

UNIVERSITÉ DE LIÈGE

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Automatic Multispeaker Voice Cloning Across Languages

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1 Abstract

To do when I'll have a good overview of the project. Try to answer:

- What is the goal of the application? What are its requirements, what is the setting, what kind of data are we going to use it on?
- What is zero-shot voice cloning? How does it fit in here (difference between an online and offline approach)?
- What are the particularities of our implementation (both model and datasets), what are its upsides and downsides (for example: requires huge datasets but fast inference)?
- What did we ultimately achieve? How good are our results?

2 Introduction

Concise presentation of the problem

Preprocessing of text into phonemes?

SOTA ON MULTISPEAKER TTS:

Previous state of the art in TTS include hidden Markov models (HMM) based speech synthesis, which is a statistical parametric speech synthesis (SPSS) method. HMMs are trained to synthesize mel-frequency cepstral coefficients (MFCC) with energy, their delta and delta-delta coefficients [1]. The result is passed through a vocoder¹ such as MLSA [2]. The spectral parameters, pitch parameters and state durations of the model are conditioned on the linguistic contexts **define context** such that different contexts are clustered by a decision tree and a distribution is learned for each cluster [6]. It is thus possible to modify the voice generated by conditioning on a speaker or tuning these parameters with adaptation or interpolation techniques (e.g. [5]), effectively making HMM-based speech synthesis a multispeaker TTS system. **Compare with concatenative?** see [7]

Improvements to HMM-based speech synthesis were later brought by feed-forward and recurrent deep neural networks (DNN) as a result of progress in both hardware and software. Several authors propose to replace the decision trees by a DNN, arguing for better data efficiency [7, 3, 4]. Some demonstrate improved speech quality for a similar number of parameters [7, 4].

Wavenet:

Breakthrough in TTS with raw waveform gen

Take images from <https://deepmind.com/blog/wavenet-generative-model-raw-audio/> ?

Dilated causal convolutions

Condition on a speaker identity

Tacotron

Deep voice (1, 2, 3 + few samples), Tacotron 2

SV2TTS

Extensions?

¹Specifically in TTS, some authors define a vocoder as a voice encoder that retrieves speech parameters to be used in synthesis. The more common definition however, is that of a function that generates a raw audio waveform from temporal features such as MFCC. This is the one we will use. **Review this**

References

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