a traditional Chinese building and contains the following text:

**Xi'an Information**  
 Hosting City: Xi'an, China

Xian, the eternal city, records the great changes of the Chinese nation just like a living history book. Called Chang'an in ancient times, Xian is one of the birthplaces of the ancient civilization in the Yellow River Basin area of the country. During Xian's 3,100 year development, 13 dynasties such as Western Zhou (11th century BC - 771 BC), Qin (221 BC - 206 BC), Western Han (206 BC - 24 AD) and Tang (618 - 907) placed their capitals here. So far, Xian enjoys equal fame with Athens, Cairo, and Rome as one of the four major ancient civilization capitals.

*It was more powerful than Rome...  
 Bigger than Constantinople...  
 It was the mother of all cities...  
 Richer than Alexandria...  
 It was the center of the world...*

On the right, a "Globalvoice English Pro" software window is open. The window title is "Globalvoice English" and it displays the same text as the browser window. The software interface includes a menu bar with options like "文書", "対話", "ウェブ", "単語帳", "設定", and "HELP". Below the menu is a toolbar with icons for "新規文書", "開く", "保存", "別名で保存", "印刷", "モニタモード", and "ファイル名 新規文書 - 無題". At the bottom of the window, there are playback controls (play, stop, pause, fast forward, fast backward) and three speaker icons labeled "KATE", "PAUL", and "MIYU". The Windows taskbar at the bottom shows the system tray with the date "2011/09/19" and time "7:53".

# Use of re-synthesis technology

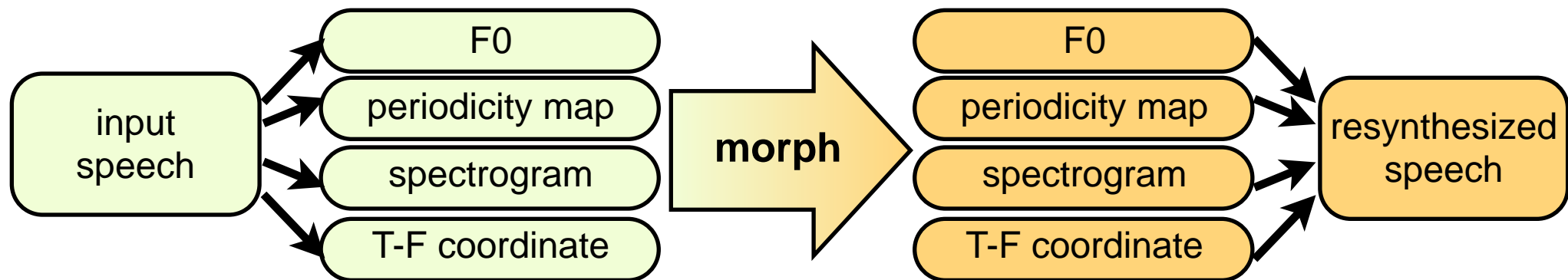
- STRAIGHT [Kawahara'06]

- High-quality analysis-resynthesis tool

- Decomposition of speech into

- Fundamental frequency, spectrographic representations of power, and that of periodicity

- High-quality speech morphing tool



- Spectrographic representation of power

- F0 adaptive complementary set of windows and spline based optimal smoothing

- Instantaneous frequency based F0 extraction

- With correlation-based F0 extraction integrated

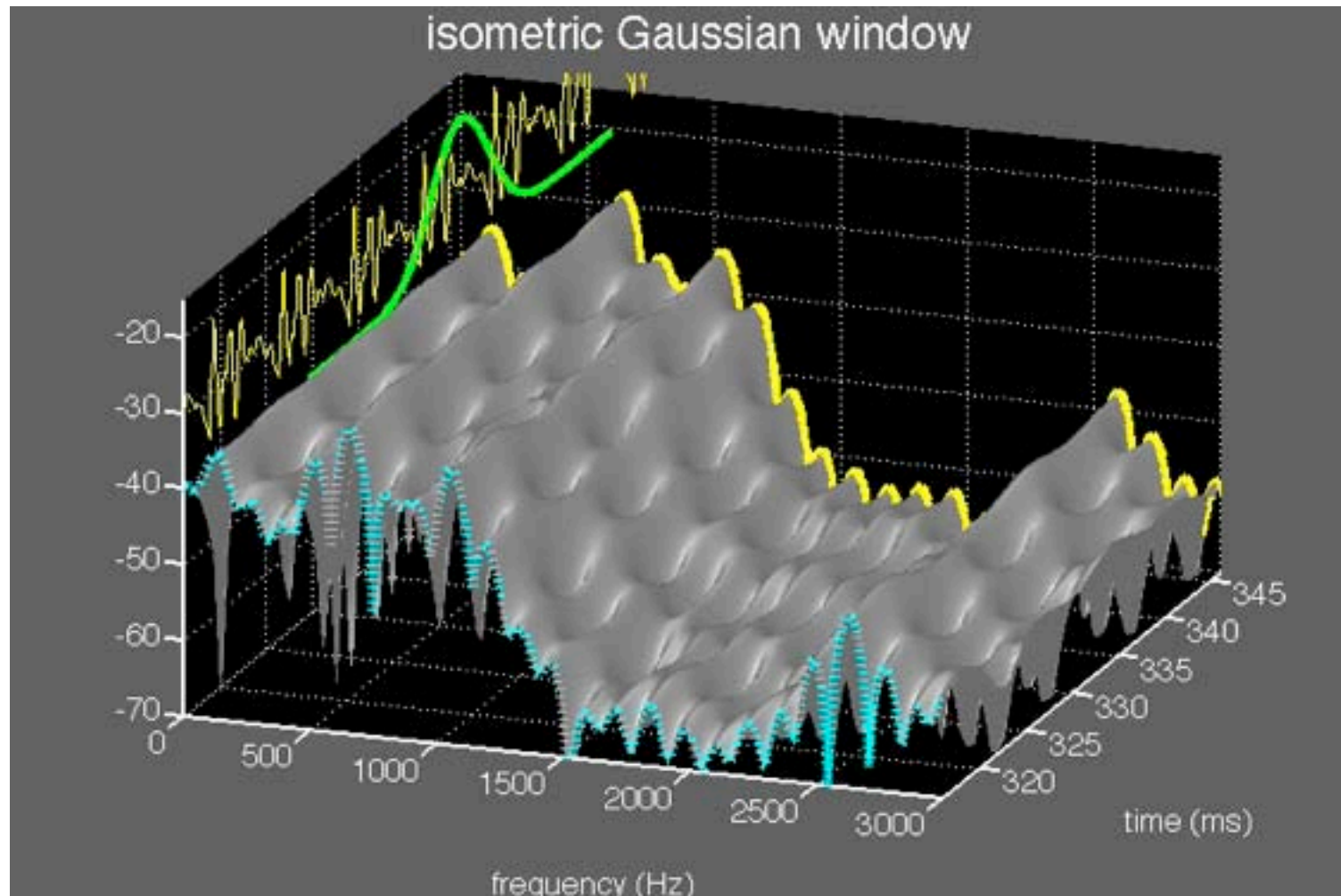
- Spectrographic representation of periodicity

- Harmonic analysis based method



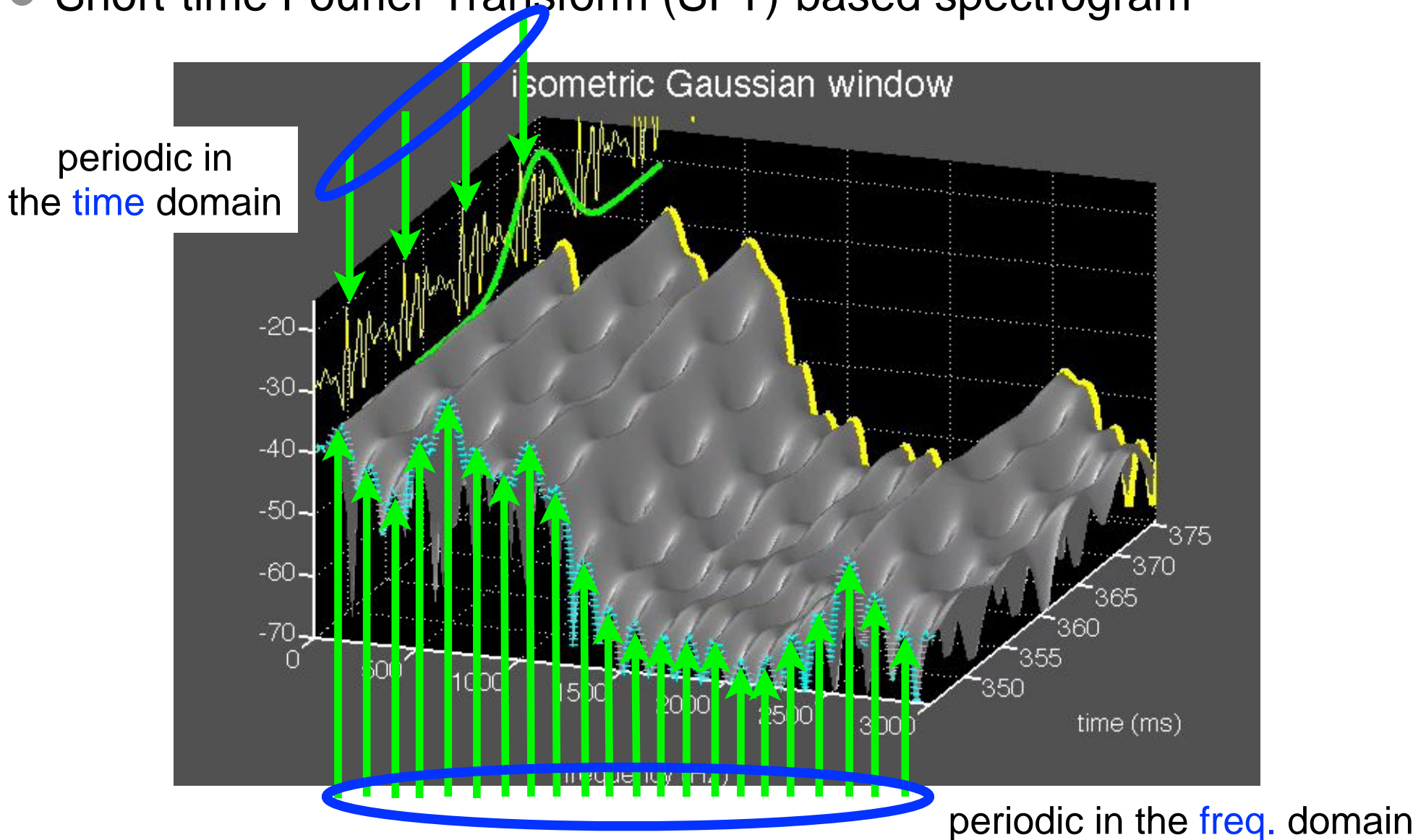
# Representation based on SFT

- Short-time Fourier Transform (SFT)-based spectrogram



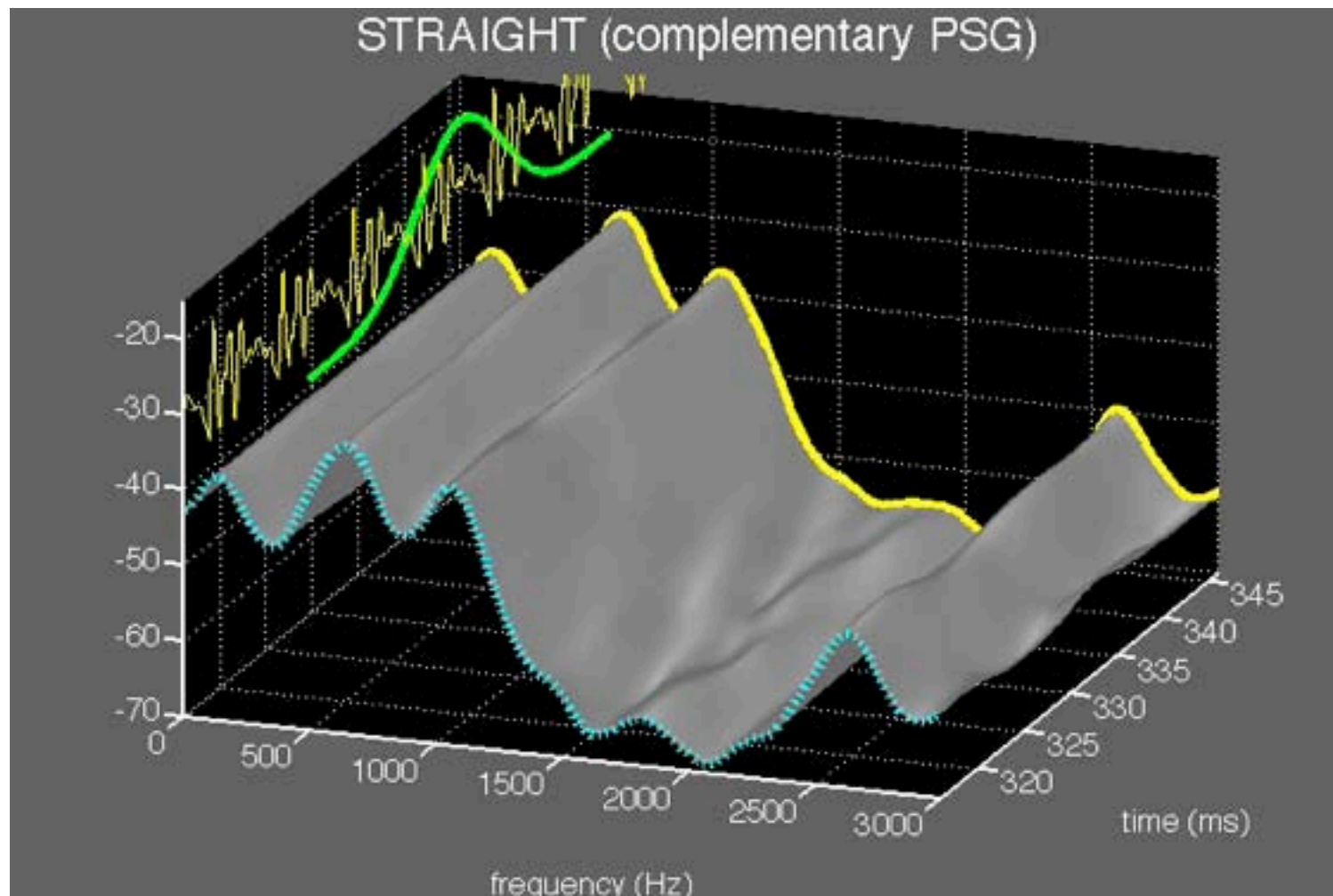
# Representation based on SFT

- Short-time Fourier Transform (SFT)-based spectrogram



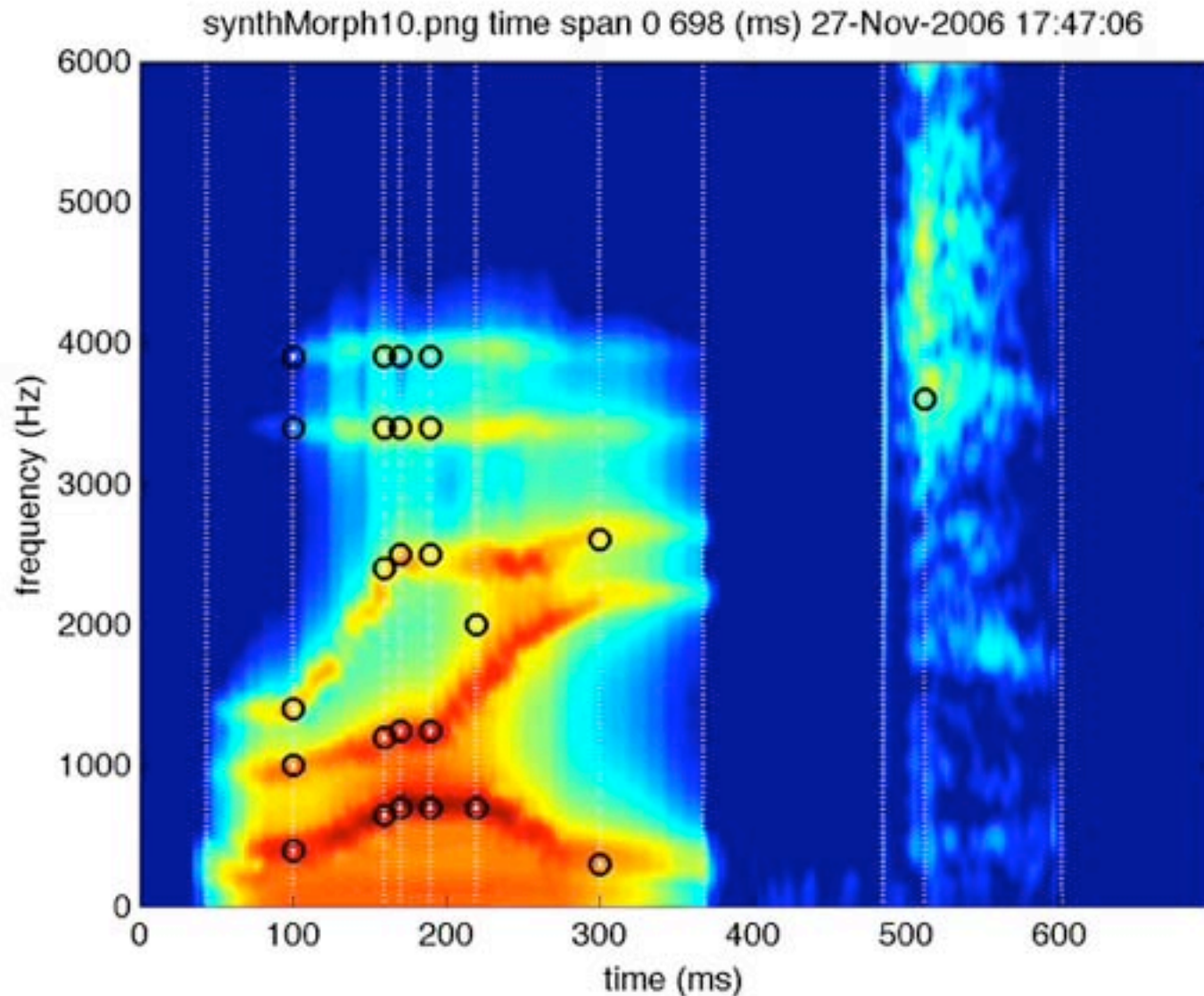
# Representation based on STRAIGHT

- Spline-based optimum smoothing reconstructs the underlying smooth time-frequency representation.



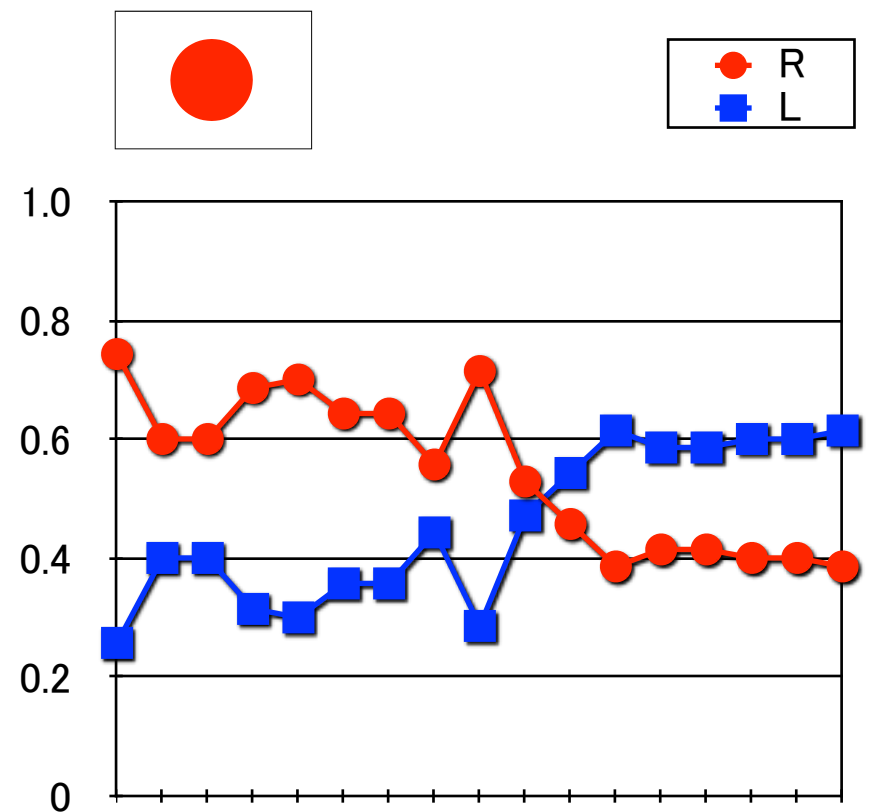
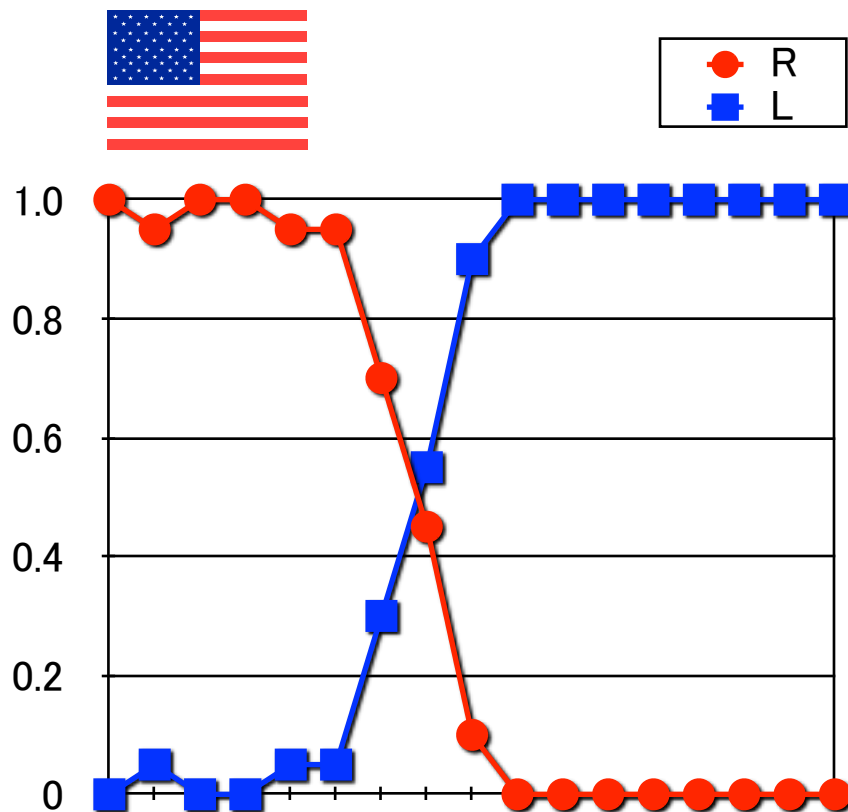
# Use of morphed utterances

- R to L morphing bet. r/l-ight generated by Klatt synthesizer [Kubo+'98]



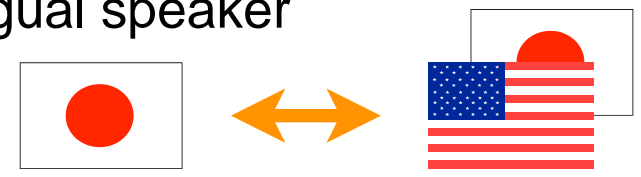
# Use of morphed utterances

- Results of categorical listening tests [Kubo+'98]
  - 1 American listener
  - 7 Japanese listeners
  - Probability of perceiving R or L in the presented sounds

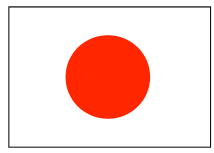


# Use of morphed utterances

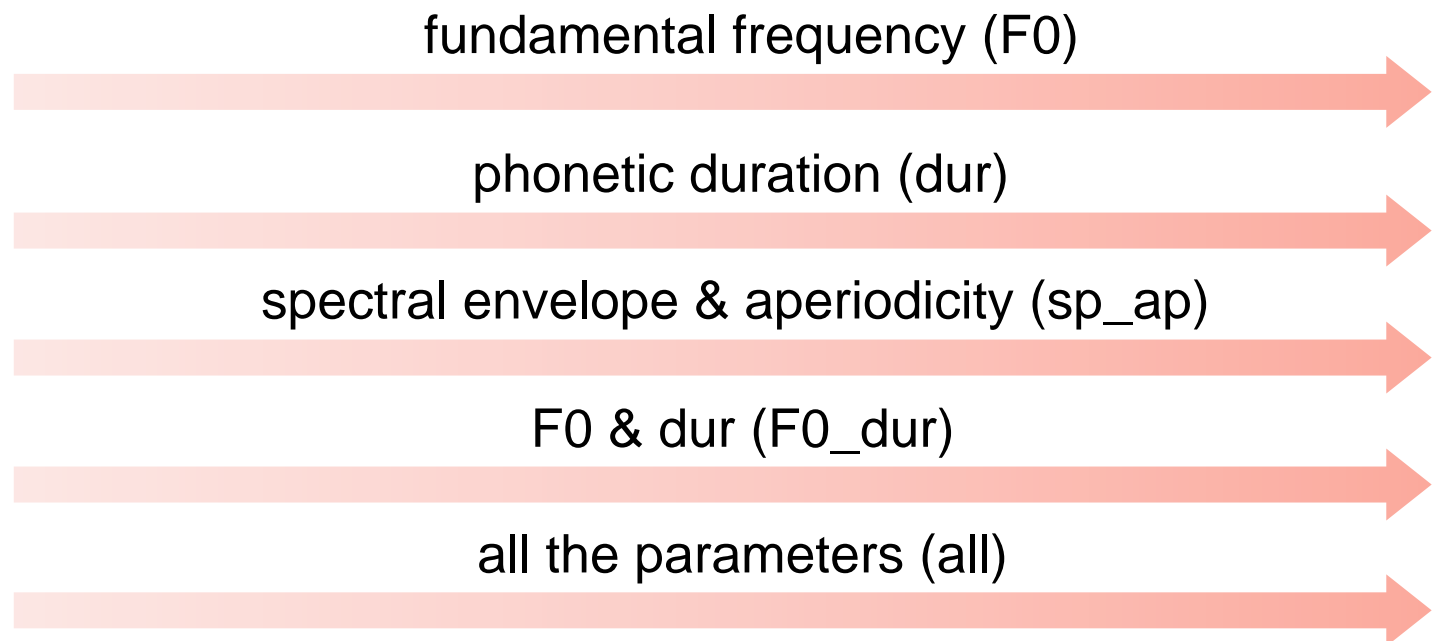
- Morphing of a native utterance and its accented version [Kato+'11]
  - Use of a pair of word utterances spoken by a bilingual speaker
    - Normal Tokyo Japanese
    - Heavily American accented Japanese



***igaku*** (medical science)



1.



0      0.25      0.5      0.75      1

morphing rate

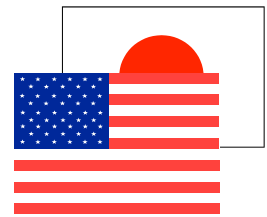
3.

4.

5.

6.

7. = 2.

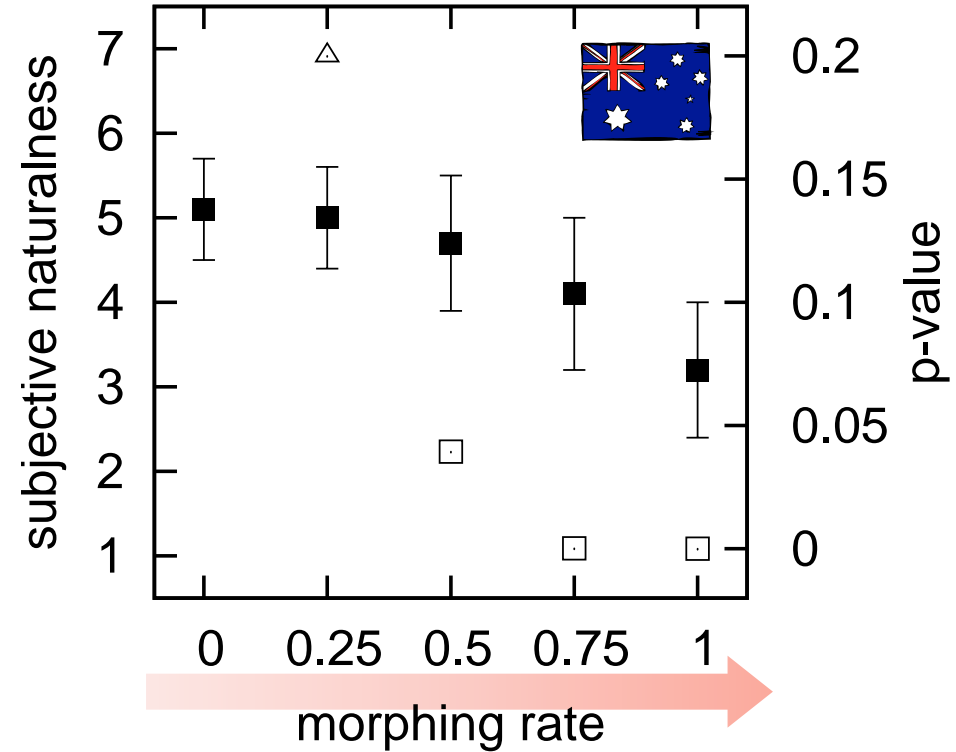
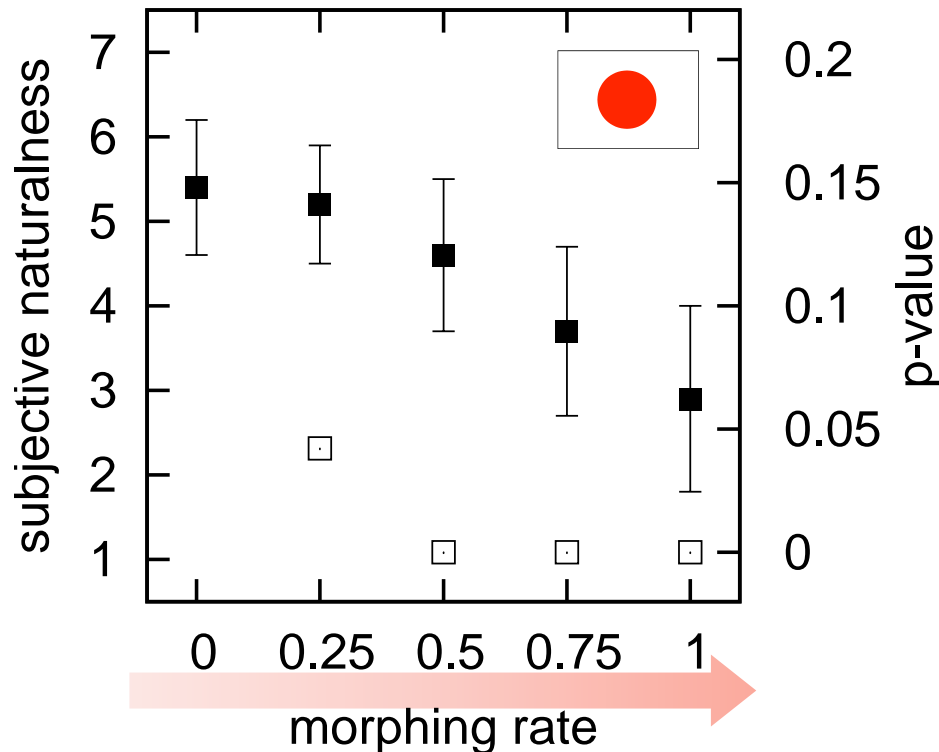


2.

# Use of morphed utterances

- Prosodic insensitivity of foreign listeners [Kato+'11]
  - 42 Japanese listeners 
  - 15 Australian listeners 
  - Judgement of naturalness as Tokyo Japanese

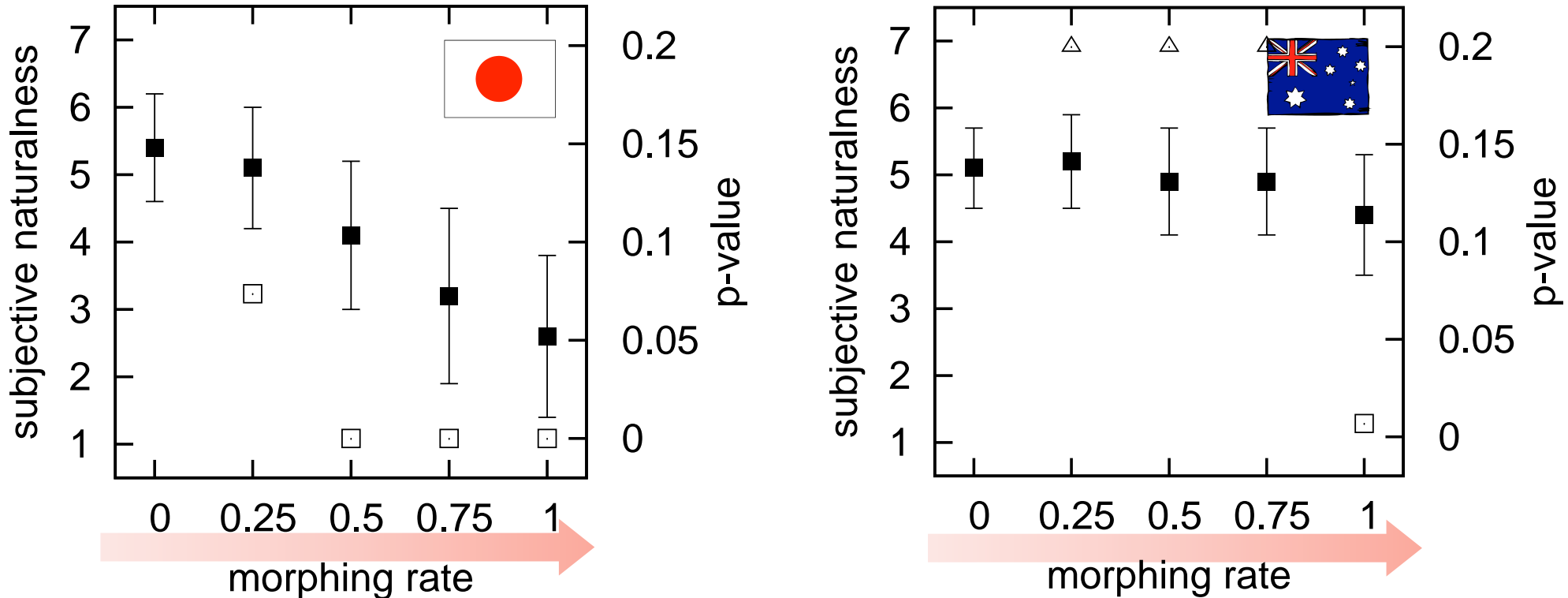
## Morphing only in terms of duration



# Use of morphed utterances

- Prosodic insensitivity of foreign listeners [Kato+'11]
  - 42 Japanese listeners 
  - 15 Australian listeners 
  - Judgement of naturalness as Tokyo Japanese

## Morphing only in terms of F0





# Feedback in a learner's own voice

- Prosodic correction of a learner's utterance [Hirose+'03]
  - The corrected version is given to a learner of Japanese as feedback
  - The feedback is generated in his/her own voice.
    - PSOLA (Pitch Synchronous OverLap Add)-based implementation
  - Easy comparison between a bad example and a good one.

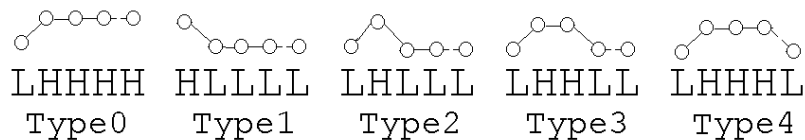


Figure 1: Binary description of 4-mora Japanese pitch accent patterns. The fifth circle point in each pattern represents pitch level of the attached particle. Type 0 can be distinguished from type 4 by the particle's pitch level.

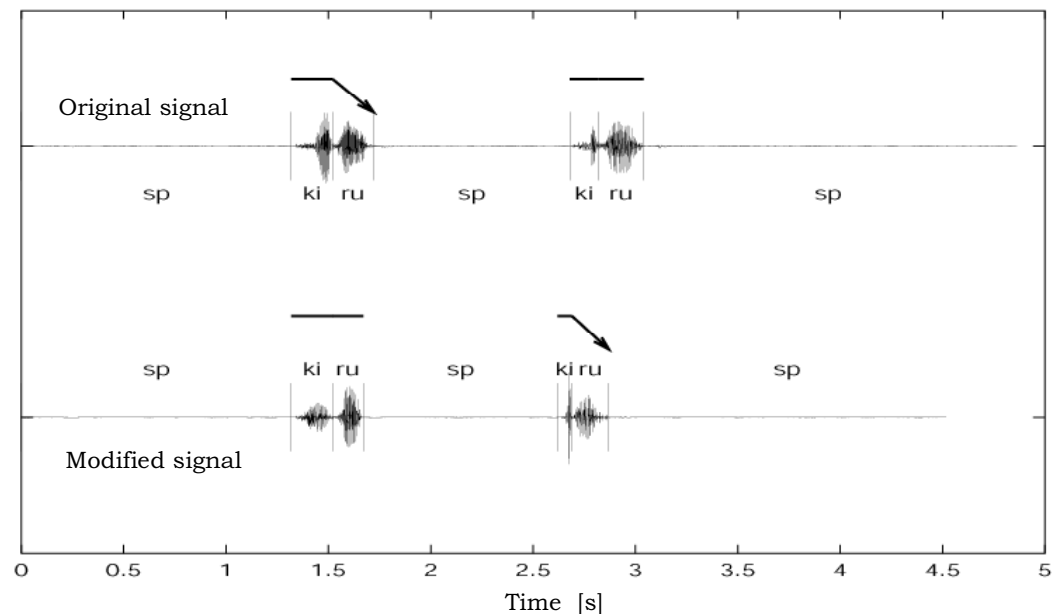


Figure 3: An example of visual feedback for the couple of homonyms "kiru (to wear)" and "kiru (to cut)."

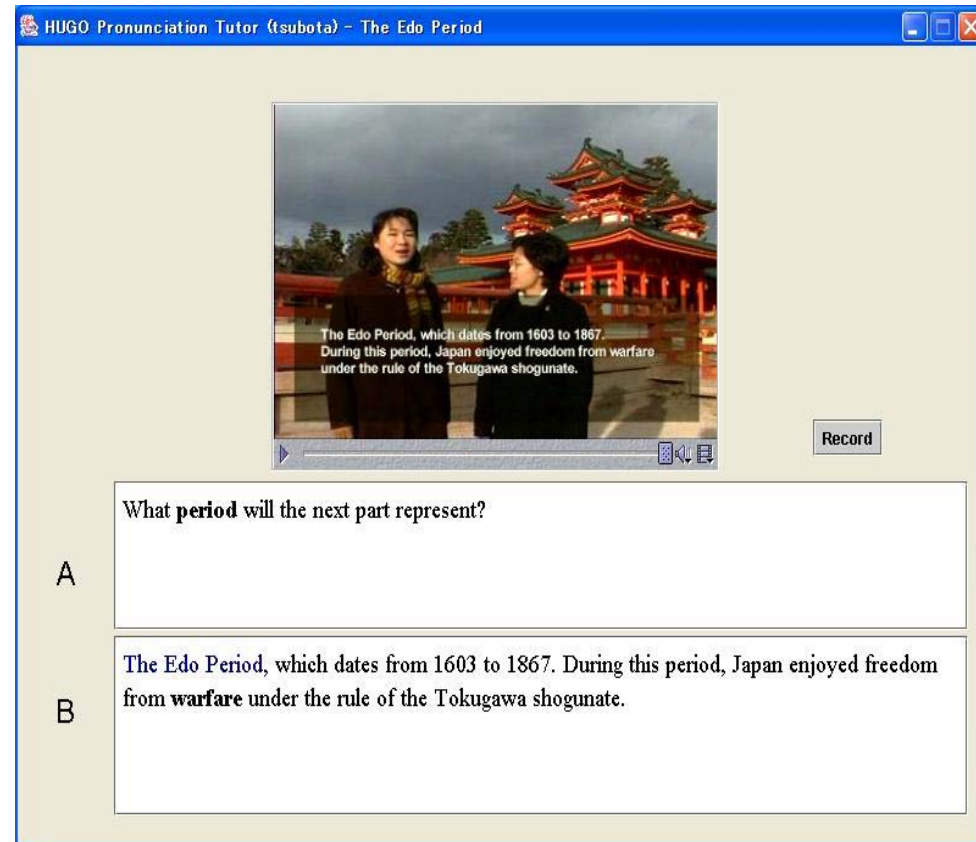


# OUTLINE

- Introduction (TK)
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- Prosodic Aspect (NM)
- Speech Synthesis Tech. for CALL (NM)
- CALL Systems (TK)
- Database for CALL (NM)

# English CALL System: HUGO @Kyoto Univ. [Tsubota, Imoto, Raux 2002]

- For Japanese college students, so that they can introduce Japanese cultures
- Dedicated acoustic model & error prediction scheme for Japanese students
- Deployed and used in classrooms



HUGO Pronunciation Tutor (tsubota) - The Edo Period

The Edo Period, which dates from 1603 to 1867. During this period, Japan enjoyed freedom from warfare under the rule of the Tokugawa shogunate.

Record

What **period** will the next part represent?

A

B

The Edo Period, which dates from 1603 to 1867. During this period, Japan enjoyed freedom from **warfare** under the rule of the Tokugawa shogunate.

# English CALL System: HUGO @Kyoto Univ.

- Goal: Pinpointing the pronunciation errors which degrade intelligibility and providing effective feedback
- Practice consists of two phases
  1. Dialogue-based skit (for natural conversation)
  2. Training on specific errors detected in the first phase (using a phrase or a word)
- Pronunciation error detection
  - Segmental pronunciation ← hand-crafted phonological rules
  - Accent (Primary & Secondary Stress) ← multiple prosodic features



Record

What **period** will the next part represent?

A

The Edo Period, which dates from 1603 to 1867. During this period, Japan enjoyed freedom from **warfare** under the rule of the Tokugawa shogunate.

B

HUGO Pronunciation Tutor (tsubota) - The Edo Period

## /R-L/ substitution

Select an item

You


Model

Words

p	ih	l	y	ax	d
p	ih	r	y	ax	d
Period					

Evaluating segmentals

Performing phoneme recognition...



[Back to diagnosis](#)

## R and L

To pronounce an **L**, your **tongue** must touch the **back of your upper teeth**.

For **R**, concentrate on progressively **rounding your lips**.



## Function words

Select an item

You  
Model  
Words

•      •	•	•      •
During	this	period,

Please record the given item

## Function words

Words such as prepositions (*after, under...*), pronouns (*he, them...*) or articles (*the, her...*) are called **function words**. They don't carry any essential meaning but have a *grammatical* role. Such words are usually **not stressed** in English. Concentrate on shortening their vowels and keeping intonation and power as flat as possible on these words.

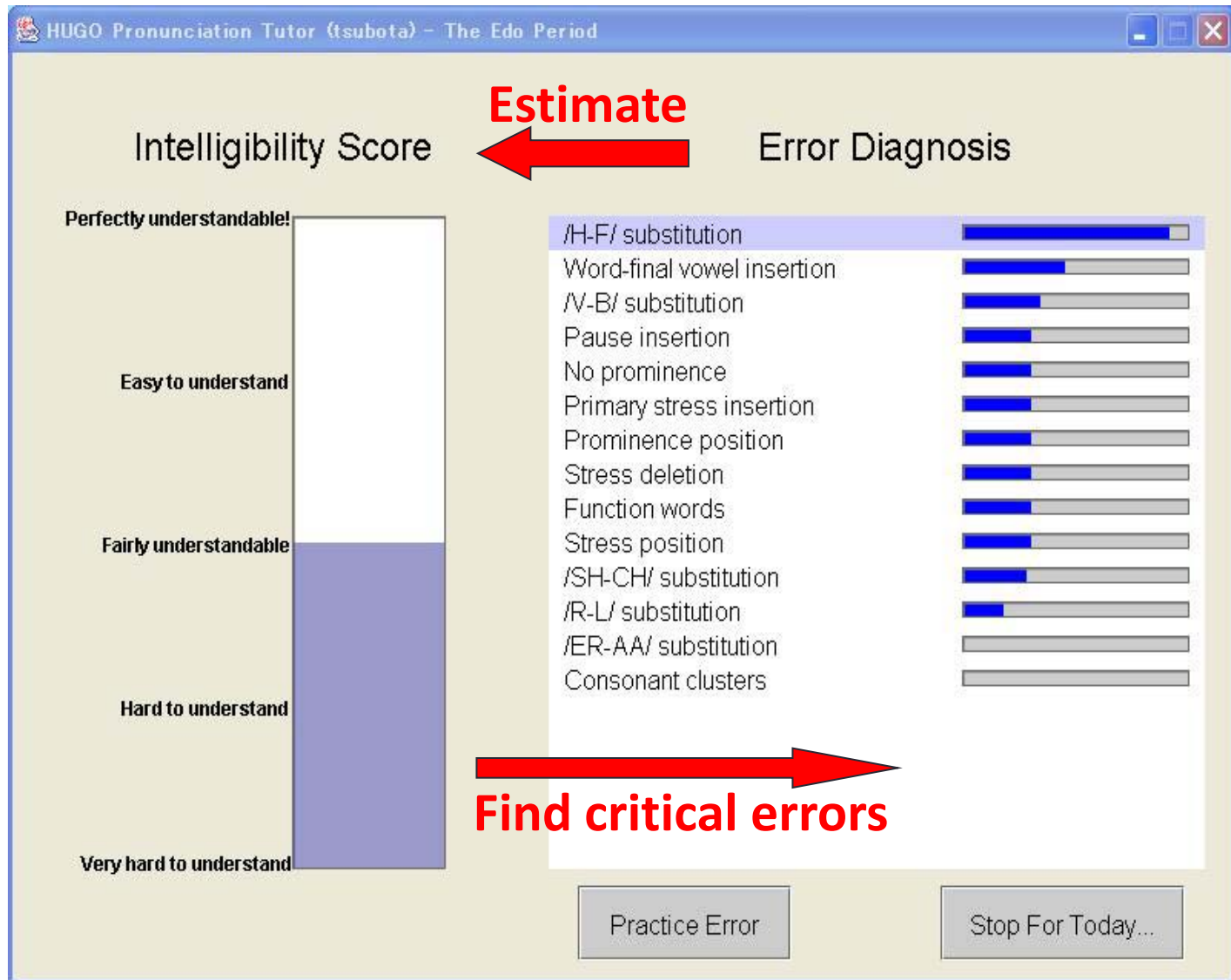


## List of Pronunciation Errors

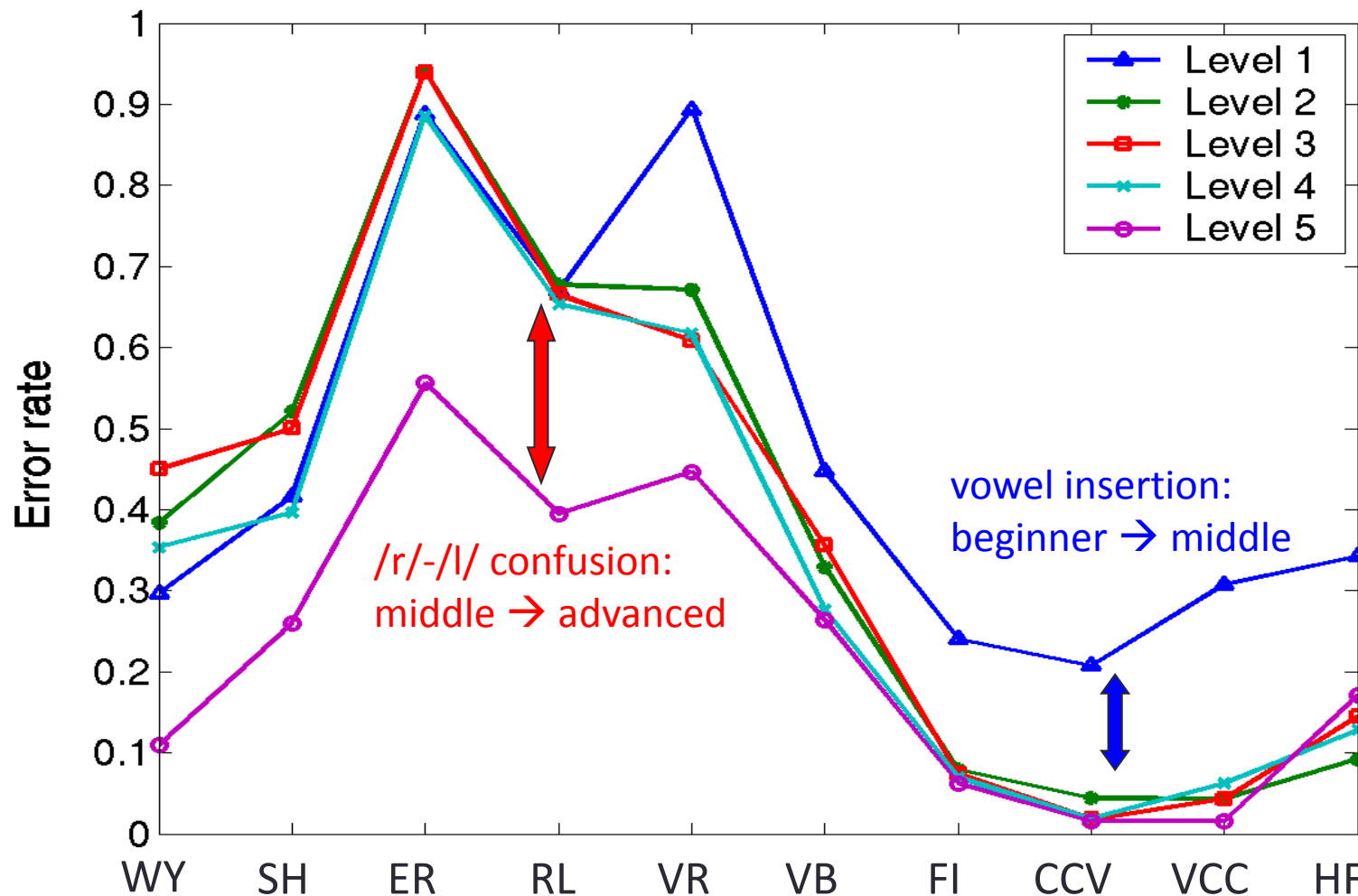
W/Y deletion ( <u>w</u> ould)	V/B substitution (pro <u>b</u> lem)
SH/CH substitution ( <u>ch</u> oose)	Final vowel insertion (le <u>t</u> )
R/L substitution ( <u>r</u> oad)	CCV-cluster insertion ( <u>a</u> ctive)
ER/A substitution (pap <u>e</u> r)	VCC-cluster insertion ( <u>s</u> tudy)
Non-reduction (stud <u>e</u> nt)	H/F substitution ( <u>f</u> ire)

- Built from literature in ESL
- Remove error patterns with low detection rate

# Intelligibility Assessment based on Error Statistics

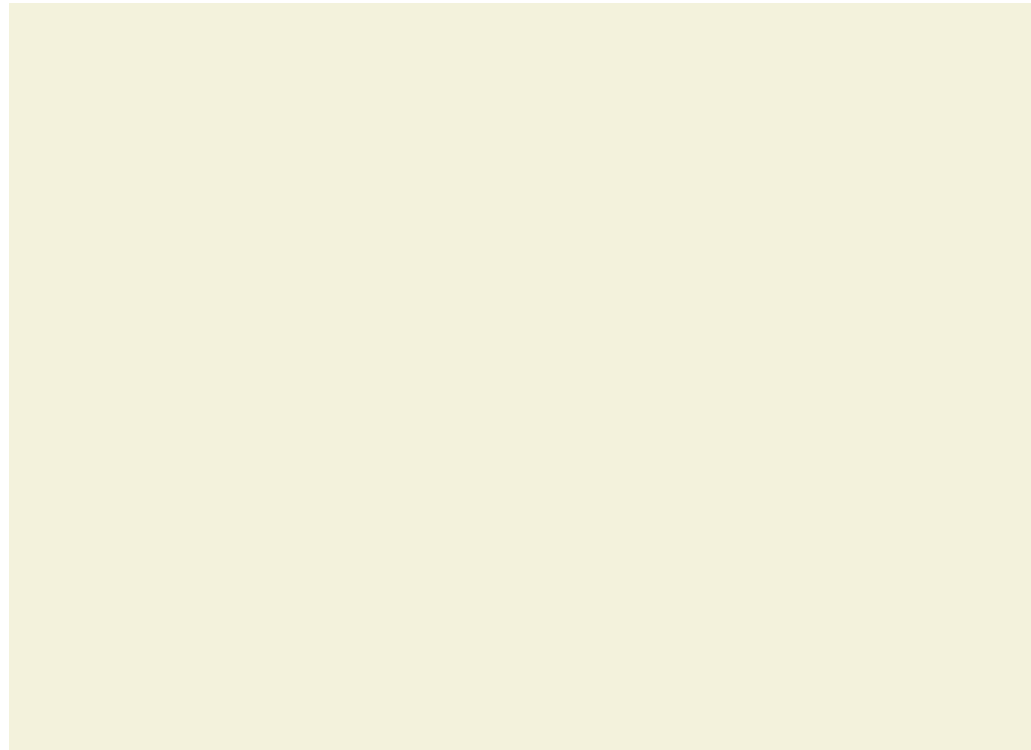


# Priority of Training on Specific Errors according to Intelligibility Level



# NativeAccent [Eskenazi 2007]

- Product of Fluency Project of CMU
- English learning
  - Error detection and feedback on articulation
  - Up to 28 L1: Japanese, Russian, French...
  - 800 exercises



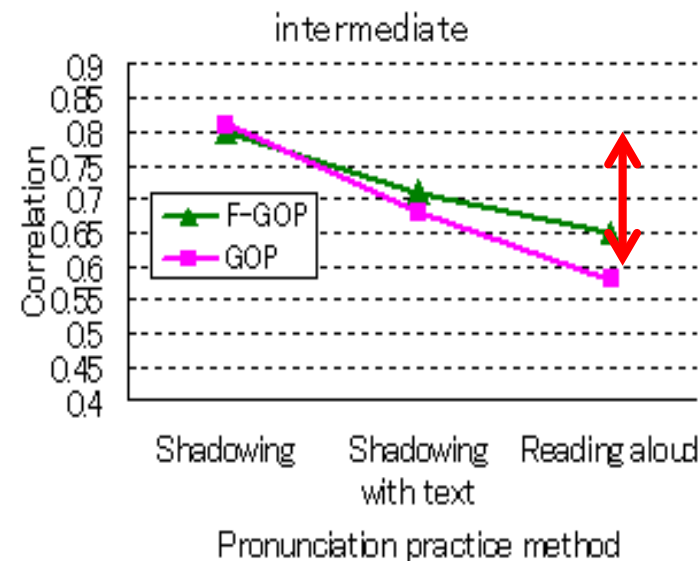
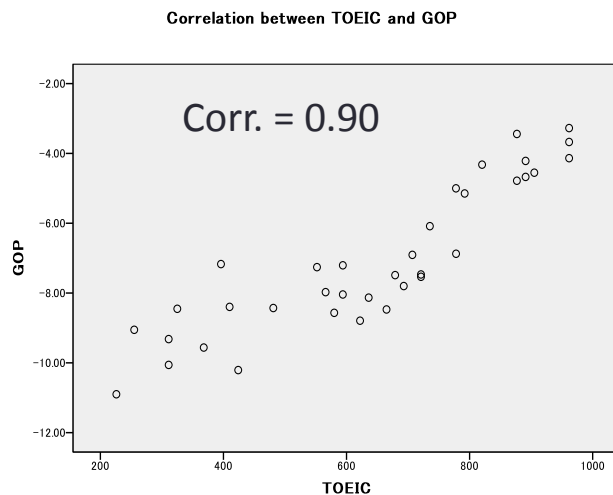
# English CALL System @ CUHK

## [Meng 2010]

- For Chinese learners of English
- Corpus: 100 Cantonese and 111 Mandarin L1
  - Reading a paragraph, words
- Pronunciation error model
  - Hand-crafted phonological rules
  - Data-driven patterns
- GOP score
- Pre-filtering based on duration models
- Synthesizing expressive speech to convey emphasis in feedback generation
- Synthesizing visual speech with articulator animation

# Shadowing Exercise [Luo 2009]

- listening and repetition of native utterances, online
  - Simultaneous training of listening and speaking skills
- High correlation between GOP and TOEIC scores (= 0.90)
  - Higher than simple reading



# ETS SpeechRater for TOEFL

## [Zechner 2007]

- Assessment of unconstrained English speech
  - TOEFL iBT Practice Online (TPO)
  - iBT Field Study
- Acoustic model: non-native speech (30hours)
- Language model: non-native speech + broadcast news
- Features: ASR results (word ID, confidence), speech rate, pause length... 40 in total
- Scoring: linear regression model
- Correlation with human rater: 0.67
  - Inter-human correlation 0.94

# Dialog-Based English CALL @POSTECH

## [Lee 2010]

- Situated dialog...(ex.) shopping
- ASR+SLU
- Example-Based Dialog Management
  - very limited domain
- Corrective feedback based on example selection
  
- Field trial on elementary school



# ETS SpeechRater for TOEFL

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# Speech database distribution sites

- Useful information source for speech databases
  - Linguistic Data Consortium (LDC, US)
    - <http://www ldc upenn edu/>
  - European Language Resources Association (ELRA, EU)
    - <http://www elra info/>
  - Speech Resource Consortium (SRC, Japan)
    - <http://research.nii.ac.jp/src/>, <http://research.nii.ac.jp/src/eng/index.html>
  - Advanced LAnGuage INformation forum (ALAGIN, Japan)
    - <http://www.alagin.jp/>, <http://www.alagin.jp/index-e.html>
  - GSK (Gengo-Shigen-Kyokai = Language Resource Association, Japan)
    - <http://www.gsk.or.jp/index.html>, [http://www.gsk.or.jp/index\\_e.html](http://www.gsk.or.jp/index_e.html)
  - Chinese Linguistic Data Consortium (C-LDC, China)
    - <http://www.chineseldc.org/>
- These sites distribute speech & language databases for general purposes.
- Only a part of the databases include non-native speech samples.

# Non-native speech data collection

- More useful information source for non-native speech data
  - “Non-native speech database” in Wikipedia
    - [http://en.wikipedia.org/wiki/Non-native\\_speech\\_database](http://en.wikipedia.org/wiki/Non-native_speech_database)
    - Based on [M. Raab+'07]
    - 42+ non-native databases are briefly described.

Corpus	Author	Available at	Language(s)	#Speakers	native Language	#Utt.	Duration	Date	Specials	Reference
AMI		EU	E		Dut and other		100h		meeting recordings	#40
ATR-Gruhn	Gruhn	ATR	E	96	C G F J Ind	15000		2004	proficiency rating	#4
BAS Strange Corpus I+II		ELRA	G	139	50 countries	7500		1998		#5
Berkeley Restaurant		ICSI	E	55	G I H C F S J	2500		1994		#41
Broadcast News		LDC	E					1997		#6
Cambridge-Witt	Witt	U. Cambridge	E	10	J I K S	1200		1999		#7
Cambridge-Ye	Ye	U. Cambridge	E	20	C	1600		2005		#8
Children News	Tomokiyo	CMU	E	62	J C	7500		2000	partly spontaneous	#6
ERJ	Minematsu	U. Tokyo	E	200	J	68000		2002	proficiency rating	#13

# Development of ERJ database

- ERJ = English Read by Japanese [Minematsu+'04]
  - Development of a database containing many pronunciation errors that are observed commonly in the English spoken by Japanese
  - A main focus is put on the errors that are made rather unconsciously.
  - Spontaneous speech is technically challenging. So read speech is focused on.
  - Target language = General American English (GAE)
- Selection of reading material
  - Word and sentence sets considering the segmental aspects of GAE
  - Word and sentence sets considering the prosodic aspects of GAE
  - In total, 807 sentences and 1009 words are prepared.

Table 1: Word and sentence sets for the segmental aspect

set	size
Phonemically-balanced words	300
Minimal pair words	600
TIMIT-based phonemically-balanced sentences	460
Sentences including phoneme sequences difficult for Japanese to pronounce correctly	32
Sentences designed for test set	100

Table 2: Word and sentence sets for the prosodic aspect

set	size
Words with various lexical accent patterns	109
Sentences with various intonation patterns	94
Sentences with various rhythm patterns	121

# Development of ERJ database

- Preparation of reading sheets
  - Many pronunciation guides are on the sheets
    - Phonemic symbols
    - Stress marks
    - Intonation curves
    - etc.

S1\_0097

She knows you, doesn't she ?

[SH IY1] [N OW1 Z] [Y UW1] [D AH1 Z AX0 N T] [SH IY1]

S1\_0105 Come to tea.

/ + - @ /

[K AH1 M] [T UW1] [T IY1]

S1\_0106 Come to tea with John.

/ + - + - @ /

[K AH1 M] [T UW1] [T IY1] [W IH1 DH] [JH AA1 N]

S1\_0107 Come to tea with John and Mary.

/ + - @ / - + - @ - /

[K AH1 M] [T UW1] [T IY1] [W IH1 DH] [JH AA1 N] [AE1 N D] [M EH1 R IY0]

# Development of ERJ database

- Selection of speakers
  - Quasi-random selection of university/college students of Japanese
  - 100 male and 100 female Japanese
  - 20 General American English (GAE) speakers
- Recording protocol
  - About 120 sentences and 220 words are assigned to each student.
    - About 400 sentences are assigned to each of the 20 American speakers.
  - Pronunciation guides are shown in the reading sheet.
  - The speakers read the material repeatedly until they “thought” that they read the material correctly.
    - *Error-free* utterances judged by the speakers themselves.
    - Still, many errors can be detected by teachers.

# Development of ERJ database

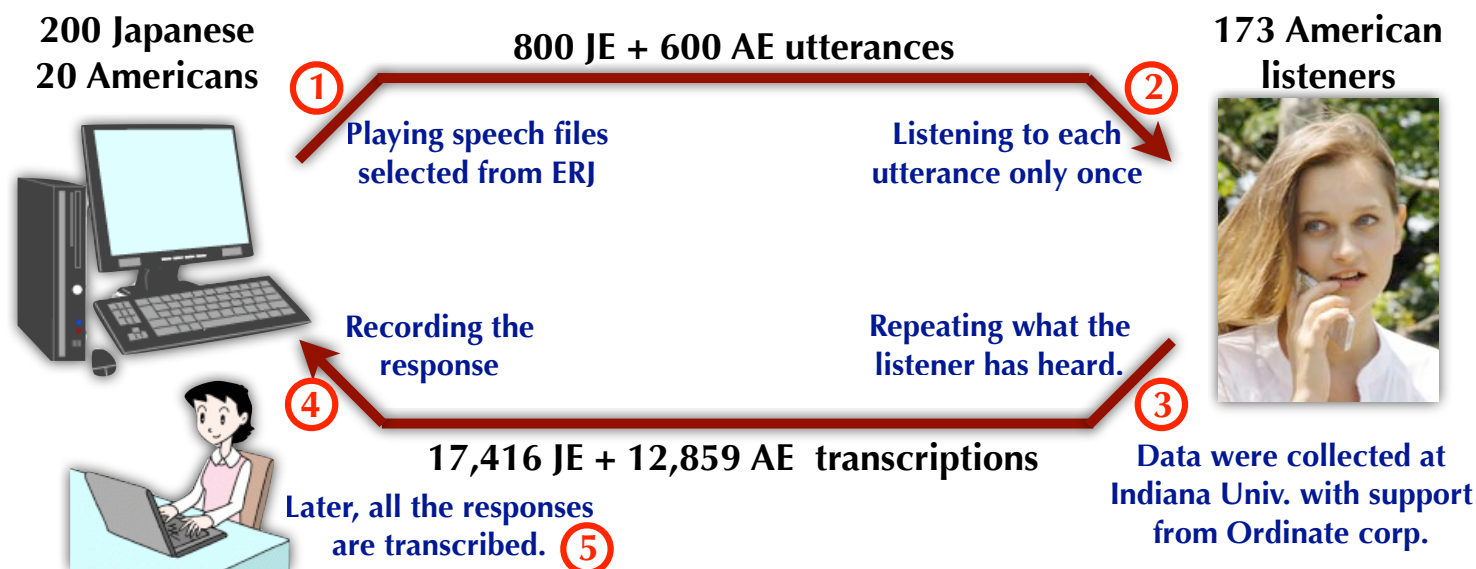
- Rating protocol
  - Five American teachers of English are asked to rate some utterances of the individual students w.r.t the three aspects of pronunciation.
    - Phonemic aspect / intonational aspect / rhythmic aspect
    - As for prosodic rating, model utterances were presented to the teachers because they claimed that the task was difficult without prosodically *perfect* utterances.
- Use of the database
  - Development of CALL systems and their modules
  - Acoustic analysis of Japanese English





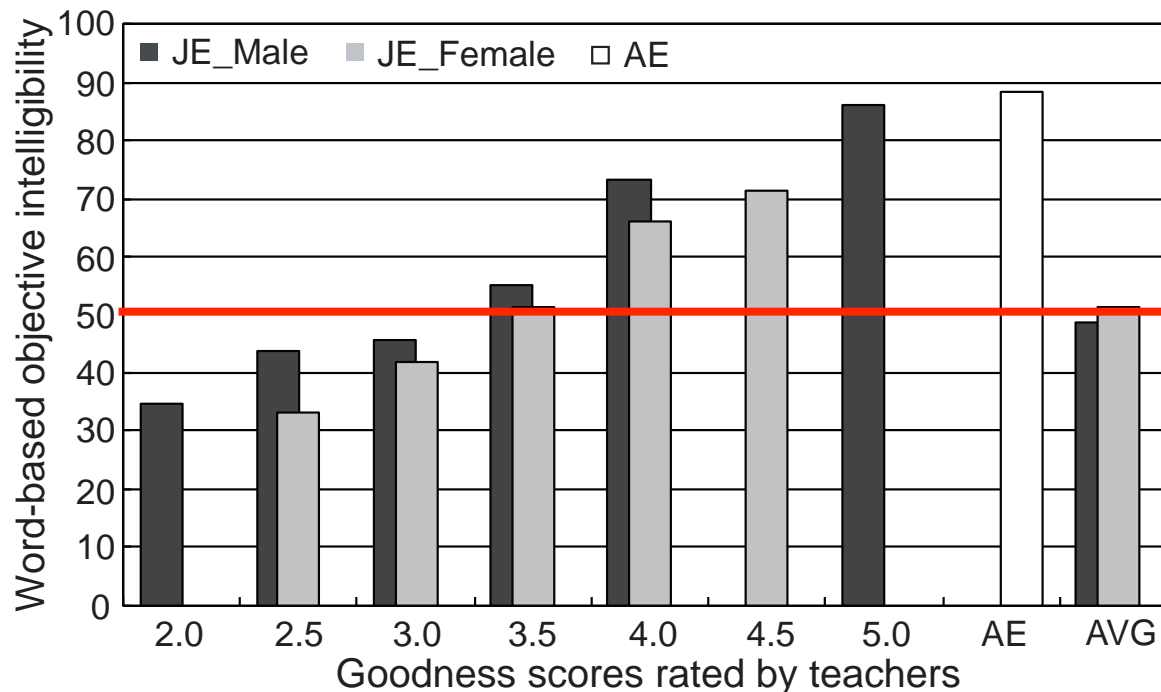
# Objective measurement of intelligibility

- How intelligible is JE? [Minematsu+'11]
  - ERJ = many read utterances judged as *error-free* by the students
    - Are these utterances understood correctly by US people?
  - A huge listening test was done using a subset of ERJ database.
    - Listeners : American with little exposure to Japanese English.
      - JE utterances are presented through a telephone line.
    - Task : just repeating what they have heard without trying to guess.
      - Presentation of each utterance was done only once.
      - Repetitive responses were transcribed by expert transcribers.



# Objective measurement of intelligibility

- How intelligible is JE? [Minematsu+'11]
  - ERJ = many read utterances judged as *error-free* by the students
    - Are these utterances understood correctly by US people?



Classification of speakers based on their proficiency scores

score	≤2.0	≤2.5	≤3.0	≤3.5	≤4.0	≤4.5	≤5.0
male	2	27	43	16	5	0	2
female	0	8	36	25	19	7	0



# Non-native speech data collection

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ERJ	Minematsu	U. Tokyo	E	200	J	68000		2002	proficiency rating	#13

- Data collection is a tough work.
  - Resource sharing is very important.

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# Japanese CALL system: CALLJ @Kyoto Univ. [Wang 2009]

- Exercise of basic sentence production (text & speech), given a image scene
- Key features
  - Dynamic generation of questions & ASR grammar network with error prediction
  - Interactive hints

The screenshot shows the CALLJ software interface for Lesson 1 - Practice. The interface is divided into several numbered sections:

- 1**: A scene with a person (S) and a flag (i) with the text "United Kingdom" and "is not...".
- 2**: Grammar Form options: 1) Objects, 2) Statement, 3) Positive, 4) Negative, 5) Also.
- 3**: Table Points: 1041 (100%).
- 4**: Session Record: Questions: 24, Answered: 0%, Score: 0%, Time: 0h 0m 12s.
- 5**: Answer table with columns: Subject, Particle, Description, Verb.
- 6**: Recommended Hint, Reveal All Hints, Reveal [Length].
- 7**: REC, Notes, Skip buttons.

H.Wang, C.J.Waple, and T.Kawahara.

Computer assisted language learning system based on dynamic question generation and error prediction for automatic speech recognition. *Speech Communication*, Vol.51, No.10, pp.995--1005, 2009.

# Japanese CALL system: CALLJ @Kyoto Univ.

## How to Try

- Windows only.
0. (Unzip CALLJ1.5.zip).
  1. Move to the directory **CALLJ**.
  2. Click “**StartCALLJ**”.
  3. Create your account by clicking “**New**” in login window for the first time.
- You need some knowledge on Japanese.