Speech Synthesis



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Lecture 19



Project Preliminary Report

- 27th October, 2019.

 - precisely how it applies to your problem. 5 points
 - parts you will rely on existing implementations.

• Preliminary project report will contribute towards 5% of your final grade. Deadline is on

• Define the following for your project: 1) Input-output behaviour of your system 5 points 2) Evaluation metric 3) At least two existing (or related) approaches to your problem

• Propose a model and an algorithm for the problem you're tackling and give detailed descriptions for both. Do not provide generic descriptions of the model. Describe

• Describe how much of your algorithm has been implemented. If you are using existing APIs/libraries, clearly demarcate which parts you will be implementing and for which **5** points

• Describe the experiments you are planning to run. If you have already run any 5 points preliminary experiments, please describe them along with reporting your initial results.





Text-To-Speech (TTS) Systems Storied History



- Von Kempelen's speaking machine (1791)
 - Bellows simulated the lungs

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- Rubber mouth and nose; nostrils had to be covered with two fingers for non-nasals
- Homer Dudley's VODER (1939)
 - First device to synthesize speech sounds via electrical means
- Gunnar Fant's OVE formant synthesizer (1960) Formant synthesizer for vowels ۲
- Computer-aided speech synthesis (1970s)
 - Concatenative (unit selection) •
 - Parametric (HMM-based and NN-based)





Loud speake andom noise Relaxation ascillator Voiced snergy switch





All images from http://www2.ling.su.se/staff/hartmut/kemplne.htm

Speech synthesis or TTS systems

- quality speech waveform for a given word sequence
- TTS systems are typically divided into two parts: •
 - A. Linguistic specification
 - B. Waveform generation

• Goal of a TTS system: Produce a natural-sounding high-

- Constructed using a large amount of speech data
- Referred to as corpus-based TTS systems
- Two prominent instances of corpus-based TTS:
 - 1. Unit selection and concatenation
 - 2. Statistical parametric speech synthesis

Current TTS systems

Unit Selection Synthesis

Unit selection synthesis or Concatenative speech synthesis

- Synthesize new sentences by selecting sub-word units from a database of speech
 - Optimal size of units?
 Diphones?
 Half-phones?

All segments



Unit selection synthesis

• Target cost between a candidate, u_i , and a target unit t_i :

$$C^{(t)}(t_i, u_i) =$$

Concatenation cost between candidate units:

$$C^{(c)}(u_{i-1}, u_i) = \sum_{k=1}^{q} w_k^{(c)} C_k^{(c)}(u_{i-1}, u_i),$$

• Find string of units that minimises the overall cost:

$$\hat{u}_{1:n} = \arg\min_{u_{1:n}} \left\{ C(t_{1:n}, u_{1:n}) \right\}$$

$$C(t_{1:n}, u_{1:n}) = \sum_{i=1}^{n} C^{(t)}(t_i, u_i) + \sum_{i=2}^{n} C^{(c)}(u_{i-1}, u_i)$$

$$\sum_{j=1}^{p} w_j^{(t)} C_j^{(t)}(t_i, u_i),$$

Unit selection synthesis



Target cost is pre-calculated using a clustering method

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Statistical Parametric Speech Synthesis

Parametric Speech Synthesis Framework



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Estimate acoustic model given speech utterances (O), word sequences (W)*

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\hat{\lambda} = \arg \max_{\lambda} p(O|W, \lambda)
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* Here W could refer to any textual features relevant to the input text



Parametric Speech Synthesis Framework



Synthesize speech from ô

۲

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HMM-based speech synthesis



Speech parameter generation

 $\hat{o} = \arg\max_{o} p(o|w, \hat{\lambda})$

 $\hat{q} = a \mathbf{i}$

 $\hat{o} = a$ Let's explore this first

- Generate the most probable observation vectors given the HMM and w:
 - $= \arg\max_{o} \sum_{\forall q} p(o, q | w, \hat{\lambda})$ $\approx \arg \max_{o} \max_{q} p(o, q | w, \hat{\lambda})$ $= \arg\max_{o} \max_{q} p(o|q, \hat{\lambda}) p(q|w, \hat{\lambda})$
- Determine the best state sequence and outputs sequentially:

$$\operatorname{rg\,max}_{q} p(q|w, \hat{\lambda})$$
$$\operatorname{rg\,max}_{o} p(o|\hat{q}, \hat{\lambda})$$

Determining state outputs

 $= \arg m a$

where $\boldsymbol{o} = [\boldsymbol{o}_1^{\top}, \dots, \boldsymbol{o}_T^{\top}]^{\top}$ is a state-output vector sequence to be generated, $q = \{q_1, \ldots, q_T\}$ is a state sequence, and $\mu_{q} = [\mu_{q_1}^{\top}, \dots, \mu_{q_T}^{\top}]^{\top}$ is the mean vector for q.



Mean -

 $\hat{o} = \arg \max p(o|\hat{q}, \hat{\lambda})$

$$\operatorname{ax} \mathcal{N}(o; \mu_{\hat{q}}, \Sigma_{\hat{q}})$$



Variance

Adding dynamic features to state outputs

$$oldsymbol{o}_t = egin{bmatrix} oldsymbol{c}_t^ op, \Delta oldsymbol{c}_t^ op \end{bmatrix}^ op$$

State output vectors contain both static (c_t) and dynamic (Δc_t) features



where
$$\Delta oldsymbol{c}_t = oldsymbol{c}_t - oldsymbol{c}_{t-1}$$

o and c arranged in matrix form

Speech parameter generation

Introducing dynamic feature constraints: •

If the output distributions are single Gaussians: •

• Then, by setting $\partial \log \mathcal{N}(o; \mu_{\hat{q}}, \Sigma_{\hat{q}}) / \partial c = 0$

 $\hat{o} = \arg \max p(o|\hat{q}, \hat{\lambda})$ where o = Wc

- $p(o|\hat{q}, \hat{\lambda}) = \mathcal{N}(o; \mu_{\hat{q}}, \Sigma_{\hat{q}})$
 - we get:
- $W^T \Sigma_{\hat{a}}^{-1} W c = W^T \Sigma_{\hat{a}}^{-1} \mu_{\hat{q}}$

Synthesis overview





Speech parameter generation

 $\hat{o} = \arg \max$ 0 $= \arg \max$ 0 $\approx \arg \max$ 0 $= \arg \max$ 0

Let's explore this next

 $\hat{q} = a$

 $\hat{o} = a$

Generate the most probable observation vectors given the HMM and w:

$$\begin{aligned} &= p(o|w, \hat{\lambda}) \\ &= \sum_{\forall q} p(o, q|w, \hat{\lambda}) \\ &= \max_{q} p(o, q|w, \hat{\lambda}) \\ &= \max_{q} p(o|q, \hat{\lambda}) P(q|w, \hat{\lambda}) \end{aligned}$$

Determine the best state sequence and outputs sequentially:

$$\mathop{\mathrm{rg\,max}}_{q} p(q|w,\hat{\lambda})$$

 $\mathop{\mathrm{rg\,max}}_{o} p(o|\hat{q},\hat{\lambda})$



Implicitly modelled by state self-transition probabilities

$$p_k(d) = a_{kk}^{d-1} \cdot (1 - a_{kk})$$

PMFs of state durations are geometric distributions

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 $\log P(\boldsymbol{d} \mid$

• geometric distributions?



State durations are determined by maximising:

$$\lambda) = \sum_{j=1}^{N} \log p_j(d_j),$$

What would this solution look like if the PMFs of state durations are

Explicit modeling of state durations

• Each state duration is explicitly modelled as a single density of state i:



where

$$\chi_{t_0,t_1}(i) = (1 - \gamma_{t_0-1}(i)) \cdot \prod_{t=t_0}^{t_1} \gamma_t(i) \cdot (1 - \gamma_{t_1+1}(i)),$$

and $\gamma_t(i)$ is the probability of being in state i at time t

Gaussian. The mean $\xi(i)$ and variance $\sigma^2(i)$ of duration

$$\sum_{t_0}^{T} \chi_{t_0,t_1}(i)(t_1 - t_0 + 1),$$

$$\sum_{t_0}^{T} \sum_{t_1=t_0}^{T} \chi_{t_0,t_1}(i),$$

$$\sum_{t_0}^{T} \chi_{t_0,t_1}(i)(t_1 - t_0 + 1)^2,$$

$$\sum_{t_0}^{T} \sum_{t_1=t_0}^{T} \chi_{t_0,t_1}(i),$$

$$\sum_{t_0}^{T} \sum_{t_1=t_0}^{T} \chi_{t_0,t_1}(i),$$

Determining state durations

maximize:

 $\log P(\mathbf{d}|\lambda, T)$

Gaussian $\mathcal{N}(\cdot;\xi_k,\sigma_k^2)$



During synthesis, for a given speech length T, the goal is to

$$\Gamma) = \sum_{k=1}^{K} \log p_k(d_k) \qquad \dots (1)$$

under the constraint that $T = \sum d_k$ k=1

We saw that each duration density $p_k(d_k)$ can be modelled as a single

State durations, d_k , $k = 1 \dots K$, which maximise (1) are given by:

$$+ \rho \cdot \sigma^{2}(k) \\ + \sum_{k=1}^{K} \xi(k) \right) / \sum_{k=1}^{K} \sigma^{2}(k)$$

Synthesis using duration models

Context Dependent Duration Models

Context Dependent HMMs



Synthesis **State Duration** Densities Sentence

HMM

State Duration

Mel-Cepstrum





Image from Yoshimura et al., "Duration modelling for HMM-based speech synthesis", ICSLP '98

Recap: HMM-based speech synthesis



DNN-based speech synthesis



Image from Zen et al., "Statistical Parametric Speech Synthesis using DNNs", 2014





- Input features about linguistic contexts, numeric values (# of words, duration of the phoneme, etc.)
- Output features are spectral and excitation parameters and their delta values
- Listening test results

Table 1. Preference scores (%) between speech samples from the HMM and DNN-based systems. The systems which achieved significantly better preference at p < 0.01 level are in the bold font.

HMM	DNN			
(α)	(#layers × #units)	Neutral	<i>p</i> value	Z
15.8 (16)	38.5 (4 × 256)	45.7	$< 10^{-6}$	
16.1 (4)	27.2 (4 × 512)	56.8	< 10 ⁻⁶	
12.7 (1)	36.6 (4 × 1 024)	50.7	< 10 ⁻⁶	





Image from Zen et al., "Statistical Parametric Speech Synthesis using DNNs", 2014

1e+07

RNN-based speech synthesis

- Access long range context in both forward backward directions using biLSTMs
- Inference is expensive; inherently have large latency



Image from Fan et al., "TTS synthesis with BLSTM-based RNNs", 2014