Speech recognition using Kaldi

Thesis about implementation Kaldi ASR for Alex SDS

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Goals of thesis

Improve speech recognition for Alex Spoken Dialogue Systems
Particularly public transport information application (800 899 998).

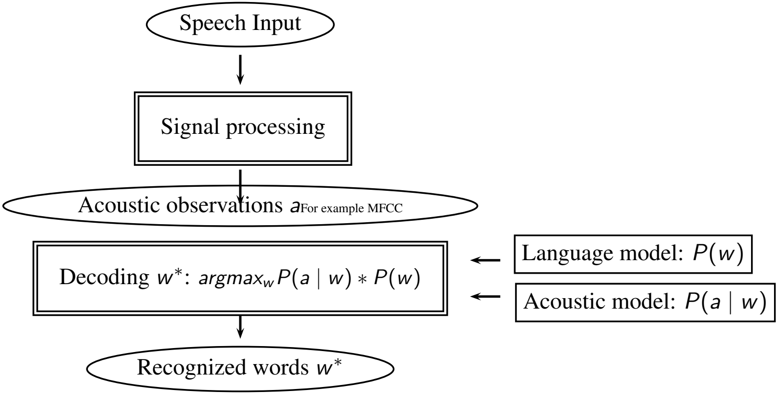
Goals of the thesis were:

- to build acoustic models using the Kaldi toolkit,
- to develop new real-time recogniser which supports incremental speech recognition,
- to integrate the recogniser into our Alex SDS.

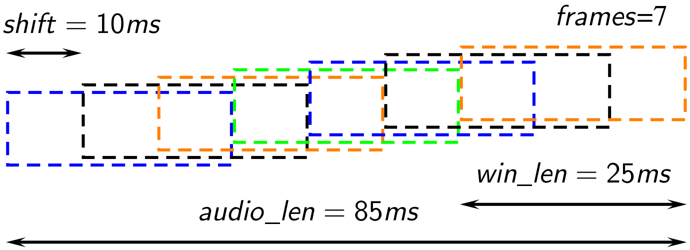
Content

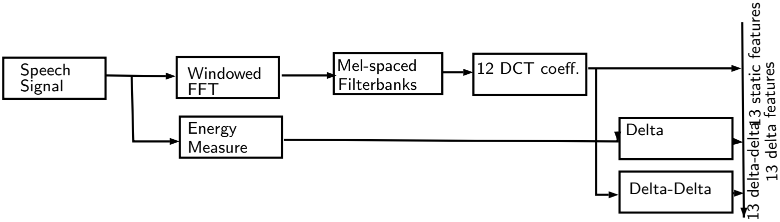
1. Task
2. ASR introduction
3. Evaluation in Public Transport Information domain
4. On-line recogniser
5. Acoustic modelling
6. Summary
7. Details

ASR components



Acoustic features, features preprocessing





Continuous Speech recognition

Pattern matching

HMM — speech time series modelling (phones/triphones for words)

- We trained several HMM acoustic models.

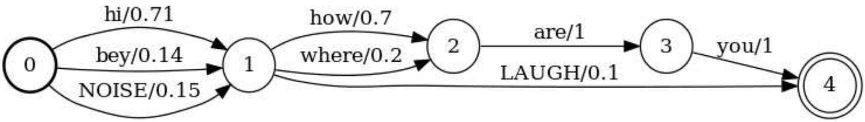
Graph search - decoding

Viterbi algorithm — dynamic programming

- We search for best parameters (beam, max-active-states).
- Normalise its output.
- Change interface.

Output formats

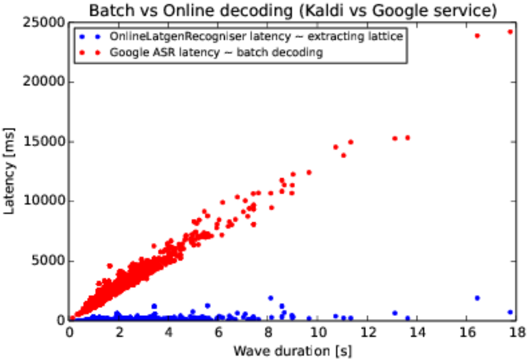
```
0.5 hi how are you
0.2 hi where are you
0.1 bey how are you
```



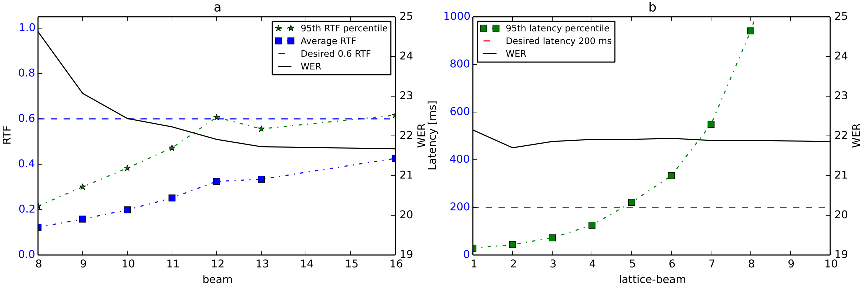
Evaluation measures

- Real Time Factor (RTF) of decoding – the ratio of the recognition time to the duration of the audio input,
- Latency – the delay between utterance end and the availability of the recognition results,
- Word Error Rate (WER).

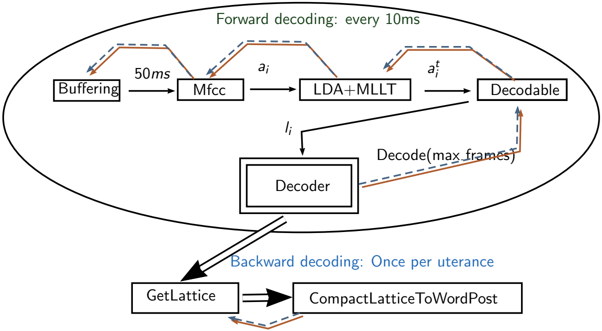
On-line vs batch decoding



Public Transport Information domain - results



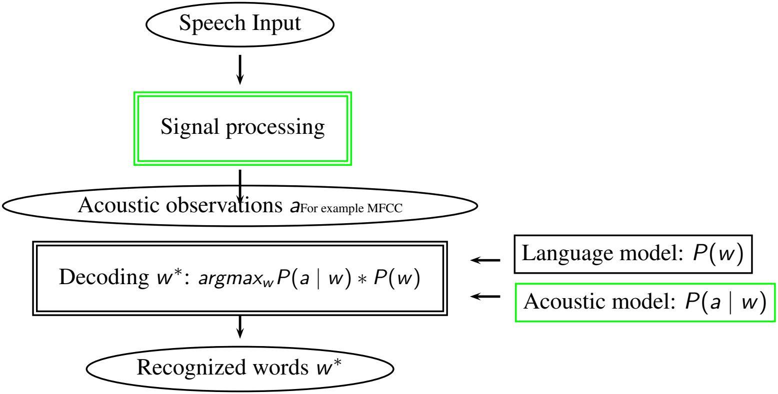
Components for on-line decoding



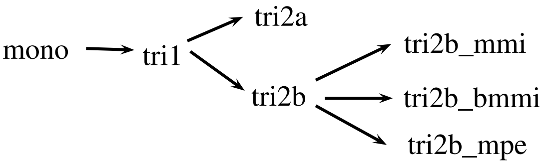
(Py)OnlineLatgenRecogniser interface

- *AudioIn* – queueing new audio for pre-processing
- *Decode* – decoding a fixed number of audio frames
- *PruneFinal* – preparing internal data structures for lattice extraction
- *GetLattice* – extracting a word posterior lattice
- *GetBestPath* – extracting a one best word sequence
- *Reset* – preparing the recogniser for a new utterance

Acoustic modeling



Acoustic models training



Training method name	Script shortcut
Monophone	mono
Triphone	tri1
$\Delta + \Delta\Delta$	tri2a
LDA+MLLT	tri2b
LDA+MLLT+MMI	tri2b_mmi
LDA+MLLT+bMMI	tri2b_bmmi
LDA+MLLT+MPE	tri2b_mpe

Vystadial dataset

Collected by UFAL Dialogue system group.

dataset	audio[hour]	# sentences	# words
English			
training	41:30	47,463	178,110
development	01:45	2,000	7,376
test	01:46	2,000	7,772
Czech			
training	15:25	22,567	126,333
development	01:23	2,000	11,478
test	01:22	2,000	11,204

ASR training results

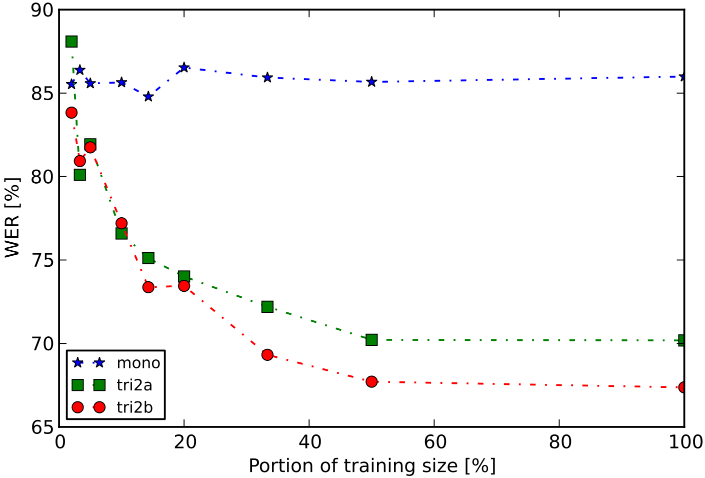
language/method	bigram
Czech	
tri $\Delta + \Delta\Delta$	56.6
tri LDA+MLLT	53.9
tri LDA+MLLT+MMI	49.5
tri LDA+MLLT+bMMI	49.3
tri LDA+MLLT+MPE	49.2
English	
tri $\Delta + \Delta\Delta$	16.2
tri LDA+MLLT	15.8
tri LDA+MLLT+MMI	10.4
tri LDA+MLLT+bMMI	10.2
tri LDA+MLLT+MPE	11.1

HTK and Kaldi acoustic models

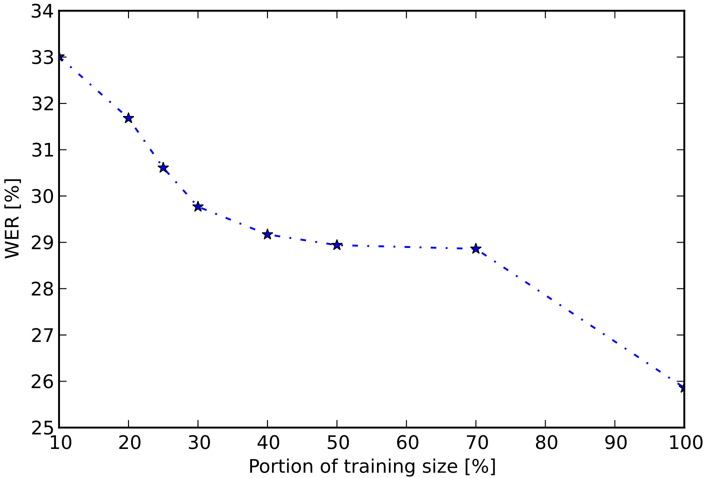
HTK method	**bigram**
Czech	
tri $\Delta + \Delta\Delta$	60.4
English	
tri $\Delta + \Delta\Delta$	17.5

Kaldi method	**bigram**
Czech	
tri $\Delta + \Delta\Delta$	56.6
English	
tri $\Delta + \Delta\Delta$	16.2

Acoustic model accuracy based training data size



Speech recognition accuracy based on LM training data size



Achievements

- Working real-time on-line speech recogniser
- Developed acoustic modeling scripts for Czech and English - accepted to Kaldi svn trunk
- Integration of ASR into Alex Dialogue Systems Framework
- Improved speech recognition for toll-free line 800 899 998

Results

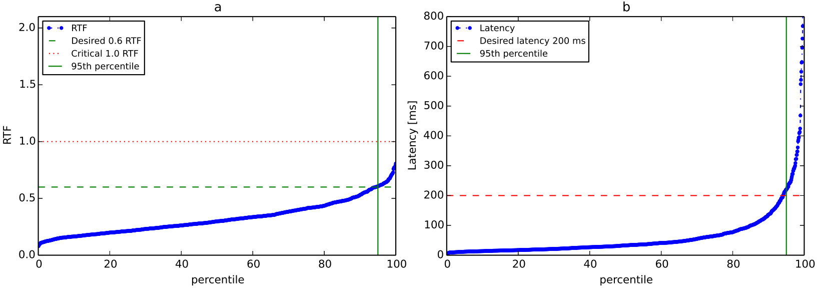
- WER 22, latency under 200 ms on Public Transport Information domain (Czech)
- WER 50 for Czech on Vystadial dataset (Czech - complex domain)
- WER 12 for English on Vystadial dataset

Functional (Py)OnlineLatgenRecogniser demo

```python
d = PyOnlineLatgenRecogniser()
d.setup(argv)
while audio_to_process():
    d.audio_in(get_raw_pcm_audio())
    dec_t = d.decode(max_frames=10)
    while dec_t > 0:
        decoded_frames += dec_t
        dec_t = d.decode(max_frames=10)
d.prune_final()
lik, lat = d.get_lattice()
```

Speed - RTF and Latency

Fast enough for 95 % of utterances.



Problem

Spoken dialogue systems needs speech recognition
OpenJulius — crashes, PocketSphinx — no posteriors, RWTH decoder — license
Cloud based services Google and Nuance — no customisation + license issues

Semiring

Name	$\mathcal{K}$	$\oplus$	$\otimes$	$\bar{0}$	$\bar{1}$
Real	$[0, \infty)$	$+$	*	0	1
Log	$(-\infty, \infty)$	$-log(e^{-x} + e^{-y})$	$+$	∞	0
Tropical	$(-\infty, \infty)$	min	$+$	∞	0

Links and references

Thank you for your attention!

Related links

- Thesis and this slides
  https://github.com/oplatek/kaldi-thesis
- OnlineLatgenRecogniser implementation and AM training scripts
  https://github.com/UFAL-DSG/pykaldi
- Alex implementation
  https://github.com/UFAL-DSG/alex
- Contact & CV
  http://www.linkedin.com/in/ondrejplatek

References

- Vystadial dataset – Matěj Korvas, Ondřej Plátek, Ondřej Dušek, Lukáš Žilka, and Filip Jurčíček, Free English and Czech telephone speech corpus shared under the CC-BY-SA 3.0 license, Proceedings of the Eight International Conference on Language Resources and Evaluation (LREC 2014), 2014, p. To Appear.