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Low-Latency Incremental Text-to-Speech Synthesis with Distilled Context Prediction Network

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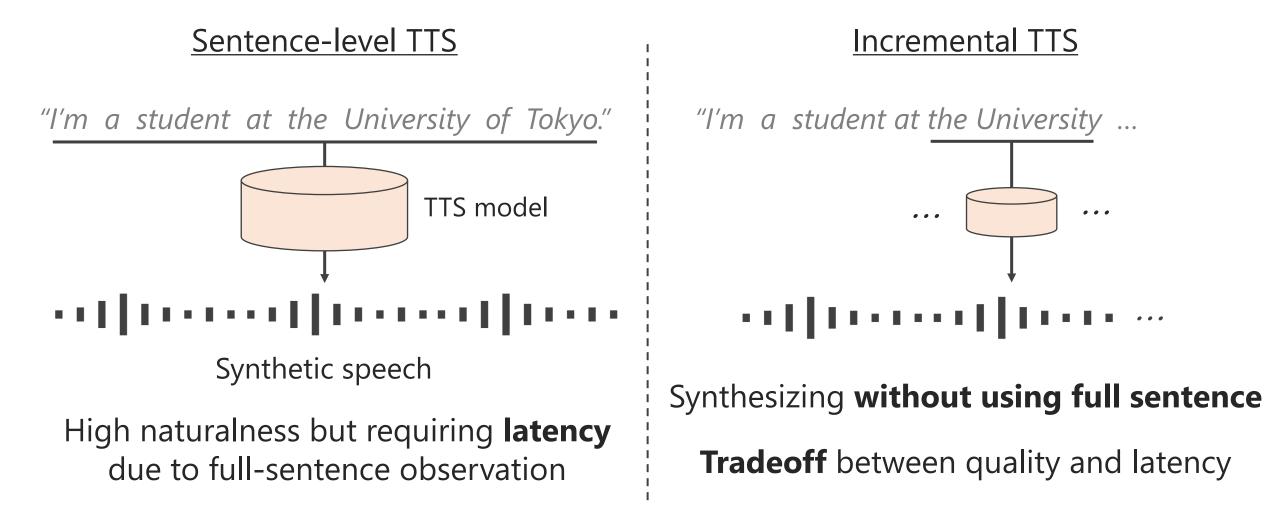




Incremental Text-to-Speech Synthesis

High-quality and low-latency streaming TTS is needed for real-time speech generation

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Incremental TTS with Pseudo Lookahead 3/13

Related work 1 [Saeki+, 2021]

Pseudo lookahead with language model Synthesizing current speech segment with **unobserved future context**

(Imitating human's incremental reading)

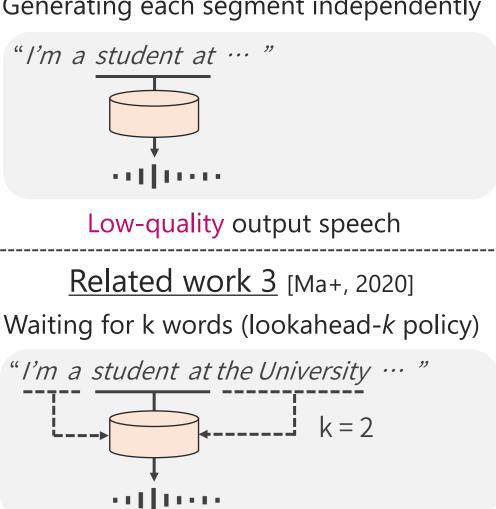
Pretrained LM (GPT2 [Radford+, 2019])

"I'm a student at Tohoku University, Japan."

(1) [1111]

Achieving high naturalness without waiting for observation of future context

<u>Related work 2</u> [Yanagita+, 2019] Generating each segment independently



Need waiting time of subsequent words

Motivation and Concept of Proposed Method 4/13

Challenge of previous method using pseudo lookahead with language model

- Huge processing time due to inference of GPT-2 at each time step
- Need to achieve faster speech synthesis than human's speaking speed

Proposed method: Fast listen-while-predict framework with language model distillation

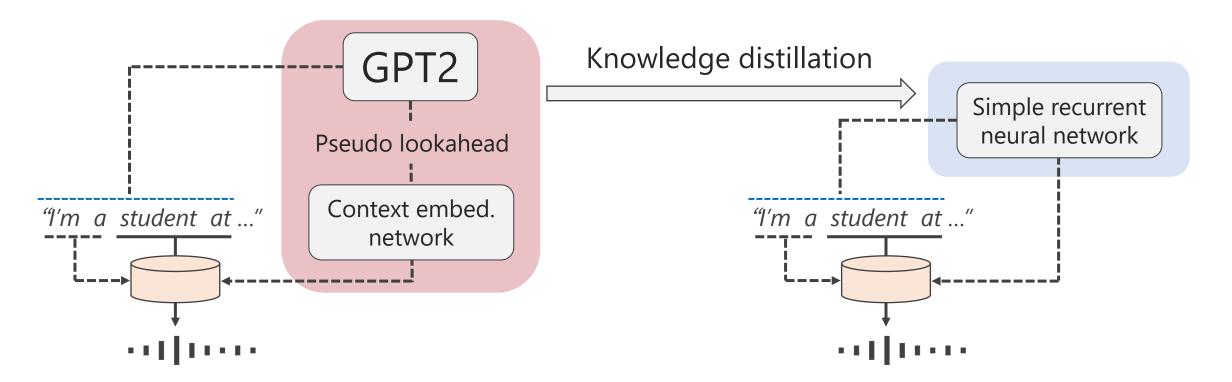
- knowledge distillation from GPT2 + contextual embedding to single lightweight model
- Directly predicting future context from observed words for fast inference

Overview of Proposed Method

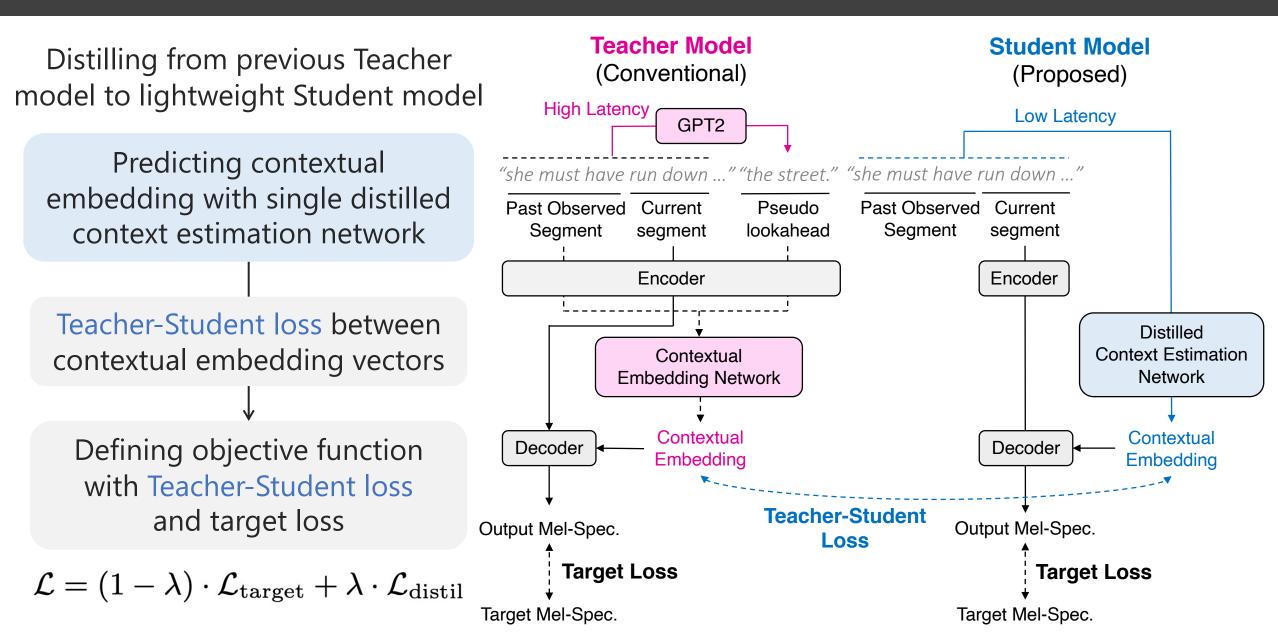
Inspired by task-specific knowledge distillation of language model (BERT) [Tang+, 2019]

- Distilling from BERT to lightweight recurrent model without attention
- Student model achieves **comparable performance** to Teacher model in various tasks

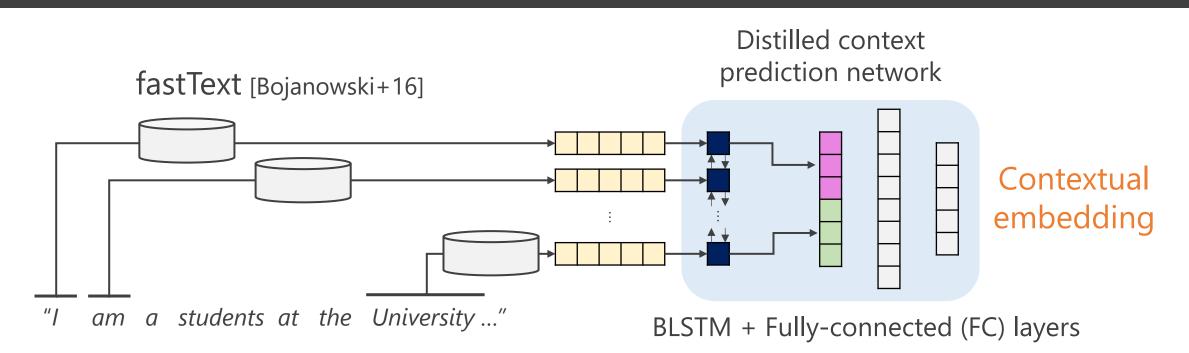
Our proposed method aims to perform task-specific knowledge distillation of GPT2 for context estimation task of text-to-speech synthesis



Proposed Teacher-Student Training Framework 6/13



Architecture of Distilled Context Prediction Network 7/13



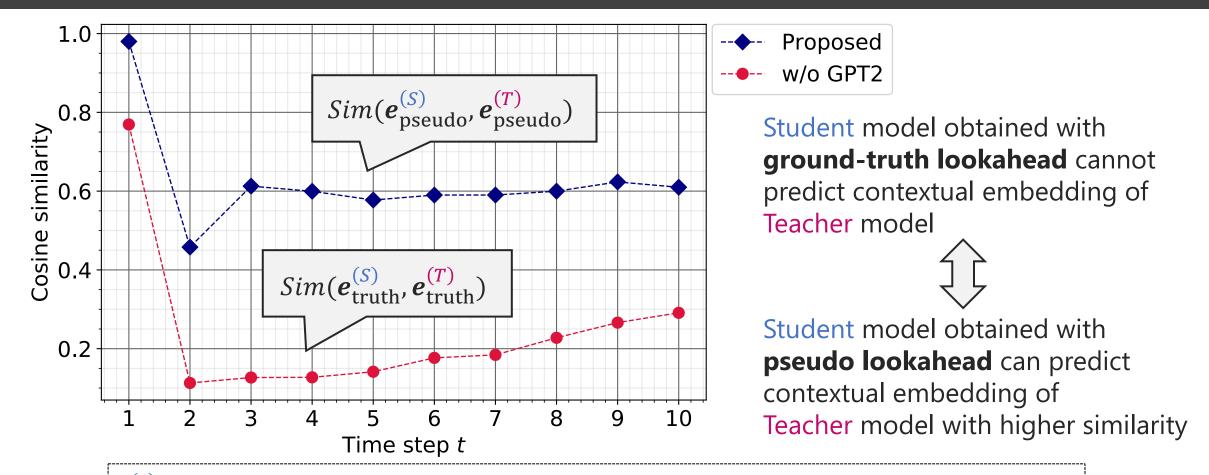
Obtaining distributed representation of observed words with pretrained fastText [Bojanowski+, 2016]

Estimating contextual embedding with lightweight recurrent model without attention

Compared three model sizes: *small, medium, large*

• (BLSTM-hidden, FC-hidden) = {(100, 200), (300, 600), (500, 1000)}

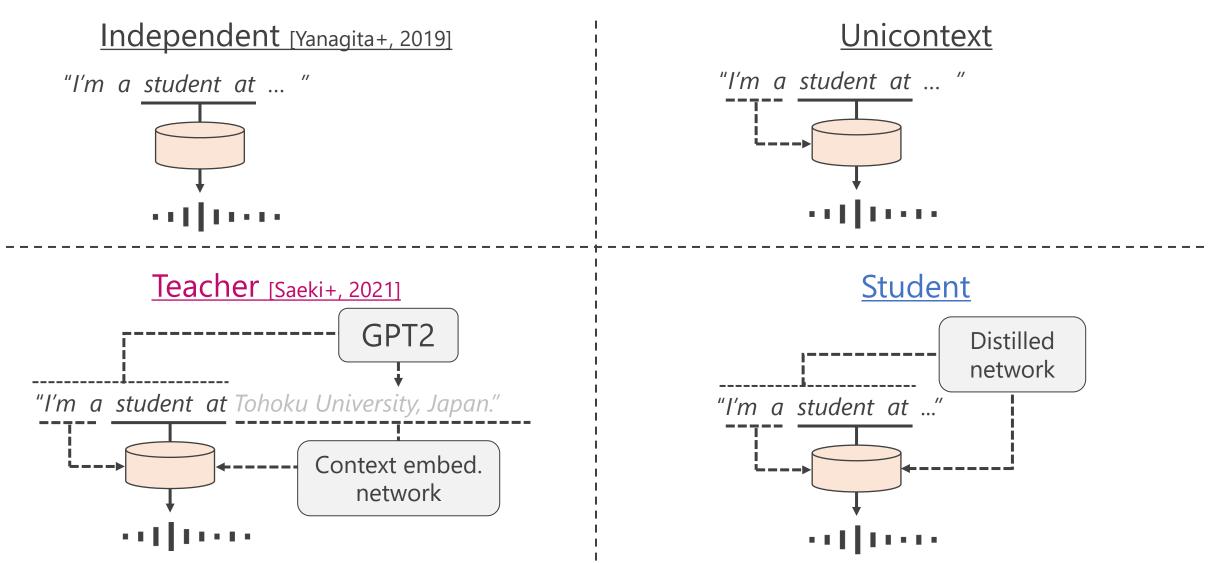
Discussion on Knowledge Distillation with Language Model 8/13



 $e_{\text{pseudo}}^{(S)}$: Contextual embedding with Student model trained using **pseudo lookahead** $e_{\text{pseudo}}^{(T)}$: Contextual embedding with Teacher model trained using **pseudo lookahead** $e_{\text{truth}}^{(S)}$: Contextual embedding with Student model trained using **ground-truth lookahead** $e_{\text{truth}}^{(T)}$: Contextual embedding with Teacher model trained using **ground-truth lookahead**

Experimental Evaluation

Corpus: LJSpeech [Ito+, 2017] (22.05 kHz)



Evaluation Results on Synthetic Speech quality

Objective evaluation: Calculated character error rate (CER) and word error rate (WER) of synthetic speech

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Subjective evaluation: Mean opinion score (MOS) test on naturalness evaluated by 40 native speakers

	Full sentence	Unicontext	Teacher	Student w/o target loss	Student w/ target loss
CER (↓)	5.5 %	20.8 %	7.8 %	8.4 %	12.7 %
WER (↓)	18.2 %	49.4 %	22.2 %	22.2 %	33.8 %
MOS (1)	3.82	3.10	3.51	3.47	3.39

<u>Student > Unicontext</u> & <u>Student \approx Target</u>: Student model predicted effective contextual embedding for

incremental TTS and **achieved comparable naturalness** to Teacher model

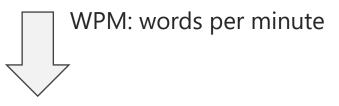
Student model performed better without target loss (correspond to results in previous work [Tang+, 2019])

Evaluation Results on Inference Speed

Independent, Unicontext $\approx 0.15s / step$ Teacher $\approx 1.5s / step$ Student $\approx 0.15s / step$

Student achieved around 10 times faster inference than Teacher

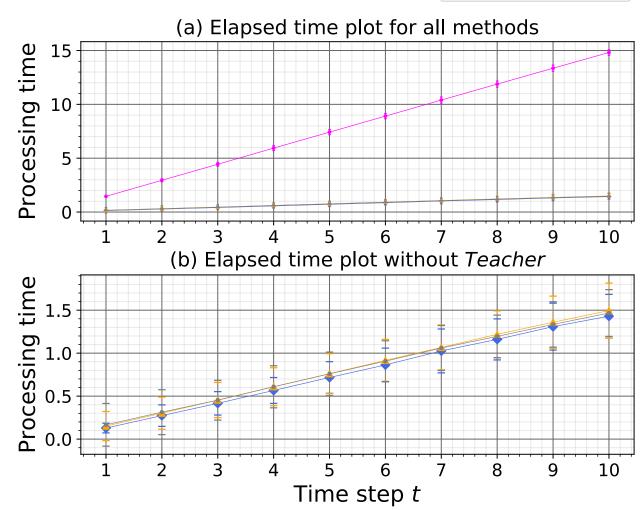
<u>Average English speaker</u>: **180** WPM <u>Teacher</u>: **80** WPM <u>Student</u>: **800** WPM

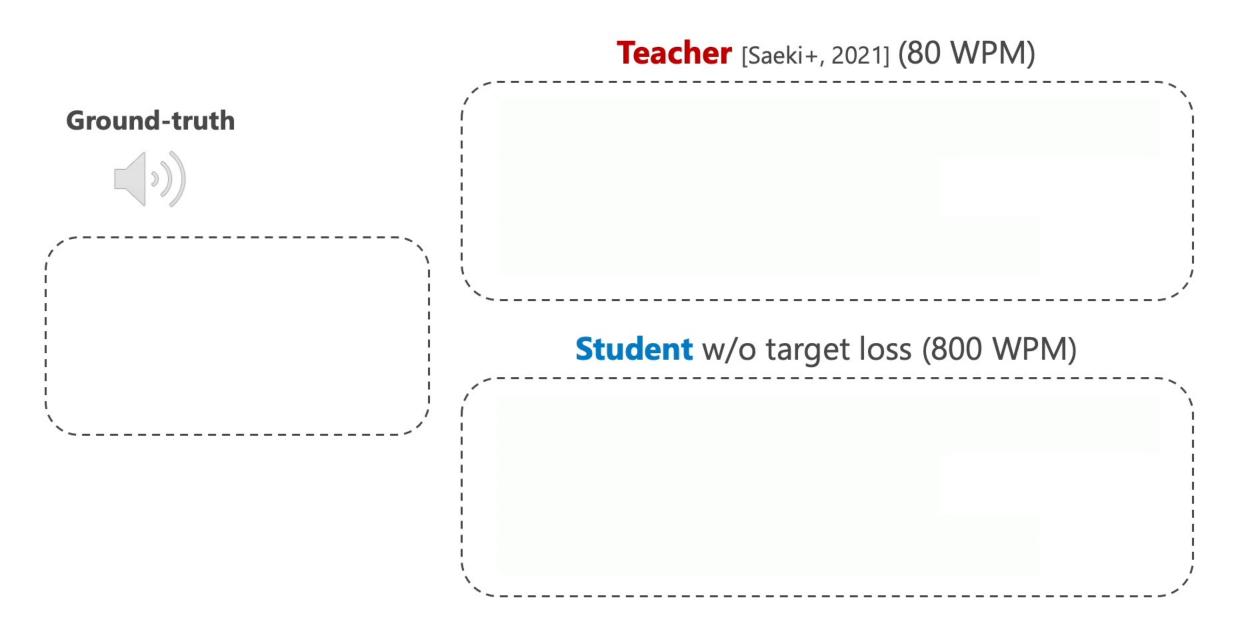


Student achieves inference speed which can be available to **real-time application** while achieving **comparable quality** to Teacher Synthesized two words per step

Used a Nvidia RTX 1080Ti GPU

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Summary and Conclusion

Research goal

Low-latency and high-quality streaming TTS for real-time speech generation

Proposed method

Fast listen-while-predict framework that estimates future context with lightweight model Knowledge distillation of context estimation model with GPT2 to single recurrent model

Evaluation results

Student model predicted effective contextual embedding for incremental TTS Student model achieved **comparable synthetic speech quality to Teacher model** Student model achieved much **faster speaking speed than human English speaker**

Future work

Further improving synthetic speech quality for equivalent quality to sentence-level TTS