Tinglong Zhu

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EDUCATION

Carnegie Mellon University Pittsburgh, PA
- Master of Science in Intelligent Information Systems (MIIS-21); GPA: 3.93/4.0 Aug 2022- May 2024

Duke Kunshan University / Duke University (Dual degree) Kunshan, China
- Bachelor of Science; Major: Data Science GPA: 3.89/4.0 Aug 2018- May 2022

PUBLICATIONS

[1] **Tinglong Zhu**, Xingming Wang, Xiaoyi Qin, Ming Li (2022) Source Tracing: Detecting Voice Spoofing. *Asia-Pacific Signal and Information Processing Association (APSIPA) SS03: Security Techniques of Speaker Recognition 2022*
[2] **Tinglong Zhu**, Xiaoyi Qin, Ming Li (2021) Binary Neural Network for Speaker Verification. *Conference of the International Speech Communication (Interspeech) 2021*
[3] Xingming Wang, Xiaoyi Qin, Ming Li, **Tinglong Zhu**, Chao Wang, Shilei Zhang (2021) The DKU-CMRI System for the ASVspoof 2021 Challenge: Vocoder Based Replay Channel Response Estimation. *ASVspoof 2021 Workshop*

INDUSTRY EXPERIENCE

TikTok Seattle, WA
Machine Learning Engineer, Global-E-Commerce Search Team Jun 2024 – Present
- Achieve 1.5% GPMs gain through adding a generalized retrieval
- Improve fine rank CTCVR model's AUC by 0.5% via upgrading model structure and adding new features
- Develop business strategies, resulting in approximately a 5% improvement on OPMs
- Design, train and deploy fine rank repurchase model to promote repurchase rate
- Build training and online serving data pipeline for newly added features

Alibaba Hangzhou, China
Research Intern, DAMO Academy Speech Team May 2023 – Aug 2023
- Took advantage of Residual Vector Quantization (RVQ), Neural Codec and LLM techniques to design and implement a new lightweight (for lower inference cost), and multi-task Text-to-Speech (TTS) system
- Reimplemented SoundStream codec (based on EnCodec) and SoundStorm model (350M Conformer)
- Wrote codebase (from scratch) for training/evaluating TTS systems (with cluster-compatible I/O)

Microsoft Beijing, China
Research Intern, Cloud + AI Speech Team May 2021 – Aug 2021
- Conducted joint training of ASR TTS ASV models to enlarge data sources of small languages, and enhanced human-computer interaction for more languages and dialects worldwide
- Built pipeline (Listen, Attend and Spell) for Automatic Speech Recognition based on ESPnet, pipeline for Automatic Speaker Verification (ASV) based on SEResNet34 and pipeline for Text-to-Speech (Tacotron2)
- Trained models on Azure clusters (multi-nodes training)
- Achieved following performance improvement: TTS system: Word Error Rate (WER): Reduced from 16.5% to 15.5%. Mean Cosine Similarity: Improves from 0.55 to 0.73

RESEARCH EXPERIENCE

Project: Source Tracing: Detecting Voice Spoofing, *Duke Kunshan University* Feb 2022 – Aug 2022
Proposed a Training Strategy for Classifying Attributes of Spoof Systems (RawNet2, ResNet34 based systems)
- Achieved 88.41%, 77.54%, 51.46% accuracy for Conversion, Waveform Generator, and Speaker Representation attributes, respectively on RawNet2 based system
- Accepted by *APSIPA ASC 2022 Special Session 03* [1]

SKILLS

Programming Languages: C/C++, Python, Java, BASH, MATLAB, R
Software & Open-Source Toolkits: PyTorch, ESPnet, Hive, SQL, Linux, TensorFlow, Azure, AWS, SciPy, Matplotlib