Bibliography

- D. Amodei, S. Ananthanarayanan, R. Anubhai, J. Bai, E. Battenberg, C. Case, J. Casper, B. Catanzaro, Q. Cheng, G. Chen, J. Chen, J. Chen, Z. Chen, M. Chrzanowski, A. Coates, G. Diamos, K. Ding, N. Du, E. Elsen, and Z. Zhu, "Deep speech 2: End-to-end speech recognition in english and mandarin," in *ArXiv*, 12 2015.
- [2] L. Lugosch, M. Ravanelli, P. Ignoto, V. Tomar, and Y. Bengio, "Speech model pre-training for end-to-end spoken language understanding," *ArXiv*, vol. abs/1904.03670, 2019.
- [3] Y.-A. Chung, H. Tang, and J. Glass, "Vector-quantized autoregressive predictive coding," in *INTERSPEECH*, 2020.
- [4] A. H. Liu, Y.-A. Chung, and J. Glass, "Non-autoregressive predictive coding for learning speech representations from local dependencies," *ArXiv*, vol. abs/2011.00406, 2020.
- [5] Y.-P. Chen, R. Price, and S. Bangalore, "Spoken language understanding without speech recognition," in 2018 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), 2018, pp. 6189–6193.
- [6] J. Poncelet and H. Van hamme, "Multitask learning with capsule networks for speech-to-intent applications," in *ICASSP 2020 - 2020 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 05 2020, pp. 8494–8498.

- [7] Y. Karunanayake, U. Thayasivam, and S. Ranathunga, "Sinhala and tamil speech intent identification from english phoneme based asr," 2019 International Conference on Asian Language Processing (IALP), pp. 234–239, 2019.
- [8] B. Zhen, X. Wu, Z. Liu, and H. Chi, "On the importance of components of the mfcc in speech and speaker recognition," in *INTERSPEECH*, vol. 37, 01 2000, pp. 487–490.
- [9] V. Këpuska and H. Elharati, "Robust speech recognition system using conventional and hybrid features of mfcc, lpcc, plp, rasta-plp and hidden markov model classifier in noisy conditions," *Journal of Computer and Communications*, vol. 03, pp. 1–9, 01 2015.
- [10] K. Aida-zade, C. Ardil, and S. Rustamov, "Investigation of combined use of mfcc and lpc features in speech recognition systems," *Signal Processing*, 01 2007.
- [11] S. Shum, N. Dehak, R. Dehak, and J. Glass, "Unsupervised speaker adaptation based on the cosine similarity for text-independent speaker verification," *Odyssey*, 01 2010.
- [12] D. Snyder, D. Garcia-Romero, G. Sell, D. Povey, and S. Khudanpur, "X-vectors: Robust dnn embeddings for speaker recognition," in 2018 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), 04 2018, pp. 5329–5333.
- [13] Y. Shi, Q. Huang, and T. Hain, "H-vectors: Utterance-level speaker embedding using a hierarchical attention model," in *ICASSP 2020 - 2020 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 05 2020, pp. 7579–7583.
- [14] A. Senior and I. Lopez-Moreno, "Improving dnn speaker independence with ivector inputs," in 2014 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), 05 2014, pp. 225–229.

- [15] C. yu, A. Ogawa, M. Delcroix, T. Yoshioka, T. Nakatani, and J. Hansen, "Robust i-vector extraction for neural network adaptation in noisy environment," in *INTERSPEECH*, 09 2015.
- [16] S. Garimella, A. Mandal, N. Strom, B. Hoffmeister, S. Matsoukas, and S. H. K. Parthasarathi, "Robust i-vector based adaptation of dnn acoustic model for speech recognition," in *INTERSPEECH*, 2015.
- [17] Y. Miao, H. Zhang, and F. Metze, "Towards speaker adaptive training of deep neural network acoustic models," in *INTERSPEECH*, 2014.
- [18] X. Cui, V. Goel, and G. Saon, "Embedding-based speaker adaptive training of deep neural networks," in *INTERSPEECH*, 08 2017, pp. 122–126.
- [19] N. Tomashenko, A. Caubrière, and Y. Estève, "Investigating Adaptation and Transfer Learning for End-to-End Spoken Language Understanding from Speech," in *Proc. Interspeech 2019*, 2019, pp. 824–828. [Online]. Available: http://dx.doi.org/10.21437/Interspeech.2019-2158
- [20] Y.-A. Chung, W.-N. Hsu, H. Tang, and J. R. Glass, "An unsupervised autoregressive model for speech representation learning," *ArXiv*, vol. abs/1904.03240, 2019.
- [21] Y.-A. Chung and J. R. Glass, "Generative pre-training for speech with autoregressive predictive coding," *ICASSP 2020 - 2020 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, pp. 3497–3501, 2020.
- [22] B. Schuller, S. Steidl, A. Batliner, F. Burkhardt, L. Devillers, C. Müller, and S. Narayan, "Paralinguistics in speech and language - state-of-the-art and the challenge," *Computer Speech and Language, Special Issue on Paralinguistics in Naturalistic Speech and Language*, 01 2013.
- [23] B. Belean, "Comparison of formant detection methods used in speech processing applications," *AIP Conference Proceedings*, vol. 1565, pp. 85–89, 11 2013.

- [24] J. P. Teixeira and A. Gonçalves, "Algorithm for jitter and shimmer measurement in pathologic voices," *Procedia Computer Science*, vol. 100, pp. 271 – 279, 2016.
- [25] X. Li and X. Wu, "Modeling speaker variability using long short-term memory networks for speech recognition," in *INTERSPEECH*, 2015.
- [26] Y. zhao, J. Li, X. Wang, and Y. Li, "The speechtransformer for large-scale mandarin chinese speech recognition," in *ICASSP 2019 - 2019 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 05 2019, pp. 7095–7099.
- [27] Z. Fan, J. Li, S. Zhou, and B. Xu, "Speaker-aware speech-transformer," 2019 IEEE Automatic Speech Recognition and Understanding Workshop (ASRU), pp. 222–229, 2019.
- [28] J. Pan, D. Liu, G. Wan, J. Du, Q. Liu, and Z. Ye, "Online speaker adaptation for lvcsr based on attention mechanism," 2018 Asia-Pacific Signal and Information Processing Association Annual Summit and Conference (APSIPA ASC), pp. 183–186, 2018.
- [29] W.-N. Hsu, Y. Zhang, and J. R. Glass, "Unsupervised learning of disentangled and interpretable representations from sequential data," in *NIPS*, 2017.
- [30] W.-N. Hsu and J. R. Glass, "Extracting domain invariant features by unsupervised learning for robust automatic speech recognition," 2018 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 5614–5618, 2018.
- [31] S. Feng and T. Lee, "Improving unsupervised subword modeling via disentangled speech representation learning and transformation," in *INTERSPEECH*, 2019.
- [32] J. Chorowski, R. J. Weiss, S. Bengio, and A. van den Oord, "Unsupervised speech representation learning using wavenet autoencoders," *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, vol. 27, pp. 2041–2053, 2019.

- [33] S. Schneider, A. Baevski, R. Collobert, and M. Auli, "wav2vec: Unsupervised pre-training for speech recognition," in *INTERSPEECH*, 2019.
- [34] A. T. Liu, S.-W. Li, and H. yi Lee, "Tera: Self-supervised learning of transformer encoder representation for speech," *ArXiv*, vol. abs/2007.06028, 2020.
- [35] P.-H. Chi, P.-H. Chung, T. Wu, C.-C. Hsieh, S.-W. Li, and H. yi Lee, "Audio albert: A lite bert for self-supervised learning of audio representation," *ArXiv*, vol. abs/2005.08575, 2020.
- [36] A. T. Liu, S. Yang, P.-H. Chi, P.-C. Hsu, and H. yi Lee, "Mockingjay: Unsupervised speech representation learning with deep bidirectional transformer encoders," *ICASSP 2020 - 2020 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, pp. 6419–6423, 2020.
- [37] E. Morais, H. Kuo, S. Thomas, Z. Tuske, and B. Kingsbury, "End-to-end spoken language understanding using transformer networks and self-supervised pretrained features," *ArXiv*, vol. abs/2011.08238, 2020.
- [38] E. Palogiannidi, I. Gkinis, G. Mastrapas, P. Mizera, and T. Stafylakis, "End-toend architectures for asr-free spoken language understanding," *ICASSP 2020 -*2020 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 7974–7978, 2020.
- [39] M. Radfar, A. Mouchtaris, and S. Kunzmann, "End-to-end neural transformer based spoken language understanding," in *INTERSPEECH*, 2020.
- [40] Y. Karunanayake, U. Thayasivam, and S. Ranathunga, "Transfer learning based free-form speech command classification for low-resource languages," in ACL, 2019.
- [41] A. Larcher, K.-A. Lee, and S. Meignier, "An extensible speaker identification sidekit in python," 2016 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 5095–5099, 2016.

- [42] A. Hannun, C. Case, J. Casper, B. Catanzaro, G. Diamos, E. Elsen, R. Prenger, S. Satheesh, S. Sengupta, A. Coates, and A. Ng, "Deepspeech: Scaling up endto-end speech recognition," *ArXiv*, 12 2014.
- [43] R. Ardila, M. Branson, K. Davis, M. Henretty, M. Kohler, J. Meyer, R. Morais, L. Saunders, F. M. Tyers, and G. Weber, "Common voice: A massivelymultilingual speech corpus," in *LREC*, 2020.
- [44] V. Panayotov, G. Chen, D. Povey, and S. Khudanpur, "Librispeech: An asr corpus based on public domain audio books," 2015 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), pp. 5206–5210, 2015.
- [45] D. Buddhika, R. Liyadipita, S. Nadeeshan, H. Witharana, S. Jayasena, and U. Thayasivam, "Voicer: A crowd sourcing tool for speech data collection," in 2018 18th International Conference on Advances in ICT for Emerging Regions (ICTer), 2018, pp. 174–181.