Flexible speech synthesis based on hidden Markov models

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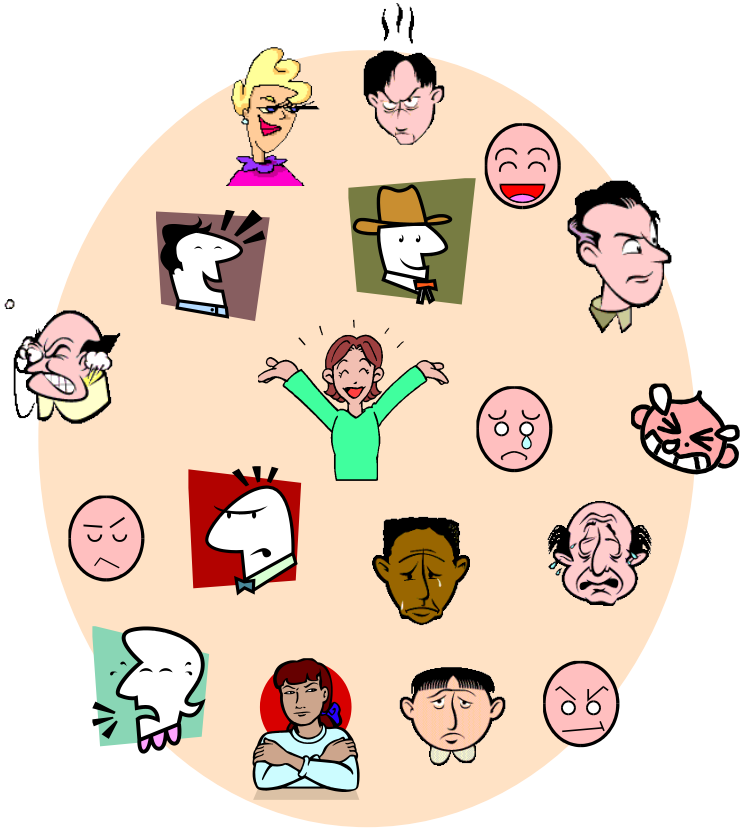
Towards human-like talking machines

For realizing natural human-computer interaction, speech synthesis systems are required to have an ability to generate speech with:

- arbitrary speaker's voice
- speaking styles
  (e.g., reading, conversational)
- emotional expressions
  (e.g., happy, angry, sad)
- emphasis
- and so on



History

Rule-based, *formant synthesis* (~'90s)

– Hand-crafting each phonetic units by rules

Corpus-based, *concatenative synthesis* ('90s~)

– Concatenate speech units (waveform) from a database

- Single inventory: diphone synthesis
- Multiple inventory: unit selection synthesis

Corpus-based, *statistical parametric synthesis*

– Source-filter model + statistical acoustic model

- Hidden Markov model: HMM-based synthesis

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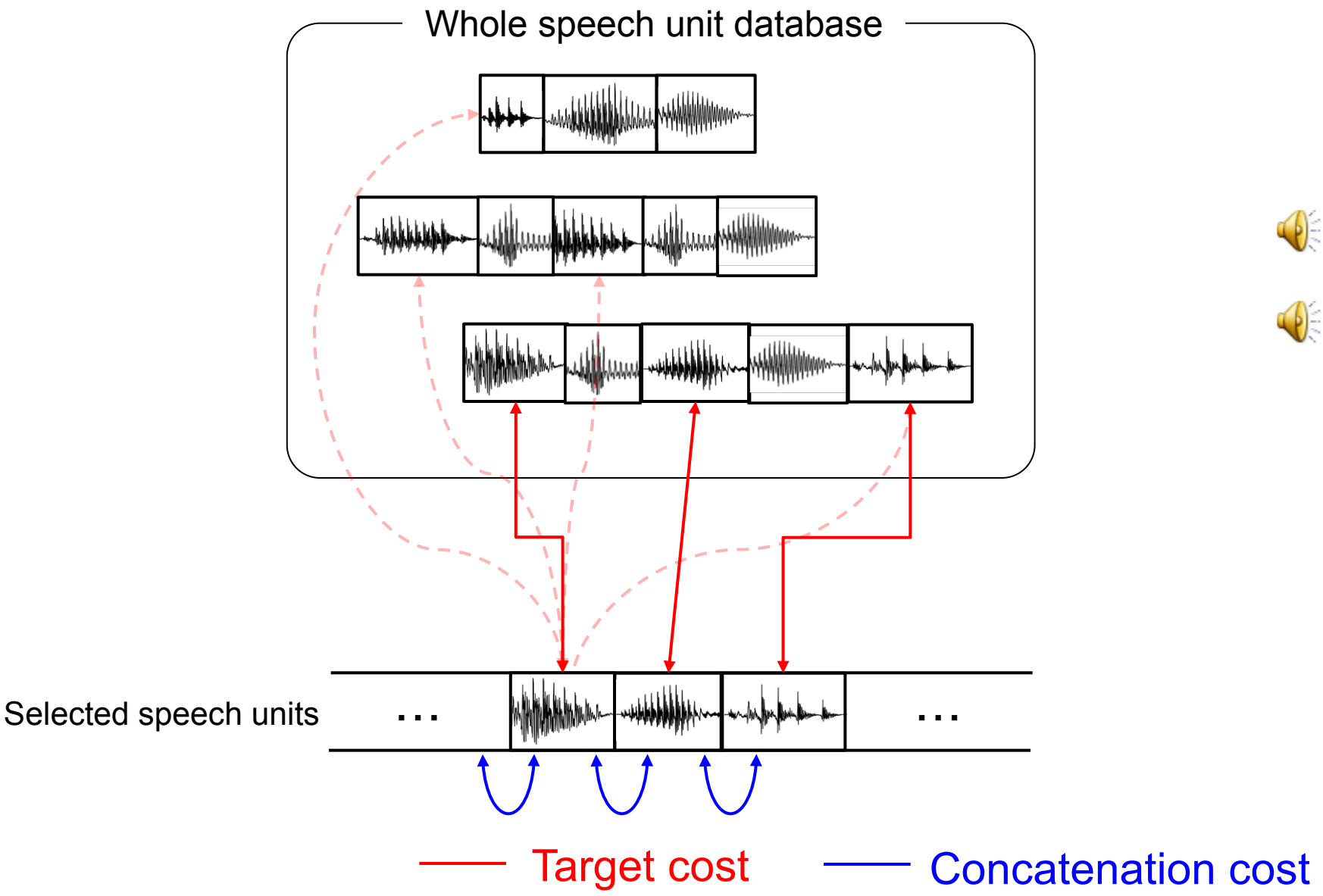
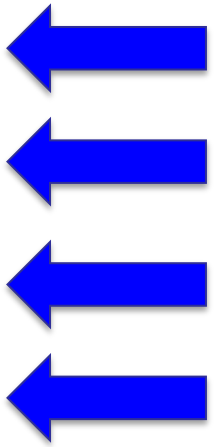
General unit-selection synthesis scheme

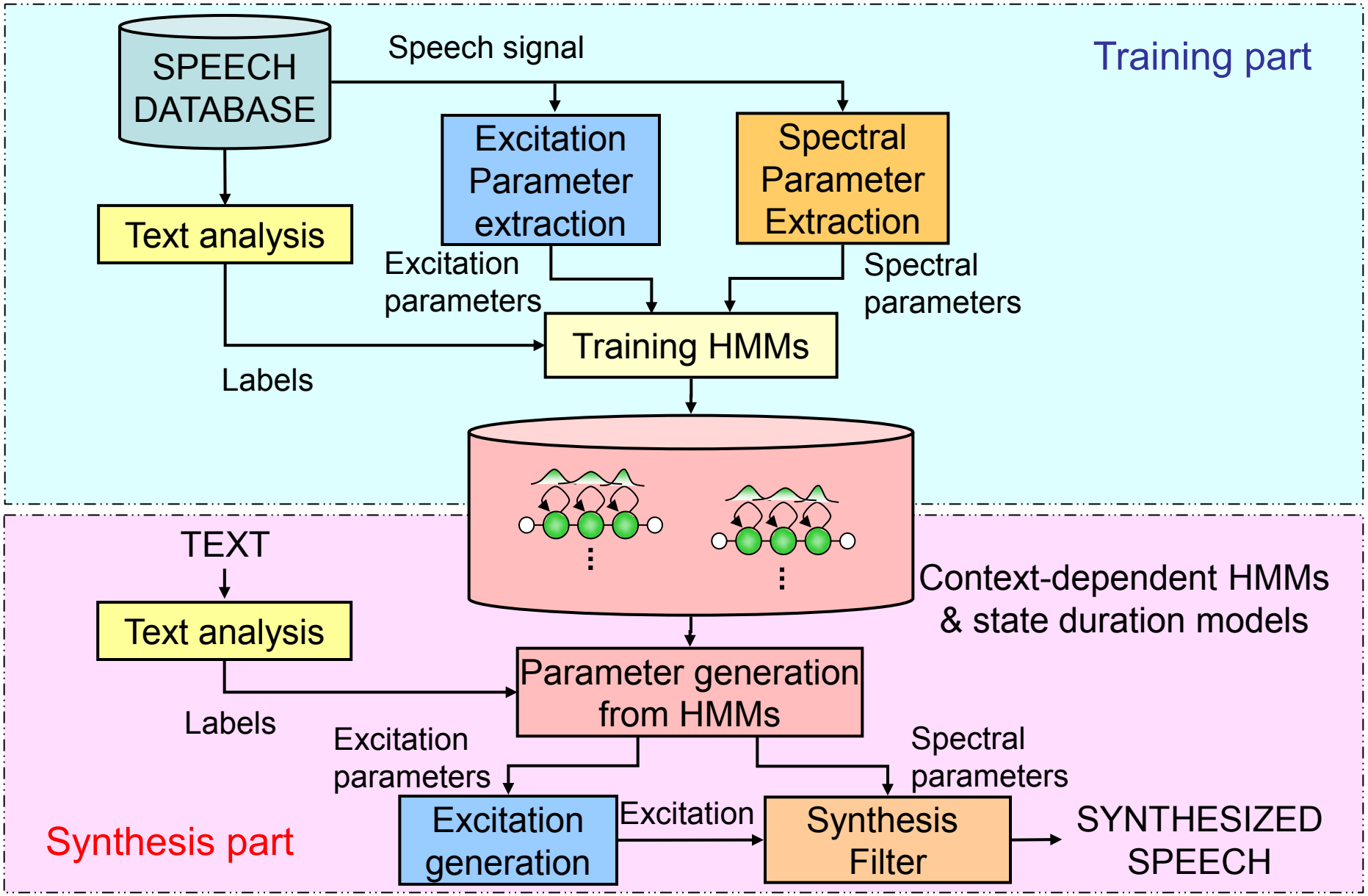


Overview of this talk

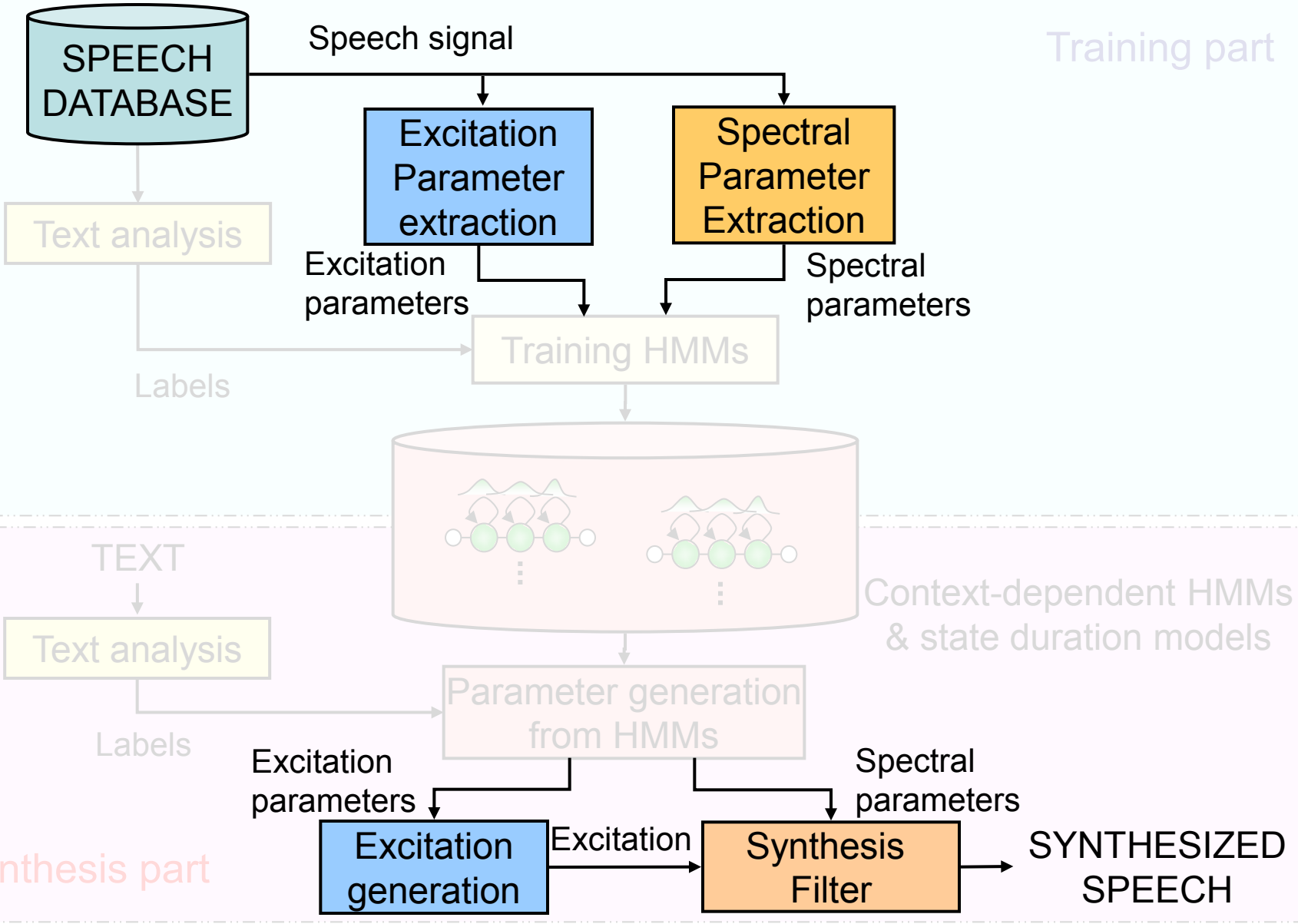
1. Introduction and background
2. Basic techniques in the system
3. Examples demonstrating its flexibility
4. Discussion and conclusion



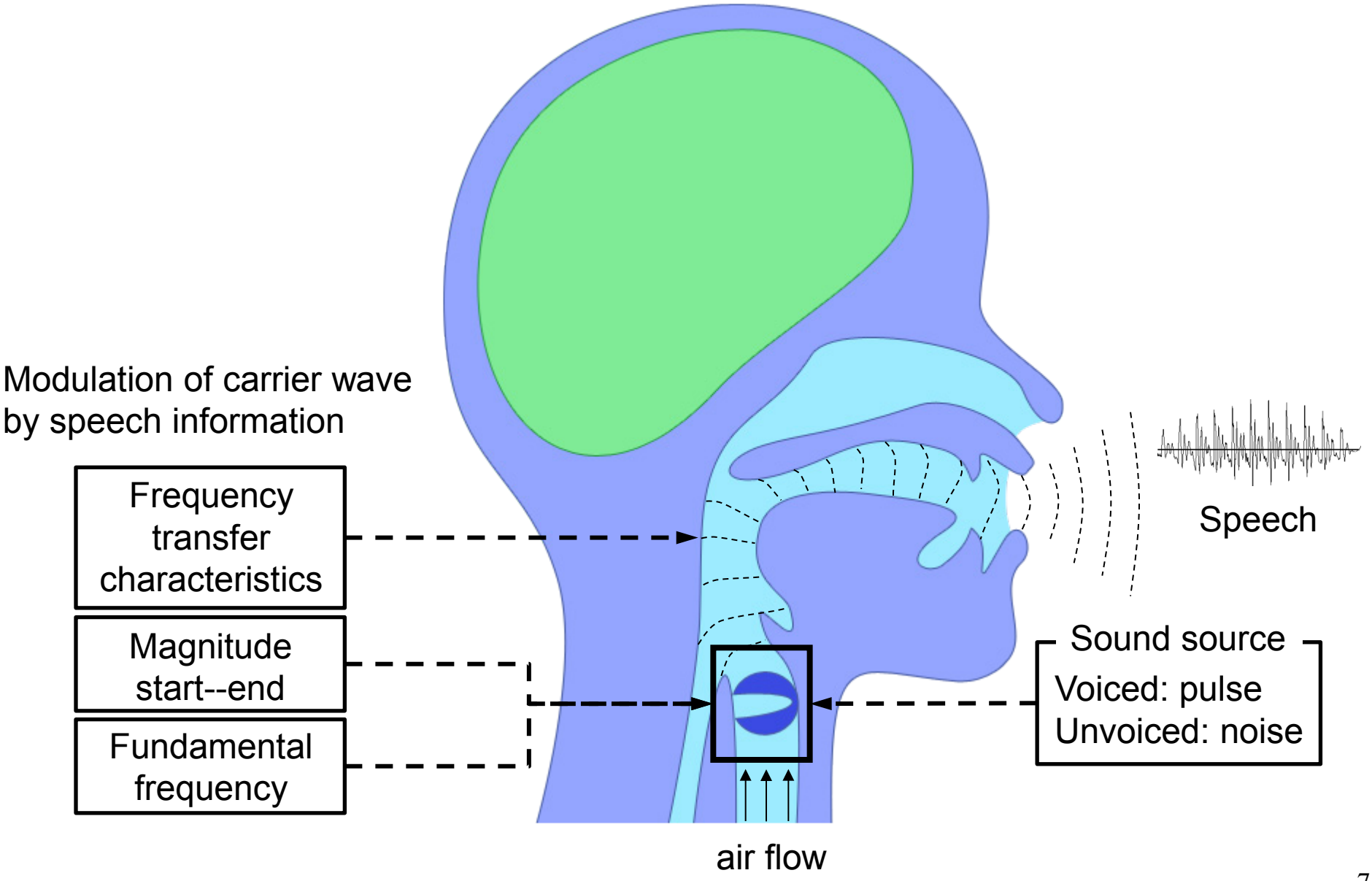
HMM-based speech synthesis system



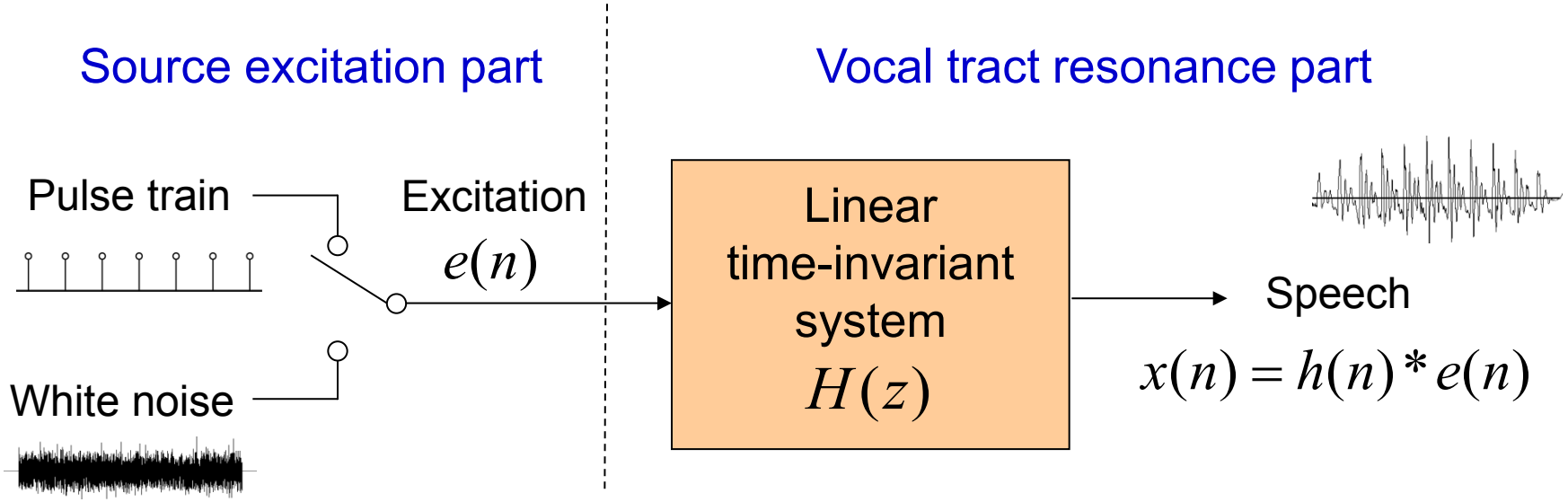
HMM-based speech synthesis system



Human speech production



Source-filter model

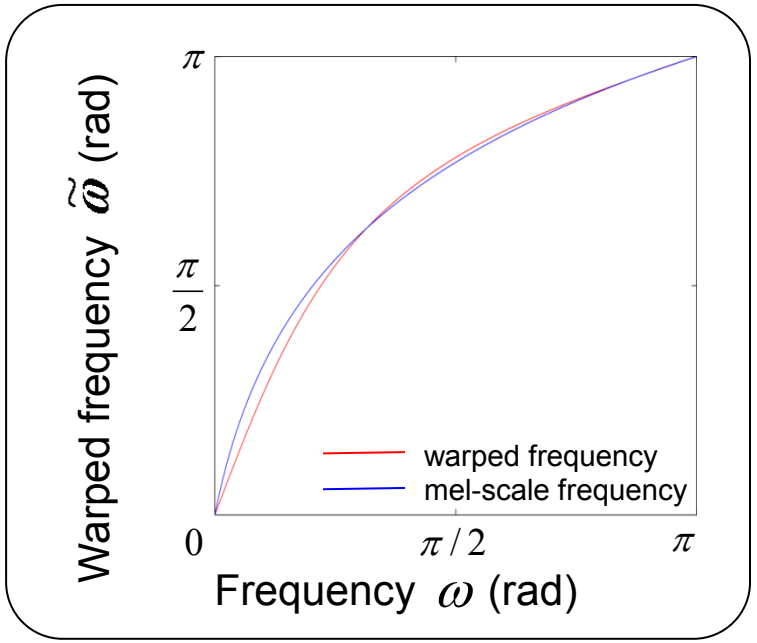


ML estimation of spectral parameter

Mel-cepstral representation of speech spectra

$$H(z) = \exp \sum_{m=0}^{M} c(m) \tilde{z}^{-m}$$

$$\tilde{z}^{-1} = \frac{z^{-1} - \alpha}{1 - \alpha z^{-1}} = e^{-j\tilde{\omega}}$$



ML-estimation of mel-cepstrum

$$\boldsymbol{c} = \arg\max_{\boldsymbol{c}} p(\boldsymbol{x} \mid \boldsymbol{c})$$

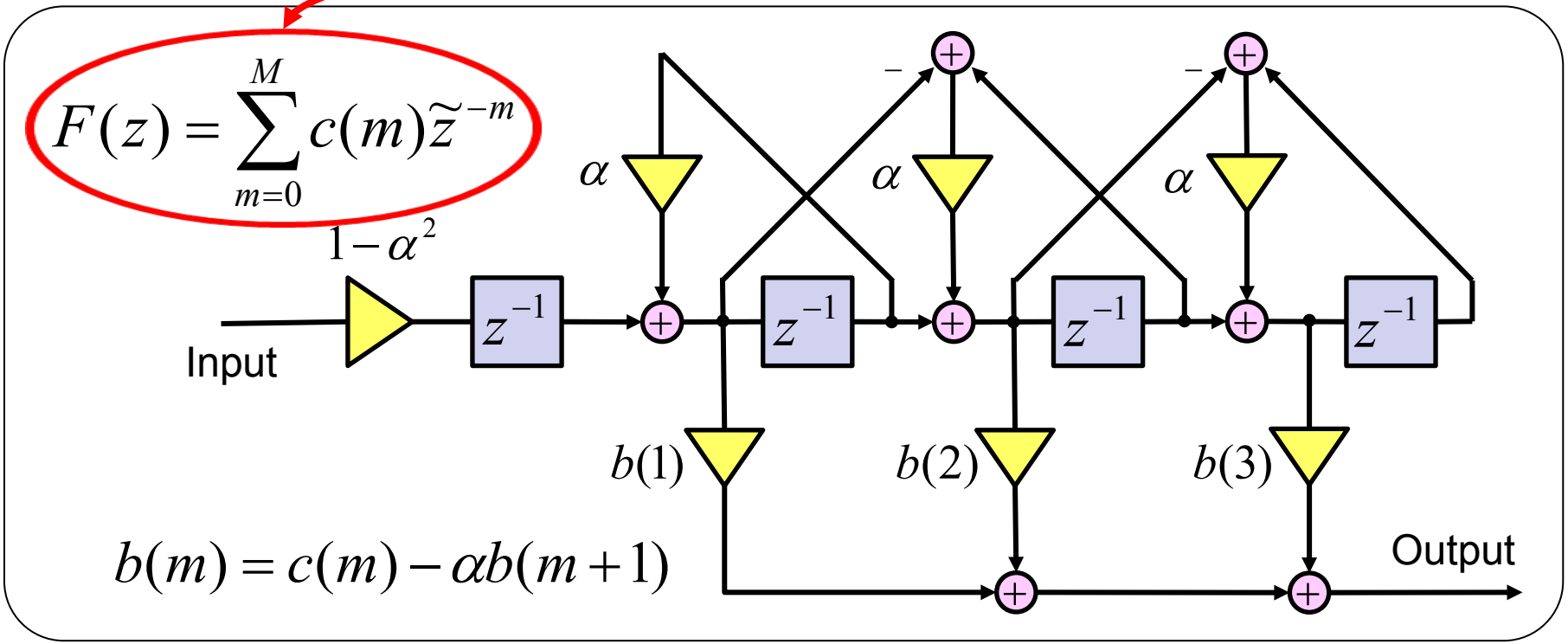
$\boldsymbol{x}$: short segment of speech waveform (Gaussian process)

$\boldsymbol{c}$: mel-cepstrum

Synthesis Filter

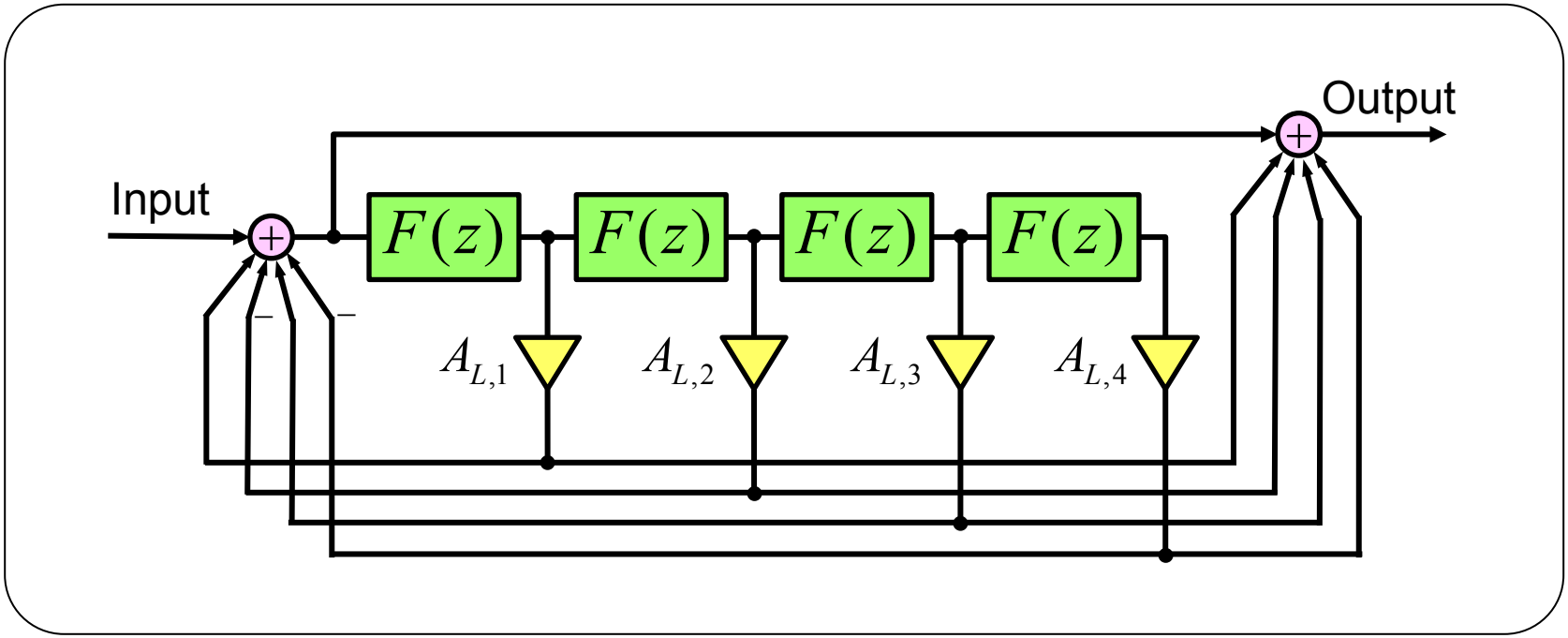
$$H(z) = \exp \sum_{m=0}^{M} c(m)\tilde{z}^{-m}$$

$$F(z) = \sum_{m=0}^{M} c(m)\tilde{z}^{-m}$$

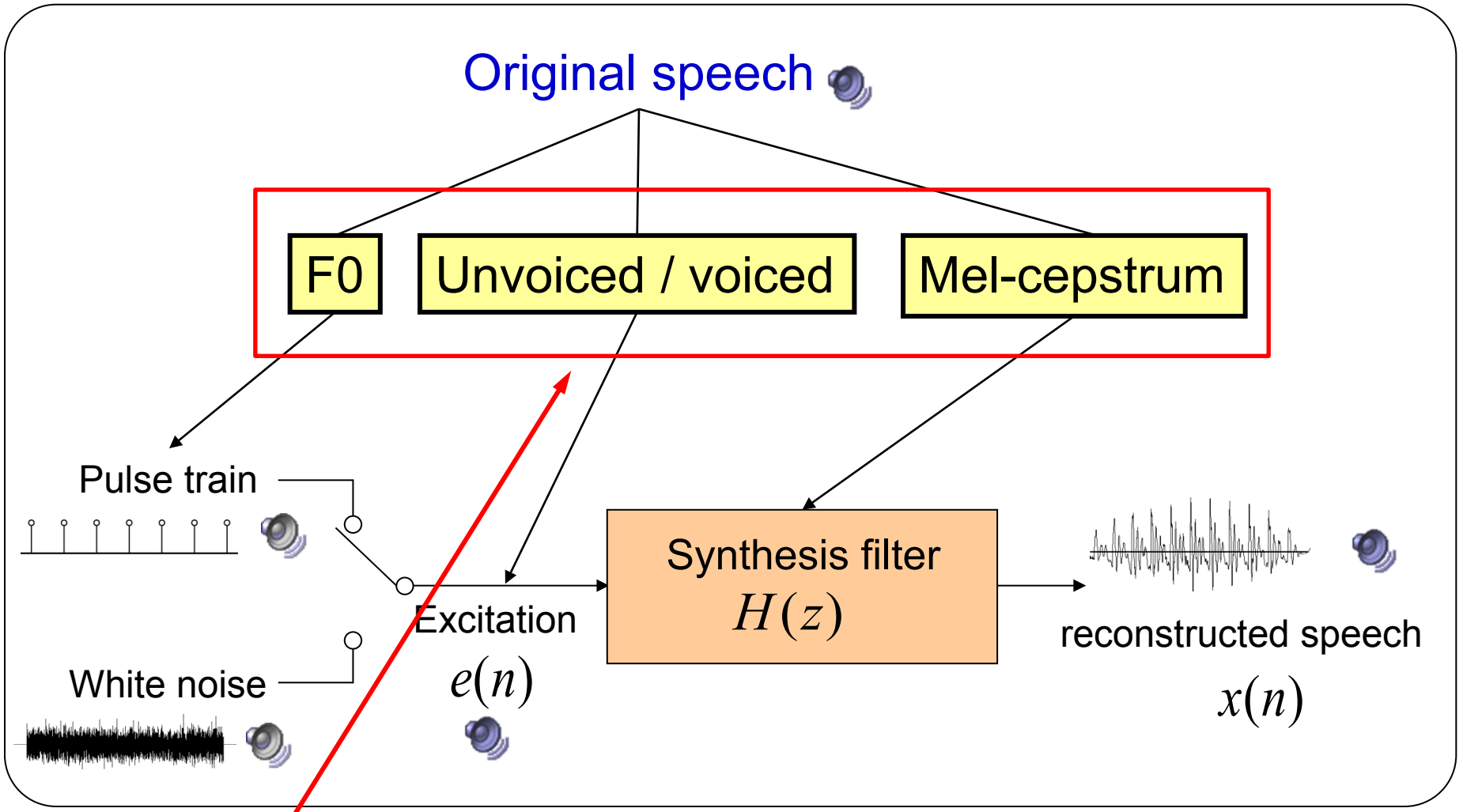


Structure of MLSA filter

$$H(z) = \exp F(z) \cong \frac{1 + \sum_{l=1}^{L} A_{L,l} \{F(z)\}^l}{1 + \sum_{l=1}^{L} A_{L,l} \{-F(z)\}^l}$$

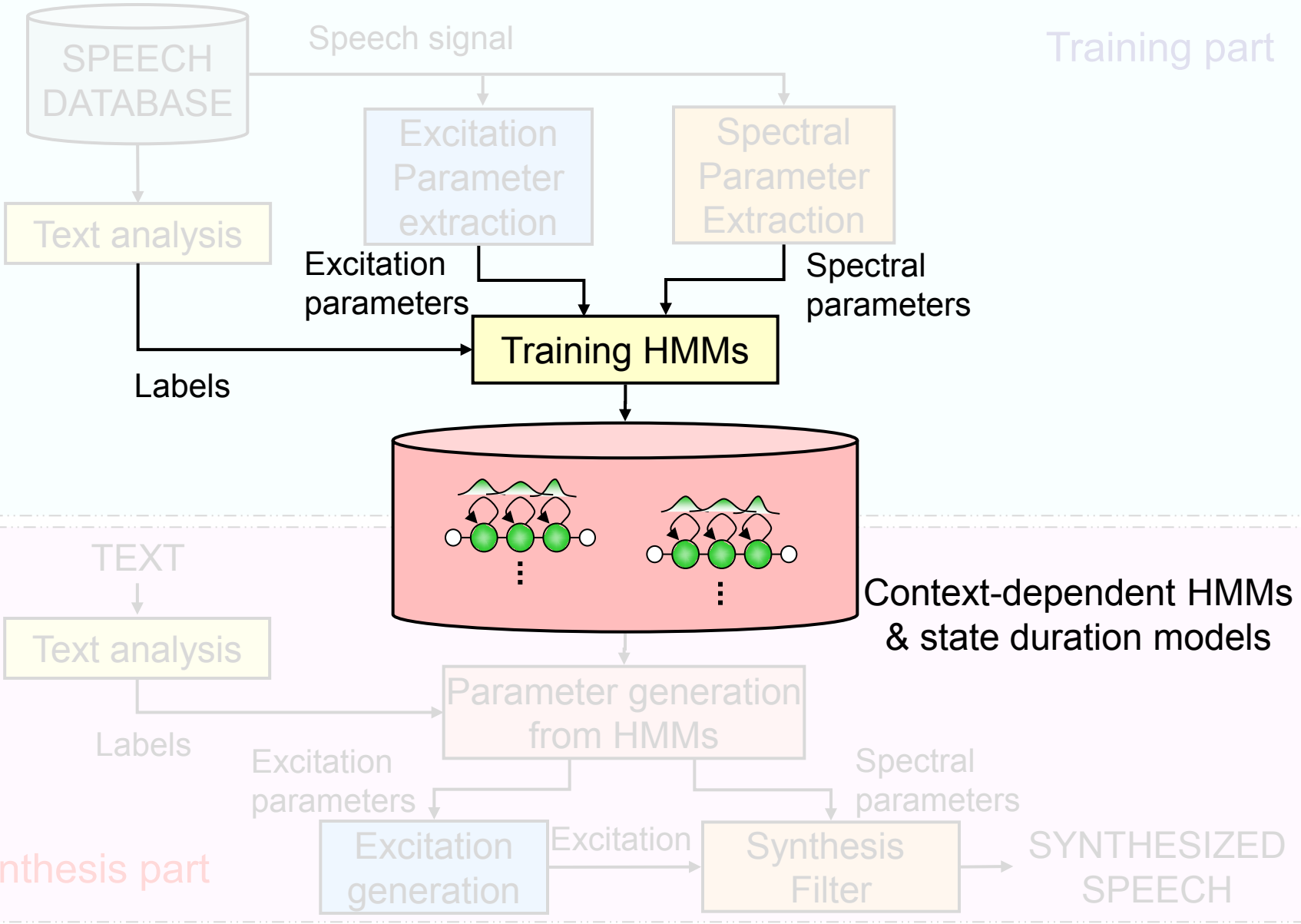


Waveform reconstruction



These speech parameters have to be estimated from the text to be synthesized → use of HMM

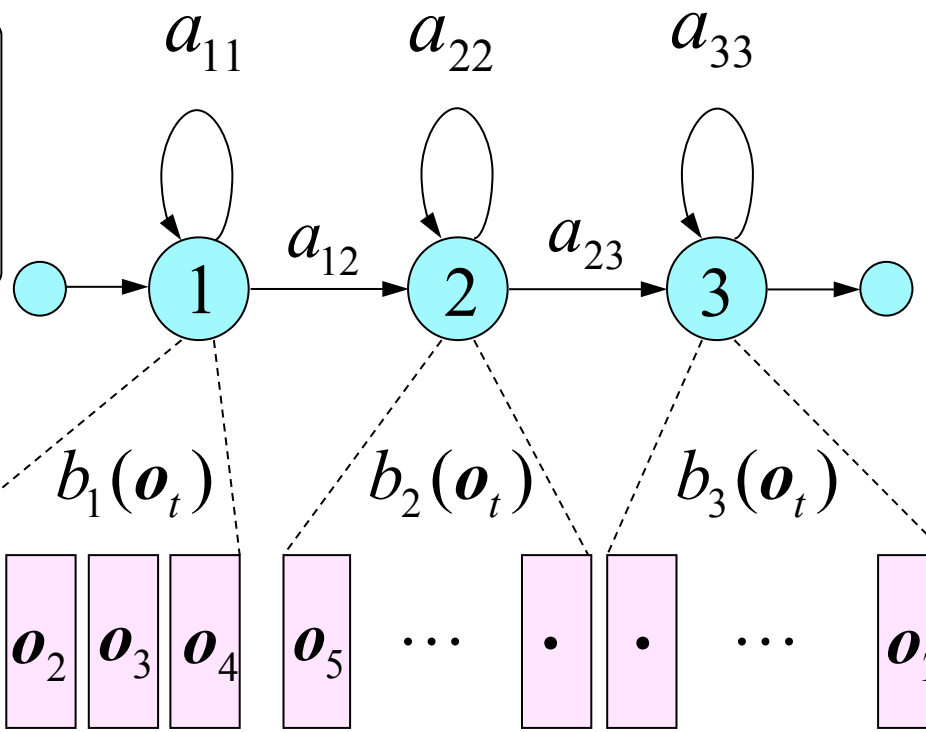
HMM-based speech synthesis system



Hidden Markov model (HMM)

a_{ij} : state transition probability

$b_q(\boldsymbol{o}_t)$: output probability

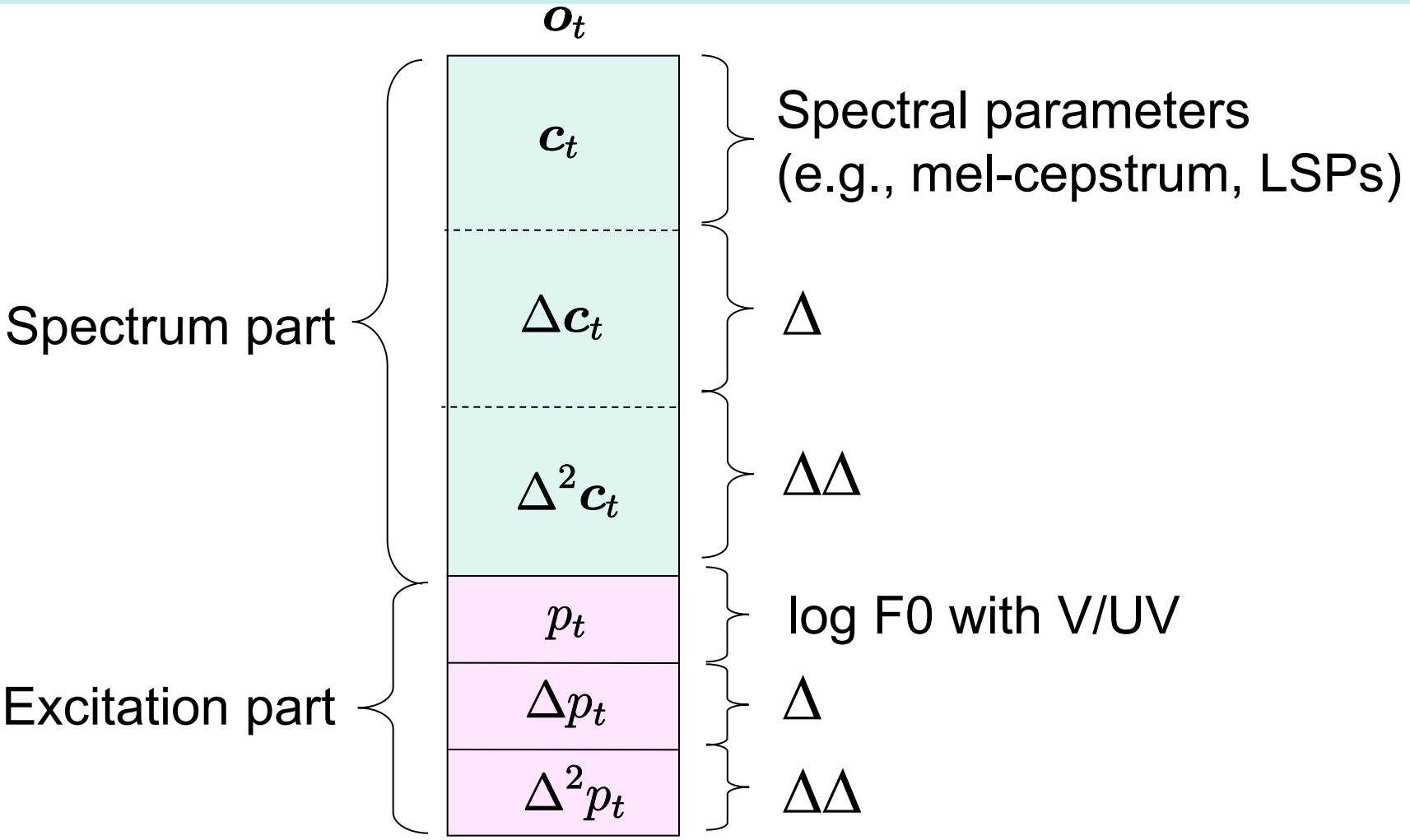


Observation sequence $\boldsymbol{o}$

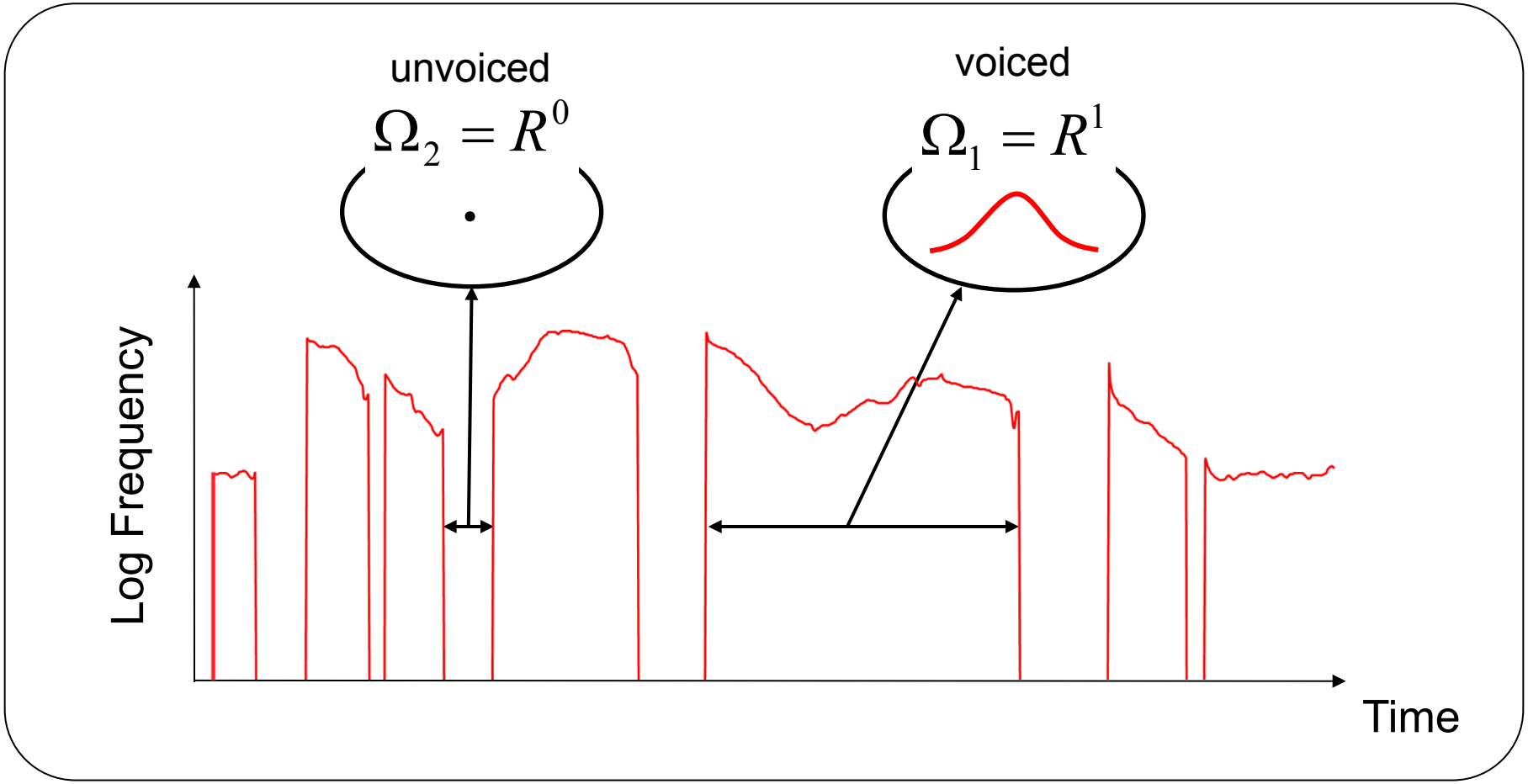
State sequence $\boldsymbol{q}$

$\boldsymbol{q}$: 1 1 1 1 2 … 2 3 … 3

Structure of state output (observation) vector



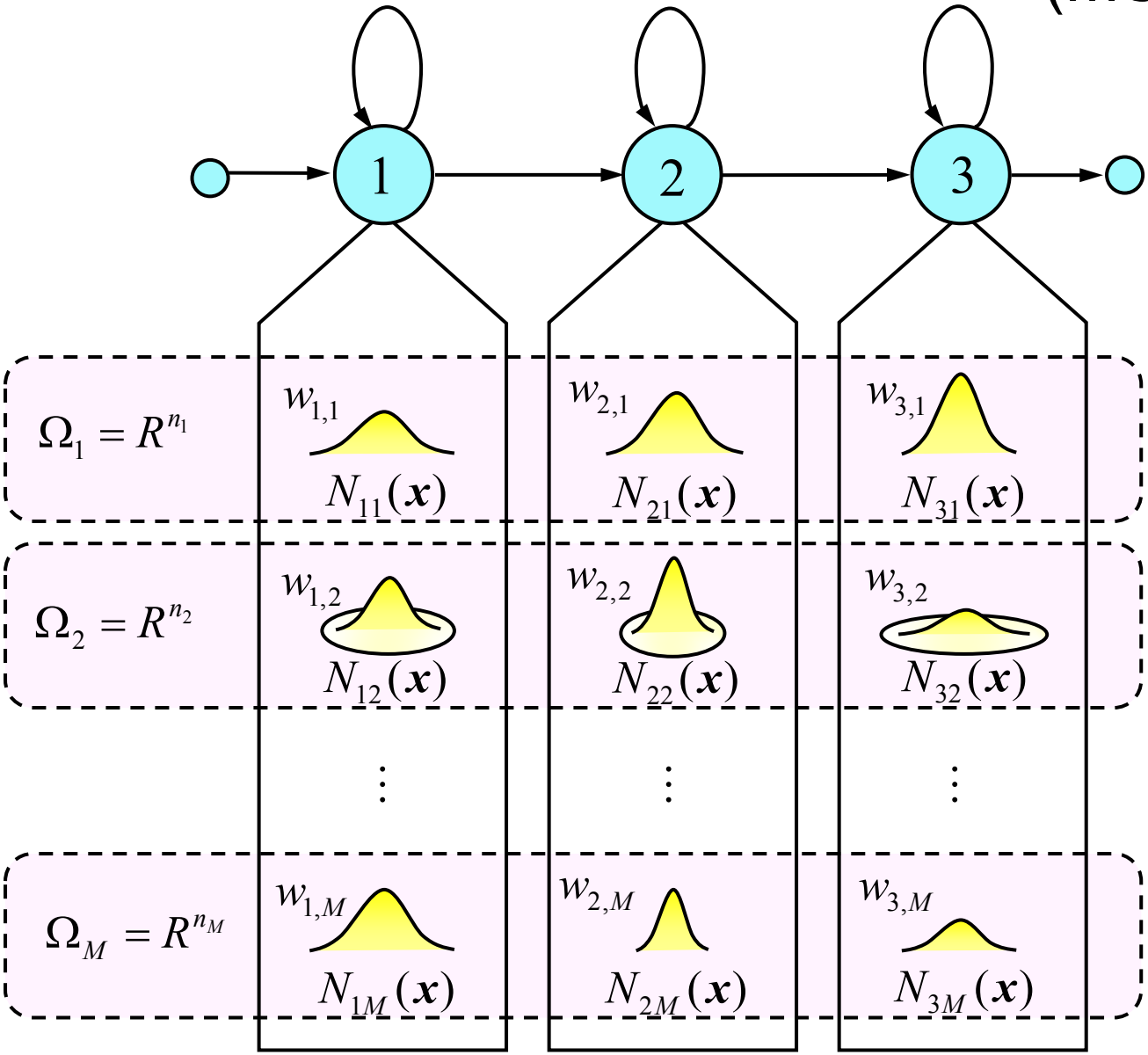
Observation of F0



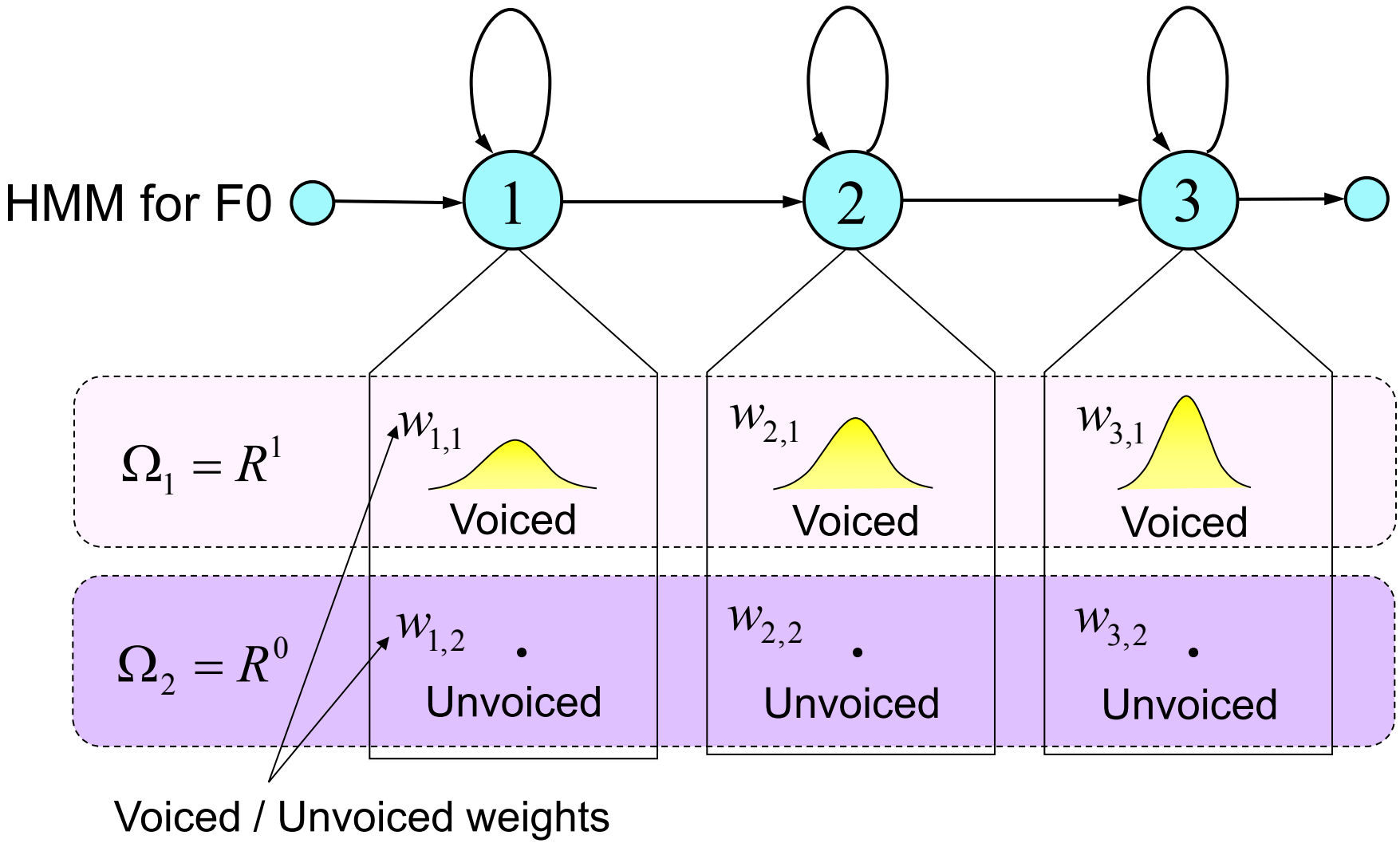
Unable to model by continuous or discrete distributions

⇒ Multi-space distribution HMM (MSD-HMM)

Multi-space probability distribution HMM (MSD-HMM)



MSD-HMM for F0 modeling



Contextual factors

Phoneme

- {preceding, succeeding} two phonemes
- current phoneme

Syllable

- # of phonemes in {preceding, current, succeeding} syllable
- {accent, stress} of {preceding, current, succeeding} syllable
- Position of current syllable in current word
- # of {preceding, succeeding} {accented, stressed} syllable in current phrase
- # of syllables {from previous, to next} {accented, stressed} syllable
- Vowel within current syllable

Word

- Part of speech of {preceding, current, succeeding} word
- # of syllables in {preceding, current, succeeding} word
- Position of current word in current phrase
- # of {preceding, succeeding} content words in current phrase
- # of words {from previous, to next} content word

Phrase

- # of syllables in {preceding, current, succeeding} phrase

.....

Huge # of combinations ⇒ Difficult to have all possible models

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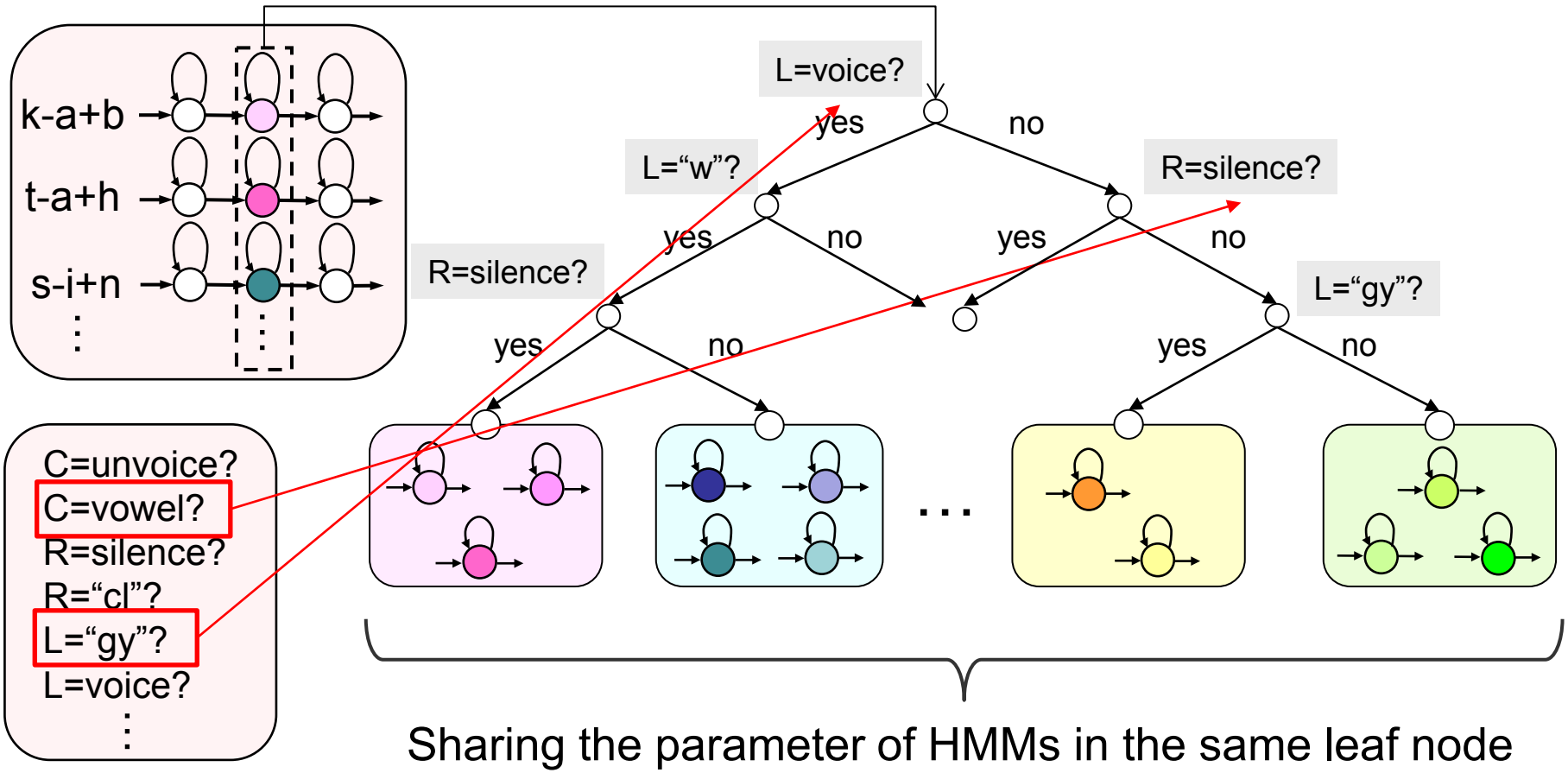
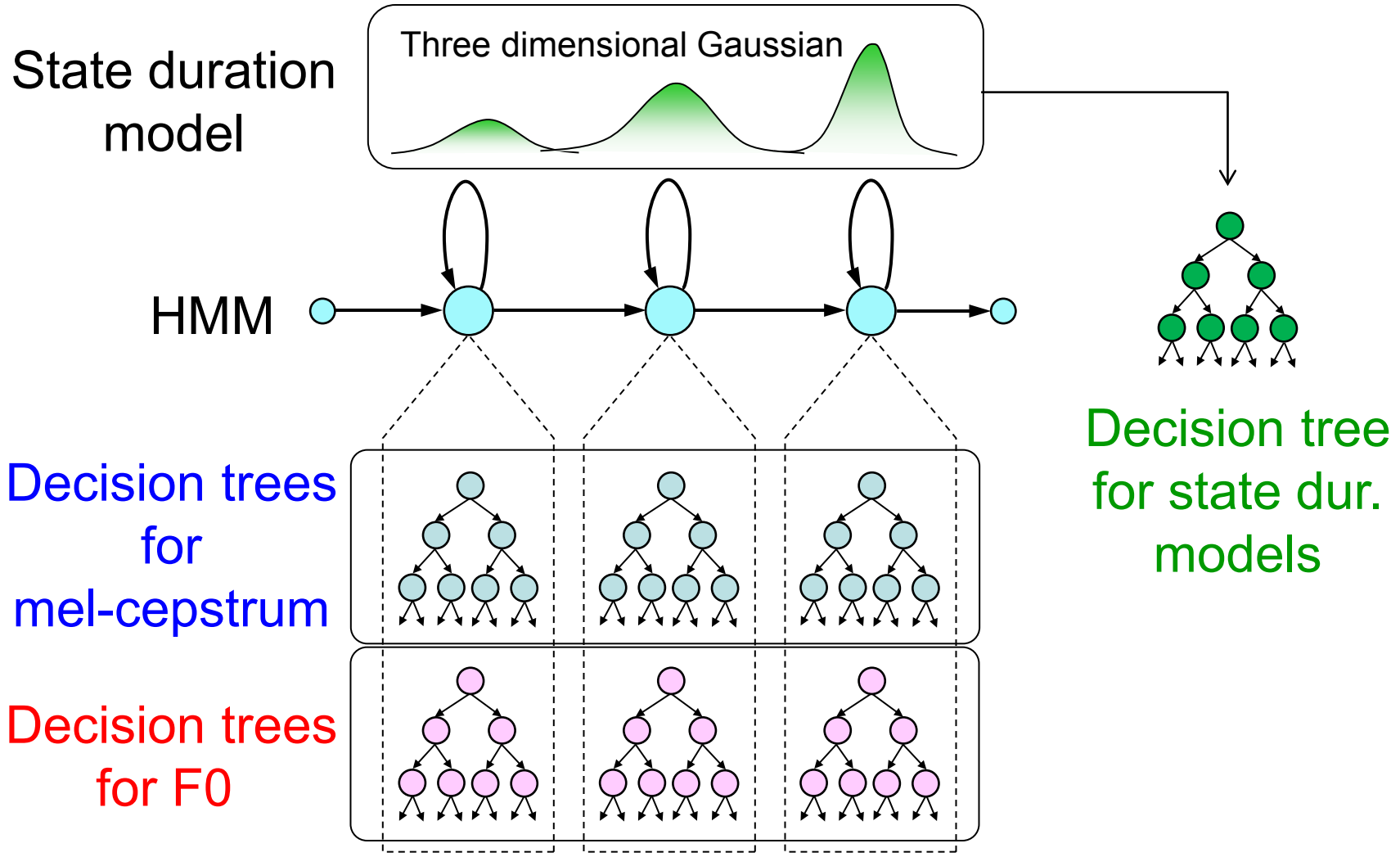
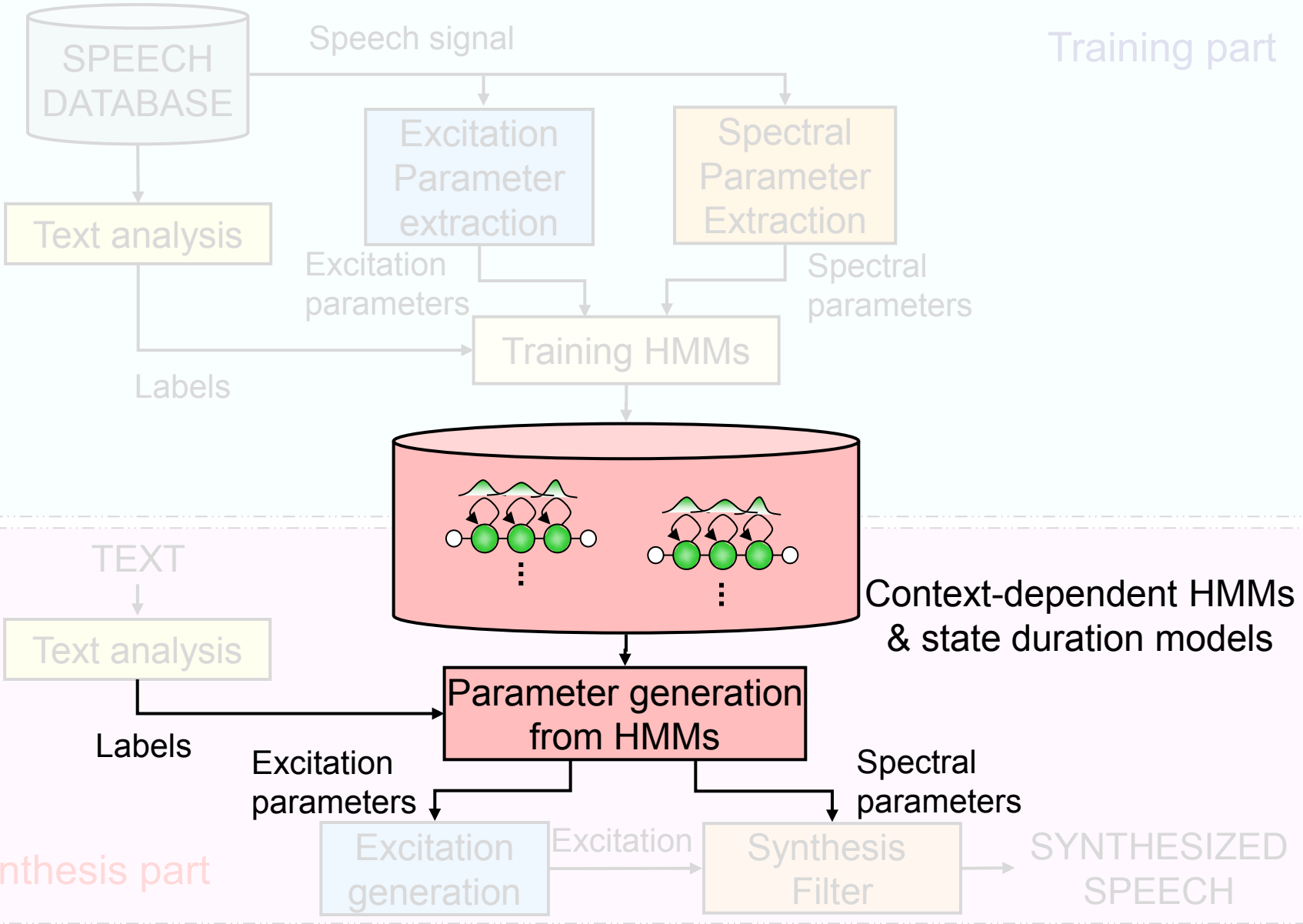
Decision tree-based state clustering [Odell; '95]



Stream-dependent tree-based clustering (2)



HMM-based speech synthesis system



Speech parameter generation algorithm [Tokuda; '00]

For given sentence HMM, determine a speech parameter vector sequence $\boldsymbol{o} = [\boldsymbol{o}_1^\top, \boldsymbol{o}_2^\top, \ldots, \boldsymbol{o}_T^\top]^\top$ which maximizes

$$P(\boldsymbol{o} \mid \hat{\boldsymbol{l}}, \hat{\boldsymbol{\lambda}}) = \sum_{\boldsymbol{q}} P(\boldsymbol{o} \mid \boldsymbol{q}, \hat{\boldsymbol{\lambda}}) P(\boldsymbol{q} \mid \hat{\boldsymbol{l}}, \hat{\boldsymbol{\lambda}})$$

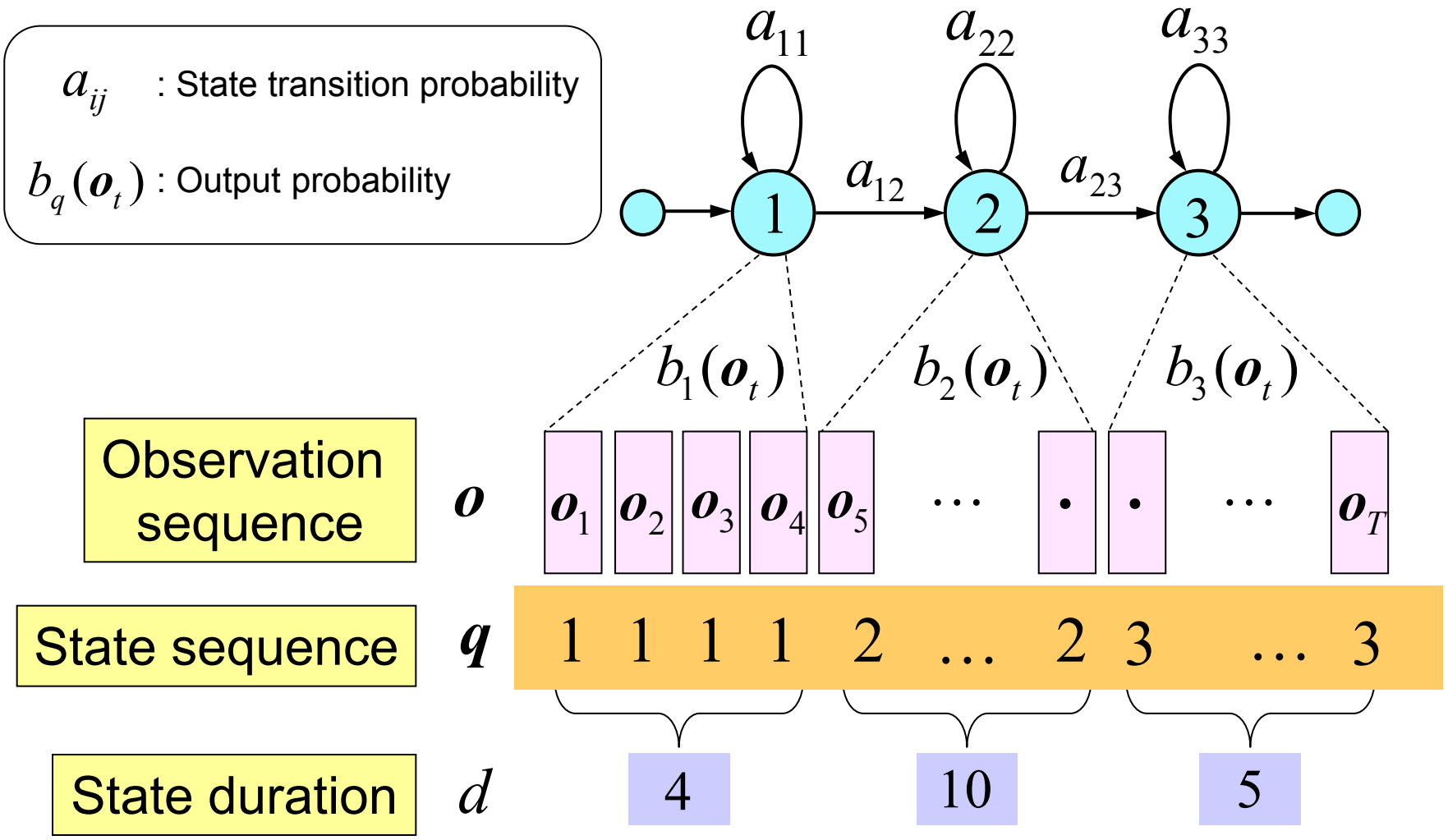
$$\approx \max_{\boldsymbol{q}} P(\boldsymbol{o} \mid \boldsymbol{q}, \hat{\boldsymbol{\lambda}}) P(\boldsymbol{q} \mid \hat{\boldsymbol{l}}, \hat{\boldsymbol{\lambda}})$$

$$\Downarrow$$

$$\hat{\boldsymbol{q}} = \arg\max_{\boldsymbol{q}} P(\boldsymbol{q} \mid \hat{\boldsymbol{l}}, \hat{\boldsymbol{\lambda}})$$

$$\hat{\boldsymbol{o}} = \arg\max_{\boldsymbol{o}} P(\boldsymbol{o} \mid \hat{\boldsymbol{q}}, \hat{\boldsymbol{\lambda}})$$

Determination of state sequence (1/3)



Determine state sequence via determining state durations

Determination of state sequence

$$P(\boldsymbol{q} \mid \hat{\boldsymbol{l}}, \hat{\boldsymbol{\lambda}}) = \prod_{i=1}^{K} p_i(d_i)$$

$p_i(\cdot)$: state-duration distribution of i-th state

d_i : state duration of i-th state

K : # of states in a sentence HMM for $\hat{\boldsymbol{l}}$

Gaussian

$$p_i(d_i) = N(d_i \mid m_i, \sigma_i^2) \quad \Rightarrow \quad \hat{d}_i = m_i$$

Speech parameter generation algorithm

For given HMM $\boldsymbol{\lambda}$,determine a speech parameter vector Sequence $\boldsymbol{o} = [\boldsymbol{o}_1^{\top}, \boldsymbol{o}_2^{\top}, \ldots, \boldsymbol{o}_T^{\top}]^{\top}$ which maximizes

$$P(\boldsymbol{o} \mid \hat{\boldsymbol{l}}, \hat{\boldsymbol{\lambda}}) = \sum_{\boldsymbol{q}} P(\boldsymbol{o} \mid \boldsymbol{q}, \hat{\boldsymbol{\lambda}}) P(\boldsymbol{q} \mid \hat{\boldsymbol{l}}, \hat{\boldsymbol{\lambda}})$$

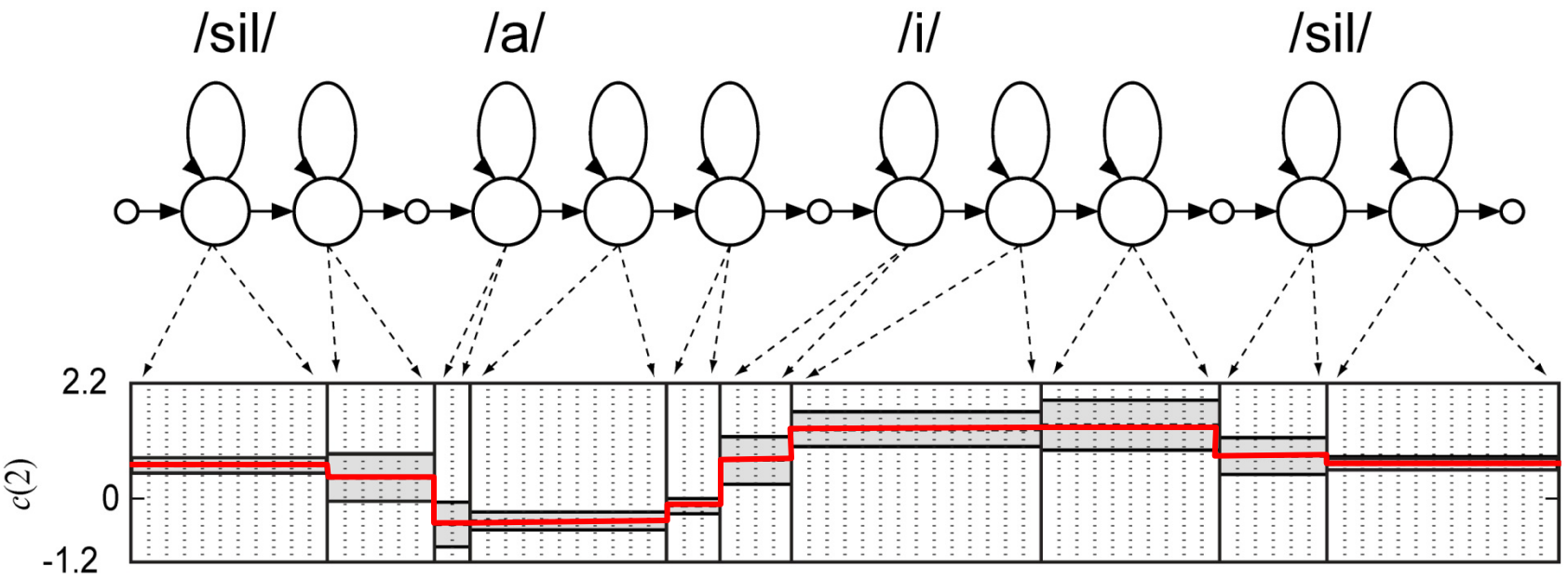
$$\approx \max_{\boldsymbol{q}} P(\boldsymbol{o} \mid \boldsymbol{q}, \hat{\boldsymbol{\lambda}}) P(\boldsymbol{q} \mid \hat{\boldsymbol{l}}, \hat{\boldsymbol{\lambda}})$$

$$\Downarrow$$

$$\hat{\boldsymbol{q}} = \arg\max_{\boldsymbol{q}} P(\boldsymbol{q} \mid \hat{\boldsymbol{l}}, \hat{\boldsymbol{\lambda}})$$

$$\hat{\boldsymbol{o}} = \arg\max_{\boldsymbol{o}} P(\boldsymbol{o} \mid \hat{\boldsymbol{q}}, \hat{\boldsymbol{\lambda}})$$

Without dynamic feature



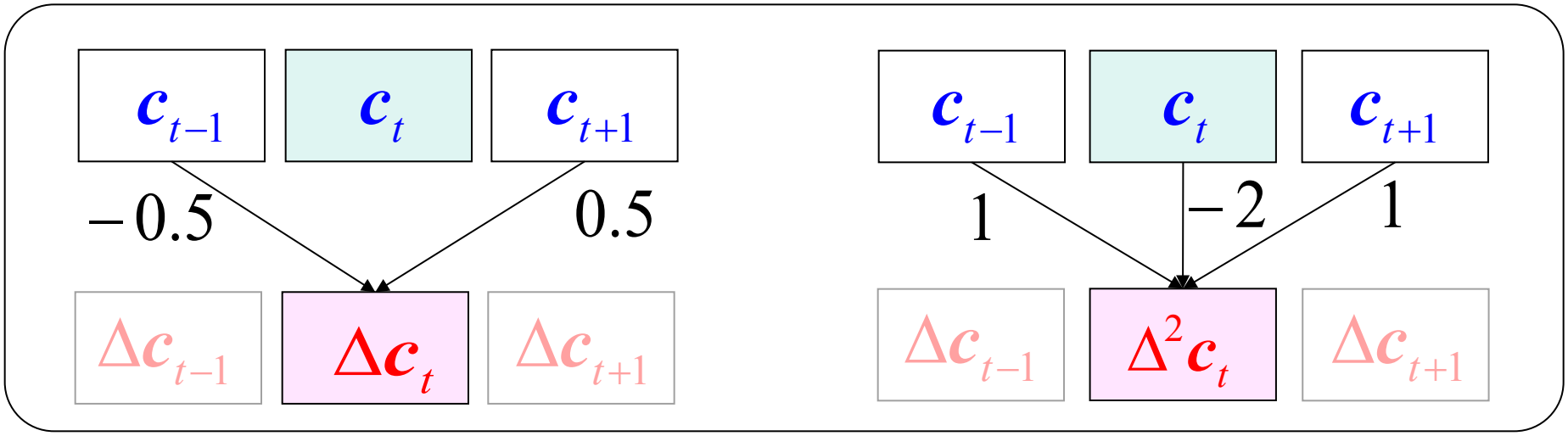
becomes a sequence of mean vectors

⇒ discontinuous outputs between states

Dynamic features

$$\Delta \boldsymbol{c}_t = \frac{\partial \boldsymbol{c}_t}{\partial t} \approx 0.5(\boldsymbol{c}_{t+1} - \boldsymbol{c}_{t-1})$$

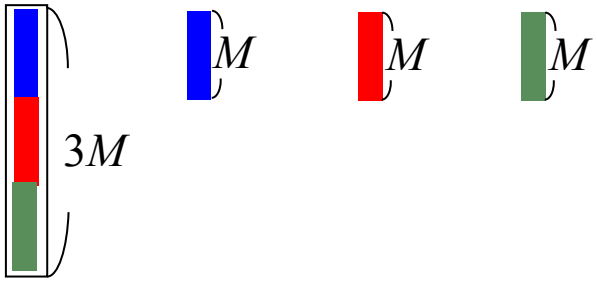
$$\Delta^2 \boldsymbol{c}_t = \frac{\partial^2 \boldsymbol{c}_t}{\partial t^2} \approx \boldsymbol{c}_{t+1} - 2\boldsymbol{c}_t + \boldsymbol{c}_{t-1}$$

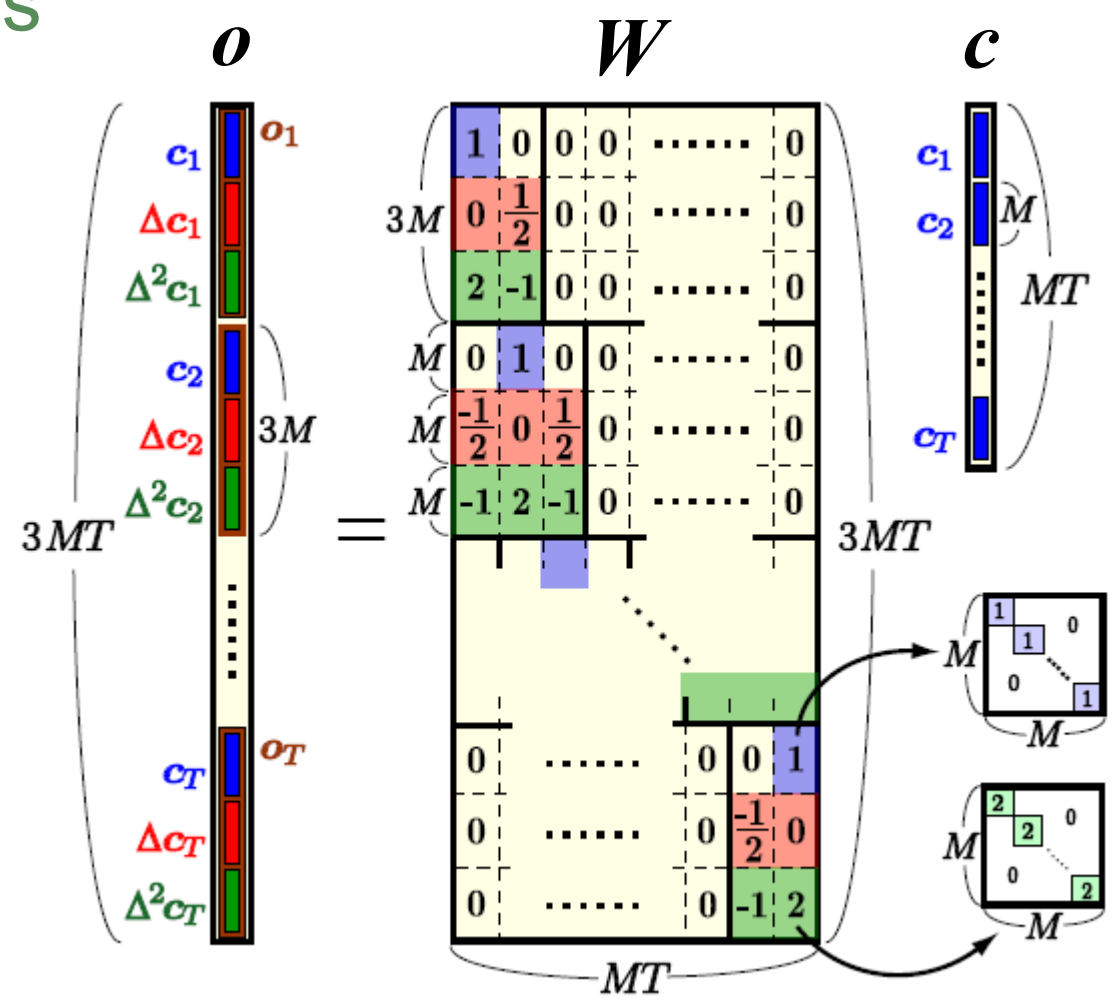


Integration of dynamic features

Relationship between speech parameter vectors & static feature vectors

$$\boldsymbol{o}_t = \left[\boldsymbol{c}_t^{\mathsf{T}}, \Delta\boldsymbol{c}_t^{\mathsf{T}}, \Delta^2\boldsymbol{c}_t^{\mathsf{T}}\right]^{\mathsf{T}}$$





Solution for the problem (1/2)

By setting

$$\frac{\partial \log P(\overbrace{\boldsymbol{O}}^{\boldsymbol{Wc}} \mid \hat{\boldsymbol{q}}, \lambda)}{\partial \boldsymbol{c}} = \boldsymbol{O},$$

we obtain

$$\boldsymbol{W}^{\mathsf{T}} \boldsymbol{\Sigma}_{\hat{\boldsymbol{q}}}^{-1} \boldsymbol{W} \boldsymbol{c} = \boldsymbol{W}^{\mathsf{T}} \boldsymbol{\Sigma}_{\hat{\boldsymbol{q}}}^{-1} \boldsymbol{\mu}_{\hat{\boldsymbol{q}}} \ ,$$

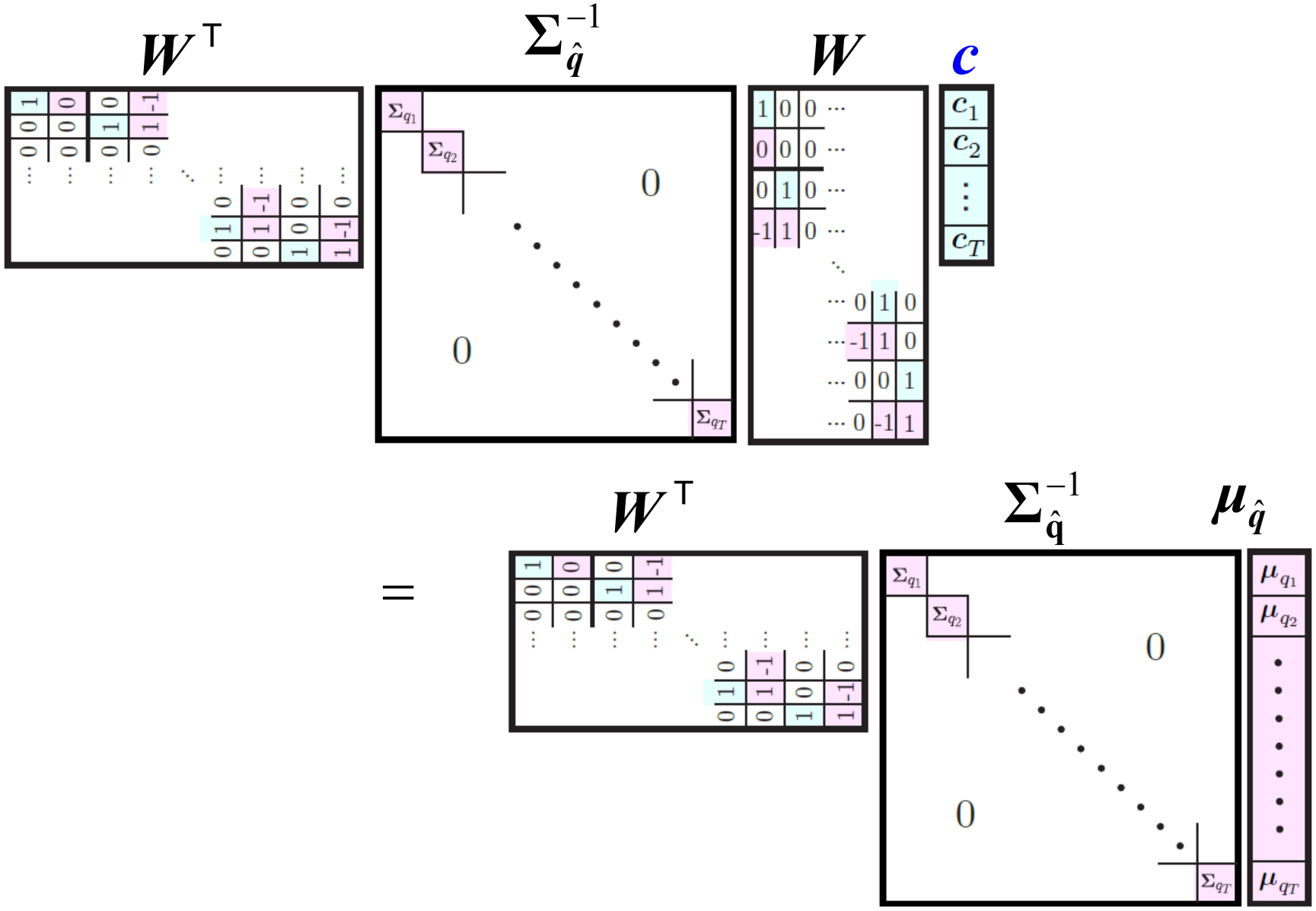
where

$$\boldsymbol{c} = [\boldsymbol{c}_1^{\mathsf{T}}, \boldsymbol{c}_2^{\mathsf{T}}, \ldots, \boldsymbol{c}_T^{\mathsf{T}}]^{\mathsf{T}}$$

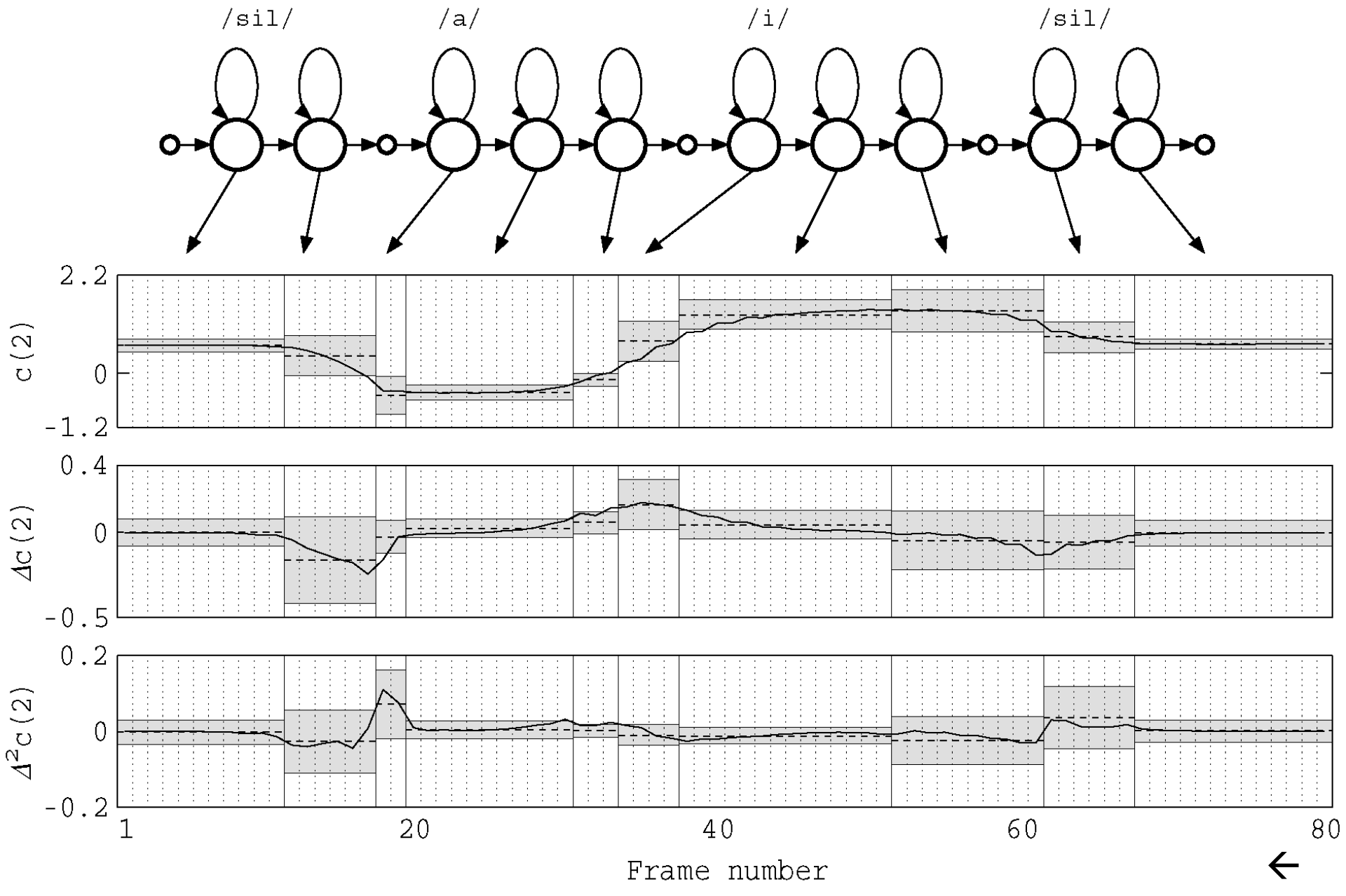
$$\boldsymbol{\mu}_{\hat{\boldsymbol{q}}} = [\boldsymbol{\mu}_{\hat{\boldsymbol{q}}_1}^{\mathsf{T}}, \boldsymbol{\mu}_{\hat{\boldsymbol{q}}_2}^{\mathsf{T}}, \ldots, \boldsymbol{\mu}_{\hat{\boldsymbol{q}}_T}^{\mathsf{T}}]^{\mathsf{T}}$$

$$\boldsymbol{\Sigma}_{\hat{\boldsymbol{q}}} = [\boldsymbol{\Sigma}_{\hat{\boldsymbol{q}}_1}^{\mathsf{T}}, \boldsymbol{\Sigma}_{\hat{\boldsymbol{q}}_2}^{\mathsf{T}}, \ldots, \boldsymbol{\Sigma}_{\hat{\boldsymbol{q}}_T}^{\mathsf{T}}]^{\mathsf{T}}$$

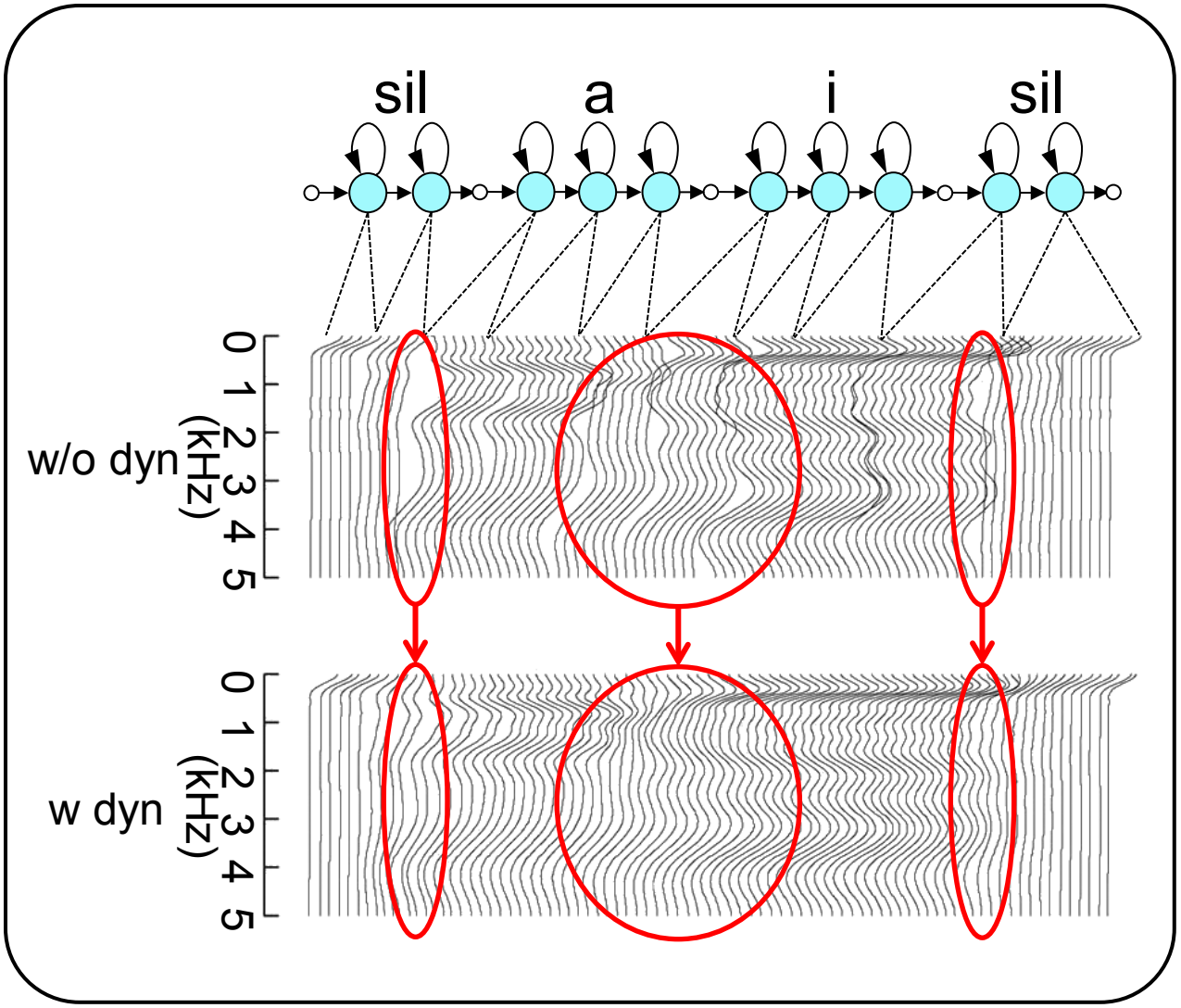
Solution for the problem (2/2)



Generated speech parameter trajectory

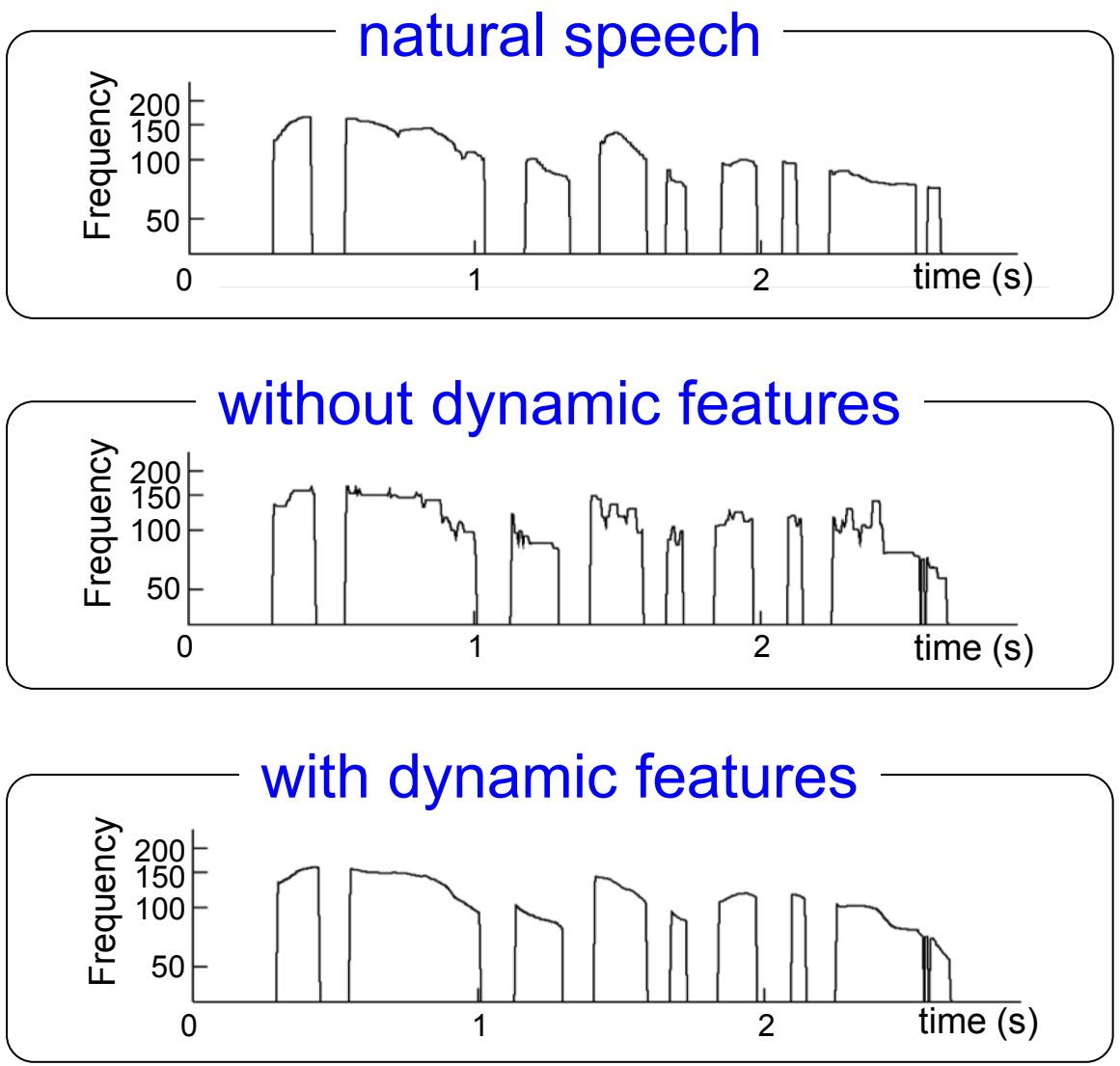


Generated spectra



Spectra changing smoothly at state boundaries

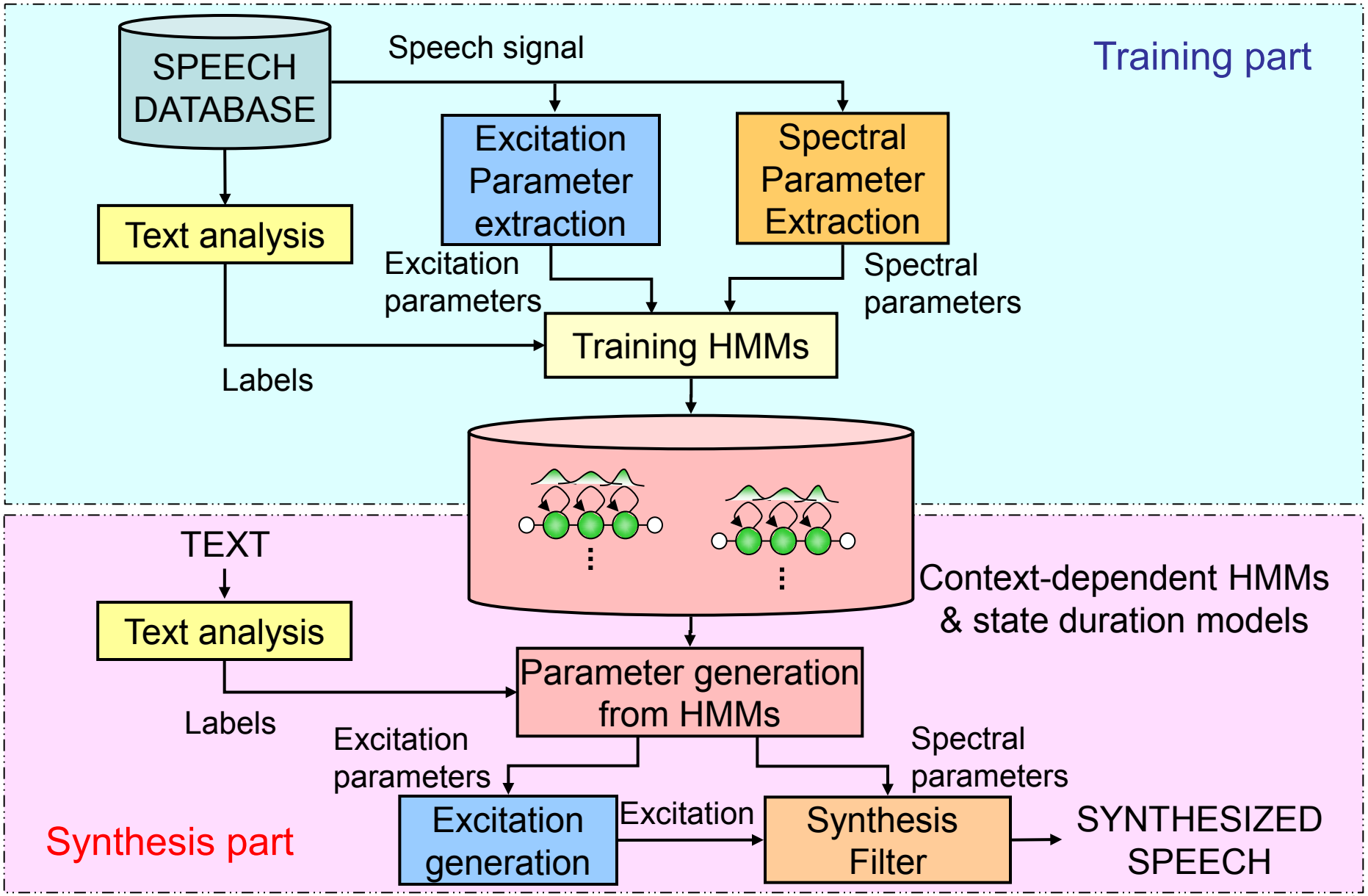
Generated F0



Effect of dynamic features

		Mel-cepstrum	
		static+ $\Delta + \Delta^2$	static
log F0	static+ $\Delta + \Delta^2$	 Smooth!	
	static		

HMM-based speech synthesis system



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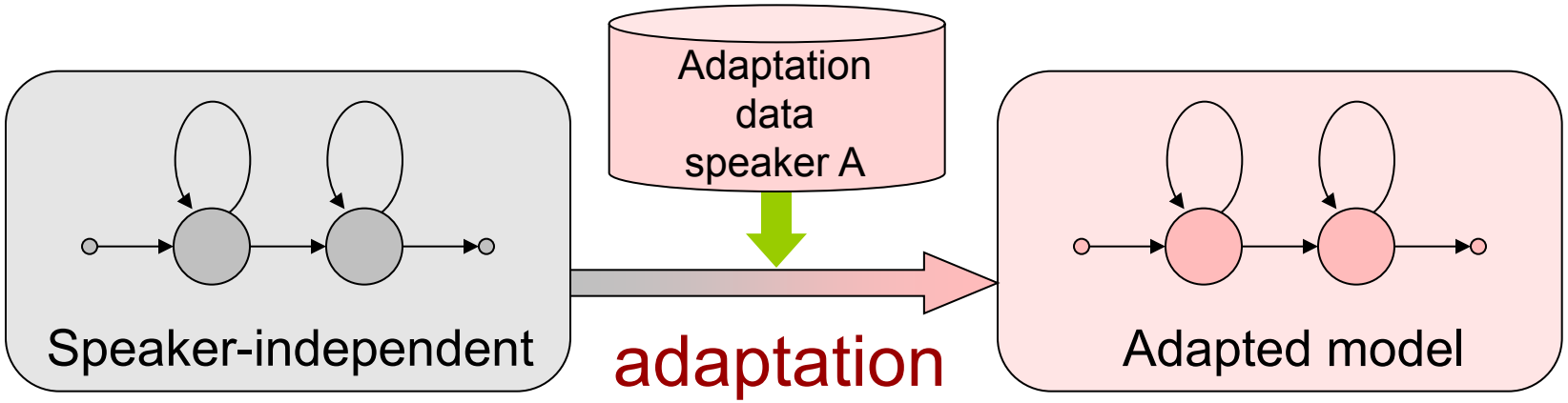
Emotional speech synthesis

text	neutral	angry
「授業中に携帯いじってんじゃねえよ！ 電源切っとけ！」 “Don’t touch your cell phone during a class! Turn off it!”		
「ミーティングには毎週参加しなさい！」 “You must attend the weekly meeting!”		

trained with 200 utterances

Speaker adaptation (mimicking voices)

MLLR-based adaptation



w/o adaptation (initial model)

Adapted with 4 utterances

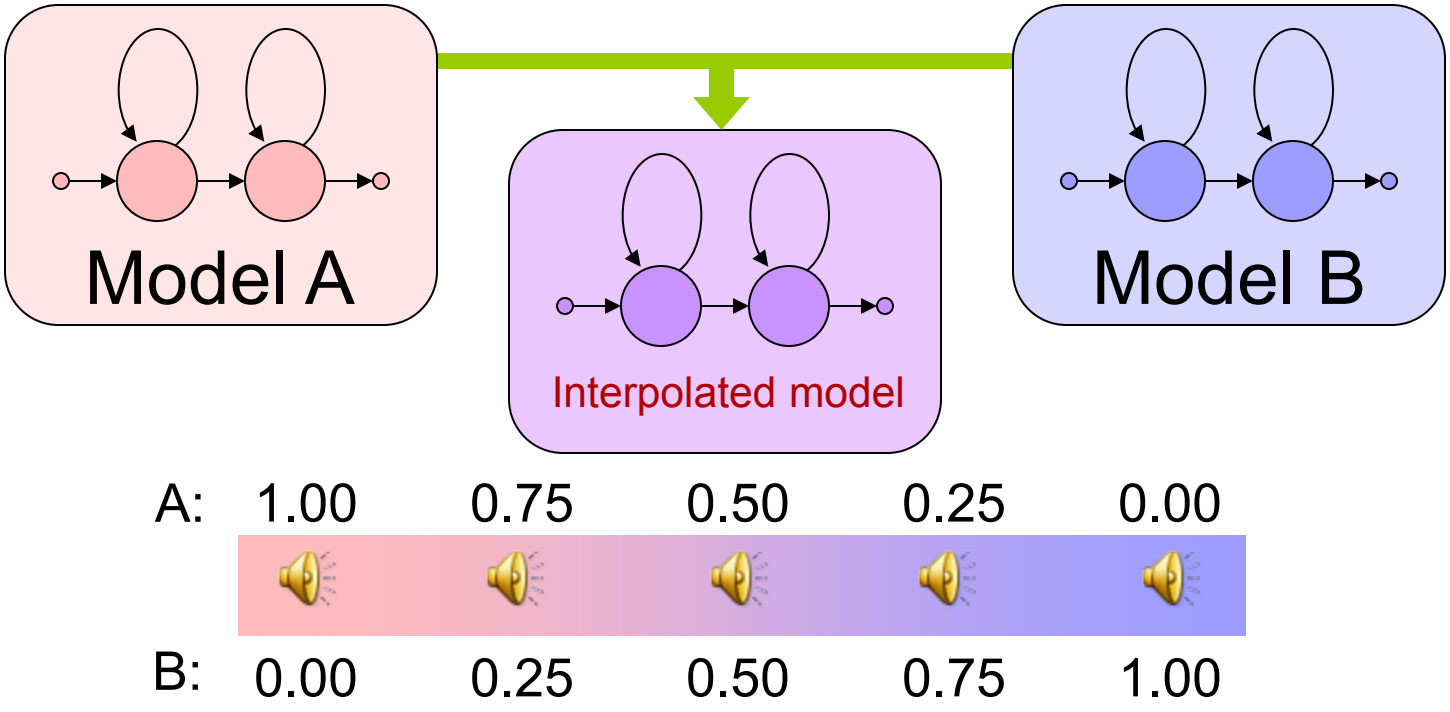
Adapted with 50 utterances

Speaker A's speaker-dependent system

?

Speaker interpolation (mixing voices)

Linear combination of two speaker-dependent models



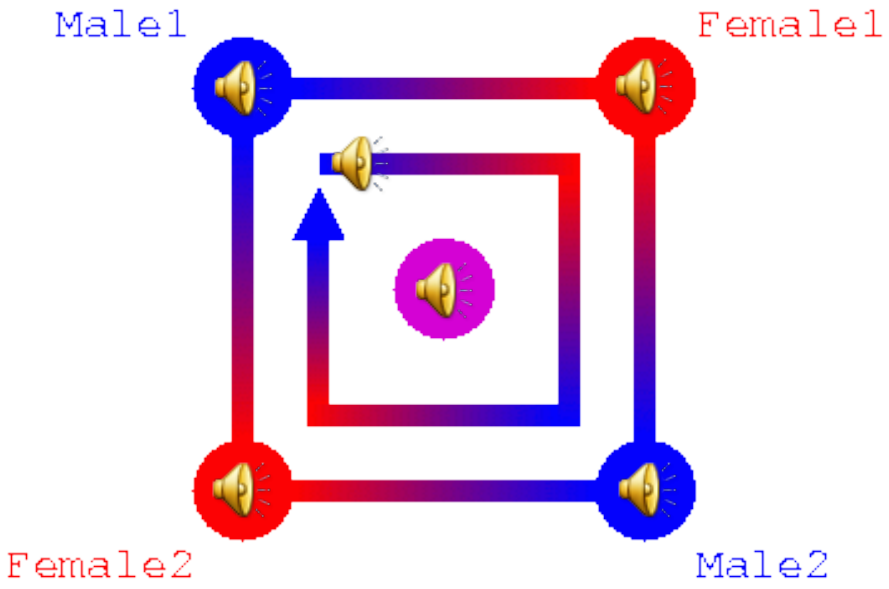
Voice morphing

Two voices:

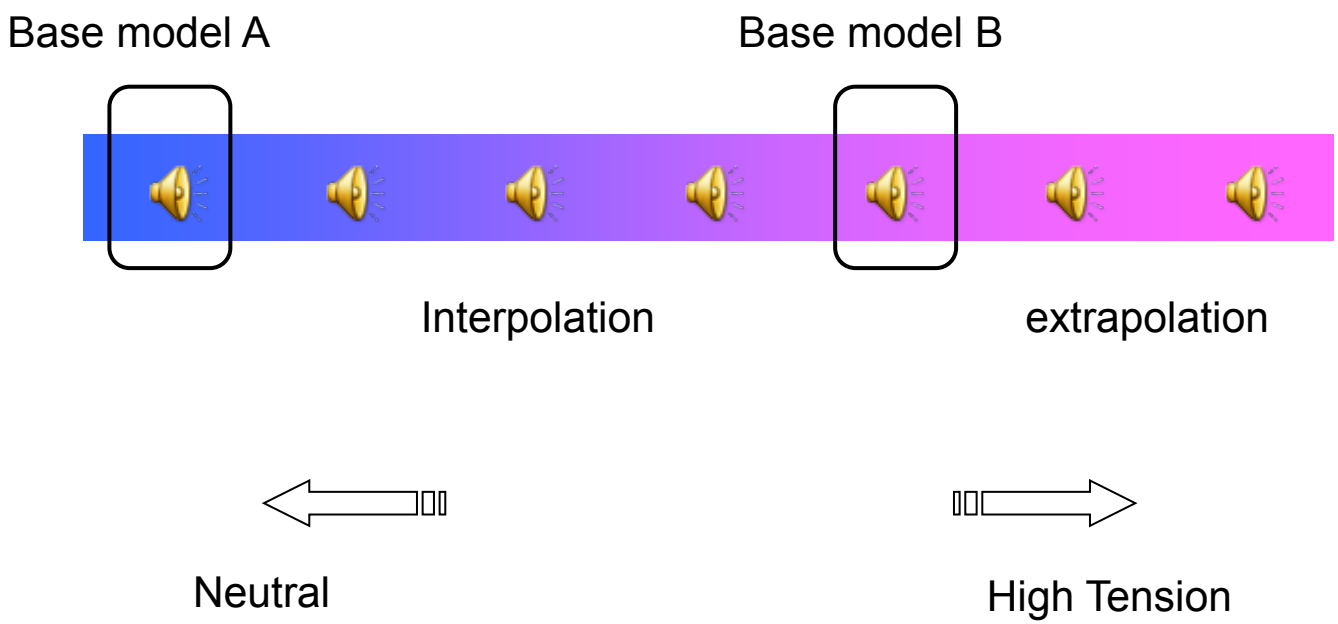
A ⇨⇨⇨⇨⇨⇨⇨⇨⇨ B

A ⇦⇦⇦⇦⇦⇦⇦⇦⇦ B

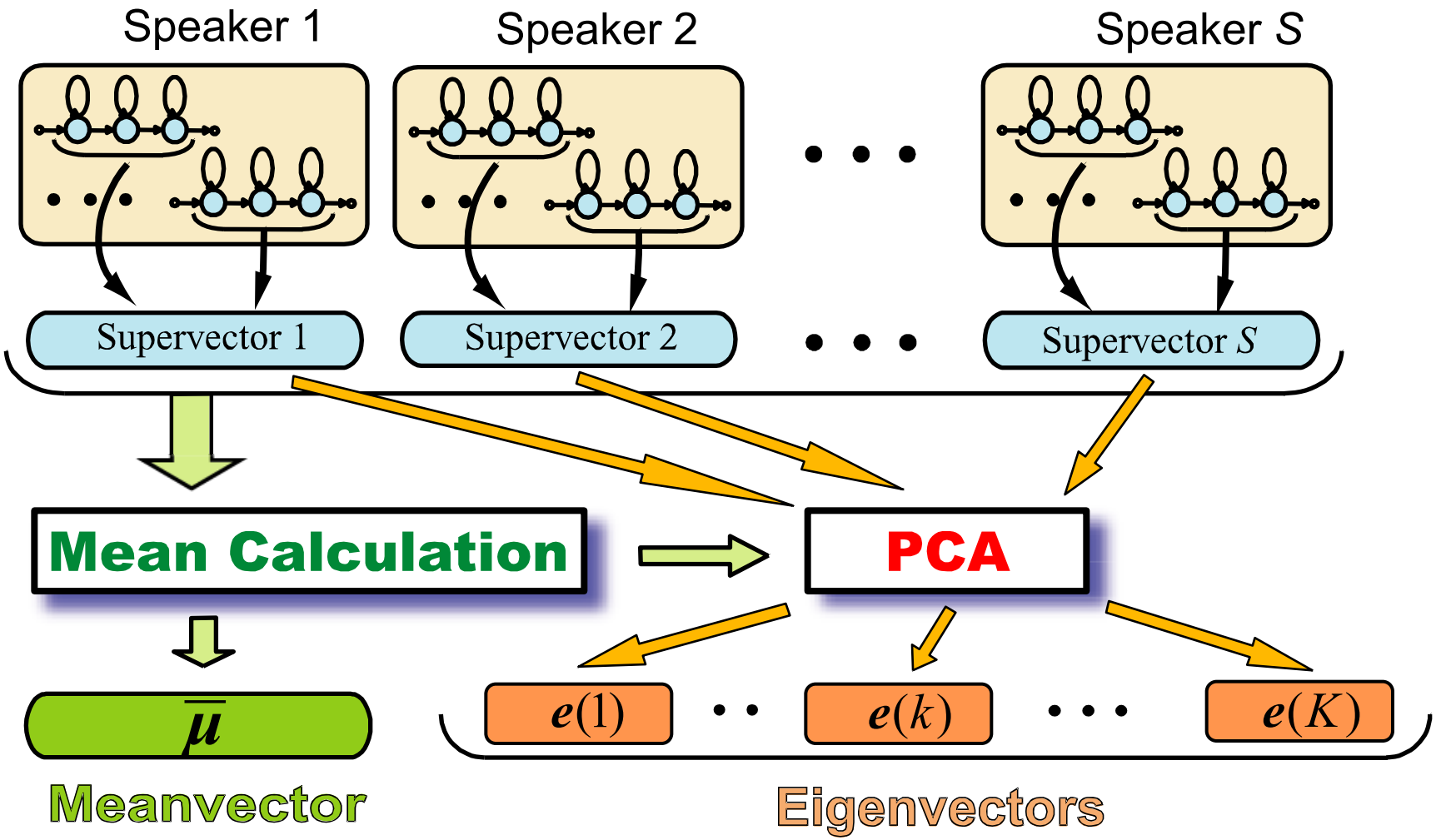
Four voices:



Interpolation of speaking styles



Eigenvoice (creating voices) [Shichiri; '02]



Click here for a demo

Multilingual speech synthesis

- Japanese 
- American English 
- Chinese (Mandarin) (by ATR) 
- Brazilian Portuguese (by Nitech, and UFRJ) 
- European Portuguese (by Nitech, Univ of Porto, and UFRJ) 
- Slovenian
  (by Bostjan Vesnicer, University of Ljubljana, Slovenia) 
- Swedish (by Anders Lundgren, KTH, Sweden) 
- German (by University of Bonn, and Nitech) 
- Korean (by Sang-Jin Kim, ETRI, Korea) 
- Finish (by TKK, Finland) 
- Baby English (by Univ of Edinburgh, UK) 
- Polish, Slovak, Arabic, Farsi, Croatian, Polyglot, etc.

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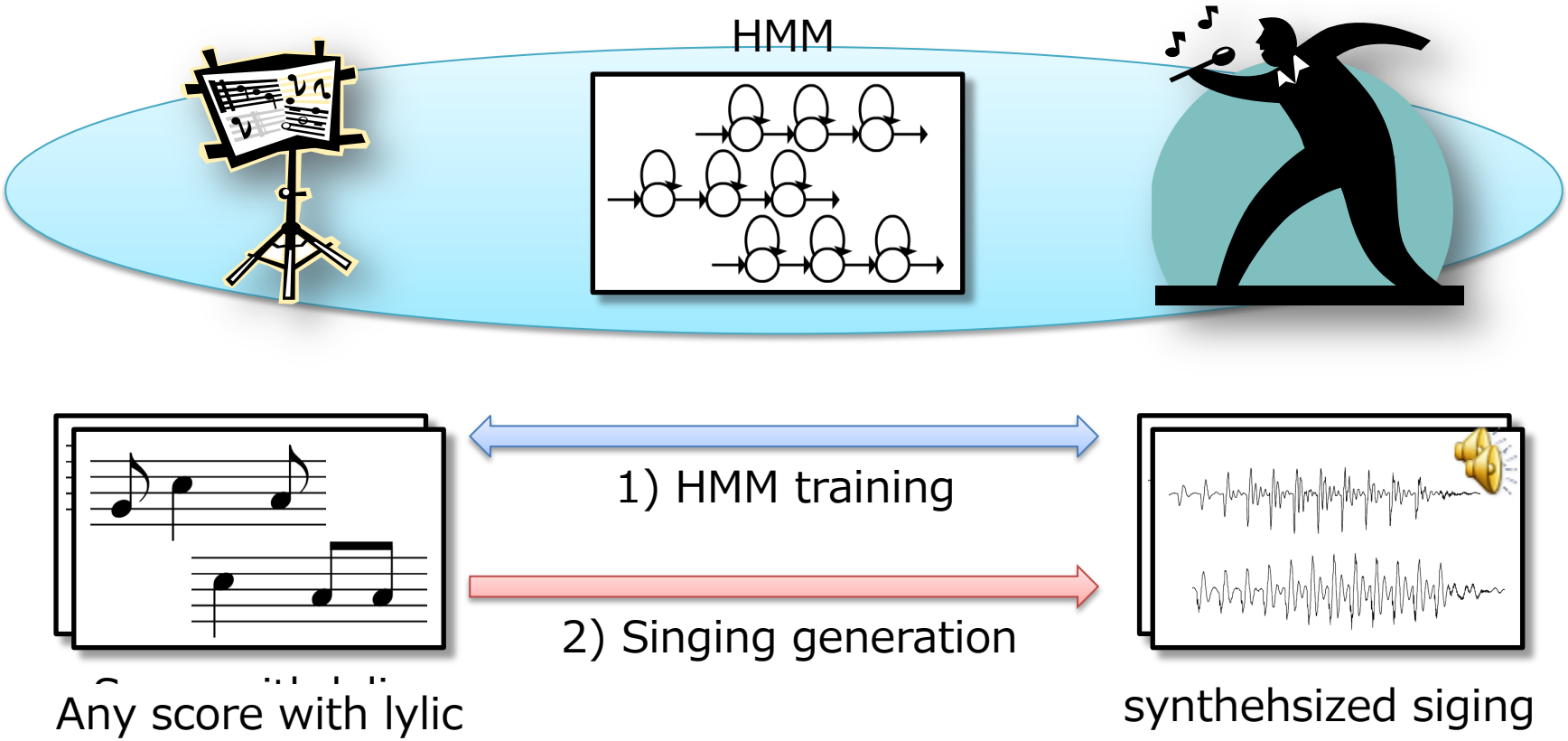
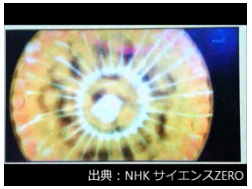
HMM-based singing synthesis



Flexibility in singing synthesis

- Sing a popular song

- Sing by a famous person's voice
  (from a TV program NHK "Science zero")

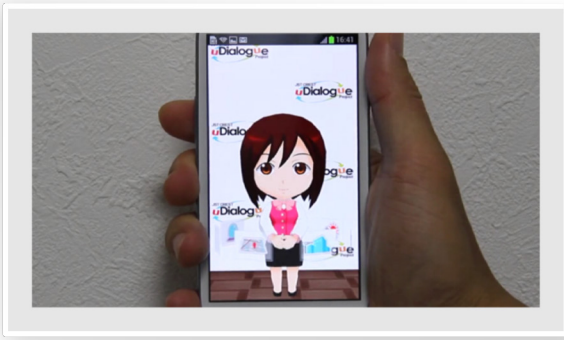


- Rap singing

MMDAgent

A toolkit for building voice interaction systems

- Fully open-source toolkit with open interfaces
- Tightly integrated speech recognition/synthesis engines
- 3-D scene rendering fully compatible with CG tools
- HMM-based flexible and expressive speech synthesis (neutral, angry, bashful, happy, sad)



Summary

Statistical approach to speech synthesis

- Whole speech synthesis process is described in a unified statistical framework
- It can provide flexibility: various voices, emotional expressions, speaking styles, etc.

Future work

- Still we have many problems should be solved:
  - Direct modeling of speech waveform
  - Importance of the database

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