

TTS

框架结构 / 结果分析 / 系统展示

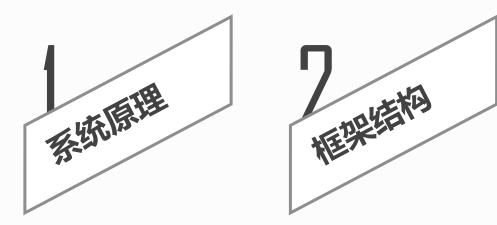
端到端语音合成

Neural network based end-to-end speech synthesis / Text To Speech

小组汇报



CONTENTS





Based on tacotron2, SV2TTS, tensorflowTTS, Real-time Voice Cloning and mockingbird

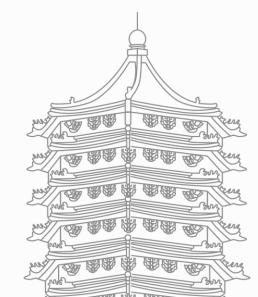












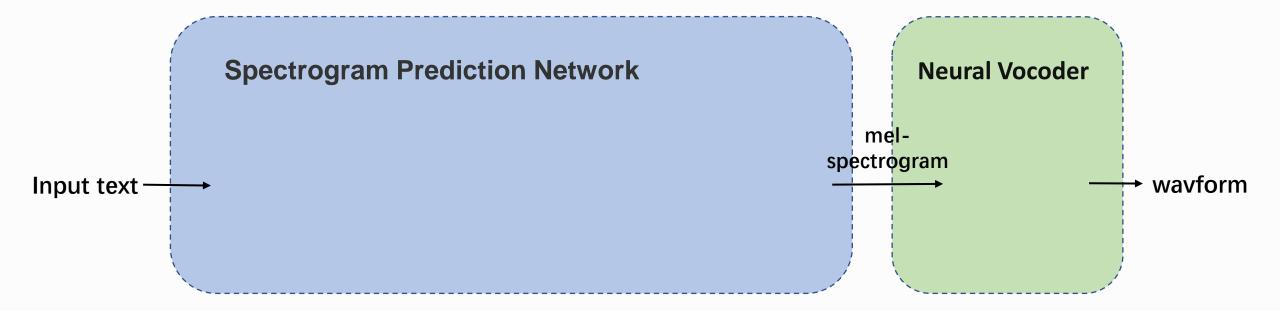


系统原理-Tacotron2

Natural TTS Synthesis by Conditioning WaveNet on Mel-Spectrogram Predictions

端到端 神经网络 语音合成 **声谱预测网络**:文本序列 → 帧级语音特征 (以梅尔频谱表示)

语音波形 神经声码器: 帧级语音特征 -





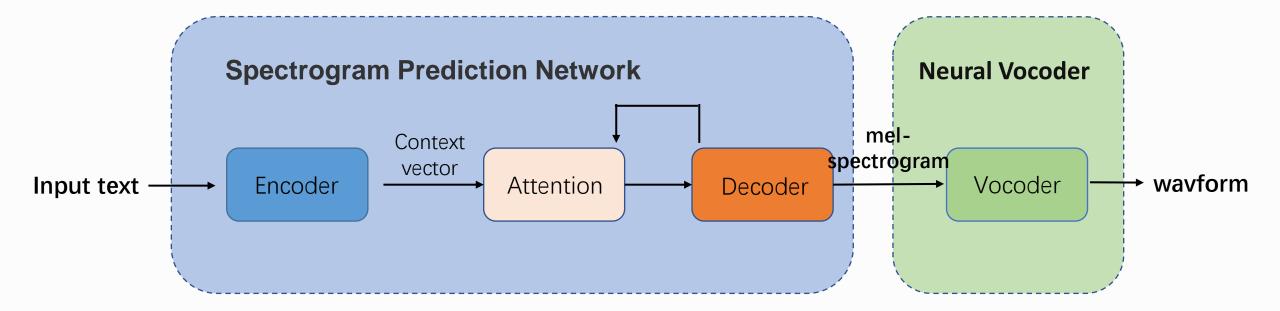


系统原理-Tacotron2

Natural TTS Synthesis by Conditioning WaveNet on Mel-Spectrogram Predictions

端到端 神经网络 语音合成 声谱预测网络: 文本序列 ── 帧级语音特征 (以梅尔频谱表示)

神经声码器: 帧级语音特征 ── 语音波形





系统原理-SV2TTS

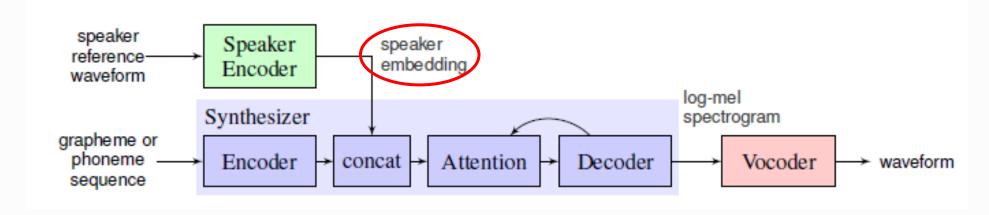
Transfer Learning from Speaker Verification to Multispeaker Text-To-Speech Synthesis

基于Google 2017年发布的论文

《Transfer Learning from Speaker Verification to Multispeaker Text-To-Speech Synthesis(SV2TTS)》

模型分为三个独立训练的组件:

- 编码器(encoder): 生成代表说话人音色的向量
- 合成器 (synthesizer):将文本转换成梅尔频谱图
- 声码器(vocoder):将梅尔频谱图转换成waveform







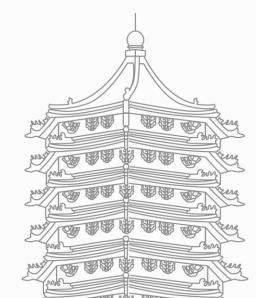
框架结构

print the presentation and make it into a film to be used in a wider field. The user can demonstrate on a projector or computer.





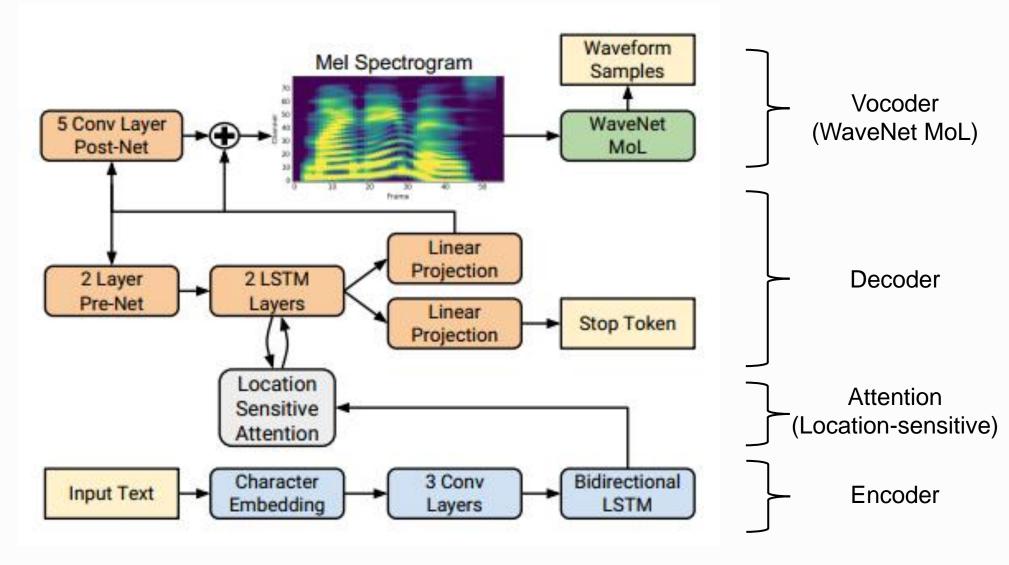






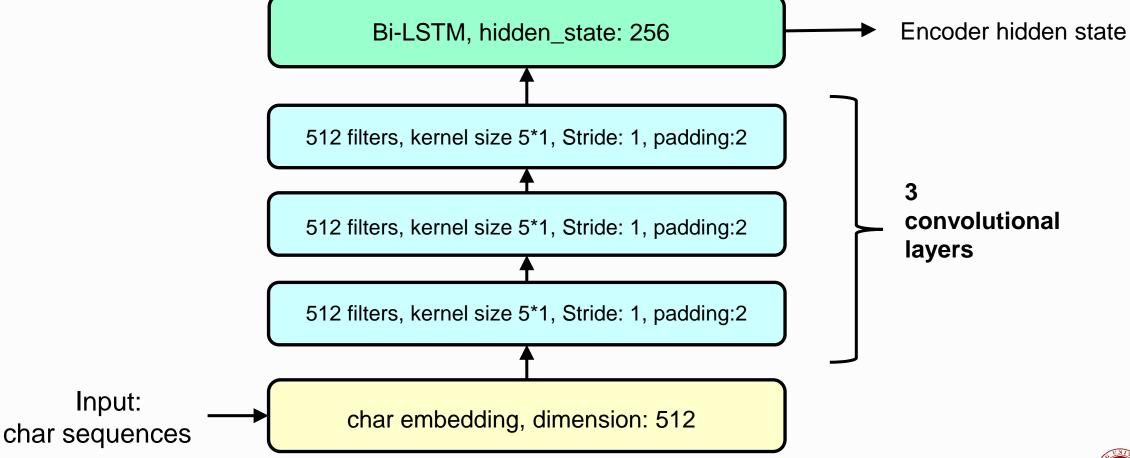
框架结构: Tacotron2

Natural TTS Synthesis by Conditioning WaveNet on Mel-Spectrogram Predictions





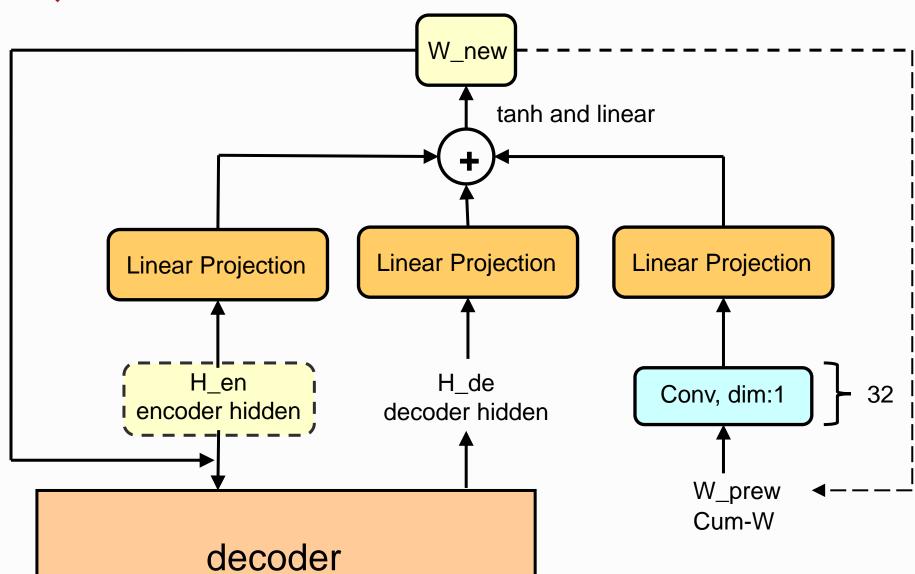
Structure: Embedding + 3*Conv_layers + Bidirectional_LSTM



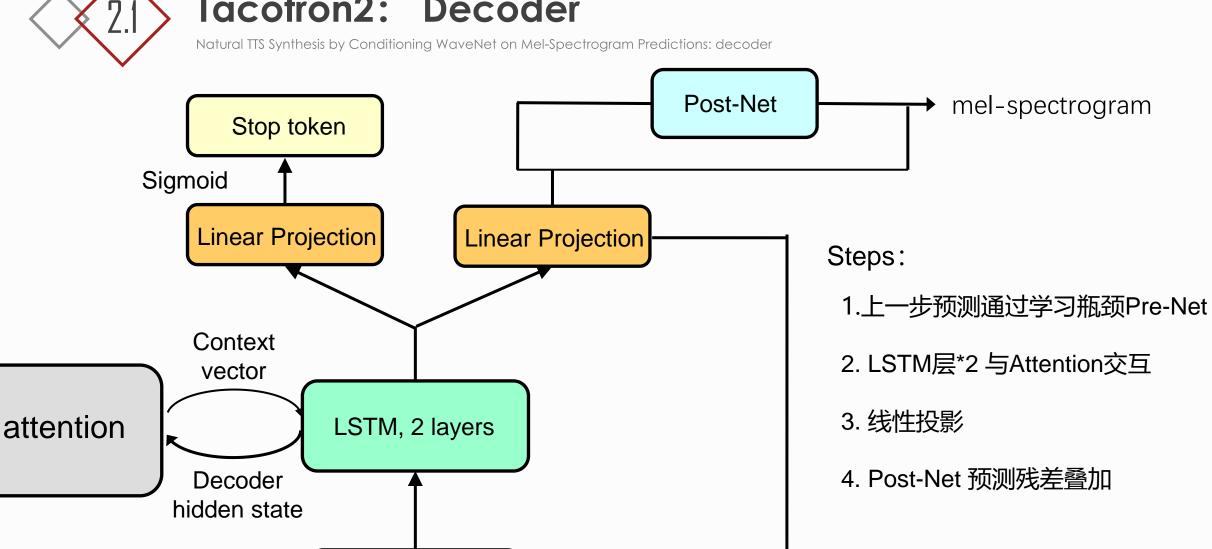


Tacotron2: Attention

Natural TTS Synthesis by Conditioning WaveNet on Mel-Spectrogram Predictions: attention







Pre-Net, 2 layers



Tacotron2: Vocoder

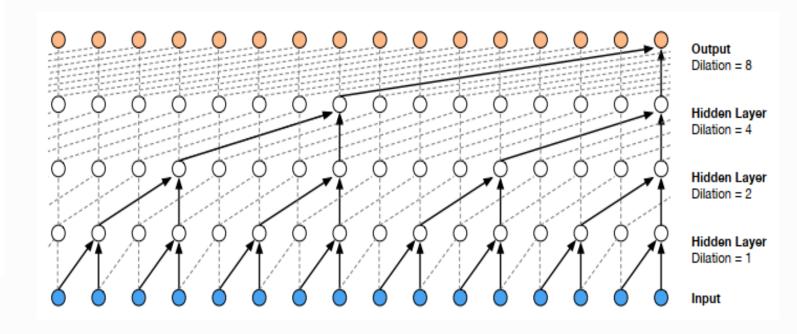
Natural TTS Synthesis by Conditioning WaveNet on Mel-Spectrogram Predictions: vocoder

Table 3: Mean Opinion Scores

Model	MOS	95% CI
Griffin Lim	1.57	± 0.04
WaveGlow	4.11	± 0.05
WaveNet	4.05	± 0.05
MelGAN	3.61	± 0.06
Original	4.52	\pm 0.04

在ljspeech上声码器主观意见分数比较

WaveNet 使用的膨胀因果卷积示意图





SV2TTS: Synthesizer & Vocoder

Transfer Learning from Speaker Verification to Multispeaker Text-To-Speech Synthesis

合成器:

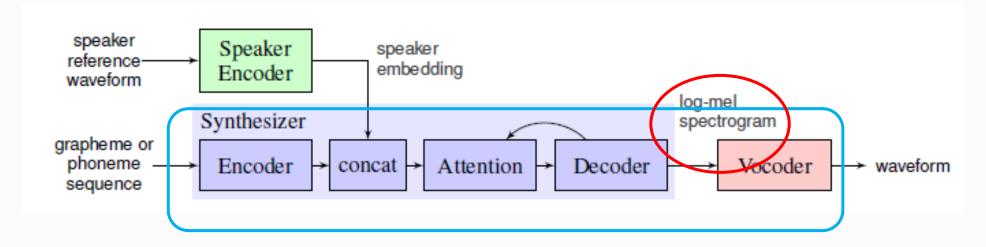
基于Tacotron2;

输入文本, 生成对应的梅尔频谱图

声码器:

论文中采用WaveNet,项目实现用了HiFiGAN;

将梅尔频谱图转换为最终的语音





SV2TTS: Speaker Encoder

Transfer Learning from Speaker Verification to Multispeaker Text-To-Speech Synthesis

数据集:

- 来自多个说话者(>18K人,约36M条语句)
- 嘲杂
- 短暂(只有几秒)
- 无文本

speaker reference waveform Speaker Encoder embedding

训练: 采用GE2E损失函数进行说话者验证

生成: speaker embedding (嵌入向量)



即广义端到端(generalized end-to-end, GE2E)损失

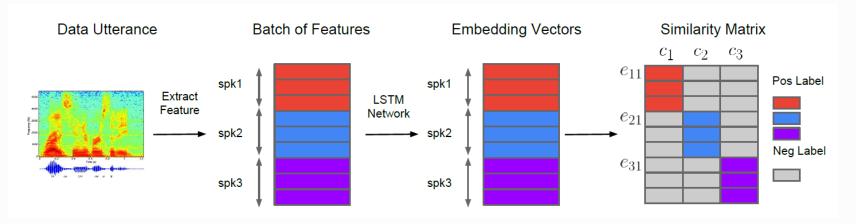
训练:

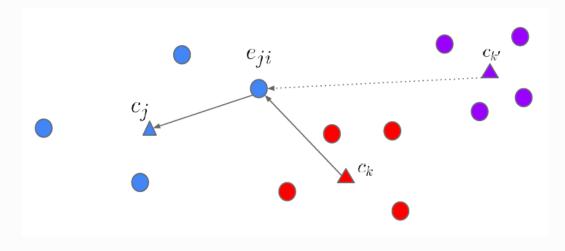
- 一个batch中包含N个说话者,平均每个说话者提供M条语句,共N*M条语句;以x_ji表示从说话人j第i条语句中提取的特征向量
- 通过LSTM网络得到输出结果 $f(\mathbf{x}_j;\mathbf{w})$,正则化后则有 $\mathbf{e}_{ji} = \frac{f(\mathbf{x}_{ji};\mathbf{w})}{||f(\mathbf{x}_{ji};\mathbf{w})||_2}$,是为说话人j 第i条语句的embedding向量
- 第j个说话者的embedding向量的中心定义为 $\mathbf{c}_k = \mathbb{E}_m[\mathbf{e}_{km}] = \frac{1}{M}\sum_{m=1}^M \mathbf{e}_{km}$
- 定义相似矩阵 Sji,k = w*cos(e_ji; c_k) + b 为每个embedding向量e_ji与所有中心c_k之间的缩放余弦相似性



训练时,我们希望每条语句的embedding向量与各自说话者的中心相似,同时远离其他说话者的

中心





两种实现方式: 两种损失

Softmax

在S_ji,k上设置softmax,使输出在k=j时等于1,否等于0;则每个embedding

向量e_ji上的损失可以定义为
$$L(\mathbf{e}_{ji}) = -\mathbf{S}_{ji,j} + \log \sum_{k=1}^{N} \exp(\mathbf{S}_{ji,k})$$

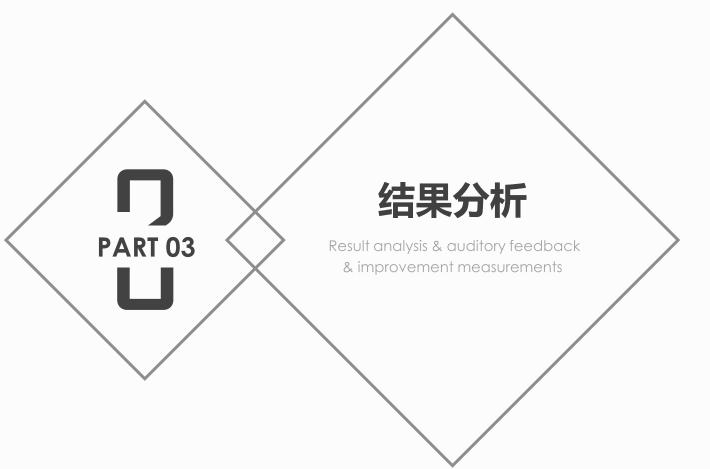
Contrast(对比度损失)

$$L(\mathbf{e}_{ji}) = 1 - \sigma(\mathbf{S}_{ji,j}) + \max_{\substack{1 \le k \le N \\ k \ne j}} \sigma(\mathbf{S}_{ji,k})$$

最终GE2E损失LG 是相似矩阵上所有损失的总和

$$L_G(\mathbf{x}; \mathbf{w}) = L_G(\mathbf{S}) = \sum_{j,i} L(\mathbf{e}_{ji})$$

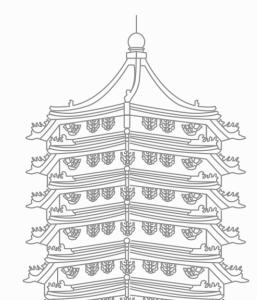


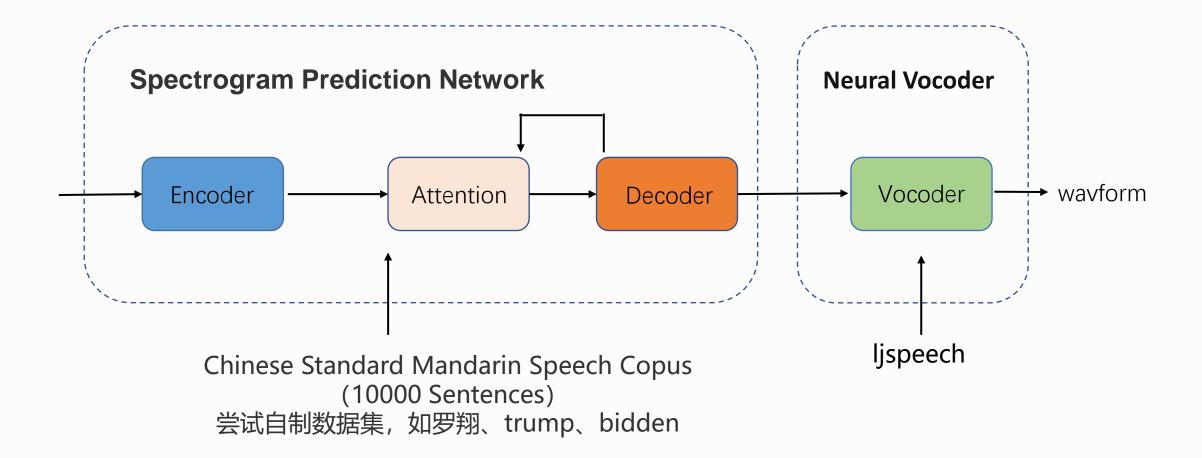








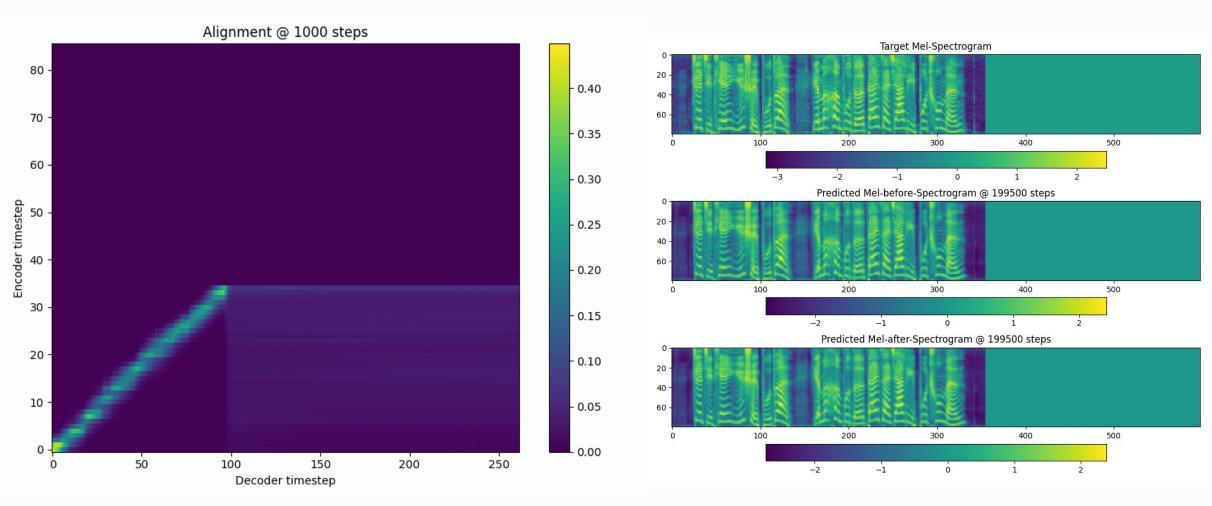






结果分析: Tacotron2

Natural TTS Synthesis by Conditioning WaveNet on Mel-Spectrogram Predictions: train





结果分析: SV2TTS

Transfer Learning from Speaker Verification to Multispeaker Text-To-Speech Synthesis

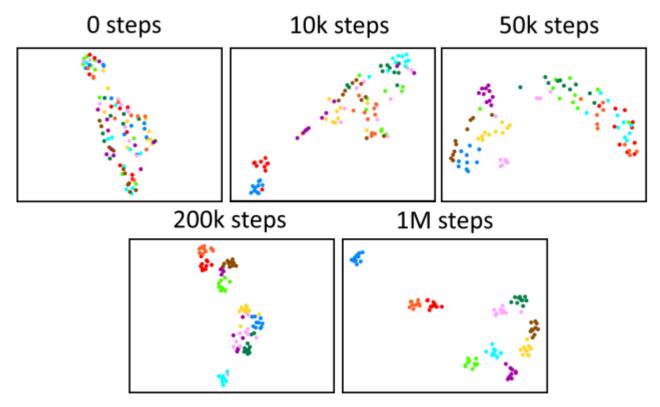


Figure 12: UMAP projections of utterance embeddings from randomly selected batches from the train set at different iterations of our model. Utterances from the same speaker are represented by a dot of the same color. We specifically omit to pass labels to UMAP, so the clustering is entirely done by the model.

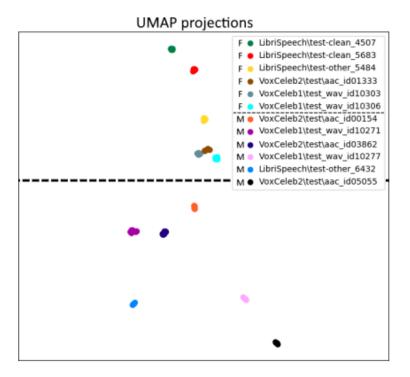
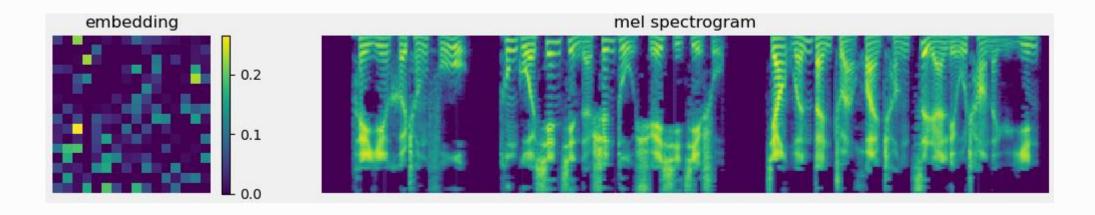
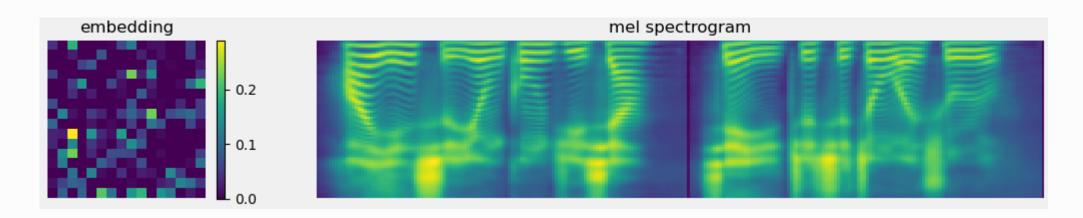


Figure 13: UMAP projections of 120 embeddings, 10 for each of the 12 speakers. Six male and six female speakers are selected at random from the test sets

结果分析: SV2TTS

Transfer Learning from Speaker Verification to Multispeaker Text-To-Speech Synthesis







听感反馈 & 改进措施

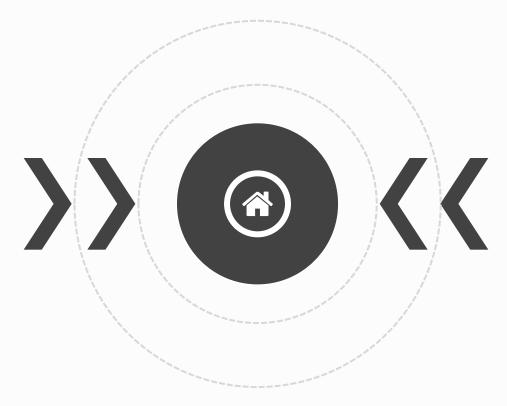
Result analysis & auditory feedback & improvement measurements

输出语音含有噪声

The generated speech is mixed with noise, which make it sound hoarse, affecting the sense of hearing

模型正则化/防止过拟合

Regularize the model to prevent over-fitting Improve vocoder



输入语音质量要求高

Input voice must be noise free and loud, otherwise the output is mixed with noise

降噪、语音增强/ 含噪样本上的训练

noise reduction and Speech enhancement train the model on more samples with noise

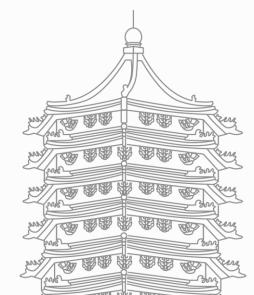














- [1] https://arxiv.org/abs/1712.05884
- [2] https://arxiv.org/abs/1308.0850
- [3] https://arxiv.org/abs/1710.10467
- [4] https://arxiv.org/abs/1806.04558
- [5] https://arxiv.org/abs/1802.08435
- [6] https://arxiv.org/abs/1609.03499
- [7] https://github.com/CorentinJ/Real-Time-Voice-Cloning
- [8] https://github.com/babysor/MockingBird



Thanks for watching! There are still many shortcomings QAQ. We welcome criticism and correction

汇报人: