Shigeki Karita

Osaka, Japan

Work Experience

NTT Communication Science Laboratories

Kyoto, Japan

Research Scientist,

April 2016 - present

Reseach and develop automatic speech recognition (ASR) and machine learning systems in C++, CUDA and Python at NTT. Mostly working with Dr. Atsunori Ogawa, Dr. Marc Delcroix at NTT, and Dr. Shinji Watanabe at JHU.

Graduate School of Engineering, Osaka University

Osaka, Japa

Teaching Assistant,

April 2015 - March 2016

Wrote several textbooks for "Programming Exercise" class that covers signal processing, statistical analysis and tiny compiler in C++ and Java, and advised students in the class with Assistant Prof. Kazuaki Nakamura.

NTT Communication Science Laboratories

Kyoto, Japan

Research Intern,

September 2015 - March 2015

Implemented a convolutional neural network (CNN) acoustic model from scratch in C++, CUDA and OpenCL, and published CNN-based reverberant robust ASR paper at ICASSP 2016 [1] with Dr. Takuya Yoshioka at NTT.

Education

Graduate School of Engineering, Osaka University

Osaka, Japan

M.Eng., Electronic and Information Engineering

April 2014 - March 2016

Advisor: Prof. Noboru Babaguchi

School of Engineering, Osaka University

Osaka, Japan

B.Eng., Electronic and Information Engineering

April 2010 - March 2014

Advisor: Prof. Noboru Babaguchi

Projects

Automatic Speech Recognition at NTT: Research Scientist

- o This project aims to research and develop automatic speech recognition (ASR) systems at NTT.
- o My research interest includes various extensions for acoustic models and end-to-end models in ASR, for example, noise robust ASR [4], far-field ASR [1], sequential training [6], semi-supervised training [9, 12].

ESPnet: end-to-end speech processing toolkit: Core Developer

- o This project aims to be an open source state-of-the-art platform for end-to-end speech processing [10].
- o I mainly develop and maintain ASR and TTS pytorch backend, and continuous integration (CI) for unittesting and sphinx documentation. For example, my early PR made ESPnet 3-4 times faster. https://github.com/espnet/espnet/pull/17

Skills

Research: Speech and signal processing, Machine learning, High performance computing.

Programming: C++, CUDA, OpenCL, D, Python, Java, Scala, Rust

Language: Japanese (native), English (full professional), German (limited, read-only)

Publications

also see https://scholar.google.com/citations?user=enV4FrIAAAAJ

International Conference (peer-reviewed).....

- [1] T. Yoshioka, <u>S. Karita</u>, and T. Nakatani, "Far-field speech recognition using cnn-dnn-hmm with convolution in time," in 2015 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 4360–4364, April 2015.
- [2] <u>S. Karita</u>, K. Nakamura, K. Kono, Y. Ito, and N. Babaguchi, "Owner authentication for mobile devices using motion gestures based on multi-owner template update," in 2015 IEEE International Conference on Multimedia Expo Workshops (ICMEW), pp. 1–6, June 2015.
- [3] S. Araki, N. Ito, M. Delcroix, A. Ogawa, K. Kinoshita, T. Higuchi, T. Yoshioka, D. Tran, <u>S. Karita</u>, and T. Nakatani, "Online meeting recognition in noisy environments with time-frequency mask based mvdr beamforming," in 2017 Hands-free Speech Communications and Microphone Arrays (HSCMA), pp. 16–20, March 2017.
- [4] S. Karita, A. Ogawa, M. Delcroix, and T. Nakatani, "Forward-backward convolutional lstm for acoustic modeling," in *Proc. Interspeech* 2017, pp. 1601–1605, 2017.
- [5] D. T. Tran, M. Delcroix, <u>S. Karita</u>, M. Hentschel, A. Ogawa, and T. Nakatani, "Unfolded deep recurrent convolutional neural network with jump ahead connections for acoustic modeling," in *Proc. Interspeech* 2017, pp. 1596–1600, 2017.
- [6] <u>S. Karita</u>, A. Ogawa, M. Delcroix, and T. Nakatani, "Sequence training of encoder-decoder model using policy gradient for end- to-end speech recognition," in 2018 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 5839–5843, April 2018.
- [7] A. Ogawa, M. Delcroix, <u>S. Karita</u>, and T. Nakatani, "Rescoring n-best speech recognition list based on one-on-one hypothesis comparison using encoder-classifier model," in *2018 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, pp. 6099–6103, April 2018.
- [8] T. Higuchi, K. Kinoshita, N. Ito, <u>S. Karita</u>, and T. Nakatani, "Frame-by-frame closed-form update for mask-based adaptive mvdr beamforming," in 2018 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 531–535, April 2018.
- [9] S. Karita, S. Watanabe, T. Iwata, A. Ogawa, and M. Delcroix, "Semi-supervised end-to-end speech recognition," in *Proc. Interspeech* 2018, pp. 2–6, 2018.
- [10] S. Watanabe, T. Hori, <u>S. Karita</u>, T. Hayashi, J. Nishitoba, Y. Unno, N. Enrique Yalta Soplin, J. Heymann, M. Wiesner, N. Chen, A. Renduchintala, and T. Ochiai, "Espnet: End-to-end speech processing toolkit," in *Proc. Interspeech* 2018, pp. 2207–2211, 2018.
- [11] M. Delcroix, S. Watanabe, A. Ogawa, <u>S. Karita</u>, and T. Nakatani, "Auxiliary feature based adaptation of end-to-end asr systems," in *Proc. Interspeech* 2018, pp. 2444–2448, 2018.
- [12] <u>S. Karita</u>, S. Watanabe, T. Iwata, M. Delcroix, A. Ogawa, and T. Nakatani, "Semi-supervised end-to-end speech recognition using text-to-speech and autoencoders," in 2019 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 6166–6170, May 2019.

Talk	c/Tutorial					 	
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[13] T. Hori, T. Hayashi, <u>S. Karita</u>, and S. Watanabe, "Advanced methods for neural end-to-end speech processing – unification, integration, and implementation," in *INTERSPEECH 2019 Tutorial*, September 2019.

Paten	ıts	 																					

Contributed to more than 5 patents at NTT