

Voice Conversion

Voice Conversion

I. Introduction to Voice Conversion

- Speech Synthesis Context (TTS)
- Overview of Voice Conversion

II. Spectrum Transformation in VC

- Gaussian Mixture Model

III. Conversion Results

- Objective Metrics & Subjective Evaluations
- Sound Samples

IV. Summary & Conclusions

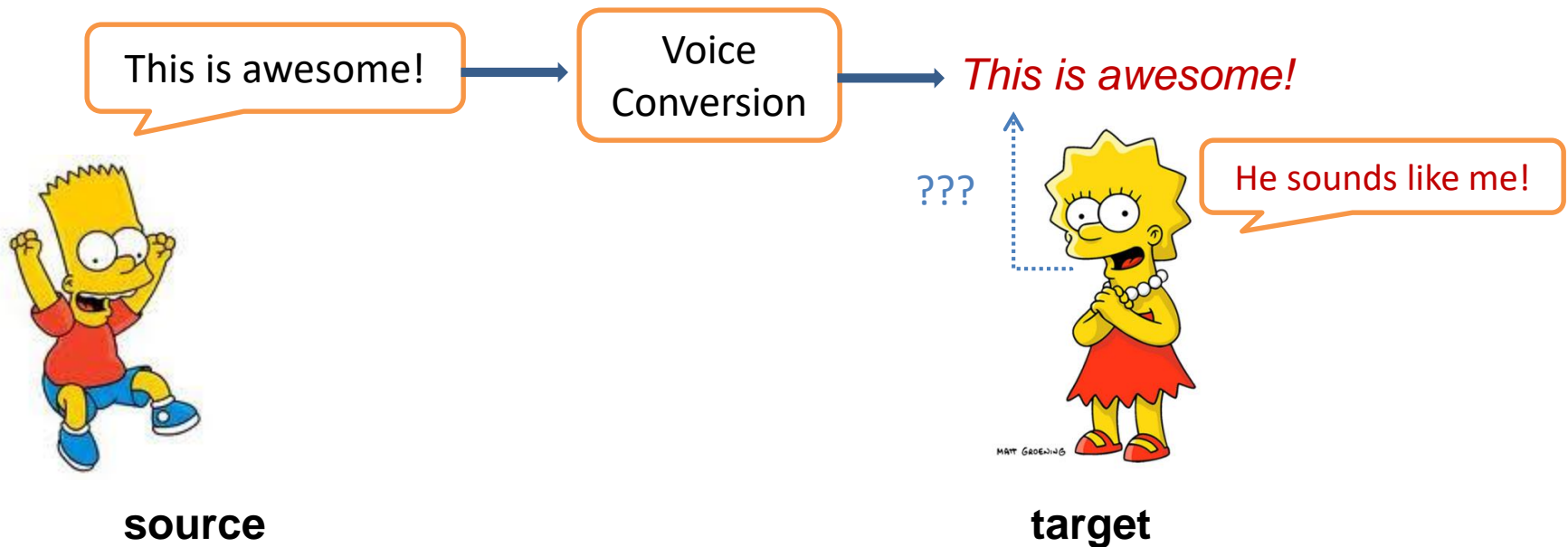
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★ I. Introduction to Voice Conversion

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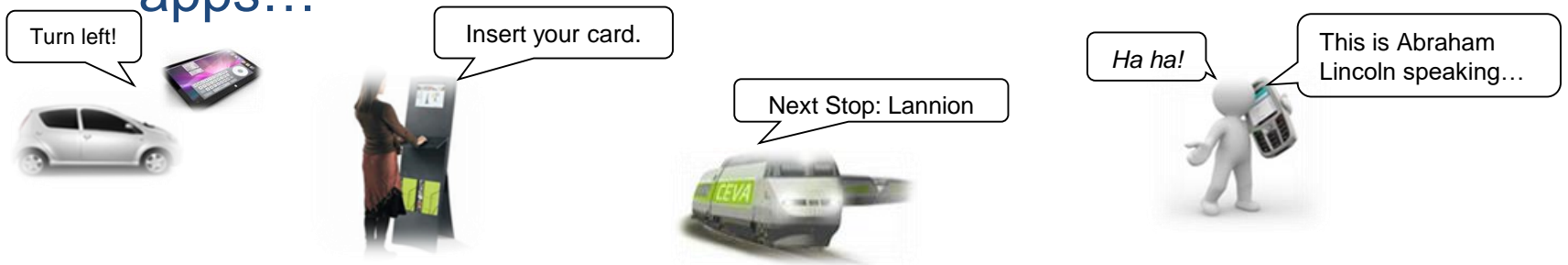
Voice Conversion (VC)

- Transform the speech of a (source) speaker so that it sounds like the speech of a different (target) speaker.



Context: Speech Synthesis

- ▶ Increase in applications using speech technologies
 - ▶ Cell phones, GPS, video gaming, customer service apps...



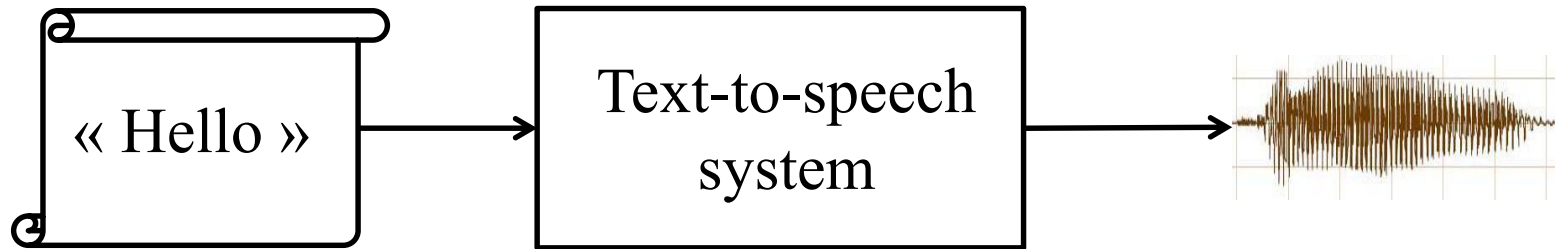
Information communicated through speech!

Text-to-Speech!

- ▶ **Text-to-Speech (TTS) Synthesis**
 - ▶ Generate speech from a given text



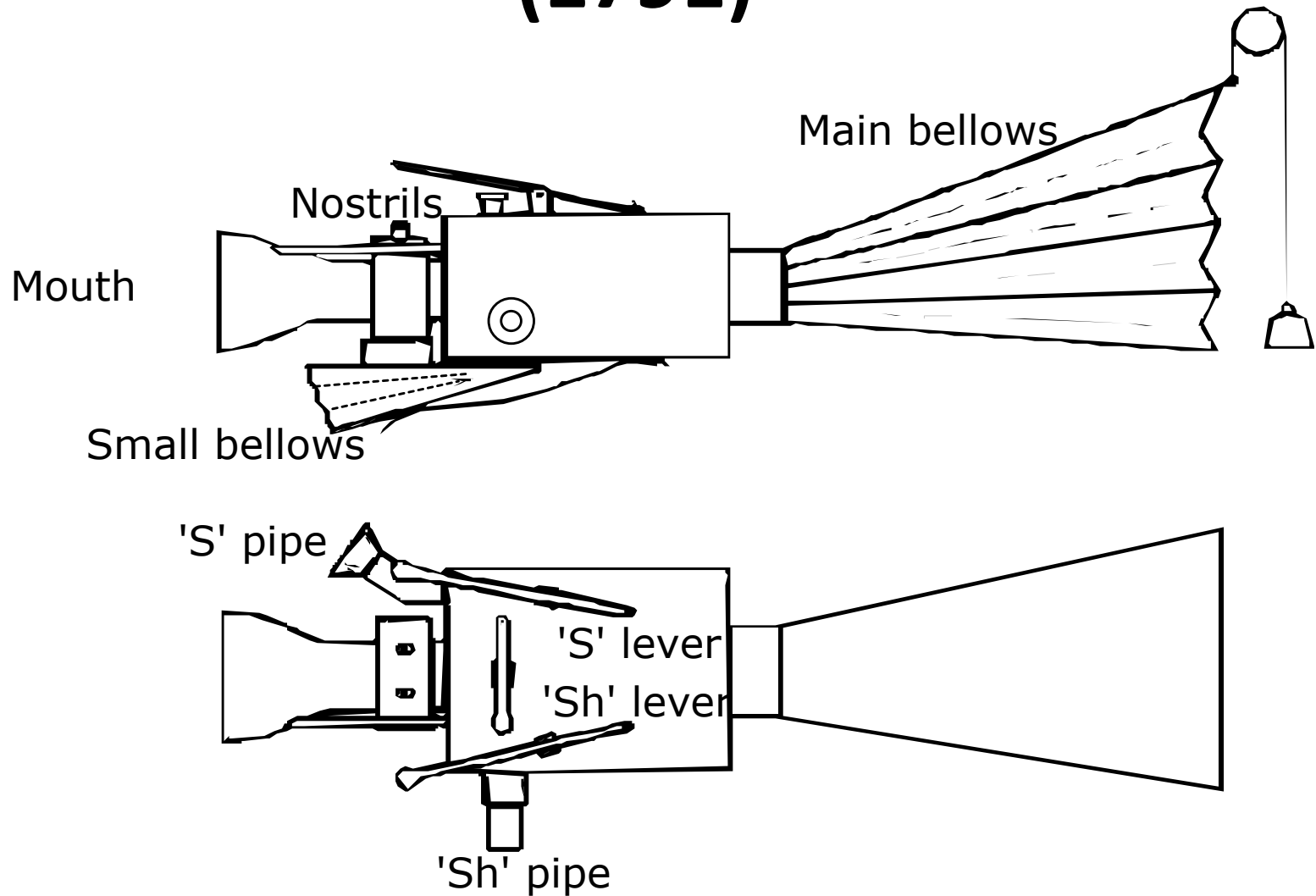
Speech Synthesis



GOAL :

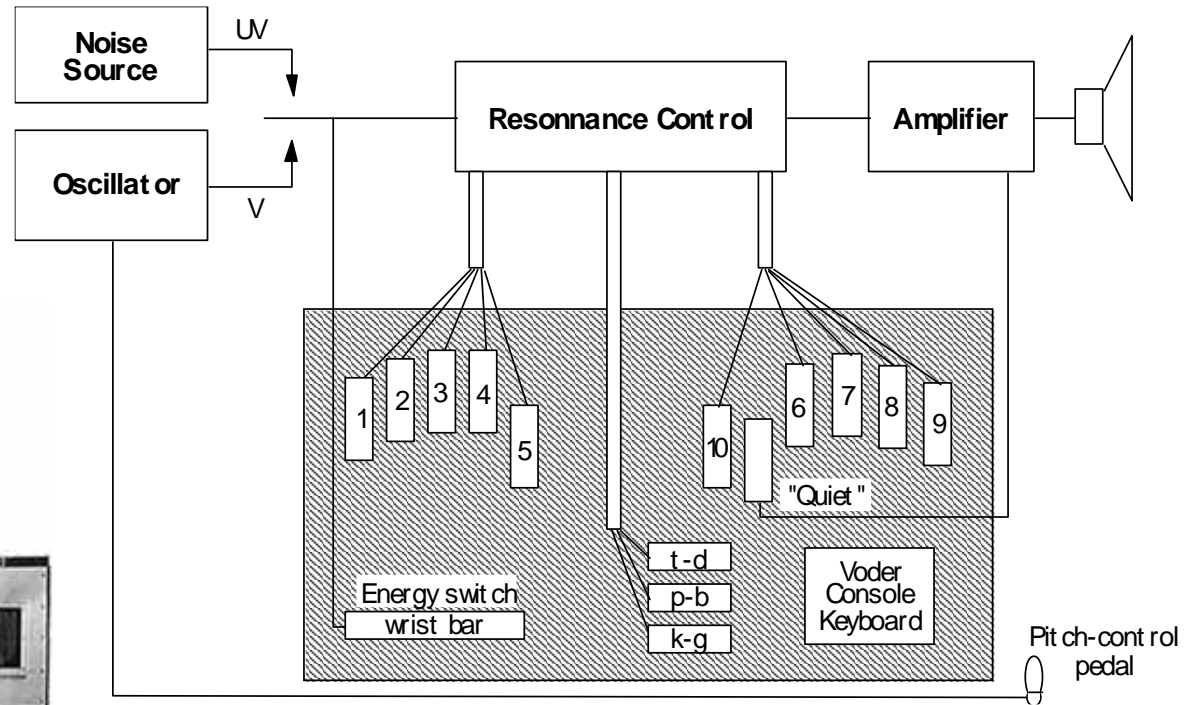
**Produce the lecture of an unknown text typed
by the user**

Von Kempelen's talking machine (1791)



Omer Dudley's Voder

(Bell Labs, 1936)



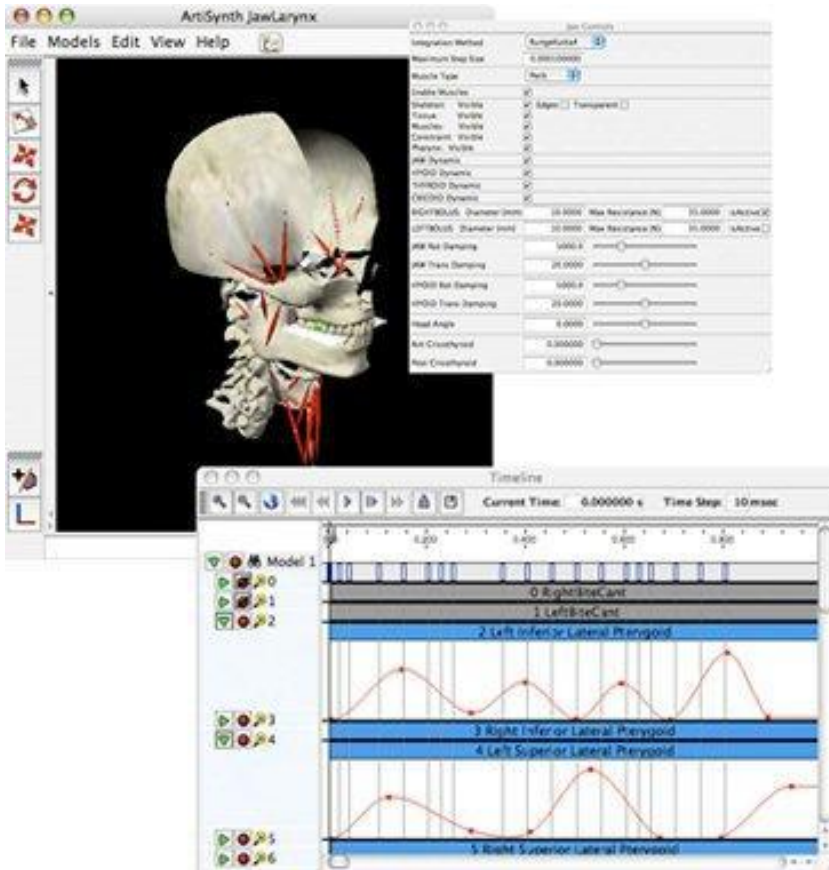
And other developments in articulatory synthesis

- Work by :

K. Stevens, G. Fant, P. Mermelstein, R. Carré (*GNUSpeech*),
S. Maeda,
J. Shroeter & M. Sondhi...

- More recently :

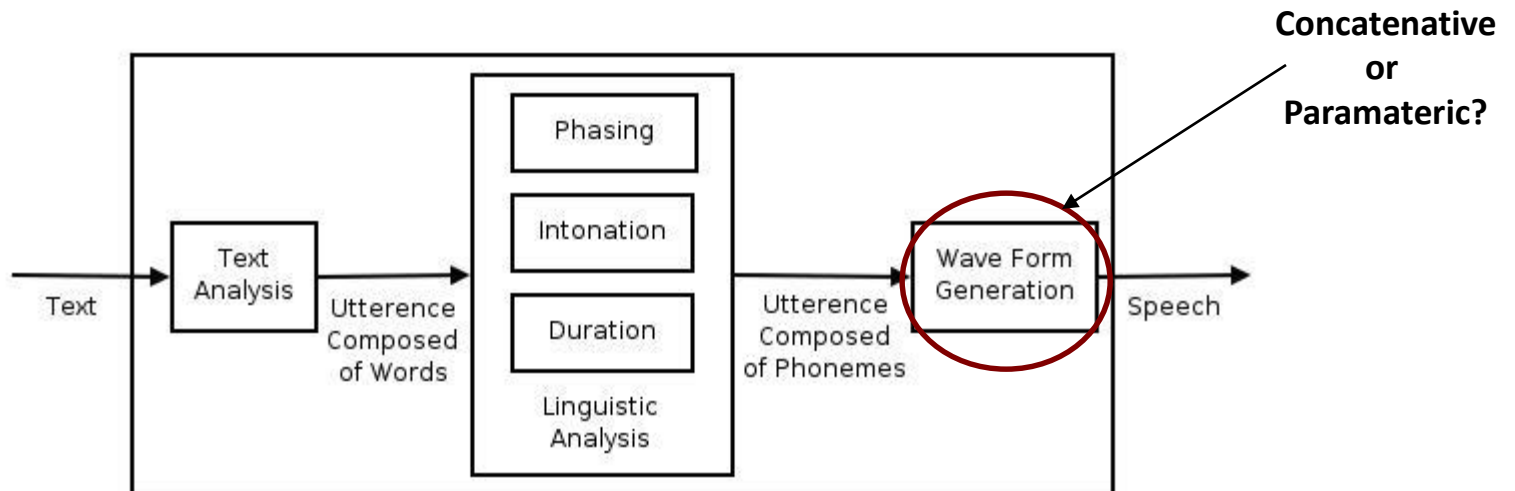
O. Engwall, S. Fels (*ArtiSynth*),
Birkholz and Kröger, A. Alwan & S. Narayanan (MRI)...



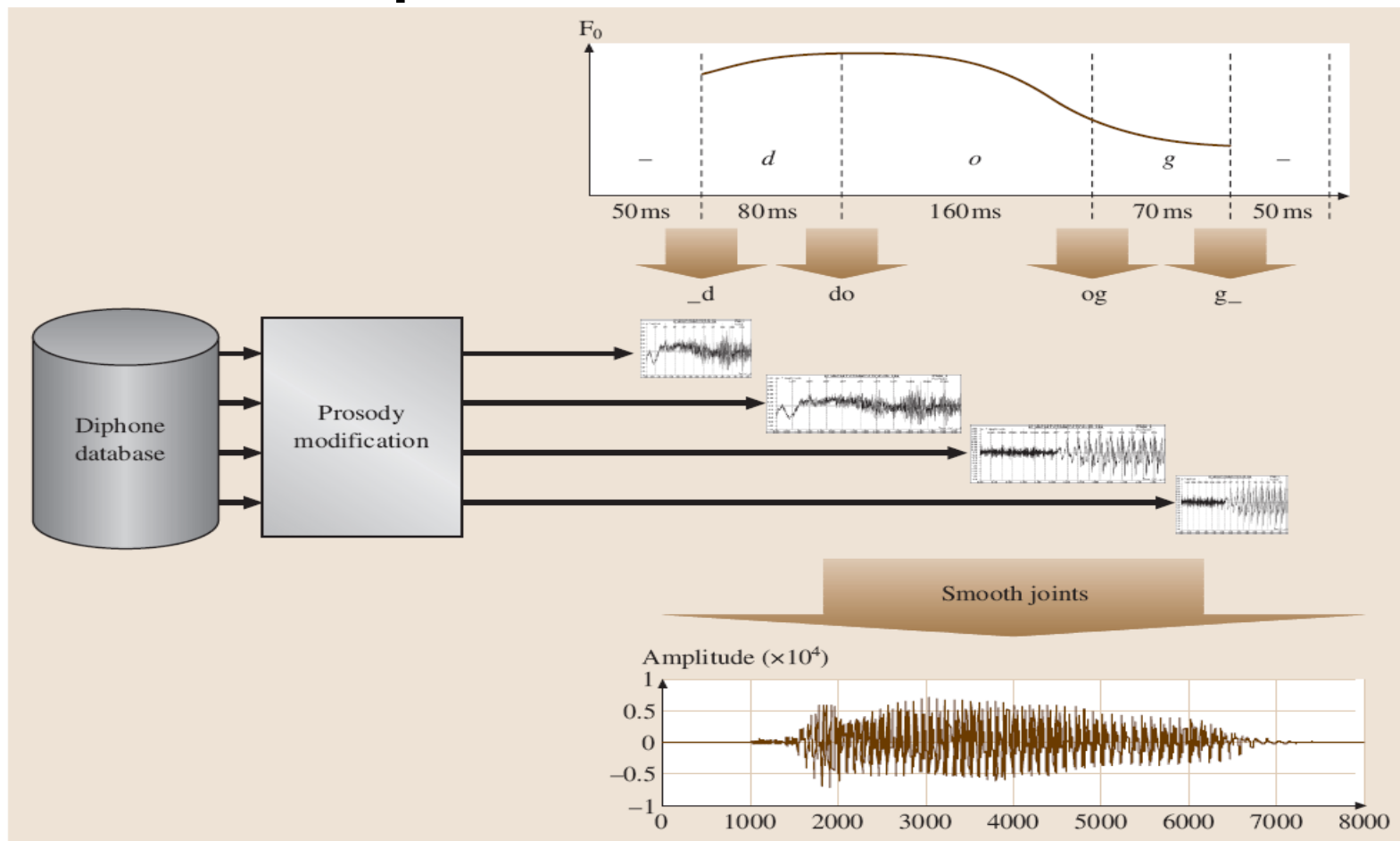
Text-to-Speech (TTS) Systems

- **TTS Approaches**

1. **Concatenative:** speech synthesized from recorded segments
 - Unit-Selection: parts of speech chosen from corpora & strung together
High-quality synthesis, but need to record & process corpora
2. **Parametric:** speech generated from model parameters
 - HMM-based: speaker models built from speech using linguistic info
Limited quality due to simplified speech modeling & statistical averaging

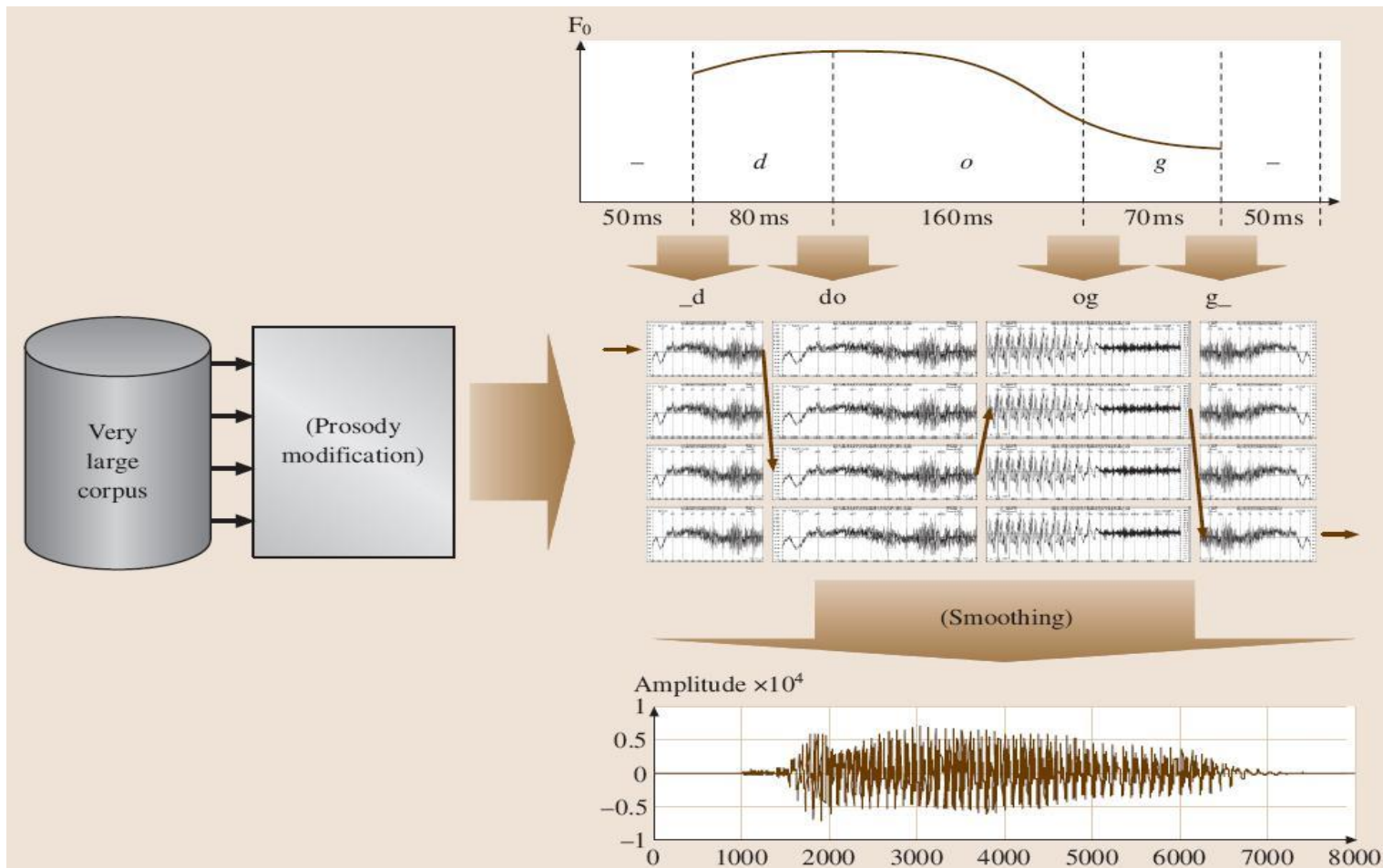


Diphone concatenation



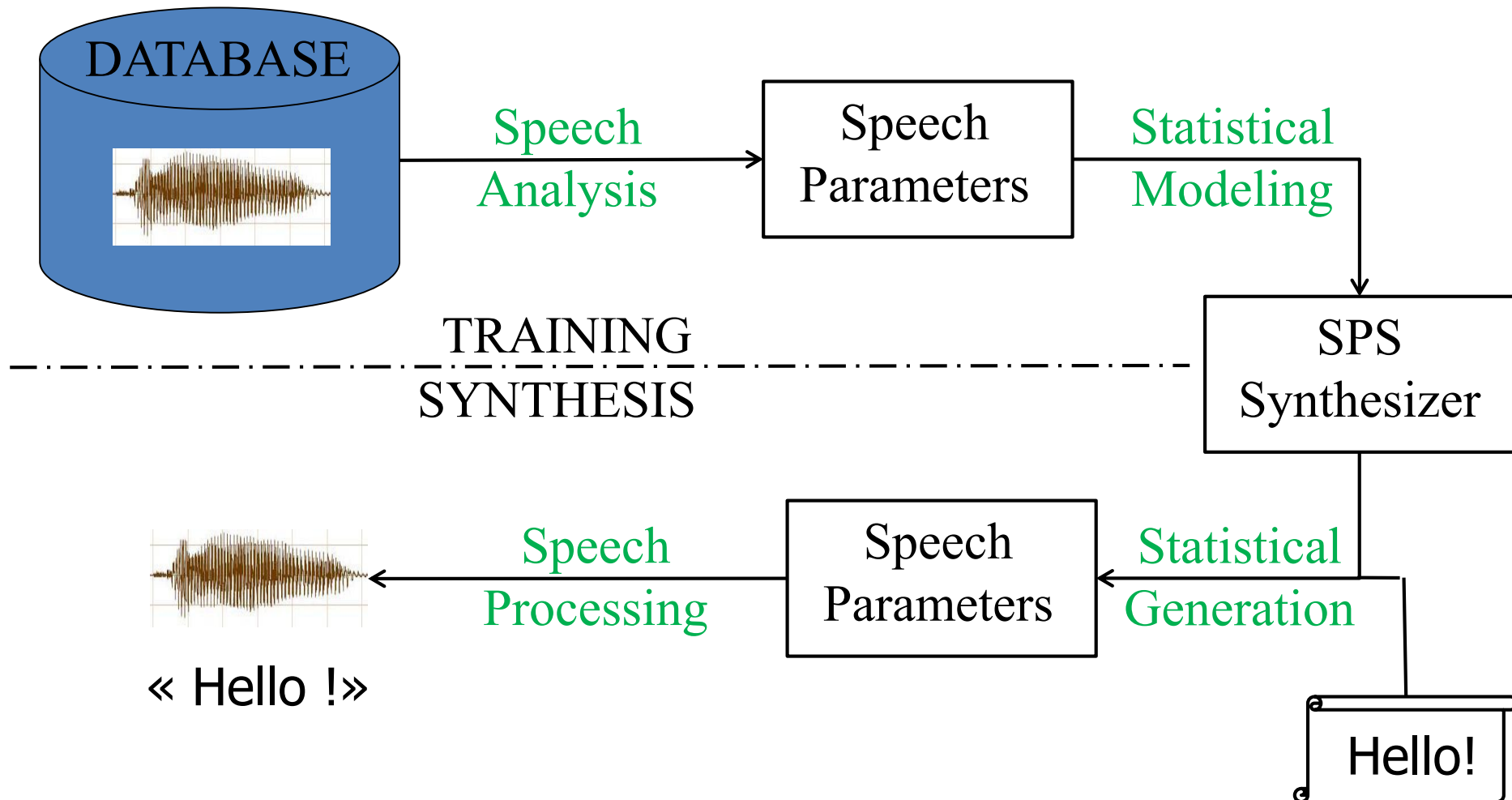
Intelligibility ✓ Naturalness ~ Mem/CPU/Voices ✓ Expressivity ✗

Unit selection



Intelligibility ✓ Naturalness ✓ Mem/CPU/Voices ~ Expressivity ~

Statistical Parametric Speech Synthesis



Voice Conversion: TTS Motivation

- Concatenative speech synthesis
 - High-quality speech
 - But, need to record & process a large corpora for each voice
- Voice Conversion
 - Create different voices by speech-to-speech transformation
 - Focus on acoustics of voices

What gives a *voice* an *identity*?

- “Voice” → notion of identity (*voice* rather than *speech*)
- Characterize speech based on different levels
 1. Segmental
 - Pitch – fundamental frequency
 - ★ • Timbre – distinguishes between different types of sounds
 2. Supra-Segmental
 - Prosody – intonation & rhythm of speech

Goals of Voice Conversion

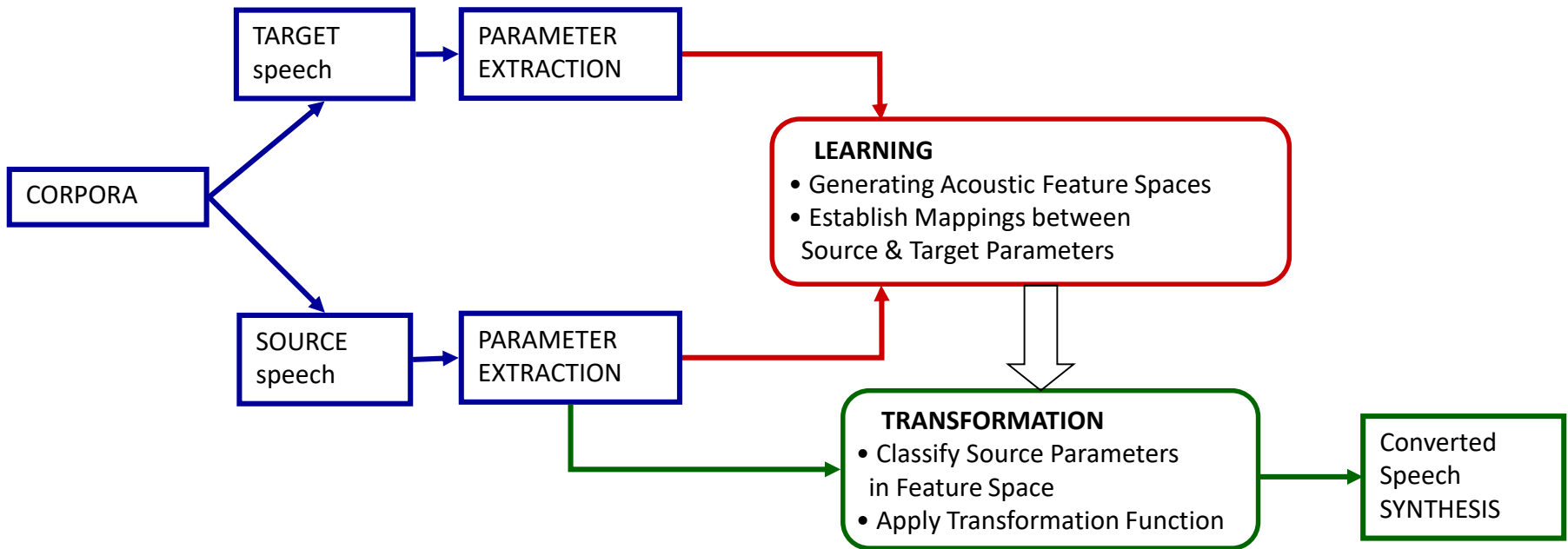
1. Synthesize High-Quality Speech
 - Maintain quality of source speech (limit degradations)
2. Capture Target Speaker Identity
 - Requires learning between source & target features

Difficult task!

- significant modifications of source speech needed that risk severely degrading speech quality...

Stages of Voice Conversion

1) Analysis, 2) Learning, 3) Transformation



- Key Parameters: the spectrum and prosody

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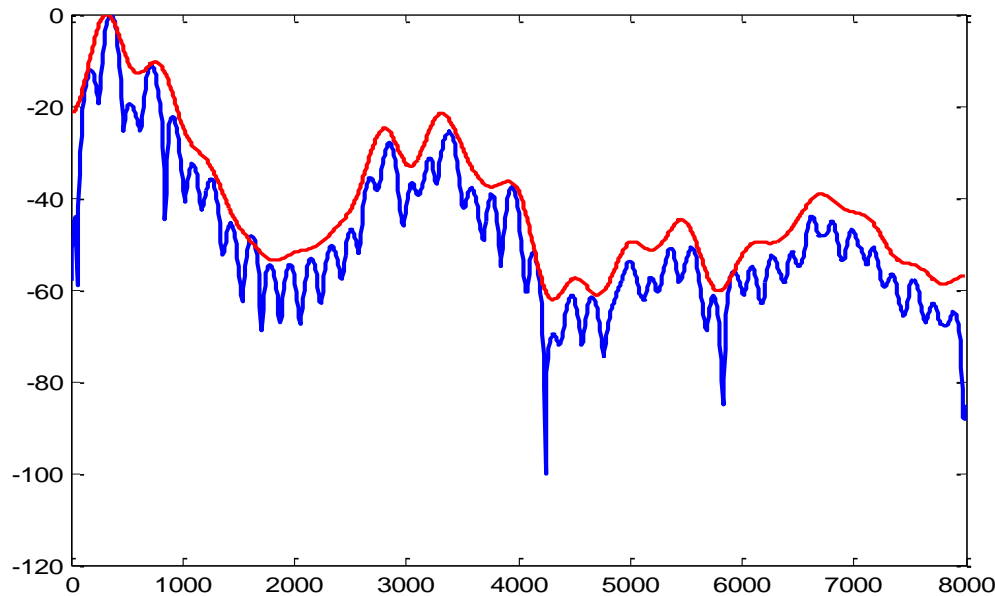


II. Spectrum Transformation in VC

- Gaussian Mixture Model

The Spectral Envelope

- Spectral Envelope: curve approximating the DFT magnitude



- Related to voice timbre, plays a key role in many speech applications:
 - Coding, Recognition, Synthesis, Voice transformation/conversion
- Voice Conversion: important for both speech quality and voice identity

Spectral Envelope Parameterization

- Two common methods

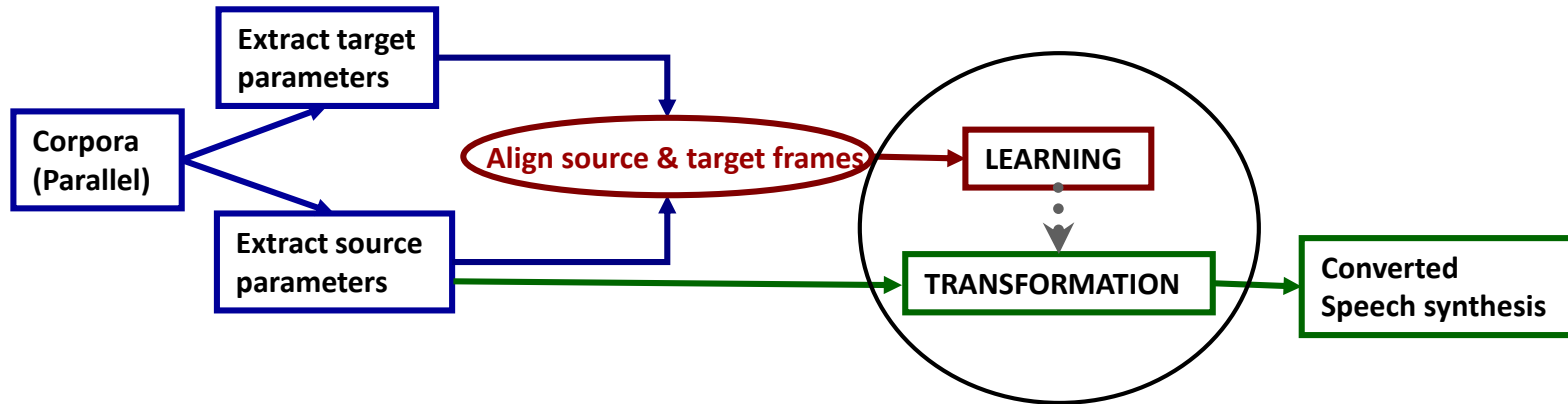
1) Cepstrum

- Discrete Cepstral Coefficients
- Mel-Frequency Cepstral Coefficients (MFCC)
 - change the frequency scale to reflect bands of human hearing

2) Linear Prediction (LP)

- Line Spectral Frequencies (LSF)

Standard Voice Conversion



Focus: Learning & Transforming the Spectral Envelope

- **Parallel corpora:** source & target utter same sentences
- Parameters are spectral features (e.g. vectors of cepstral coefficients)
- Alignment of speech frames in time

→ Standard: Gaussian Mixture Model

Voice Conversion

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★ II. Spectrum Transformation in VC

- Standard: Gaussian Mixture Model
 - Formulation
 - Limitations
 1. Acoustic mappings between source & target parameters
 2. Over-smoothing of the spectral envelope

GMM-based VC

1. Start form Minimum Mean Square Estimation (MMSE)
2. Time alignment
3. To derive the transfer function of GMM based VC.

Mean-Square Estimation(1/4)

- 如用一個 constant c 去 estimate RV \mathbf{y} , 以 MS estimation (i.e. , mean-square error 為最小之 estimation) 可如下推導

$$e = E \{ (\mathbf{y} - c)^2 \} = \int_{-\infty}^{\infty} (y - c)^2 f(y) dy$$

$$\frac{de}{dc} = - \int_{-\infty}^{\infty} 2(y - c) f(y) dy = 0$$

$$c = \int_{-\infty}^{\infty} y f(y) dy = E \{ \mathbf{y} \}$$

Mean-Square Estimation(2/4)

- 現在考慮 nonlinear MS estimation 由一個 RV \mathbf{x} 去估計另一個 RV \mathbf{y}

$$\begin{aligned} e &= E_{xy} \left\{ [\mathbf{y} - c(\mathbf{x})]^2 \right\} \\ &= \iint (y - c(x))^2 f(x, y) dx dy \\ &= \int f(x) \left[\int (y - c(x))^2 f(y|x) dy \right] dx \end{aligned}$$

$\because [\cdot]$ 為正， $f(x)$ 為正，所以只要 $[\cdot]$ 中之 $c(x)$ 使得 $[\cdot]$ 為最小 for every given x ，then e is minimum (i.e., 本來是 $\int f(x)[\cdot]dx$ 合起來考慮時要 minimum，但它等同於對每一 x ， $[\cdot]$ 皆 minimum 即可)

Mean-Square Estimation(3/4)

\therefore 要 minimum $[\cdot]$ for each given x , 而 $c(x)$ 為一 deterministic

(constant) when x is given , \therefore 由前面 case 和 $c(x) = E_y[\mathbf{y} | x]$,

再將 \mathbf{x} 可改變考慮進去 , 上式變為 $c(\mathbf{x}) = E_y[\mathbf{y} | \mathbf{x}]$

▫ 如 RVs \mathbf{y} 和 \mathbf{x} 為 independent , 則 $E_y[\mathbf{y} | \mathbf{x}] = E_y[\mathbf{y}] = \text{constant}$

Mean-Square Estimation(4/4)

1 mixture Gaussian, assume x_t and y_t are joint Gaussian, source x_t follow a Gaussian distribution.

By using MMSE, conversion function is

$$\hat{y}_t = F(x_t) = E[y_t | x_t] = \nu + \Gamma \Sigma_{xx}^{-1} (x_t - \mu_x)$$

where $\nu = \mu_y$, and $\Gamma = \Sigma_{xy}$

Gaussian Mixture Model (GMM) for VC

- **Origins:**
 - Evolved from "fuzzy" Vector Quantization (*i.e.* VQ with "soft" classification)
 - Originally proposed by [Stylianou et al; 98]
 - Joint learning of GMM (most common) by [Kain et al; 98]
- **Underlying principle:**
 - Exploit joint statistics exhibited by aligned source & target frames
- **Methodology:**
 - Represent distributions of spectral feature vectors as mix of Q Gaussians
 - Transformation function then based on MMSE criterion

Preliminaries of GMM

- We assume that the dataset X has been generated by a *parametric* distribution $p(X)$.
- Estimation of the parameters of p is known as *density estimation*.
- We consider Gaussian distribution.

Typical parameters (1)

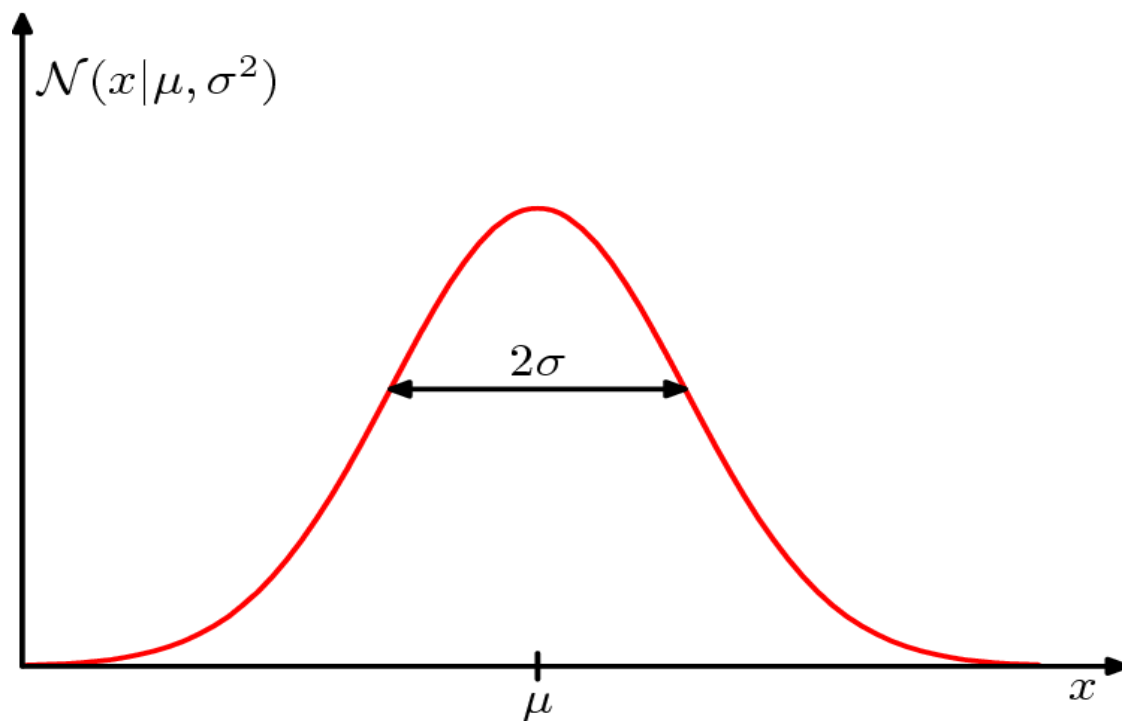
- *Mean* (μ): average value of $p(X)$, also called expectation.
- *Variance* (σ): provides a measure of variability in $p(X)$ around the mean.

Typical parameters (2)

- *Covariance*: measures how much two variables vary together.
- *Covariance matrix*: collection of covariances between all dimensions.
 - Diagonal of the covariance matrix contains the variances of each attribute.

One-dimensional Gaussian

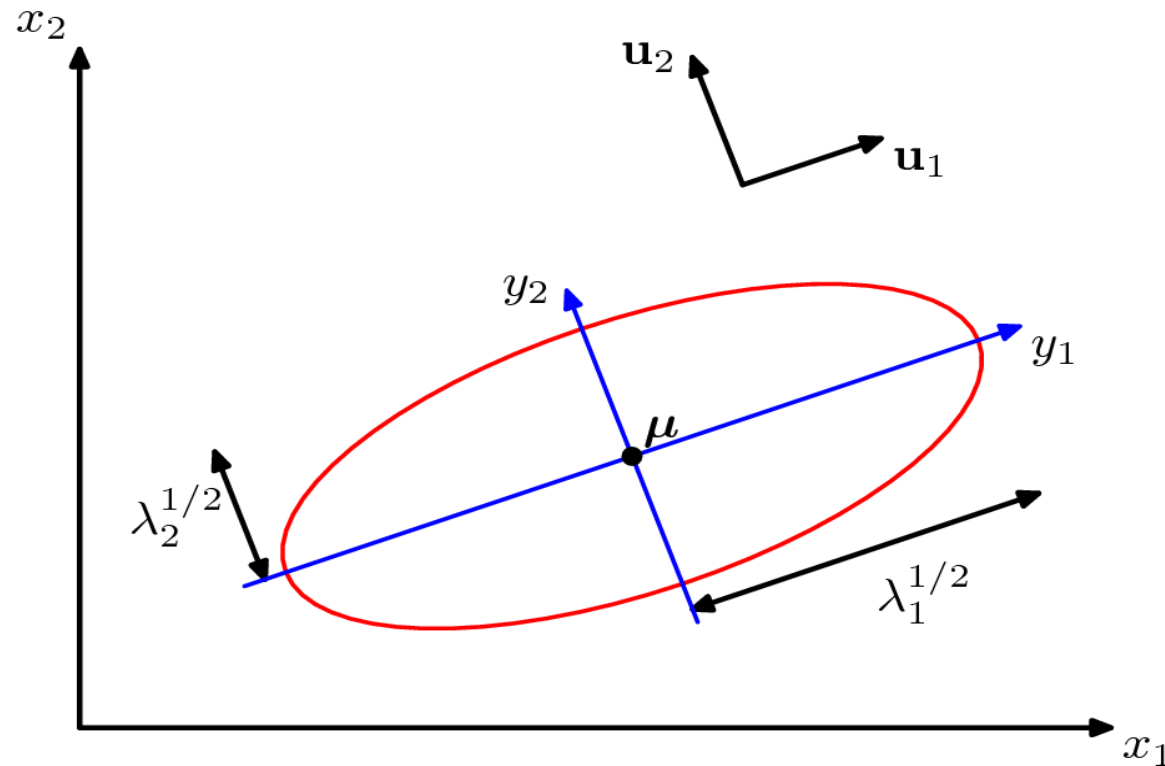
$$\text{Normal}(x | \mu, \sigma^2) = \frac{1}{\sqrt{2\pi\sigma^2}} \exp\left\{-\frac{1}{2\sigma^2}(x - \mu)^2\right\}$$



- Parameters to be estimated are the mean (μ) and variance (σ)

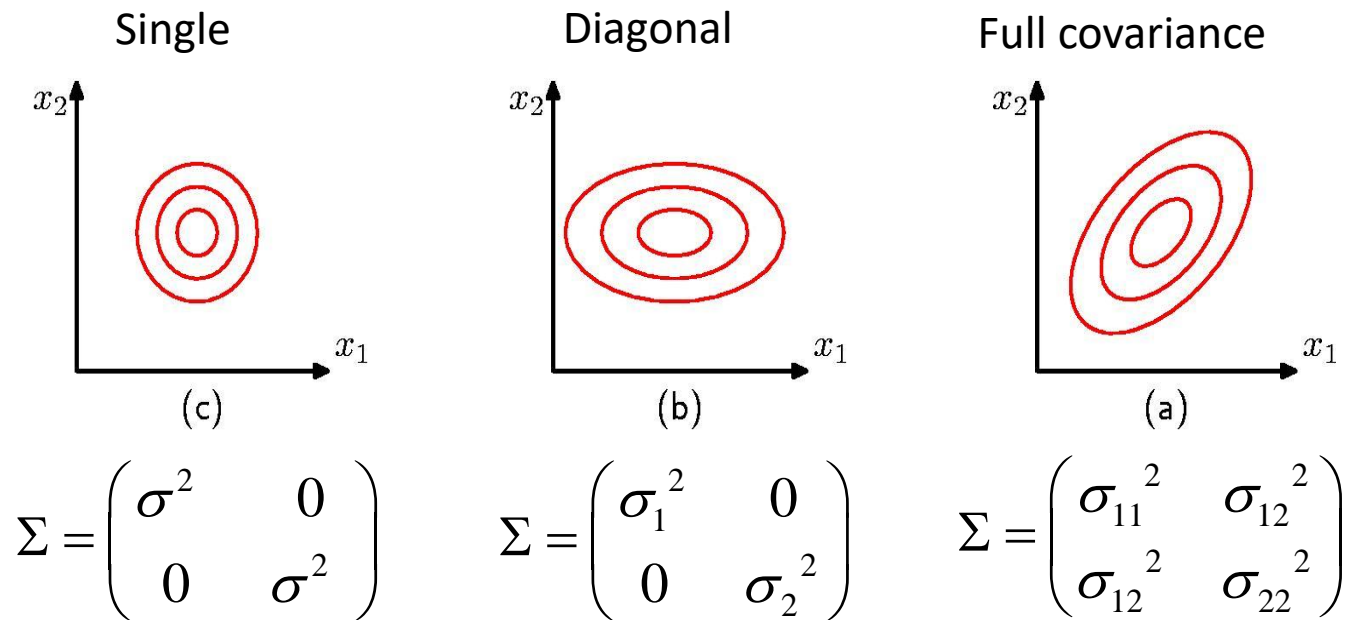
Multivariate Gaussian (1)

$$\text{Normal}(\mathbf{x} \mid \mu, \Sigma) = \frac{1}{(2\pi)^2} \frac{1}{\det(\Sigma)^{1/2}} \exp \left\{ -\frac{1}{2} (\mathbf{x} - \mu)^T \Sigma (\mathbf{x} - \mu) \right\}$$



- In multivariate case we have covariance matrix instead of variance

Multivariate Gaussian (2)



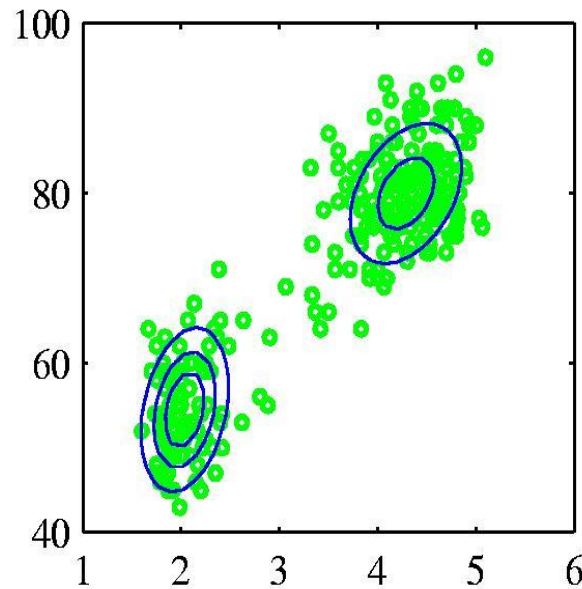
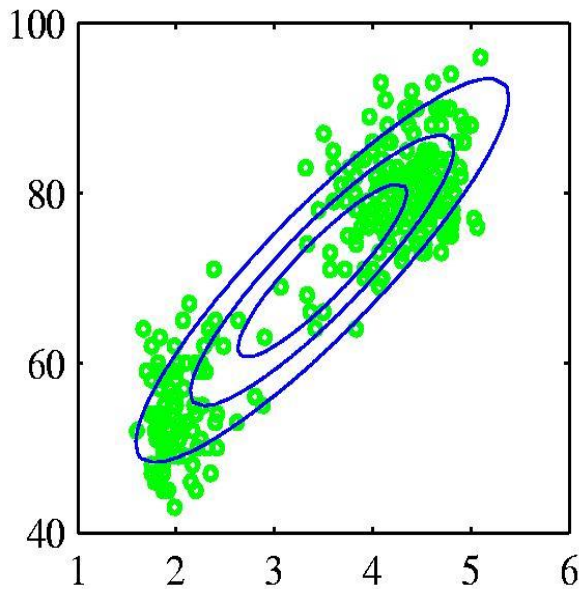
Complete data log likelihood:

$$\ln p(X) = \ln \prod_{n=1}^N \text{Normal}(\mathbf{x}_n \mid \mu, \Sigma)$$

Maximum Likelihood (ML) parameter estimation

- Maximize the log likelihood formulation
- Setting the gradient of the complete data log likelihood to zero we can find the closed form solution.
 - Which in the case of mean, is the sample average.

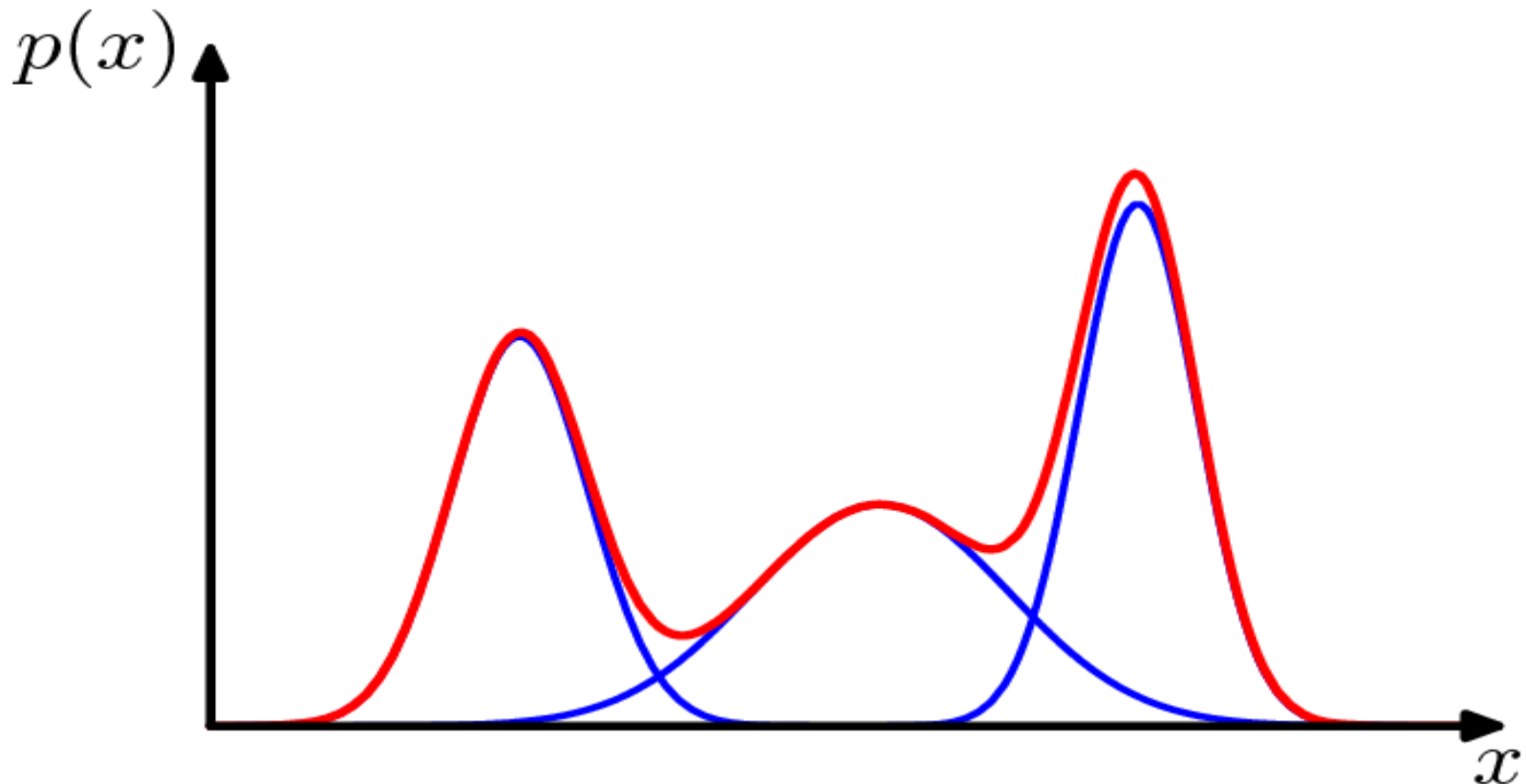
When one Gaussian is not enough



■ Real world datasets are rarely unimodal!

Mixtures of Gaussians

$$p(\mathbf{x}) = \sum_{k=1}^M \pi_k \text{Normal}(\mathbf{x} \mid \mu_k, \Sigma_k)$$



Mixtures of Gaussians (2)

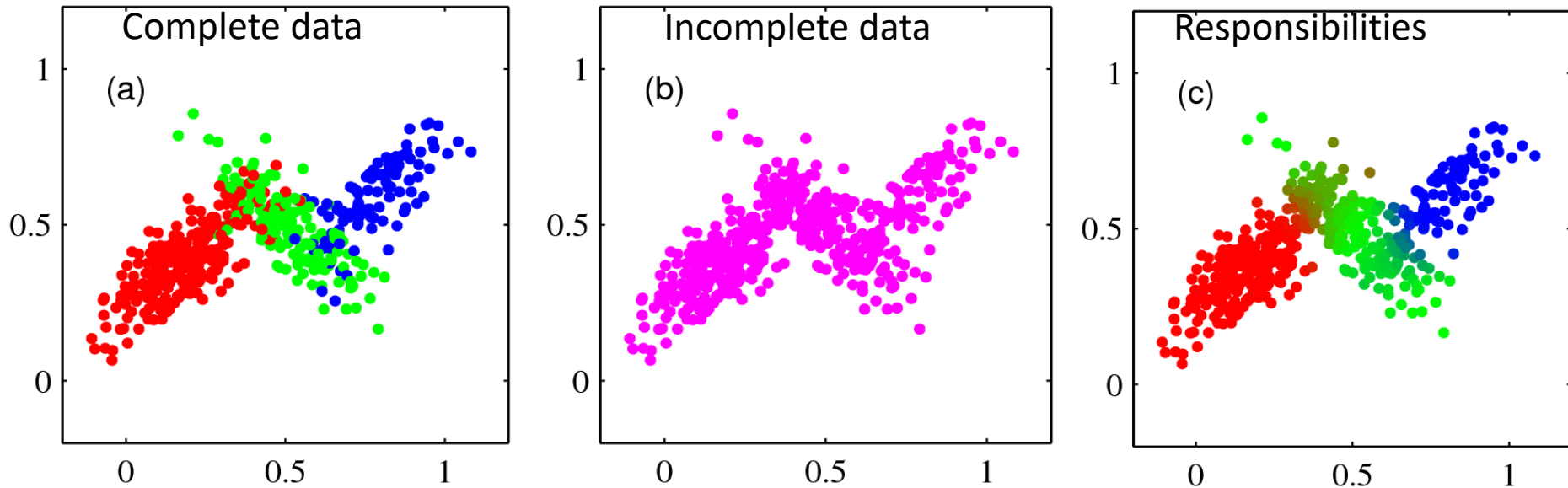
- In addition to mean and covariance parameters (now M times), we have mixing coefficients π_k .

Following properties hold for the mixing coefficients:

$$\sum_{k=1}^M \pi_k = 1 \qquad 0 \leq \pi_k \leq 1$$

It can be seen as the prior probability of the component k

Responsibilities (1)



- Component labels (red, green and blue) cannot be observed.
- We have to calculate approximations (responsibilities).

Responsibilities (2)

- Responsibility describes, how probably observation vector \mathbf{x} is from component k .
- In clustering, responsibilities take values 0 and 1, and thus, it defines the hard partitioning.

Responsibilities (3)

■ We can express the marginal density $p(\mathbf{x})$ as:

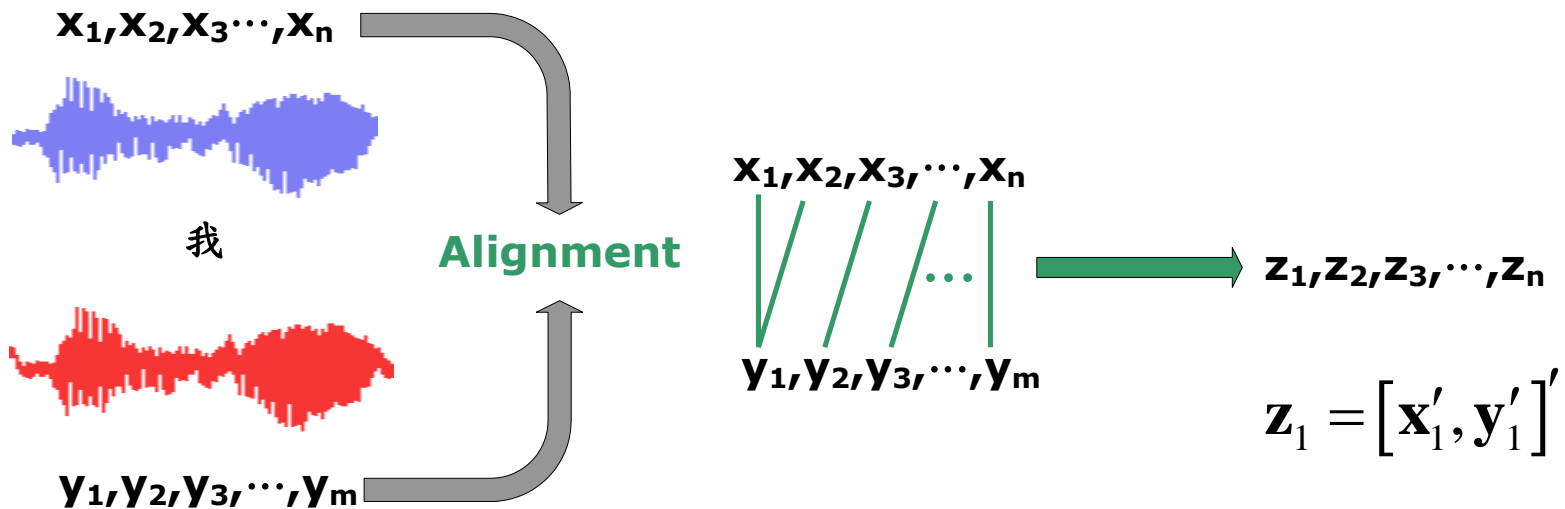
$$p(\mathbf{x}) = \sum_{k=1}^M p(k) p(\mathbf{x} | k)$$

■ From this, we can find the responsibility of the k^{th} component of \mathbf{x} using Bayesian theorem:

$$\begin{aligned} \gamma_k(\mathbf{x}) &= p(k | \mathbf{x}) \\ &= \frac{p(\mathbf{x}) p(k)}{\sum_l p(l) p(\mathbf{x} | l)} \\ &= \frac{\pi_k \text{Normal}(\mathbf{x} | \mu_k, \Sigma_k)}{\sum_l \pi_l \text{Normal}(\mathbf{x} | \mu_l, \Sigma_l)} \end{aligned}$$

GMM-based Method

- Vector sequence of source speech $\mathbf{X} = \{\mathbf{x}_1, \mathbf{x}_2, \dots, \mathbf{x}_n\}$
- Vector sequence of target speech $\mathbf{Y} = \{\mathbf{y}_1, \mathbf{y}_2, \dots, \mathbf{y}_m\}$



- Vector sequence of aligned source-target speech $\mathbf{Z} = \{\mathbf{z}_1, \mathbf{z}_2, \dots, \mathbf{z}_n\}$

GMM-based Method

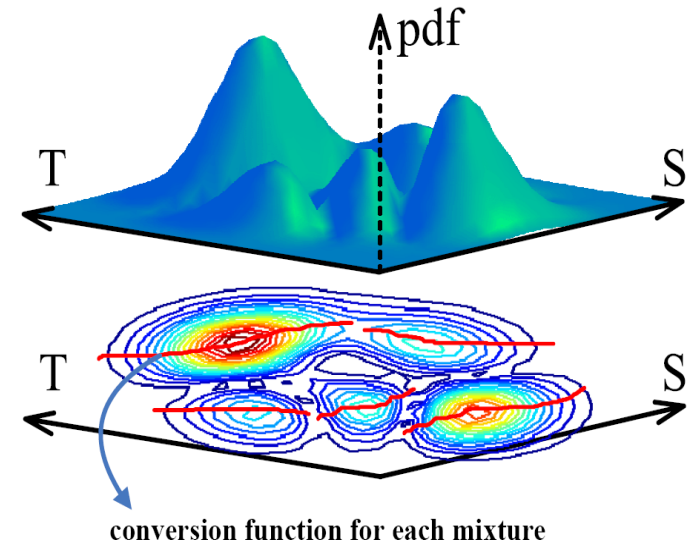
- Conversion function for a mixture

$$p(\mathbf{y} | \mathbf{x}) = \frac{1}{(2\pi)^{d/2} \det^{1/2} \left(\Sigma^{\mathbf{Y}\mathbf{Y}} - \Sigma^{\mathbf{Y}\mathbf{X}} (\Sigma^{\mathbf{X}\mathbf{X}})^{-1} \Sigma^{\mathbf{X}\mathbf{Y}} \right)} \exp \left(-\frac{1}{2} \mathbf{U} \right)$$

$$\mathbf{U} = \left(\mathbf{y} - \left(\boldsymbol{\mu}^{\mathbf{Y}} + \Sigma^{\mathbf{Y}\mathbf{X}} (\Sigma^{\mathbf{X}\mathbf{X}})^{-1} (\mathbf{x} - \boldsymbol{\mu}^{\mathbf{X}}) \right) \right)'$$

$$\left[\Sigma^{\mathbf{Y}\mathbf{Y}} - \Sigma^{\mathbf{Y}\mathbf{X}} (\Sigma^{\mathbf{X}\mathbf{X}})^{-1} \Sigma^{\mathbf{X}\mathbf{Y}} \right]^{-1} \left(\mathbf{y} - \left(\boldsymbol{\mu}^{\mathbf{Y}} + \Sigma^{\mathbf{Y}\mathbf{X}} (\Sigma^{\mathbf{X}\mathbf{X}})^{-1} (\mathbf{x} - \boldsymbol{\mu}^{\mathbf{X}}) \right) \right)$$

- GMM-based conversion function



$$\tilde{\mathbf{y}}_t = F(\mathbf{x}_t) = E[\mathbf{y}_t | \mathbf{x}_t] = \sum_{m=1}^M p(m | \mathbf{x}_t) \left[\boldsymbol{\mu}_m^{\mathbf{Y}} + \Sigma_m^{\mathbf{Y}\mathbf{X}} (\Sigma_m^{\mathbf{X}\mathbf{X}})^{-1} (\mathbf{x}_t - \boldsymbol{\mu}_m^{\mathbf{X}}) \right]$$

- Posterior probability

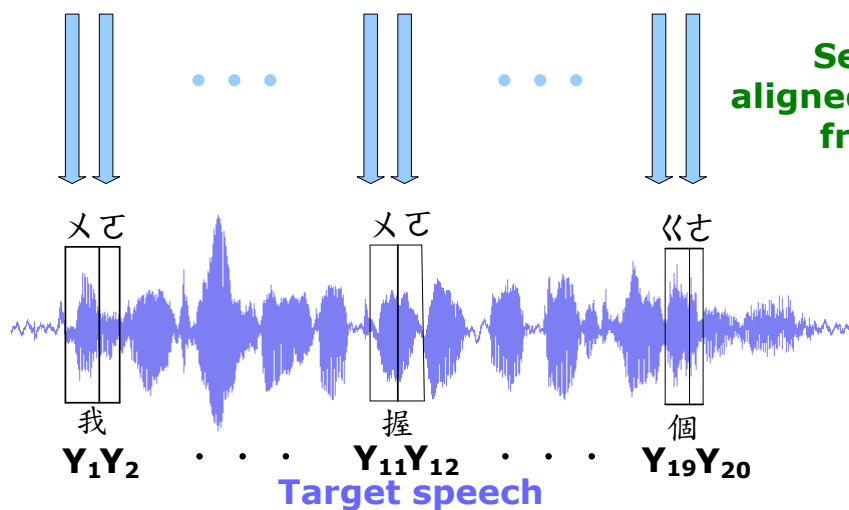
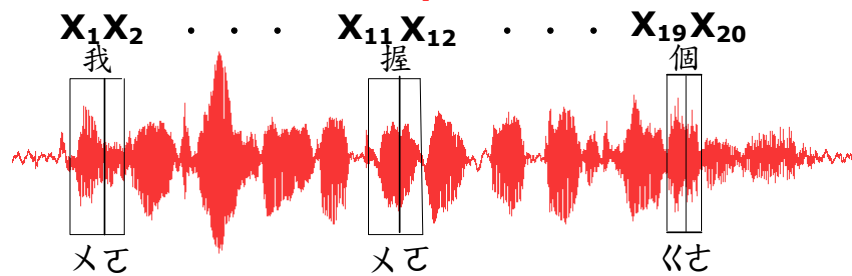
$$p(m | \mathbf{x}_t) = \frac{w_m N(\mathbf{x}_t; \boldsymbol{\mu}_m^{\mathbf{X}}, \Sigma_m^{\mathbf{X}\mathbf{X}})}{\sum_{k=1}^M w_k N(\mathbf{x}_t; \boldsymbol{\mu}_k^{\mathbf{X}}, \Sigma_k^{\mathbf{X}\mathbf{X}})}$$

GMM-based Method

- One conversion function for **all sub-syllable**

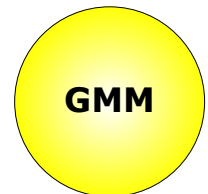
我要好好把握今天這個機會

Source speech



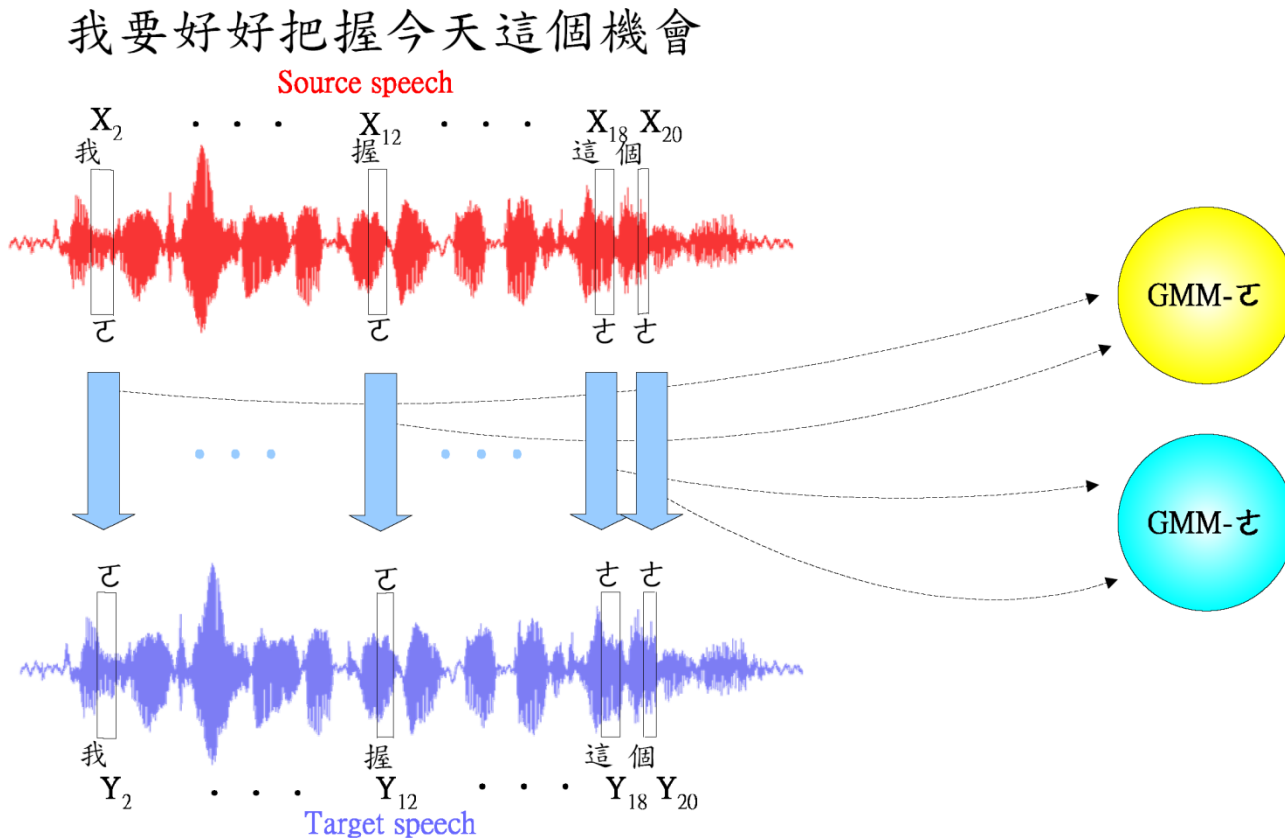
Sequence of
aligned source-target
frame pairs

Training



GMM-based Method

- One conversion function for **each sub-syllable**
 - 38 context independent final
 - 112 right context dependent initial



GMM-based Spectral Transformation

- 1) Align N spectral feature vectors in time. (discrete cepstral coeffs)

$$\text{source : } X = \{x_1, \dots, x_N\}, \quad \text{target : } Y = \{y_1, \dots, y_N\}, \quad \text{joint : } Z = (X, Y)$$

- 2) Represent PDF of vectors as mixture of Q multivariate Gaussians

$$p(z) = \sum_{q=1}^Q \alpha_q N(z; \mu_q, \Sigma_q), \quad \sum_{q=1}^Q \alpha_q = 1, \quad \alpha_q \geq 0$$

Learn $\{\alpha_q, \mu_q, \Sigma_q, q=1:Q\}$ from Expectation Maximization (EM) on Z

- 3) Transform source vectors using weighted mixture of Maximum Likelihood (ML) estimator for each component.

$$\hat{y}_n(x_n) = \sum_{q=1}^Q w_q^x(x_n) \left[\mu_q^y - \Sigma_q^{yx} \left(\Sigma_q^{xx} \right)^{-1} (x_n - \mu_q^x) \right]$$

$w_q^x(x_n)$: probability source frame belongs to acoustic class described by component q (calculated in Decoding)

GMM-Transformation Steps

1) Source frame $x_n \rightarrow$ want to estimate target vector: \hat{y}_n












2) Classify $x_n \rightarrow$ calculate $w_q^x(x_n)$

$w_q^x(x_n)$: probability source frame belongs to acoustic class described by component q (Decoding step)

3) Apply transformation function:

$$\hat{y}_n(x_n) = \underbrace{\sum_{q=1}^Q w_q^x(x_n)}_{\text{weighted sum}} \underbrace{\left[\mu_q^y - \Sigma_q^{yx} \left(\Sigma_q^{xx} \right)^{-1} (x_n - \mu_q^x) \right]}_{\text{ML estimator for class}}$$

Conversion Examples

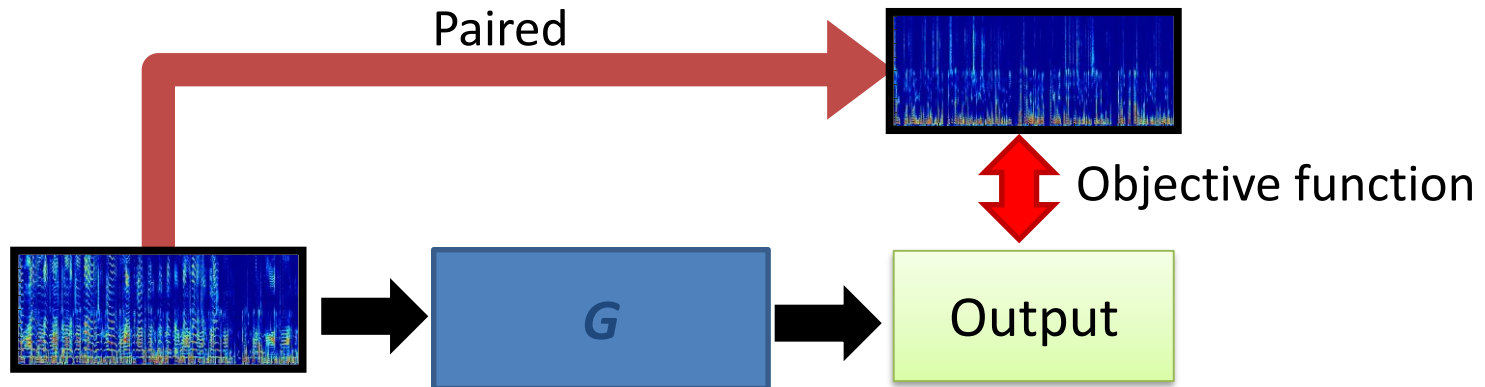
	Source	Target	GMM	DFWA	DFWE
slt → clb (FF)					
bdl → clb (MF)			 	 	



Target analysis-synthesis with
converted spectral envelopes

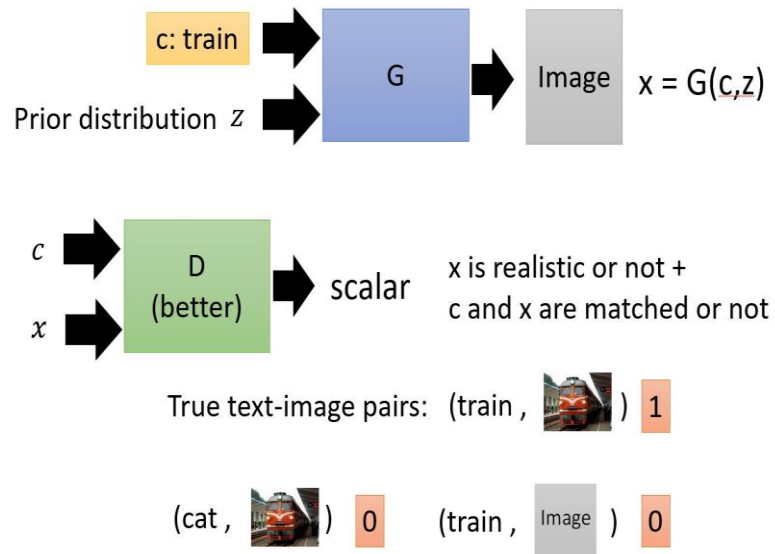
- GMM-based suffer “loss of presence”

Speech Signal Generation (Regression Task)

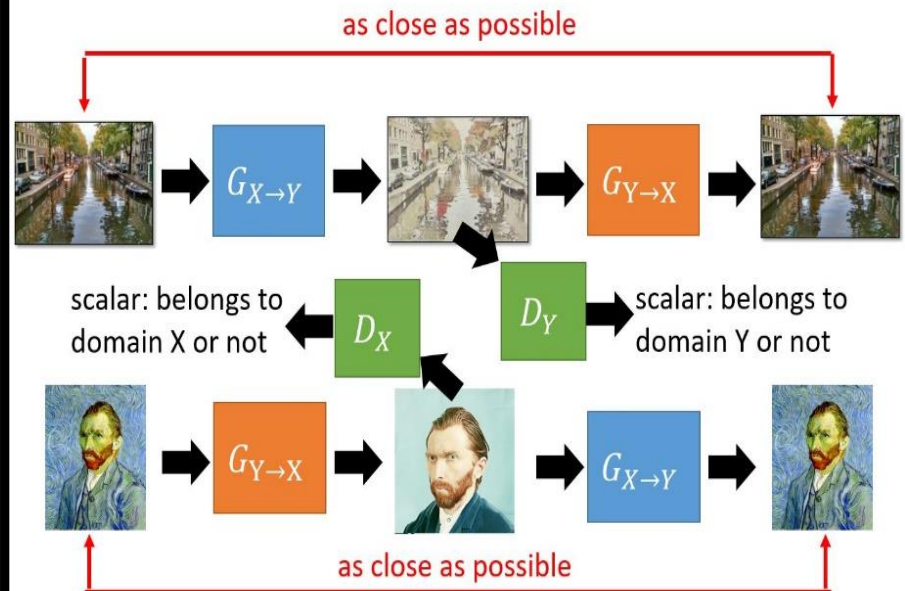


[Scott Reed, et al, ICML, 2016]

Conditional GAN

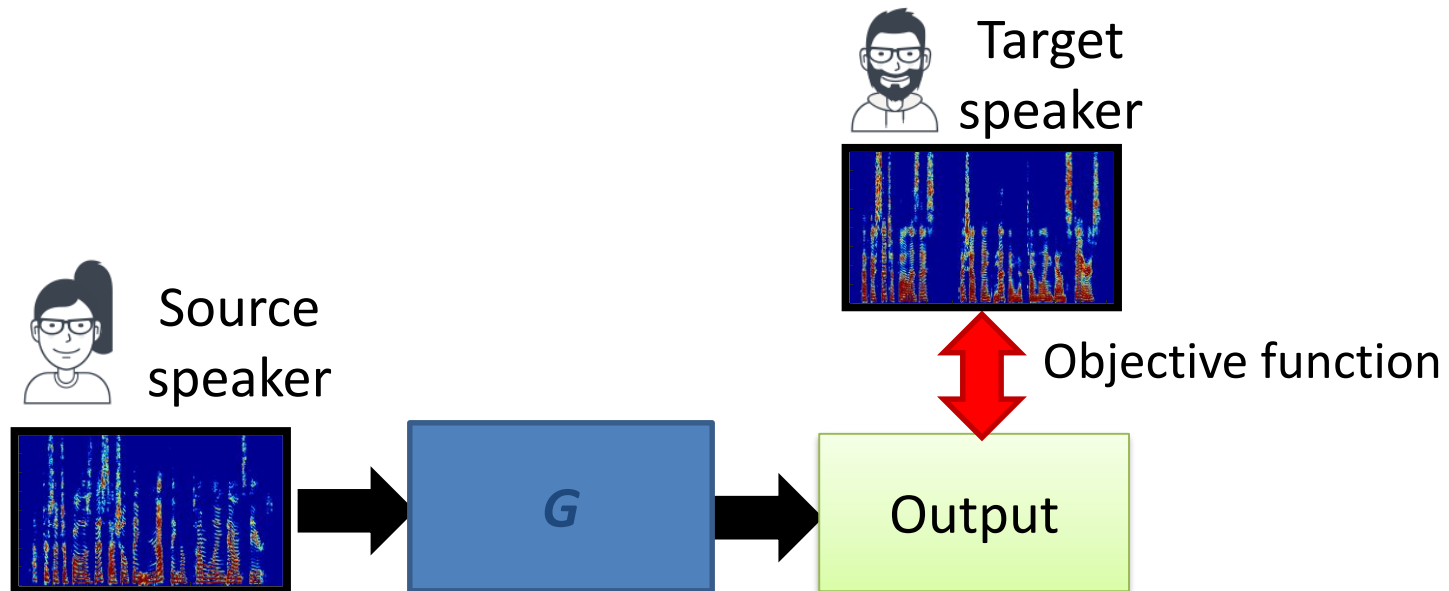


Cycle-GAN



Voice Conversion

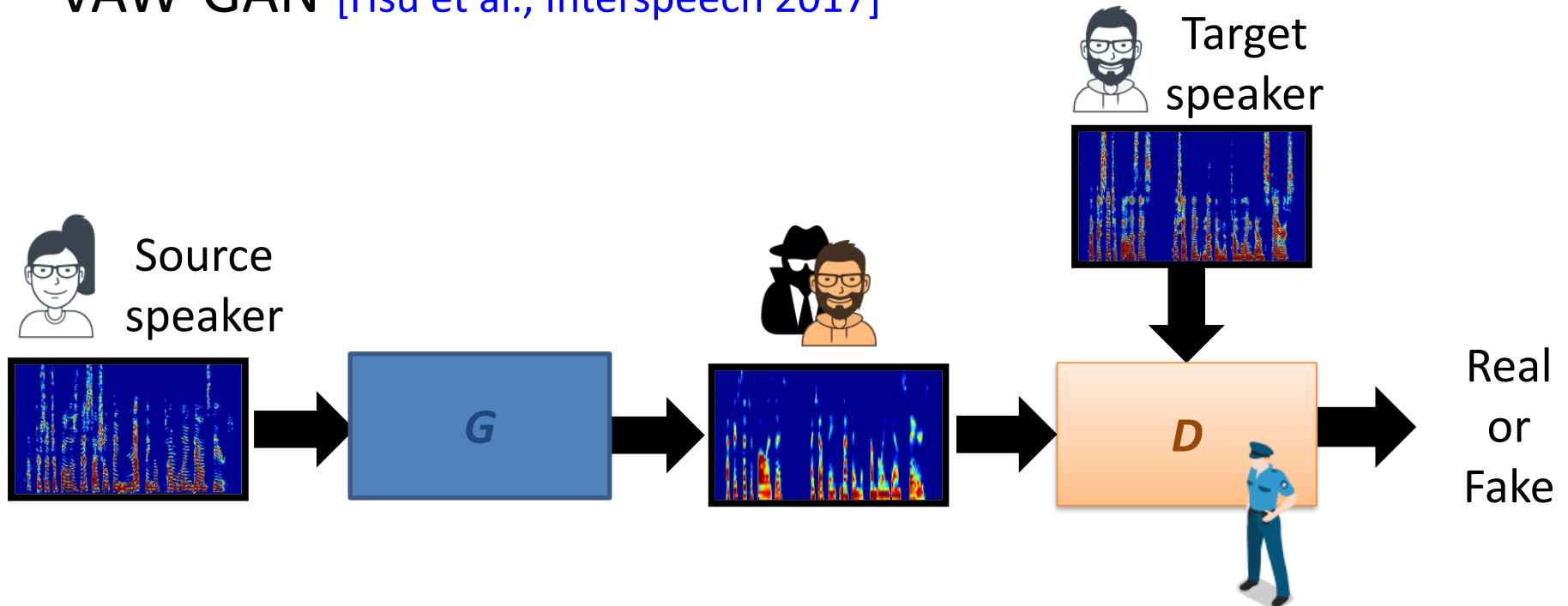
- Convert (transform) speech from source to target



- Conventional VC approaches include Gaussian mixture model (GMM) [Toda et al., TASLP 2007], non-negative matrix factorization (NMF) [Wu et al., TASLP 2014; Fu et al., TBME 2017], locally linear embedding (LLE) [Wu et al., Interspeech 2016], restricted Boltzmann machine (RBM) [Chen et al., TASLP 2014], feed forward NN [Desai et al., TASLP 2010], recurrent NN (RNN) [Nakashika et al., Interspeech 2014].

Voice Conversion

- VAW-GAN [Hsu et al., Interspeech 2017]



- Conventional MMSE approaches often encounter the “over-smoothing” issue.
- GAN is used a new objective function to estimate G .
- The goal is to increase the naturalness, clarity, similarity of converted speech.

$$V(G, D) = V_{GAN}(G, D) + \lambda V_{VAE}(x|y)$$

Voice Conversion (VAW-GAN)

- Objective and subjective evaluations

Fig. 14: The spectral envelopes.

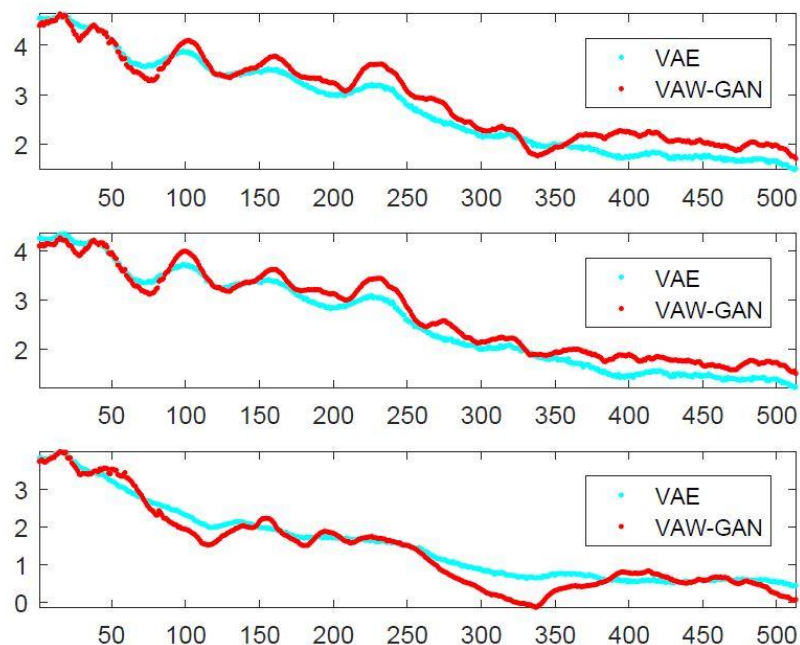


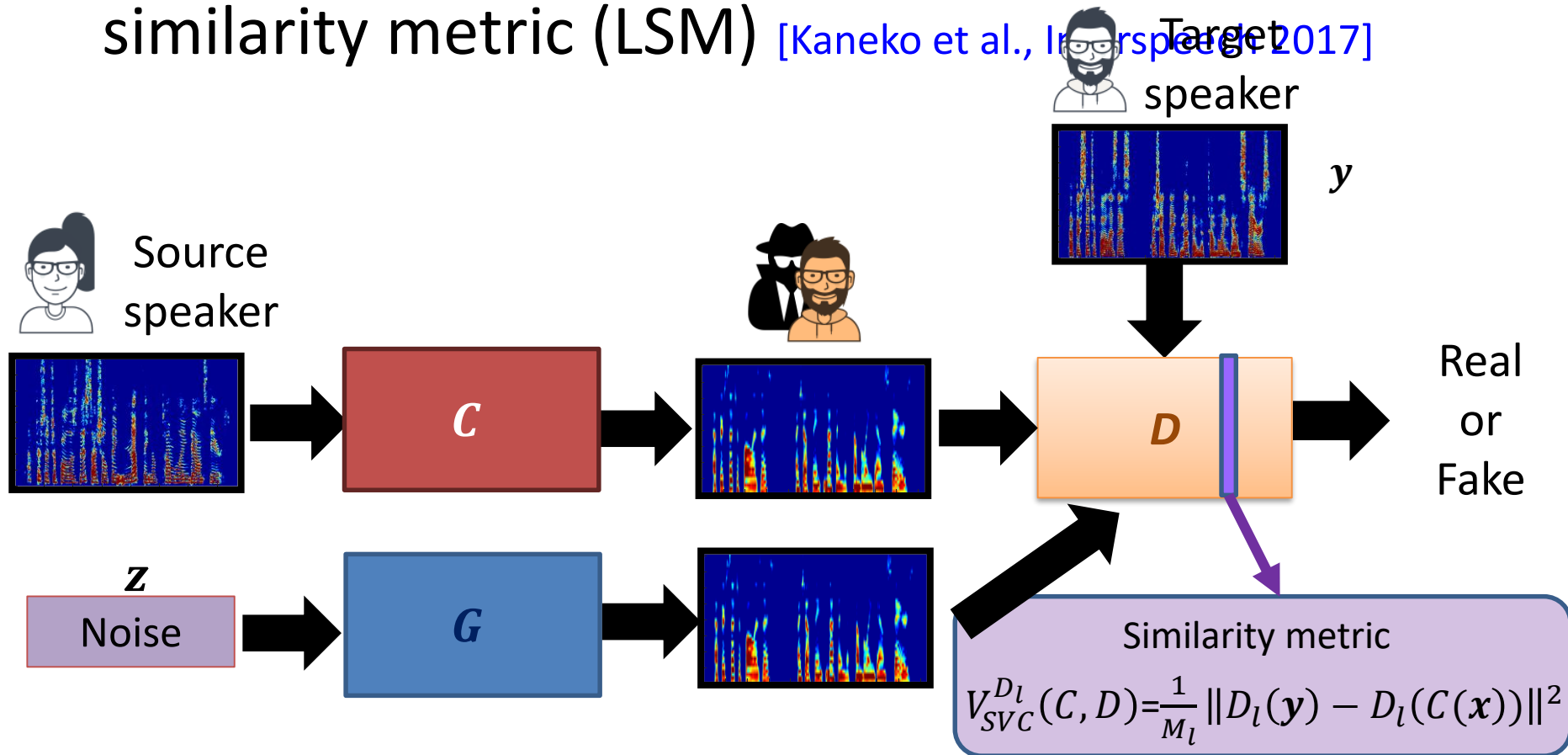
Fig. 15: MOS on naturalness.



VAW-GAN outperforms VAE in terms of objective and subjective evaluations with generating more structured speech.

Voice Conversion

- Sequence-to-sequence VC with learned similarity metric (LSM) [Kaneko et al., Interspeech 2017]

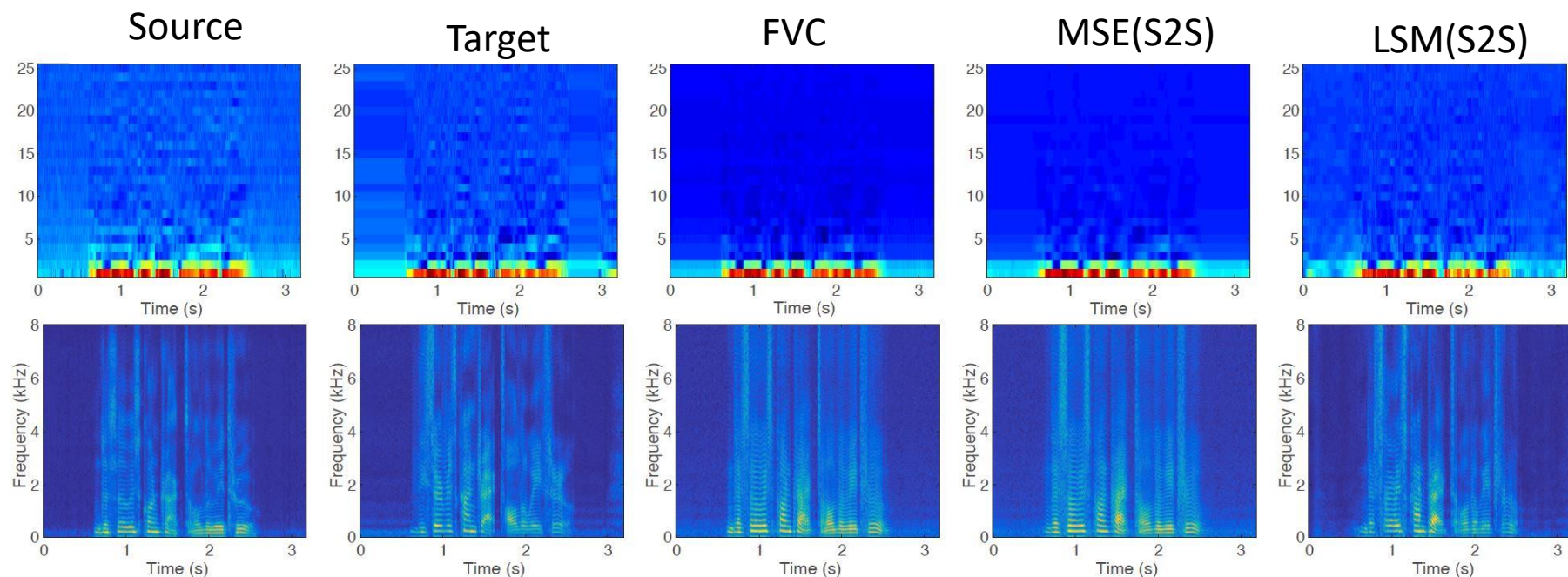


$$V(C, G, D) = V_{SVC}^{D_l}(C, D) + V_{GAN}(C, G, D)$$

Voice Conversion (LSM)

- Spectrogram analysis

Fig. 16: Comparison of MCCs (upper) and STFT spectrograms (lower).

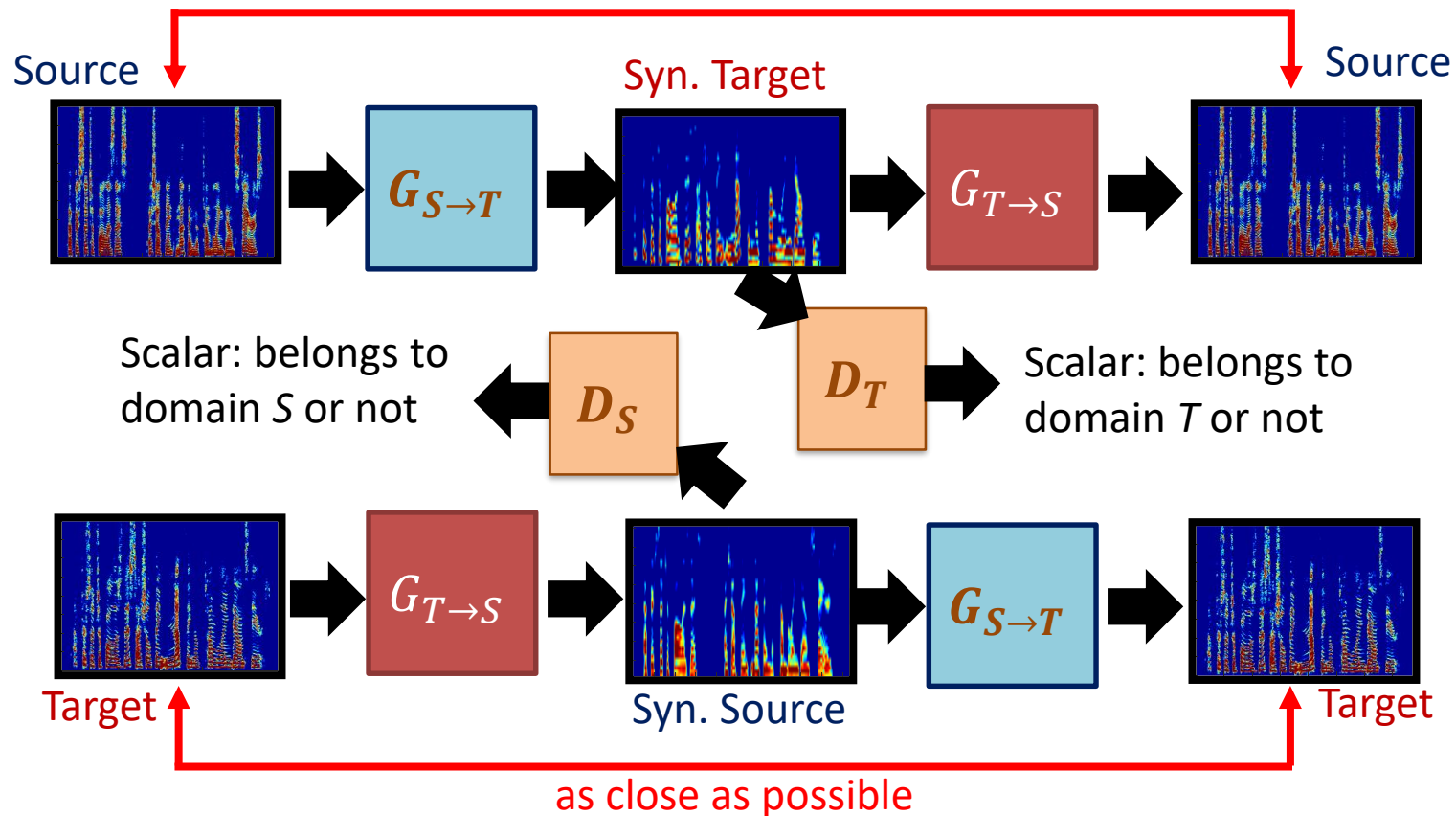


The spectral textures of LSM are more similar to the target ones.

Voice Conversion

- CycleGAN-VC [Kaneko et al., arXiv 2017]

as close as possible

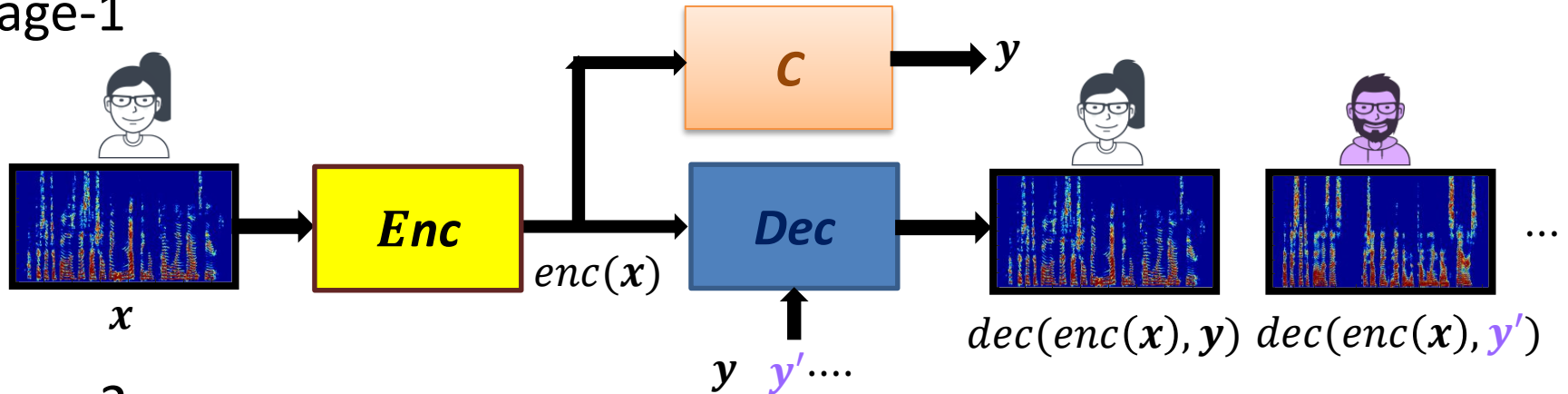


$$V_{Full} = V_{GAN}(G_{X \rightarrow Y}, D_Y) + V_{GAN}(G_{Y \rightarrow X}, D_X) + \lambda V_{Cyc}(G_{X \rightarrow Y}, G_{Y \rightarrow X})$$

Voice Conversion

- Multi-target VC [Chou et al., arxiv 2018]

➤ Stage-1



➤ Stage-2

