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Friday, August 30, 2019

Utilizing Prosody To Improve Turn Detection In Medical Conversations



Company Background

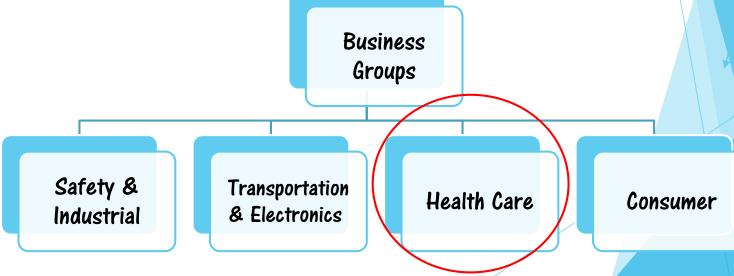


Minnesota Mining and Manufacturing Company

- multinational conglomerate corporation
- Maplewood, Minnesota, United States, 1902

Science is at the heart of everything we do.

At 3M, science is applied in collaborative ways to improve lives daily.





Company Background [...contd.]

Bio Processing

Drug Delivery
Systems

Food Safety

Health Information

Systems

Health Care

Divisions

Medical

Medical Device Components

Oral Care

Personal Health
Care

Pittsburgh Office,

merger with

MModal



Project Goal

Build models using prosody to improve turn-taking in automatic medical transcriptions, eventually to build a better version of the existing transcriber(s).



Project Background

Big picture -

Develop system(s) that automatically maintains Electronic Health Records so as to:

- relieve physicians/caregivers from the documentation burden increasing their efficiency.
- Current EHR product, FLUENCY Direct -
 - A single, cloud-hosted voice profile allows clinicians to dictate into their EHR from anywhere, any device, and any care setting.
 - Front-end speech recognition solution.
 - Even dictation takes additional doctors' time, though much reduced than data entry.



Project Background [..contd.]

Next Step-

Develop system(s) that:

- eliminates any sort of documentation overhead.
- automatically transcribes real-life medical conversations in real-time.
- extracts and summarizes useful Patient Health Information.

Current State-

Transcribes recorded medical conversations using following inhouse speech recognizers:

- Lower Frame Rate LSTM Acoustic model that uses
 LSTM based Voice Activity Detector.
- Speaker ID model that uses LSTM based D-vector embedding for speaker voice recognition and hybrid realignment.



Shortcomings

```
WAV.all/100002.wav DR 100002 0.130612 18.5959 all right and then we close it it is working and we ignore it all right so we were saying that your A1c is a little higher UH UH this UH this month UH specifically it's in WAV.all/100002.wav DR 100002 29.5306 30.8694 days ago from Quest WAV.all/100002.wav PT 100002 30.902 30.9673 yeah WAV.all/100002.wav DR 100002 32.0776 35.2939 the A1c was eight point three usually it's in the 7s WAV.all/100002.wav PT 100002 35.4898 35.7184 right
```

Issues/Flaws:

- Not all turns in the existing auto-transcripts are properly aligned or correctly detected.
- Difficult to predict/assign Speaker IDs to short turns.
- Long gaps: mostly not silence.



Motivation

Why Prosody?

- Time to think beyond just the words.
- Related literature shows that prosody:
 - is useful in predicting turns in dialogs at speech-onset or at pauses.
 - has universal utility across languages and speech genres like task-oriented face-to-face dialogs, general telephonic conversations, etc.



Related Work

Ward et al. used only prosody (pitch, volume, etc.) without any lexical annotation to build a general continuous TensorFlow LSTM RNN model to successfully predict:

- whether a speaker will yield/hold turn after a pause and at the speech onset.
- whether a speaker will continue speaking over a certain future window (0-250ms to 0-3s).

	after a 250ms pause			after a 500ms pause			
Instances % Hold	3,405 59.8%	7,54 58.8		2,079 57.6%	4,60 57.6		
Model	Skantze	Replica	Ours	Skantze	Replica	Ours	
Shift: Precision Shift: Recall Shift: F-measure	0.726 0.703 0.714	0.776 0.528 0.628	0.784 0.601 0.680	0.711 0.738 0.724	0.780 0.549 0.644	0.800 0.660 0.720	
Hold: Precision Hold: Recall Hold: F -measure	0.805 0.822 0.813	0.730 0.893 0.803	0.759 0.884 0.817	0.802 0.780 0.791	0.727 0.886 0.799	0.778 0.879 0.825	

Prediction]	English	Japanese					
Window	MAE	% reduction	MAE	% reduction				
0 to 250ms	0.11	67%	0.21	38%				
0 to 500ms	0.19	42%	0.30	12%				
0 to 1s	0.27	18%	0.36	-6%				
0 to 2s	0.34	-3%	0.40	-18%				
0 to 3s	0.36	-9%	0.41	-21%				
baseline	0.33		0.34					
Mean Absolute Error- English and Japanese MapTask								

			% reduction				
Prediction	American				Canadian		
Window	English	Japanese	Mandarin	Spanish	French		
0 to 250ms	46%	44%	53%	51%	51%		
0 to 500ms	28%	23%	36%	34%	35%		
0 to 1s	14%	3%	22%	20%	19%		
0 to 2s	4%	-5%	9%	7%	7%		
0 to 3s	0%	-8%	7%	2%	5%		
baseline	0.39	0.34	0.45	0.41	0.43		
Mean Absolute Error-Telephone Corpora							



Data

Audio Set:

- Recorded face-to-face medical conversations
- Not professional quality
- Approx. 36k dialogs (Doctor-Patient)
 [Dev: Ik, Test:Ik, rest-Train but each train batch was of 7k only]
- Remaining data is either doctor dictations (single speaker) or multi-party conversations (Doctor-Patient-Caregiver).
- Typical duration is 8-10 minutes, sometimes as long as 45-60mins.
- Mono (single channel), .wav audios Transcripts:
 - .stm files, 1 for each audio
 - generated by speech recognizers, not humans



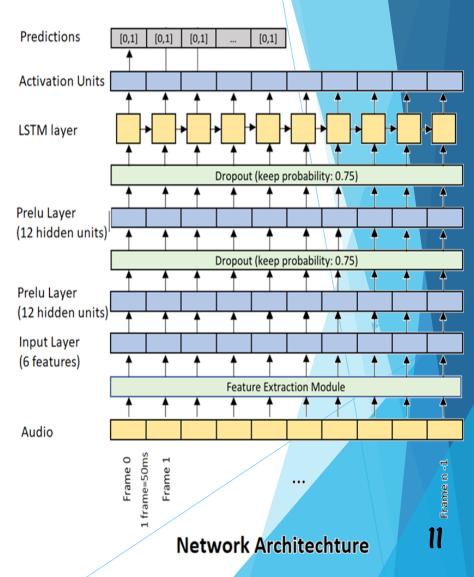
Features

- 6 low-level extracted prosodic features:
 - Absolute pitch (log Hz)
 - Relative pitch (z-normalized pitch)
 - Volume (energy in dB)
 - Cepstral Flux
 - Speaking frame (from pitch)
 - Speaking frame (from energy)
- 64 dimensions Log Filter Bank Energy¹.
- Features are extracted for every 10ms window across the entire audio.
- Pitch Tracker: YAAPT algorithm.²
- Cepstral Flux/LFBE: James Lyons's implementation.
 - 1. Zeynab Raeesy et. al. LSTM-based Whisper Detection, Amazon Alexa
 - 2. Stephen A. Zahorian and Hongbing Hu, "A spectral/temporal method for robust fundamental frequency tracking," J. Acoust. Soc. Am. 123(6), June 2008
 - 3. https://github.com/anath2110/prosodyMonsterPython.git



Key Methodology

- Labels: Turn-Ends as 1,
 Speech Zones as 0, rest
 (unknown zones) as -1
- Feature Vector: 1200 frames, each 10ms
- Prediction Window: 1 frame (i.e. next 10ms)
- Model: Many-to-Many LSTM RNN
- Activation Function: Sigmoid
- Loss Function: Cross-Entropy
- Mass the unknowns while training
 - Implementation: TensorFlow with Python





Variations

- Uni-directional LSTM RNN, 2 hidden PRelu layers of 30 hidden units and a single LSTM layer of 30 units, trained with 6 low-level prosody features.
- Uni-directional LSTM RNN, 2 hidden PRelu layers of 96 hidden units and a single LSTM layer of 96 units, trained with 64 LFBE features.
- Uni-directional LSTM RNN, 2 hidden PRelu layers of 30 hidden units and a single LSTM layer of 30 units, trained with extended feature vector that includes features from the present frame(10ms) concatenated with corresponding 10 past and 10 future frames.



Variations

- Bi-directional LSTM RNN, 2 hidden PRelu layers of 30 hidden units and a single LSTM layer of 30 units, trained with 6 low-level prosody features.
- Bi-directional LSTM RNN, 2 hidden PRelu layers of 96 hidden units and a single LSTM layer of 96 units, trained with 64 LFBE features.
- Bi-directional LSTM RNN, 2 hidden PRelu layers of 30 hidden units and a single LSTM layer of 30 units, trained with extended feature vector that includes features from the present frame(10ms) concatenated with corresponding 10 past and 10 future frames.



Results

- Models trained with LFBE features performed worse than low-level prosody features.
- Bi-directional LSTM RNN models performed better than LSTM RNN models.
- Differences in loss non-conclusive.

Future Plans

- Train models on the entire train set of dialogs and for more epochs or lower learning rate.
- Use mid-level features as well.
- Evaluate against human transcriptions.
- Compare and combine performance of :
 - Prosody models
 - Speaker ID models
 - Language Model based turn predictions



Challenges -> Solutions

Confidentiality Issues:

Working with confidential Patient Health Information data-

- prevents transfer of data from its confidential remote Linux storage.
- makes it difficult to listen to audios for spotchecking, failure analysis, etc.
- required me to undertake HIPAA training and obtain additional authorization that delayed start of project for nearly a month.

Work-arounds:

- Transfer few audios to local Windows storage, listen and delete immediately.
- Wait until properly authorized and trained.



Challenges -> Solutions

Software Installation Issues:

Working in remote LINUX server as a non-root user-

makes it difficult to install various packages, e.g. installing Python IDEs, debuggers, media-players, building TensorFlow from source using bazel.

Work-arounds:

- Use Python command-line interpreter.
- Use Vi-editors to debug.
- Open Jupyter notebook installed in your remote server from your local host browser. License Issues:

MATLAB's GNU GPL conflicted with corporate license laws.

Work-arounds:

Translate and modify to Python.



Challenges -> Solutions

Data issues:

Working with (extracting prosody features from and training deep-learning models on) excessively large amount (few TBs) of speech data resulted in:

- Hard-disk storage overheads.
- Memory (RAM) error.
- Slow computation.

Work Arounds:

- Use additional disk storage provided(3 TB).
- Delete the unknown labels and corresponding features from the respective arrays instead of masking.
- Use parallelisation
- Use GPUs



Perks

- Professionally enriching experience.
- Hands-on knowledge of using most of deep-learning techniques with real-life data.
- Co-operative supervisor and team.
- Meet with and showcase my work to the international team from UK and Germany.
- Expand my Linkedln network.
- Boost in confidence.
- Possible new ideas for my research.
- Escape Room activities.
- One-day trip passes.
- Free Lunch every Wednesdays and Fridays.
- Visit to Minnesota (3M HQs).
- Tickets to my first ever Baseball game.



Thank you for your attention!

Questions?

Comments?

