Yi-Chiao Wu 呉宜樵

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EDUCATION

2017-2021 **Nagoya University** **Nagoya, Japan**

- Ph.D. in Graduate school of Informatics (artificial intelligent system group)
- Ph.D. thesis: Incorporating prior knowledge on speech production mechanism into neural speech waveform generation
- Advisor: Tomoki Toda

2005-2011 **National Chiao Tung University** **Hsinchu, Taiwan**

- M.S. in Communication Engineering (specialized in speaker recognition)
- B.S. in Communication Engineering
- Master thesis: Speaker recognition system for intelligent home robot

2009-2011 **Industry-university Cooperative Research Project** **Hsinchu, Taiwan**

- Built a human identification system with a microphone array and face and voiceprint recognitions
- Transferred the speaker recognition system (with windows UI) to COMPAL Electronics

WORK EXPERIENCE

2020-2021 **Graduate School of Informatics, Nagoya University** **Nagoya, Japan**

Researcher

- Proposed a neural-post-filter for improving low-cost Text-To-Speech (TTS) systems.
- Proposed a pitch-dependent structure for the real-time parallel WaveGAN speech generation model
- Develop a baseline system and release the source code for Voice Conversion Challenge 2020

2017-2020 *Research Assistant*

- Proposed a pitch-dependent structure for WaveNet to improve the robustness of unseen data
- Proposed a collapsed speech detection and suppression method for WaveNet vocoder
- Got an overall performance ranking 2/12 in Voice Conversion Challenge 2018 non-parallel VC task

2019 **National Institute of Information and Communications Technology** **Kyoto, Japan**

Summer Intern

- Reduced 30% training time and model size of WaveGlow with the depthwise CNN

2015-2017 **Institute of Information Science, Academia Sinica** **Taipei, Taiwan**

Research Assistant

- Got an overall performance ranking 7/17 in Voice Conversion Challenge 2016
- Combined manifold learning techniques with an exemplar-based speaker voice conversion system
- Integrated exemplar-based post-filtering methods with neural-based speech enhancement systems

2013-2015 **Da Vinci Innovation Lab, ASUS** **Taipei, Taiwan**

Software R&D Engineer

- Developed ASUS Zenbo robot's text-independent speaker recognition engine
- Designed and implemented a speaker recognition mobile application on an Android platform
- Integrated speaker recognition with a microphone array, noise reduction, and face recognition systems

2012-2013 **Multimedia BU II, Realtek** **Hsinchu, Taiwan**

System Designer

- Worked with IC component designers to design and verify TV audio systems
- Ported audio drivers for five mass-production projects (for TOSHIBA, SONY, Skyworth, etc.)
- Worked with software engineers to pass the Dolby and DTS certification for audio systems

RESEARCH INTEREST & SKILLS

- Speech Generation: voice conversion, speech enhancement, text to speech, and speech generation
- Programming: PyTorch, TensorFlow, Python, MATLAB, C/C++, JAVA, UNIX shell script
- Google Scholar Citations: More than 600 citations
- Review experience: More than 10 international journal and conference papers