

Text-to-Speech Synthesis – Part II

Dr. Mathew Magimai Doss

Outline

Introduction

Statistical Parametric Speech Synthesis (SPSS)

Hybrid Speech Synthesis

End-to-end Speech Synthesis

Evaluation

Outline

Introduction

Statistical Parametric Speech Synthesis (SPSS)

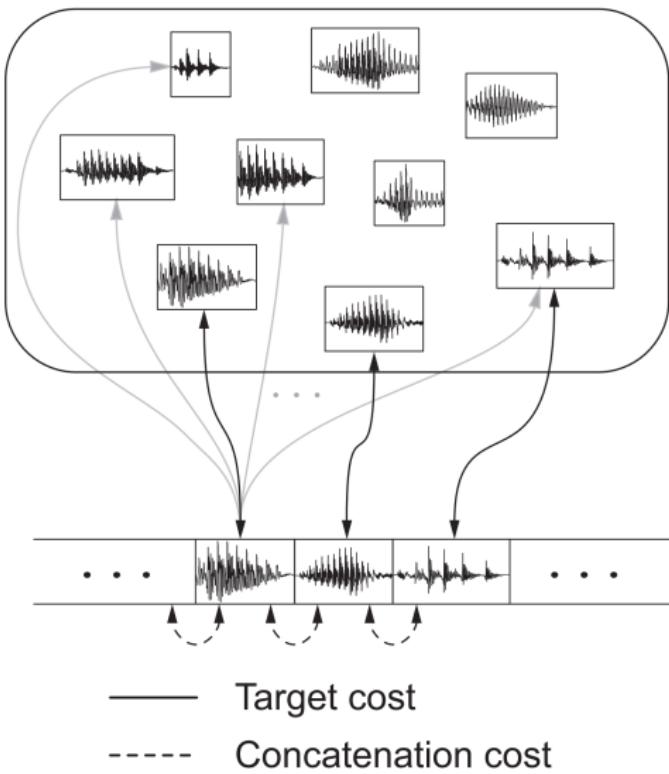
Hybrid Speech Synthesis

End-to-end Speech Synthesis

Evaluation

Last Week: Concatenative Synthesis

All segments



Zen, Tokuda & Black, 2009

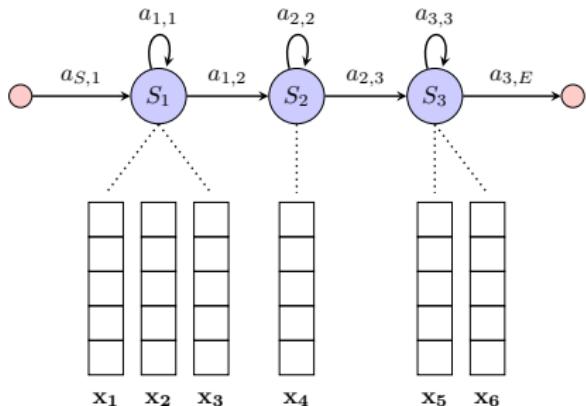
Last Week: Concatenative Synthesis

Target cost: Find best match to the target unit, in terms of

- Phonetic context
- F0, stress, phrase position, duration
- Acoustic distance

Join cost: Find a unit that can combine well with neighboring units and has

- Matching formants, energy, F0



These can be seen as emission
(*target cost*) and transition (*join*
cost) probabilities of HMMs.

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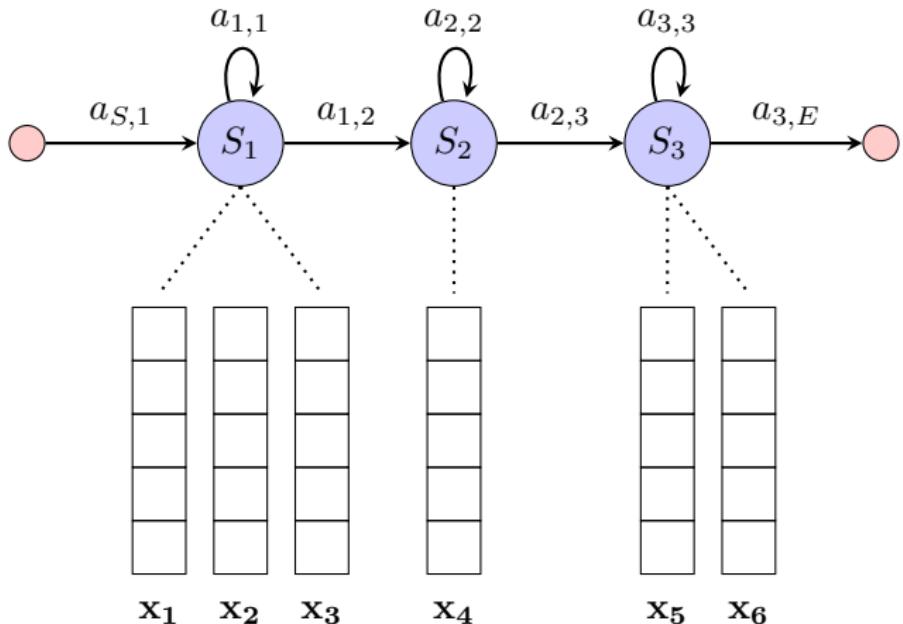
End-to-end Speech Synthesis

Evaluation

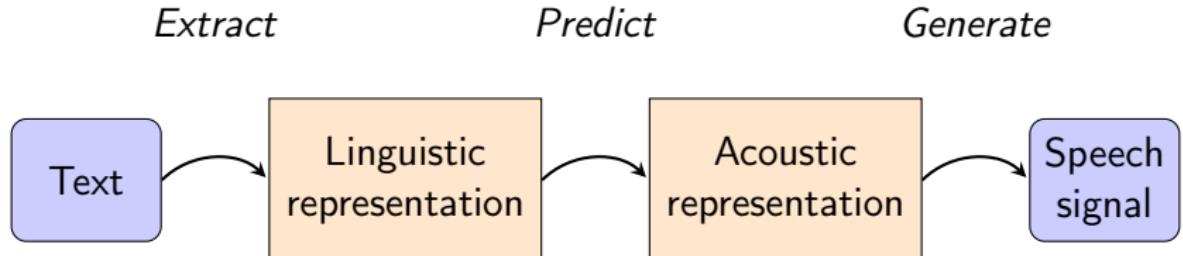
Statistical Parametric TTS

- Uses HMMs, like ASR, but to generate speech.
- Needs less training data, no need to store the unit database.
- Easy to adapt the speech.
- No artefacts from unit joints.
- Buzzy speech quality.
- Interactive online demo

HMM Synthesis

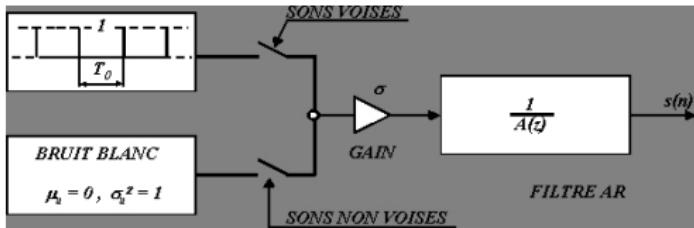


TTS: Basic Steps



Linguistic representation: Context-dependent states (with a lot of context!), representing phonemes

Acoustic representation: Spectral (*system*) and excitation (*source*) features



Linguistic representation

Context for modelling HMM states includes:

- Current, preceding, following phonemes
- Position of current phoneme in syllable
- Numbers of phonemes in current, preceding, following syllables
- Stress and accent of current, preceding, following syllables
- Number of syllables to previous, next stressed syllable
- Position of current word in phrase
- Number of words to next content word
- ...

HTS linguistic feature specification

Acoustic representation

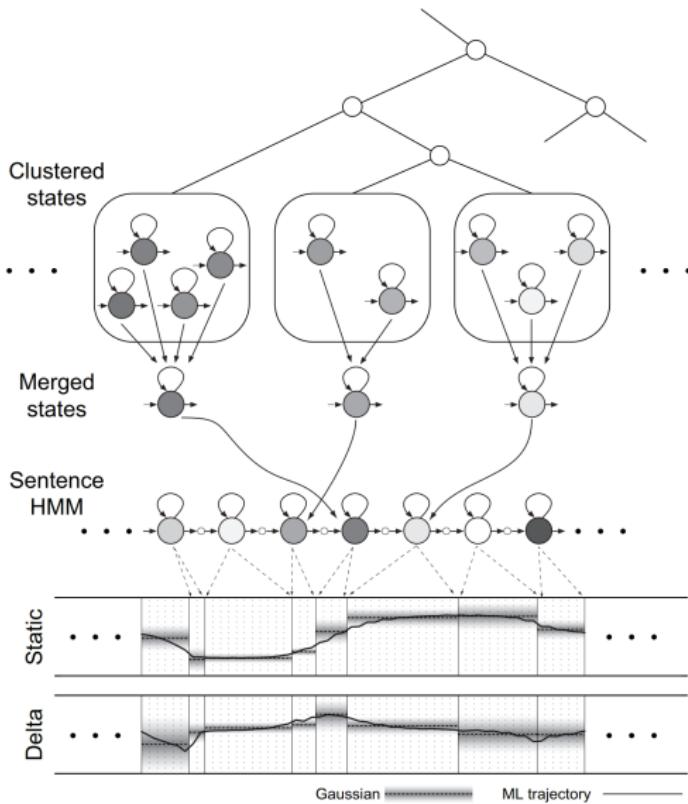
Predicted parameters:

- Spectrum: MFCCs + Δ + $\Delta\Delta$
- Excitation: $\log F_0$ + Δ + $\Delta\Delta$
- Possibly further vocoder parameters
- HMM state durations

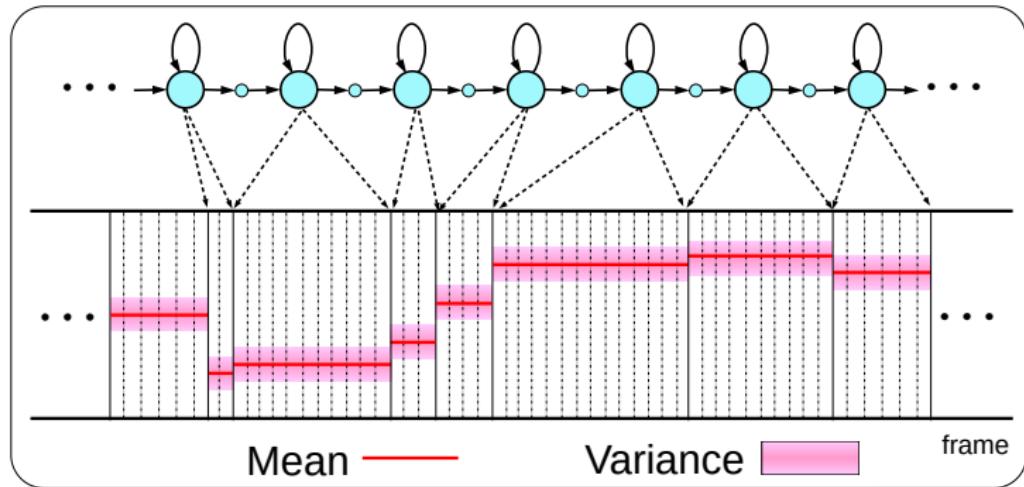
Prediction models:

- Regression trees with Gaussian probability distributions
- Neural networks

HMM Synthesis



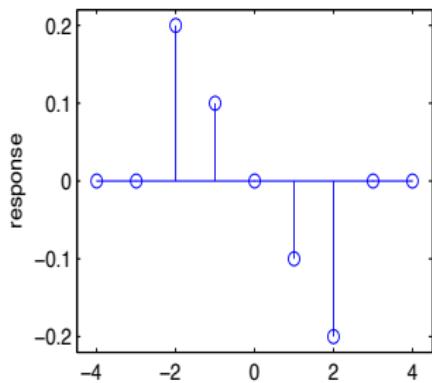
Static Features



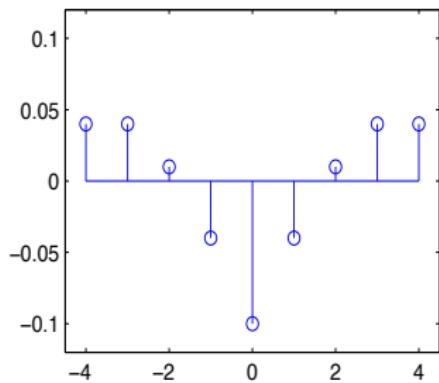
HTS slides

Recall: Temporal derivatives

$$\Delta_{c_m} = \frac{\sum_{k=1}^K k \cdot (c_{m+k} - c_{m-k})}{2 \cdot \sum_{k=1}^K k^2} \quad (1)$$



Delta (first order derivative)

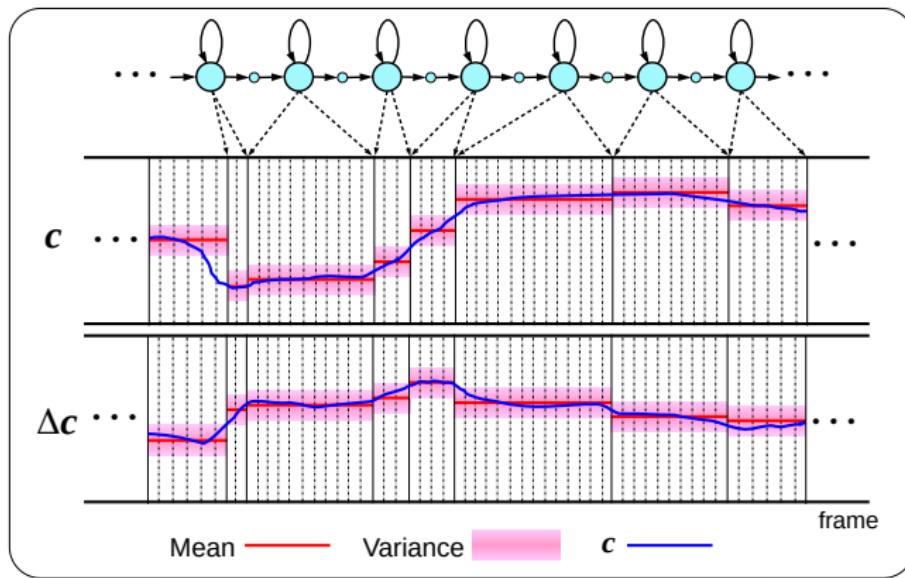


Delta-Delta (second order derivative)

- Savitzky-Golay filtering and temporal derivatives computation

With Dynamic Features

Dynamic features help to generate smooth trajectories.



HTS slides

See formulation of HMM as trajectory model

Duration Modeling

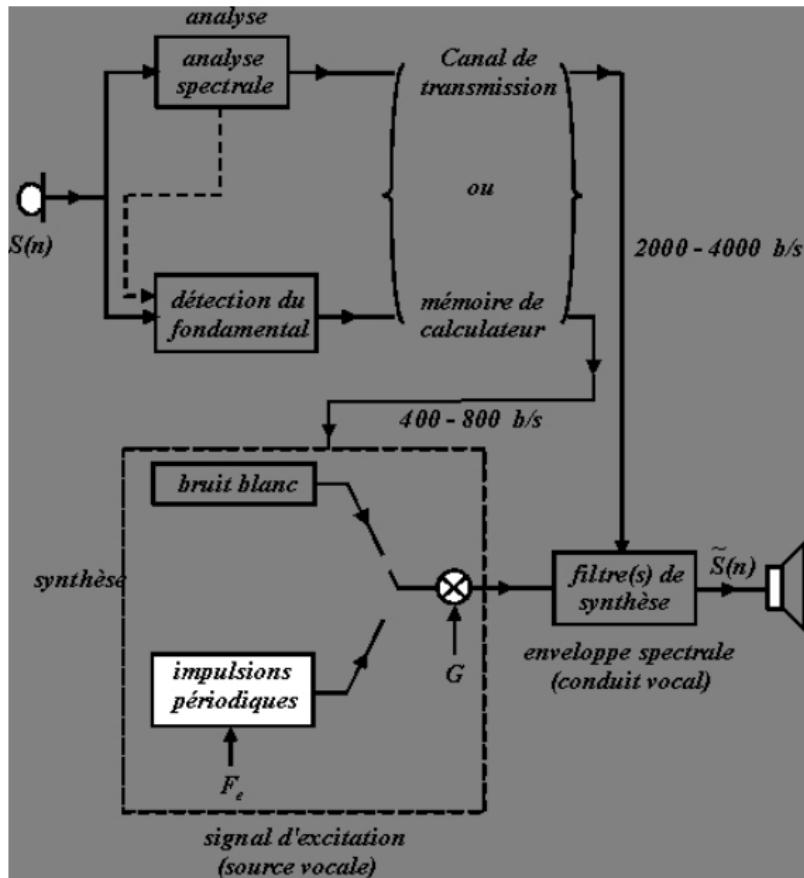
- Normal HMMs model duration through the transition probabilities of self-loops
 - Duration probabilities decay exponentially, which is inaccurate
 - Usually sufficient for ASR, but TTS needs explicit model

Hidden semi-Markov models (HSMM)

- Replace self-transitions with explicit Gaussian duration model

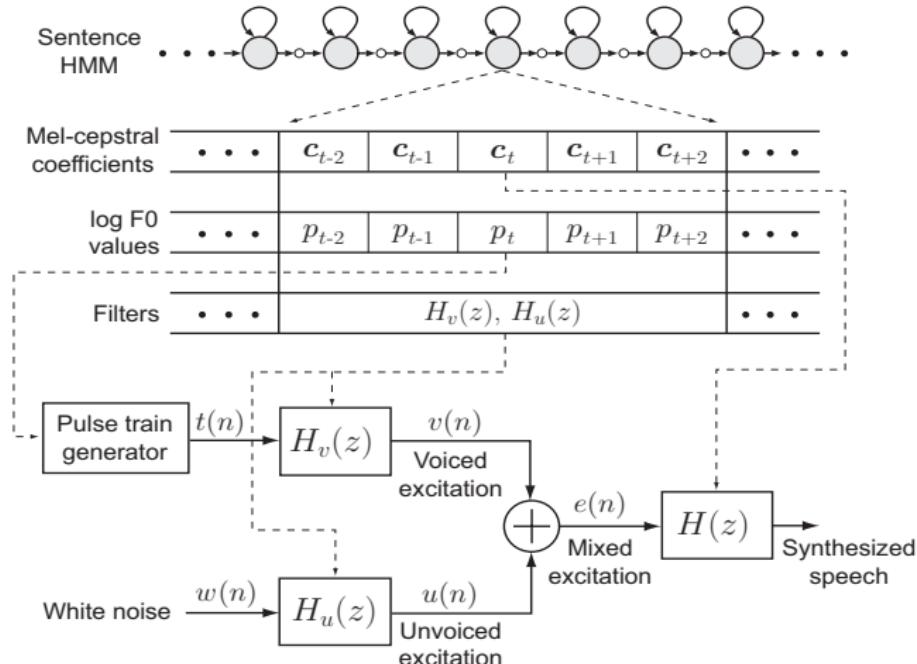
Shun-Zheng Yu, [Hidden semi-Markov models](#), Artificial Intelligence, Vol. 174, 2010, pp 215–243.

Vocoding: recall LP-based speech coding



Vocoding: applied to HMM-based TTS

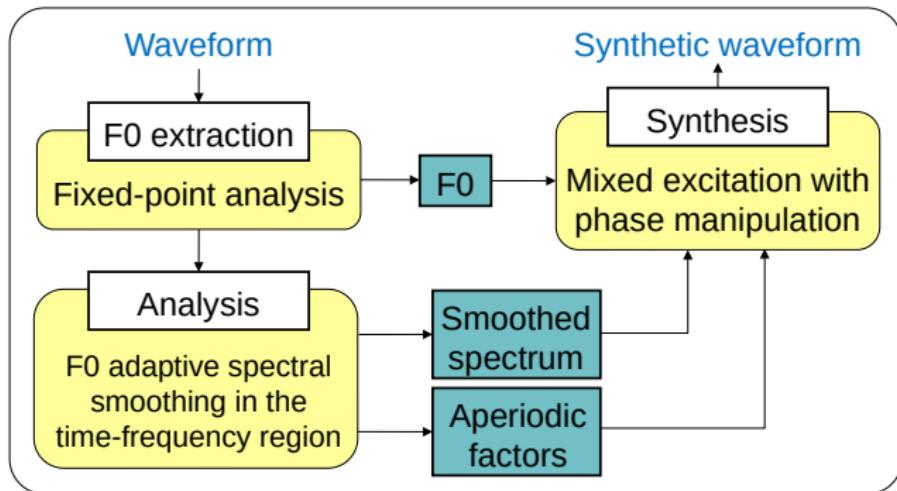
Waveform generation using source-filter model given cepstral feature and F_0 information estimates



source: Zen, Tokuda & Black, 2009

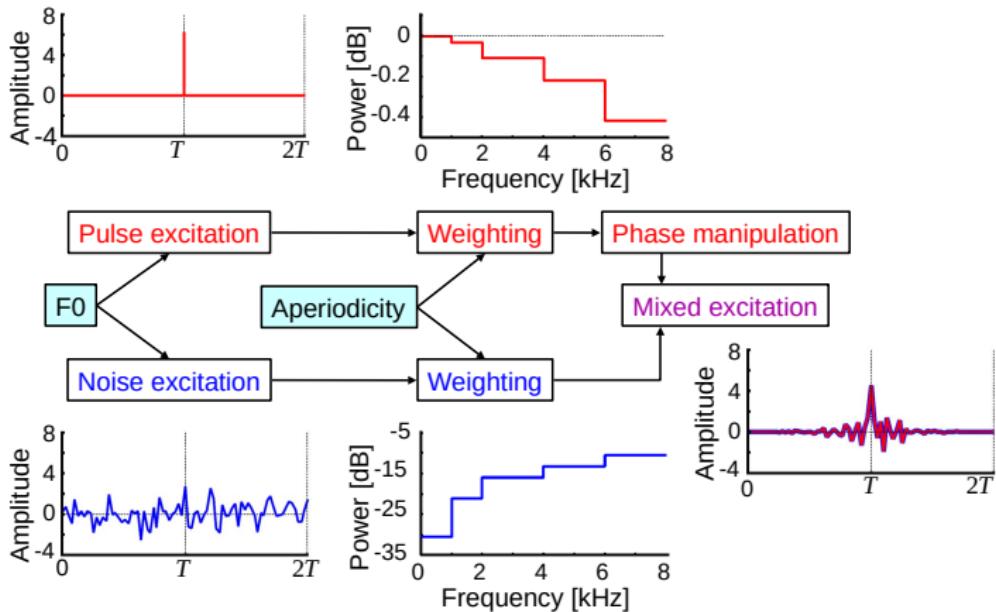
STRAIGHT Vocoder

Speech Transformation and Representation by Adaptive Interpolation of weiGHTed spectrogram

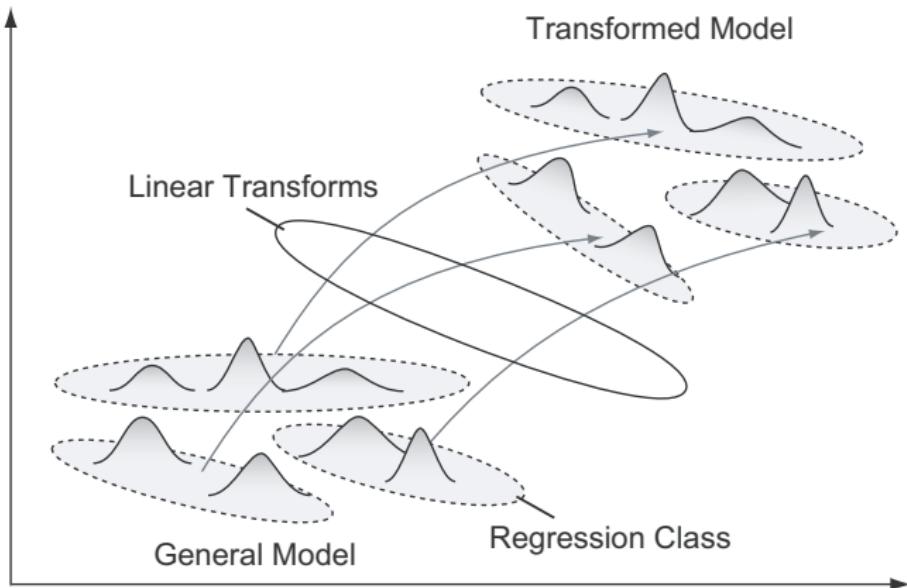


HTS slides

STRAIGHT excitation generation



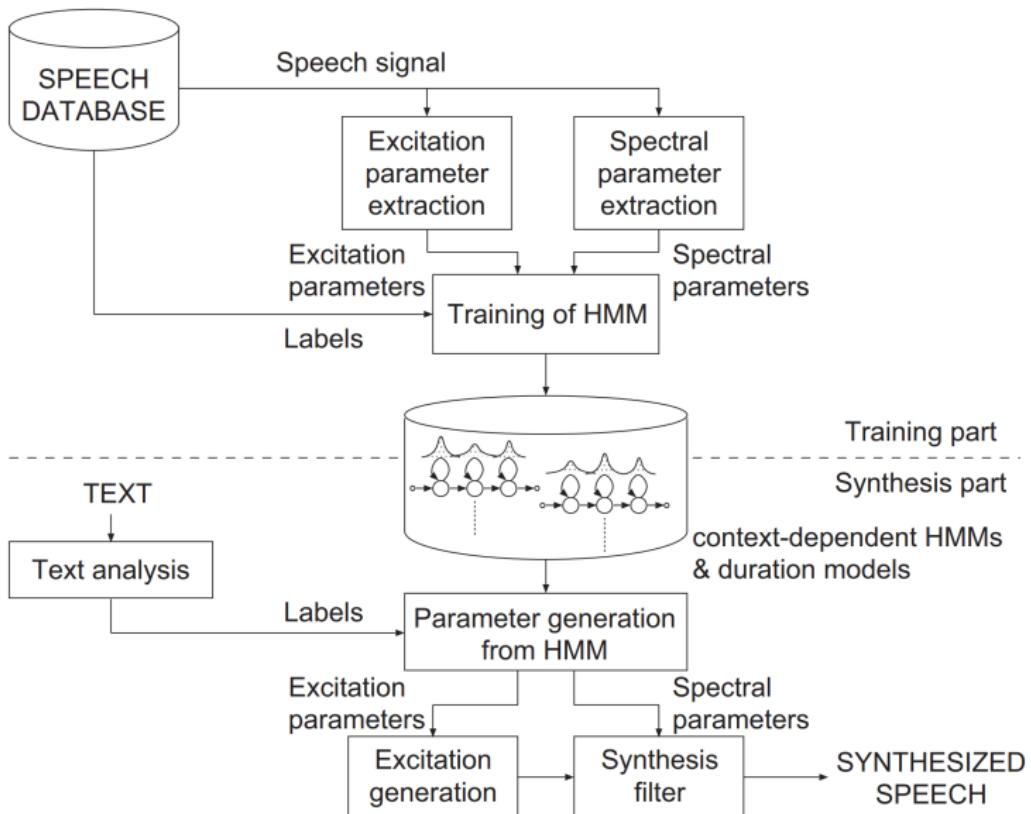
Speaker Adaptation



source: Zen, Tokuda & Black, 2009

Using adaptation techniques such as, maximum likelihood linear regression (MLLR) (applied to model parameters), constrained MLLR (applied to features).

HTS System Overview



Summary: HMMs for ASR vs. TTS

	ASR	TTS
Acoustic features	About 13 spectral parameters + Δ + $\Delta\Delta$	40–60 spectral parameters + Δ + $\Delta\Delta$ + source features
Frame shift	10 ms	5 ms
Modeling unit	Triphone	Quinphone with full linguistic context
States per model	3	5
State emission distribution	GMM	Single Gaussian
Duration model	HMM self-loops	Explicit model (HSMM)
Parameter estimation	Baum-Welch (EM)	Baum-Welch (EM)
Decoding	Viterbi search	Not usually required
Generation	Not required	Maximum likelihood

Dines et al., 2010 and King, 2011

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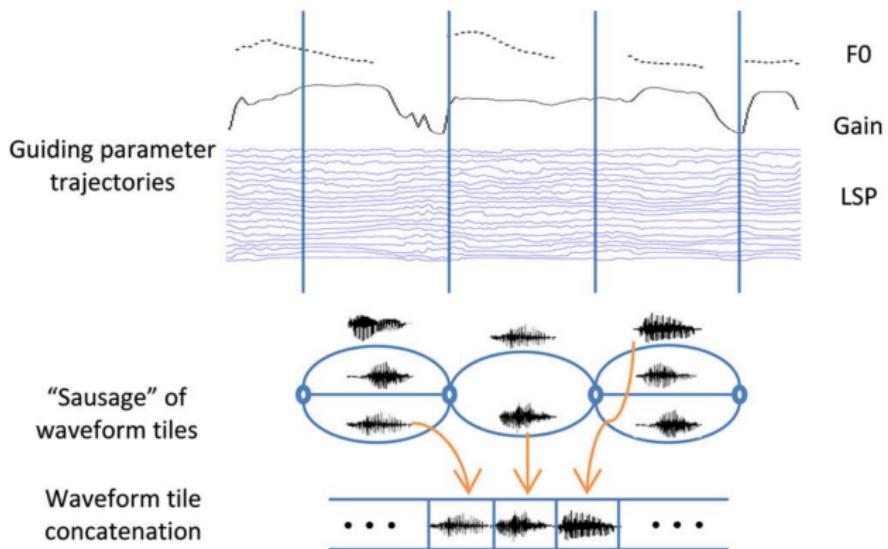
End-to-end Speech Synthesis

Evaluation

Hybrid TTS

Statistically driven unit selection synthesis:

- Like SPSS, but replace the vocoder with unit concatenation
- Like unit selection, but select units based on predicted acoustic parameters



Qian, Soong & Yan, 2012

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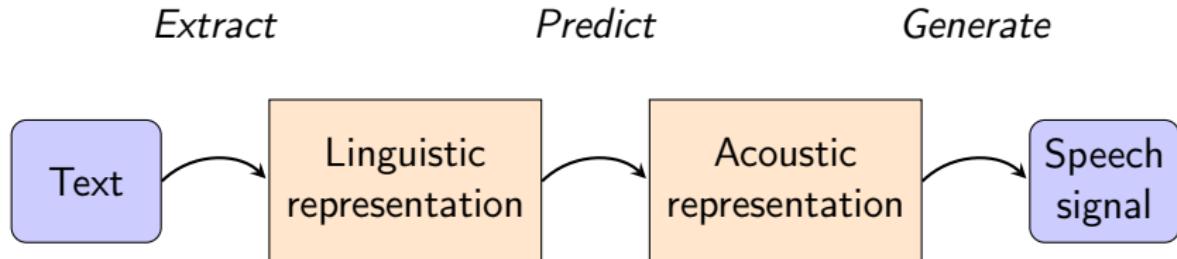
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End-to-end TTS

Aims to replace hand-crafted TTS components with neural networks, in particular:

- NLP pre-processing pipeline
 - Text normalization (difficult!)
 - Lexicons and grapheme-to-phoneme (G2P) conversion
- Vocoder

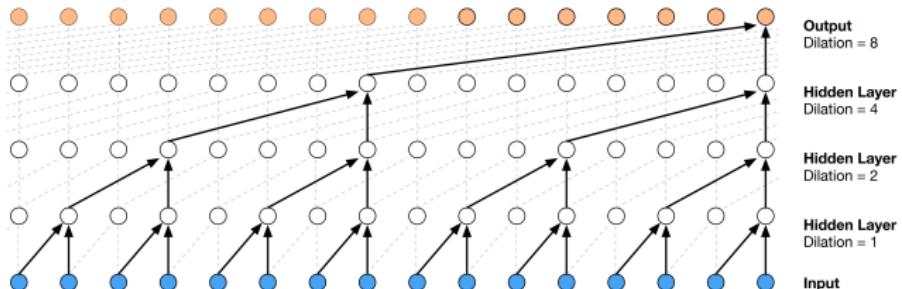


WaveNet (Neural Vocoder)

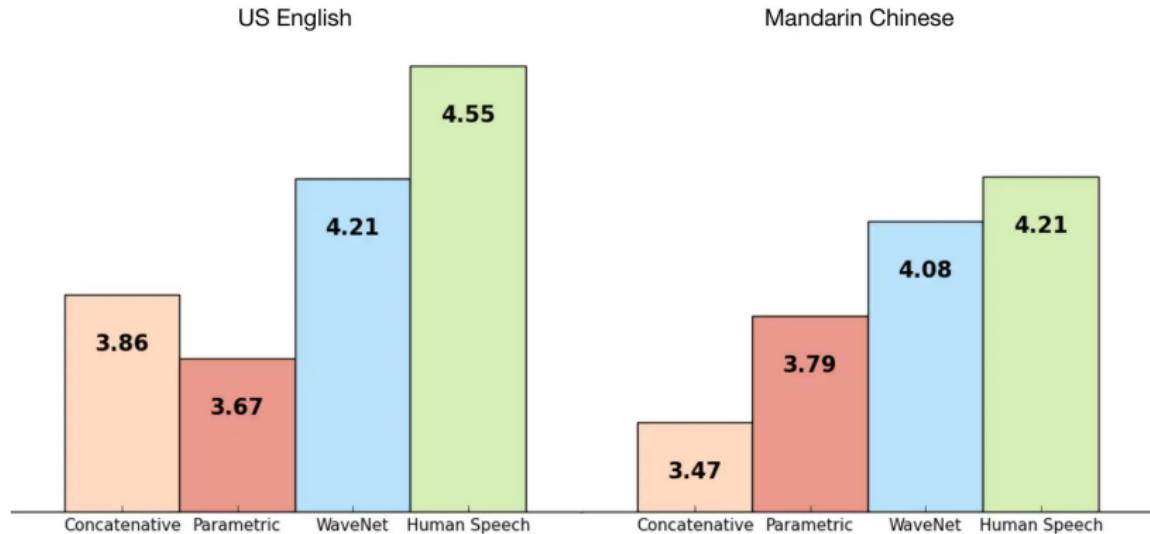
- Directly predicts speech samples ($x[1] \dots x[n] \dots x[N - 1]$) given acoustic and linguistic features f

$$\prod_{n=1}^{N-1} p(x[n]|x[n-1], \dots, x[0], f)$$

- Dilated convolutions allow covering long ranges
- Initially very slow, but now used for real-time, cloud-based TTS. Still requires non-negligible computing resources.
- Very natural speech compared to traditional SPSS ([Samples](#))
- Can train one model for multiple speakers



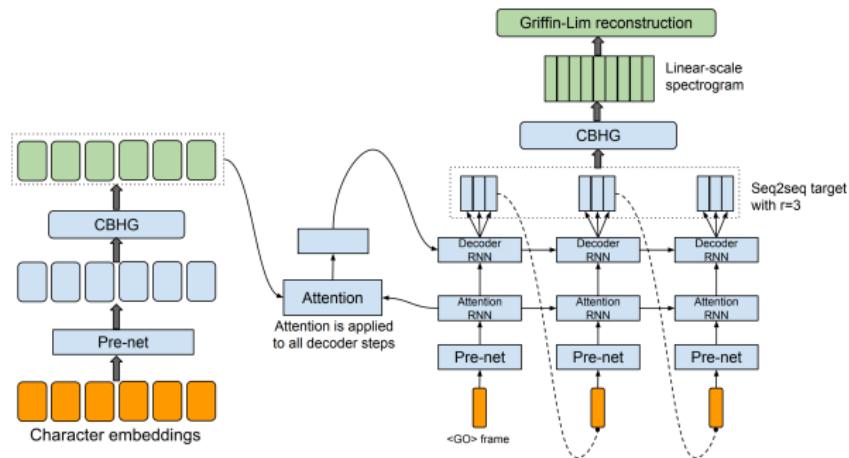
WaveNet MOS Scores



van den Oord et al., 2016

Tacotron

- Generate spectrograms directly from text (character embeddings)
 - No need for HMM alignments to train
 - No need for G2P conversion
 - Assumes text normalized input ("16" is "sixteen")
- Spectrogram inversion with Griffin-Lim algorithm (Tacotron 2 uses WaveNet)



Wang et al., 2017

TTS Training Data Requirements

Typical amounts of data required for training:

Architecture	Training data
Diphone synthesis	1 instance per diphone (total of 1000)
Unit selection	5–40 hours (Taylor, 2009), difficult to adapt to new speakers (can use voice conversion)
Statistical parametric synthesis	5+ hours for initial system, but can easily adapt to new speakers
WaveNet	25+ hours, but can combine multiple speakers and can adapt to new speakers with <10 minutes (Chen et al., 2019)

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Evaluating Synthetic Speech

- Subjective vs. objective evaluation
- Naturalness vs. intelligibility
- Other evaluation measures?

Subjective vs. Objective Evaluation

Subjective

- Tests with human listeners
- Expensive and time-consuming, but very flexible
- Not easy to reproduce

Objective

- Automatic evaluation through computers
 - E.g. with ASR systems or by measuring distances to human reference samples
- Cheap and fast, but more difficult to interpret
- Reproducible

Naturalness vs. Intelligibility

Naturalness

- Material: Target domain text or phonetically balanced sentences
- Metrics: Mean opinion scores (MOS)
{Excellent, Good, Fair, Poor, Bad}, A/B preference tests

Intelligibility

- Material: Semantically unpredictable sentences
 - The dog fights under the red beach.*
 - The deaf dress sees the bear.*
 - When does the gold take the beige fear?*
 - The wheat attempts the time trembling.*
 - The real glass opens the corner.*
 - Turn the date or the hand.*
- Metric: Word error rate
- Can be made more challenging by adding noise

Blizzard Challenge

- Yearly challenge task to build TTS systems on a shared dataset
- Thorough, centrally organized listening tests
- <http://www.festvox.org/blizzard/>

Thank you for your attention!

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