# Free on-line speech recogniser based on Kaldi ASR toolkit producing word posterior lattices **Ondřej Plátek and Filip Jurčíček** {oplatek, jurcicek}@ufal.mff.cuni.cz

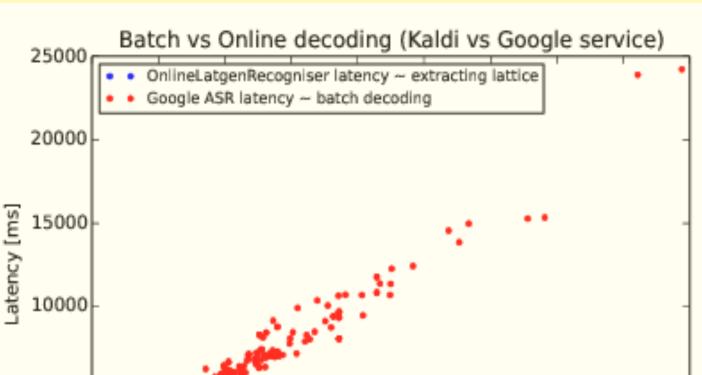
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#### **Motivation: Batch versus Online Decoding**

# **Batch Decoding**

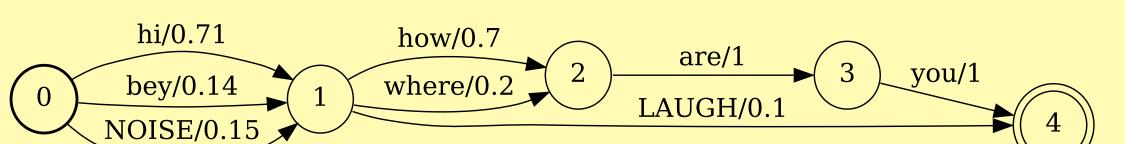
- Waits for the end of the utterance to start decoding
- Latency increases linearly with the utterance length

# **Online Decoding**



### **OnlineLatgenRecognizer Design**

- Simple and responsive
- Robust
- Guaranteed latency
- Iterative decoding
- Supports LDA + MLLT, bMMI, MPE
- Straightforward C++ interface
- Python extension
- Outputs Word Posterior Lattices



- Incremental processing in small chunks
- Result: **low latency**

## The Kaldi ASR Toolkit

- Based on Finite-State Transducers
- State-of-the-art acoustic modelling techniques
- Wave duration [s]
- Well maintained by an enthusiastic community



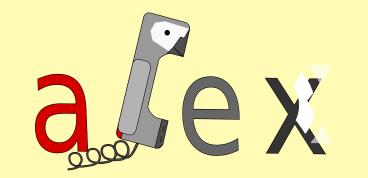
 Lacked support for online decoding

#### Motivation: Get Kaldi's high performance with low latency for use in a Spoken Dialogue System

### **Evaluation in a Spoken Dialogue System**

 Tested in production environment in the Alex spoken dialogue system framework

Czech public transport information domain



# C++API

AudioIn(audio) - Accepts audio.

#### **Decode(max frames)**

- Decodes at most max frames

**PruneFinal()** - prepares decoder for lattice extraction.

**GetLattice()** - extracts lattice

Reset() - prepare for new utterance

**GetBestPath()** - single output

# **Thin Python Wrapper**

OnlineLatgenRecogniser rec; rec.Setup(...); size\_t decoded\_now = 0, max\_decode = 10; char \*audio array = NULL;

while (recognitionOn()){ size\_t audio\_len = getAudio(audio\_array); rec.AudioIn(audio\_array, audio\_len); do { decoded\_now = rec.Decode(max\_decode);

} while(decoded\_now > 0);

rec.PruneFinal(); double tot lik; fst::VectorFst<fst::LogArc> word post lat; rec.GetLattice(&word\_post\_lat, &tot\_lik); rec.Reset();

class AsrSimplifiedInAlex: def rec\_in(self, frame): self.decoder.frame\_in(frame.payload) dec t = self.decoder.decode(max frames) while dec t > 0: frame total += dec t dec t = self.decoder.decode(max frames)

#### **Parameter grid search**

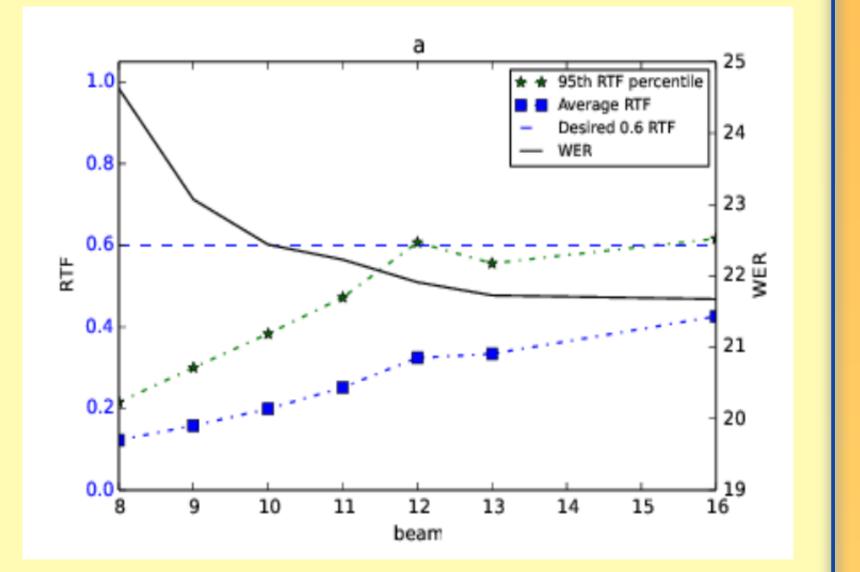
- **beam** controls a dynamic number of alternative ASR hypotheses
- max-active-states -

a threshold on the number of alternative hypotheses

• lattice-beam - level of approximation during phoneme lattice extraction

# Evaluation

- On 1000 recorded utterances from the Alex system, from previously unseen dialogues
- Utterance length varies
- WER: 22%
- Decoder latency well below 200 ms in 95% cases
- Noisy utterances slow down the decoder
- Latency and decoding speed do not depend on utterance



def hyp\_out(self): self.decoder.prune final() utt\_lik, lat = self.decoder.get\_lattice()

# **Training Scripts for Acoustic Modelling**

- Speaker-independent models for Kaldi
- LDA+MLLT+bMMI
- Advanced acoustic models retrained based on simpler models, monophones trained from flat start

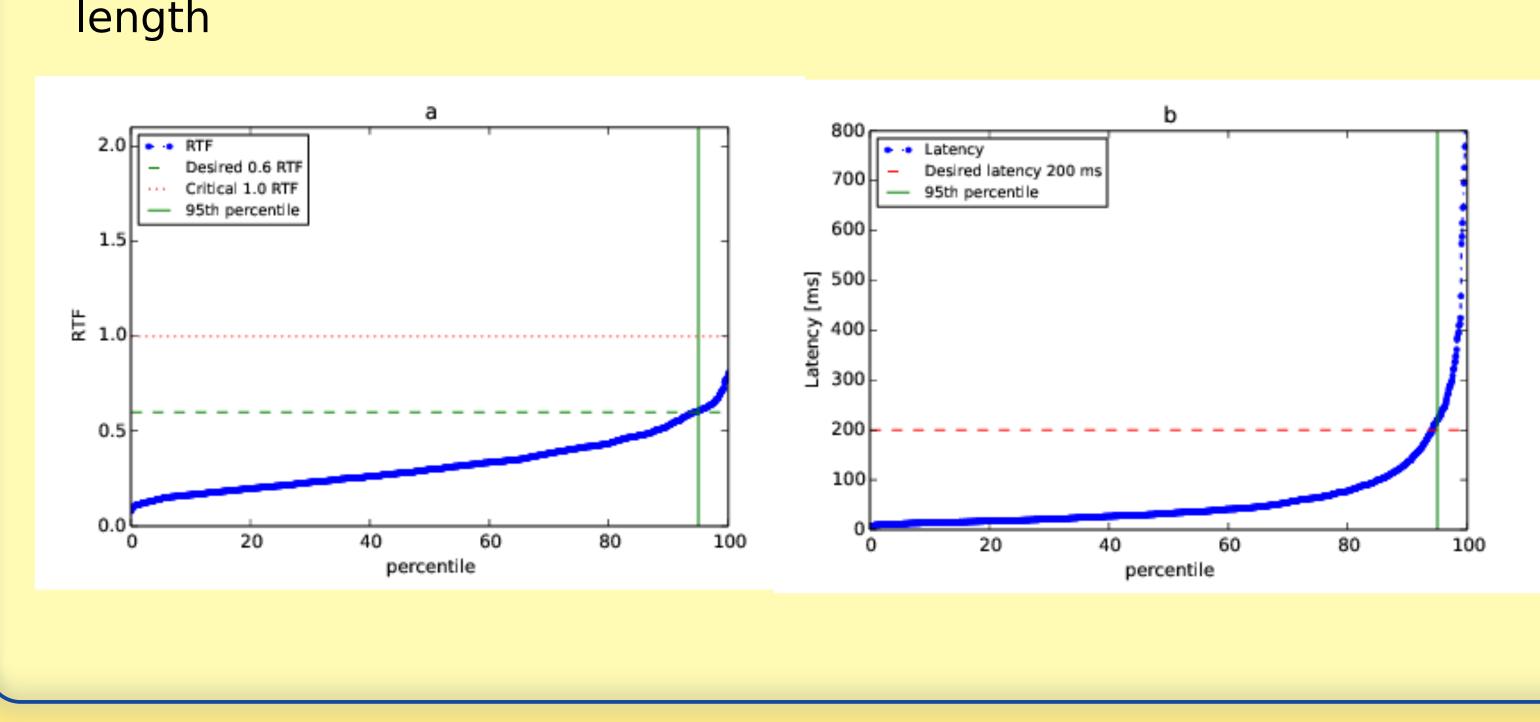
tri2a ,tri2b\_mmi mono-tri1/-tri2b -tri2b bmmi tri2b\_mpe

# **Training Data Sizes**

audio[hour] # sentences # words dataset English

#### Results

Method	bigram WER
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



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training	41:30	47,463	178,110	Method	bigram WER
development	01:45	2,000	7,376		
test	01:46	2,000	7,772	$\sim$ tri $\Delta + \Delta \Delta$	16.2
Czech				tri LDA+MLLT	15.8
training	15:25	22,567	126,333	tri LDA+MLLT+	-MMI 10.4
development	01:23	2,000	11,478	tri LDA+MLLT+	-bMMI 10.2
test	01:22	2,000	11,204	tri LDA+MLLT+	-MPE 11.1

#### Summary

- Apache 2.0 license
- Simple C++ API, easy to use Python thin wrapper
- Used in the Alex spoken dialogue system
- High quality word posterior lattices
- Training scripts for free acoustic data