Venkata Krishna Naveen, Tadala

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Education

- **University of Southern California** — Los Angeles, California
  M.S (Hons) in Electrical and Computer Engineering; ***GPA****: 3.91/4.0* — *Aug. 2018 – May. 2020*
- **National Institute of Technology, Tiruchirappalli** — Tiruchirappalli, India
  Bachelor of Technology in Electronics and Communication; ***GPA****: 3.83/4.0* — *Aug. 2014 – July. 2018*

Programming Skills

- **Programming Languages** : Python, C++, C, Bash, MATLAB
- **Software tools/ Libraries** : Kaldi, Pytorch, Tensorflow, ESPnet, OpenSMILE

Work Experience

- **ASAPP, Inc** — Mountain View, CA
  Speech Recognition intern — *January 2020 - May 2020*
  - **Alignment based data augmentation approach for ASR**: Conducted variety of experiments comparing different data augmentation techniques and proposed a new technique using alignments.
  - **Design of recipe**: Developed the pipeline for building Acoustic models from scratch
- **Sensory, Inc** — Boulder, CO
  Applied Scientist intern — *May 2019 - December 2019*
  - **Training Acoustic models**: Trained different Acoustic models for the ASR setup involving GMMs and DNN, Decisions on feature dimension, choice of alignment model and single-speaker Vs multi-speaker criterion to improve Acoustic model real-time performance have been made.
  - **Decoding for Recognition**: Designed scripts to perform recognition task given an Acoustic and Language model; Creation of FSTs and lattice decoding.

Research Experience

- **Signal Analysis and Interpretation Laboratory** — Los Angeles, CA
  USC, Research Assistant, Advisor: Dr.Shrikanth Narayanan — *Sept 2018 - Present*
- **Language Extraction and Acoustic Patterns laboratory** — Bangalore, India
  Indian Institute of Science, Research Intern, Advisor: Dr.Sriram Ganapathy — *May 2017 - July 2017*

Projects

- **Neural Speaker embedding for Speaker Adaptation in Automatic Speech Recognition**:
  - Building Deep learning architectures in *Kaldi* to generate speaker embeddings.
  - Conducting experiments to understand their representation and significance in speaker adaptation of Automatic Speech Recognition, development in *C++*, *Python* and *Bash* scripting
- **Role recognition using Machine learning**:
  - Extracted prosodic features using *OpenSMILE* and developed machine learning models using *scikit-learn* in Python.
  - Achieved 10% absolute improvement over the baseline results.
- **Speech Representation learning based on data-driven modulation filtering for Robust ASR**:
  - Implemented Convolutional Restricted Boltzmann Machine models to obtain data-driven rate and scale filters to extract temporal and spectral content within a speech file on Aurora4 dataset.
  - Compared the performance of systems based on choice of input features filtered by rate, scale and a combination of these data driven filters in unsupervised manner in Matlab.
- **Music Genre Classification**:
  - Classification of genres of music, feature extraction using adversarial methods in Keras.

Publications

[1] *Multistream CNN for Robust Acoustic Modeling* - Kyu J. Han , J Pan , **VKN Tadala** , Tao Ma, Dan Povey