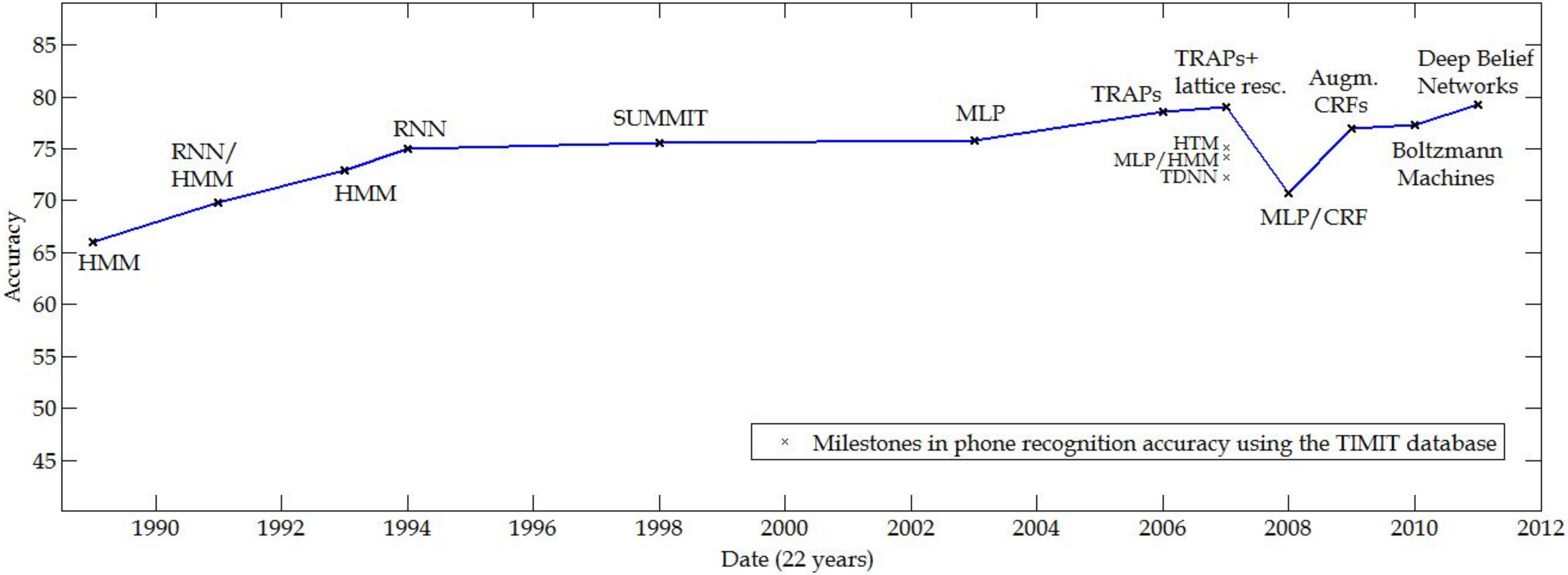
Speech Recognition in Java

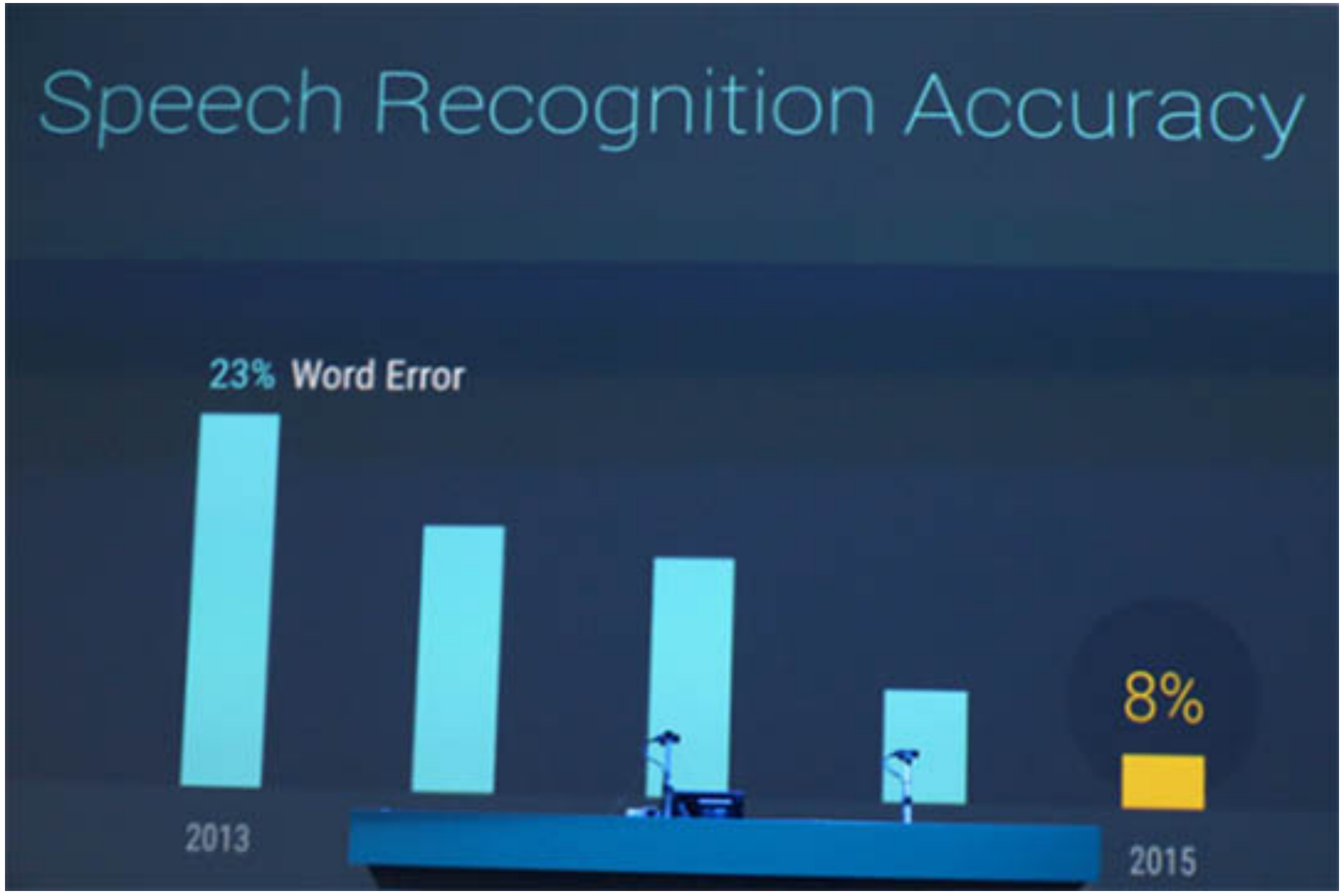
Breandan Considine

JetBrains, Inc.

Automatic speech recognition in 2011

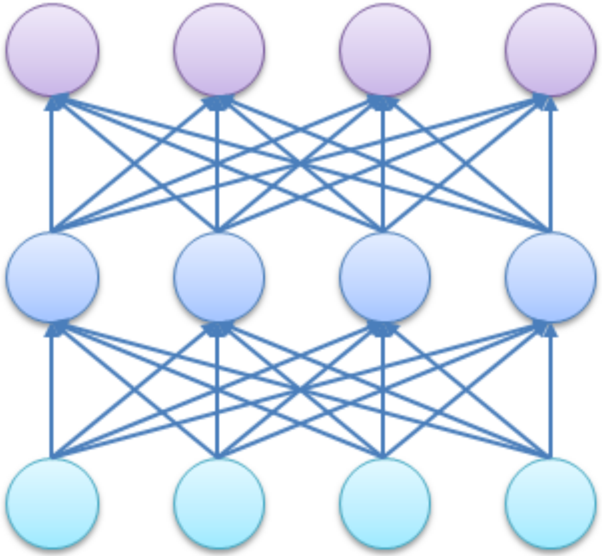


Automatic speech recognition in 2015



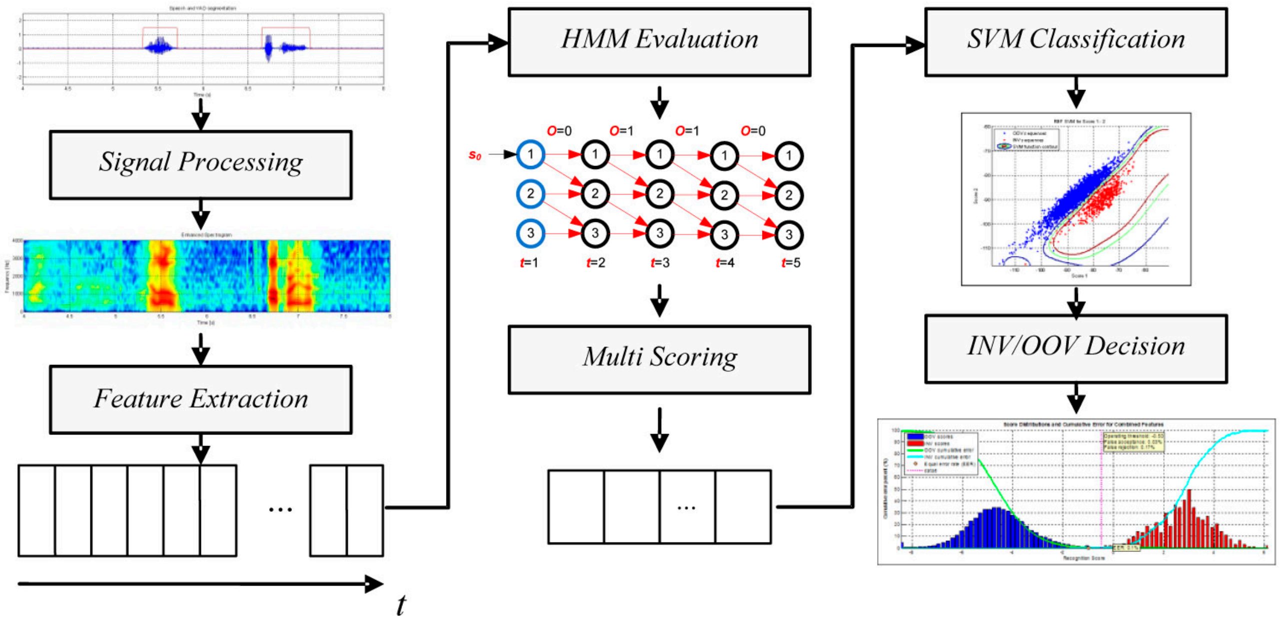
What happened?

- Bigger data
- Faster hardware
- Smarter algorithms



Traditional ASR

- Requires lots of handmade feature engineering
- Poor results: >25% WER for HMM architectures



State of the art ASR

- <10% average word error on large datasets
- DNNs: DBNs, CNNs, RBMs, LSTM
- Thousands of hours of transcribed speech
- Rapidly evolving field
- Takes time (days) and energy (kWh) to train
- Difficult to customize without prior experience

FOSS Speech Recognition

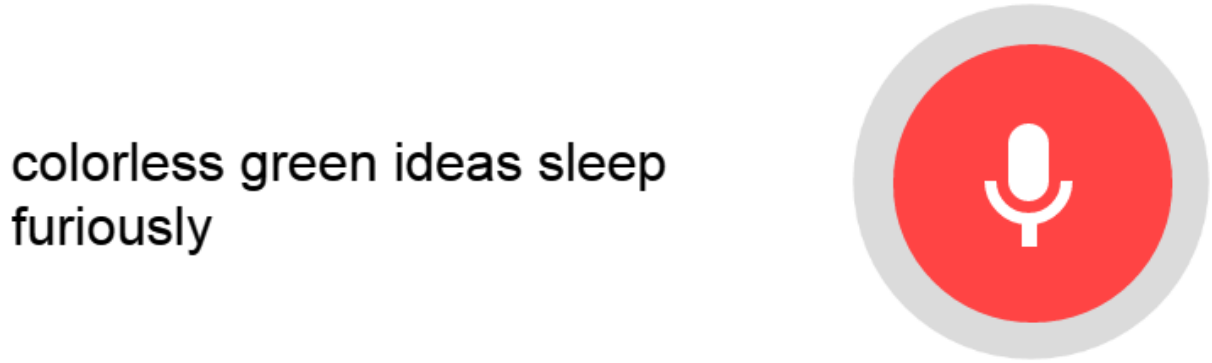
- Deep learning libraries
  - C/C++: Caffe, Kaldi
  - Python: Theano, Caffe
  - Lua: Torch
  - Java: dl4j, H2O
- Open source datasets
  - LibriSpeech – 1000 hours of LibriVox audiobooks
- Experience is required

Let’s think…

- What if speech recognition were perfect?
  - Models are still black boxes
- ASR is just a fancy input method
- How can ASR improve user productivity?
- What are the user’s expectations?
  - Behavior is predictable/deterministic
  - Control interface is simple/obvious
  - Recognition is fast and accurate

Why offline?

- Latency – many applications need fast local recognition
- Mobility – users do not always have an internet connection
- Privacy – data is recorded and analyzed completely offline
- Flexibility – configurable API, language, vocabulary, grammar



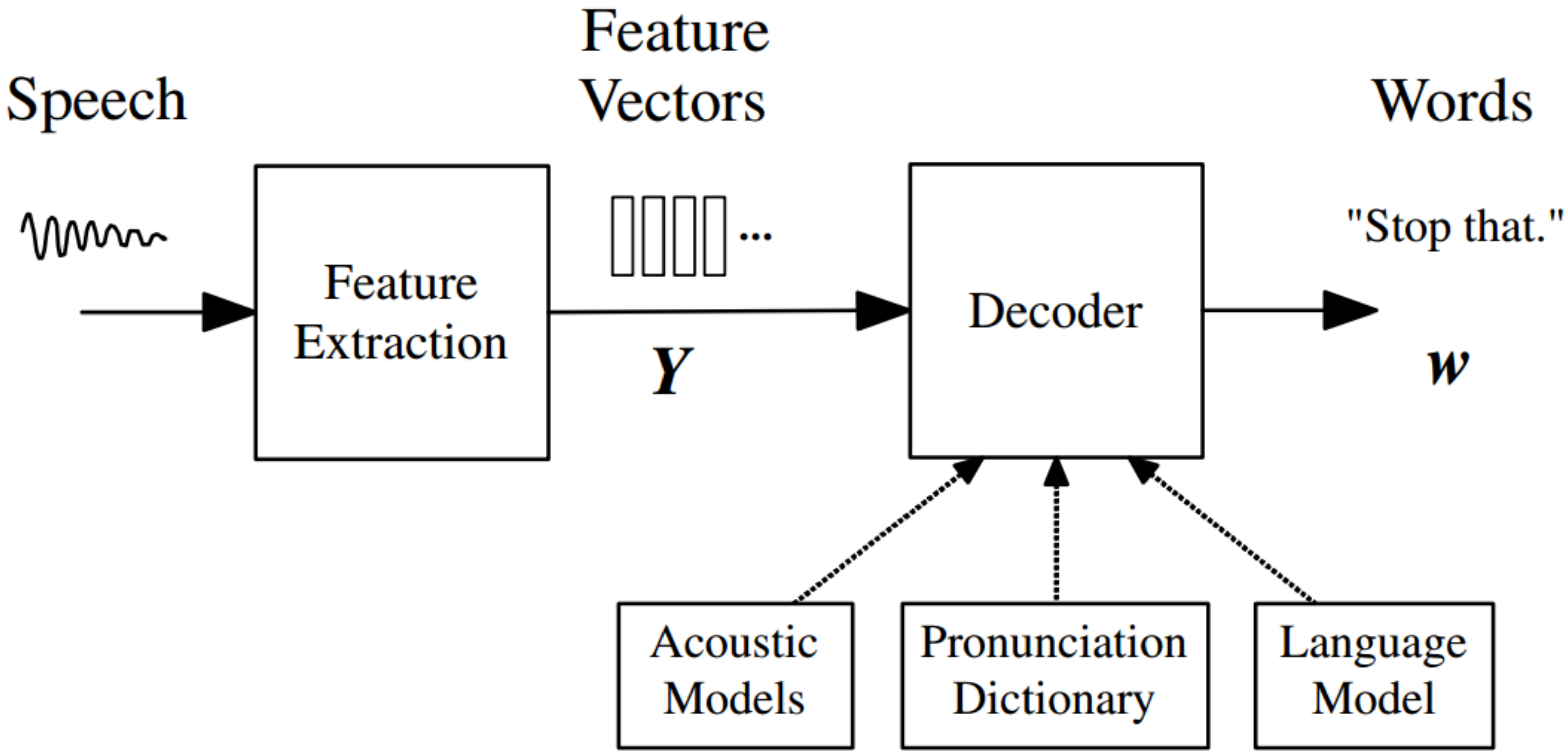
Introduction

- What techniques do modern ASR systems use?
- How do I build a speech recognition application?
- Is speech recognition accessible for developers?
- What libraries and frameworks exist for speech?

Maven Dependencies

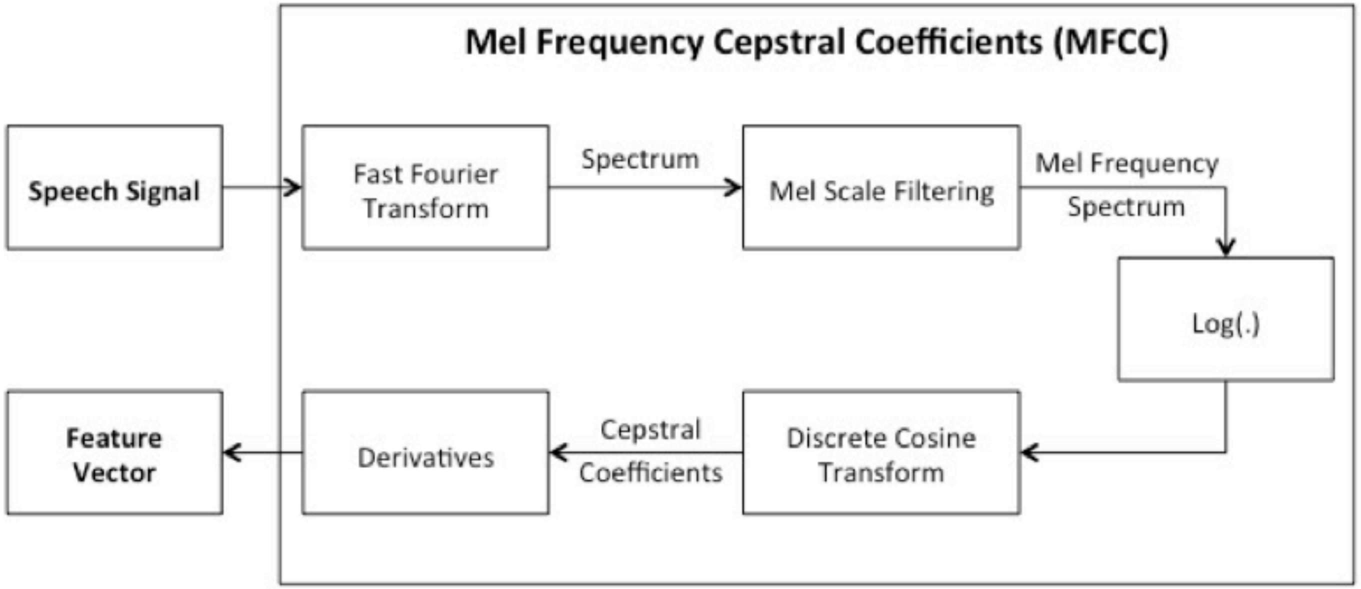
```xml
<dependency>
    <groupId>edu.cmu.sphinx</groupId>
    <artifactId>sphinx4-core</artifactId>
    <version>1.0-SNAPSHOT</version>
</dependency>

<dependency>
<groupId>edu.cmu.sphinx</groupId>
<artifactId>sphinx4-data</artifactId>
<version>1.0-SNAPSHOT</version>
</dependency>
```


Speech
Feature
Vectors
Words
Feature
Extraction
Decoder
"Stop that."
Y
w
Acoustic
Models
Pronunciation
Dictionary
Language
Model

Feature Extraction

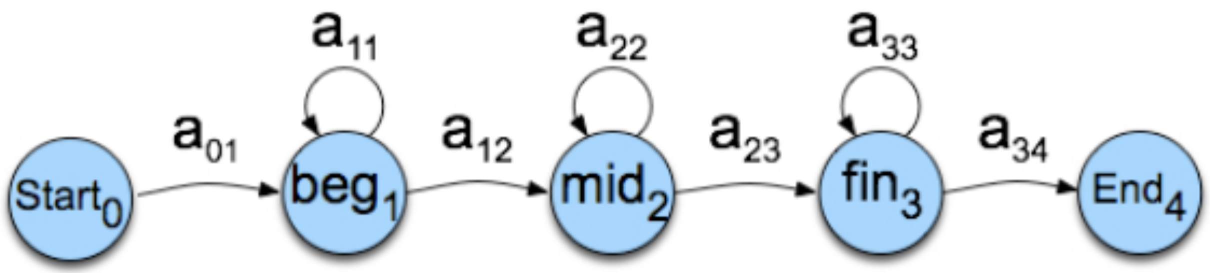
- Recording in 16kHz, 16-bit depth, mono, single channel
- 16,000 samples per second at 16-bit depth = 32KBps



Modeling Speech: Acoustic Model

- Acoustic model training is *very* time consuming (months)
- Pretrained models are available for many languages

```
config.setAcousticModelPath("resource:<directory>");
```



CMU Sphinx

Speech Recognition Toolkit

Brought to you by: air, arthchan2003, awb, bhiksha, and 5 others

Looking for the latest version? **Download pocketsphinx-5prealpha.tar.gz (33.7 MB)**

Home

Name	Modified	Size	Downloads / Week
Acoustic and Language Models	2015-10-17		885
sphinxtrain	2015-07-05		140
G2P Models	2014-07-15		71
sphinxbase	2014-06-09		743
pocketsphinx	2014-06-09		1,432
sphinx4	2014-02-20		580
cmuclmtk	2011-04-16		35
sphinx3	2009-01-01		13
sphinx2	2005-10-13		1

Totals: 9 Items

Recommended Projects

- FreeTTS
- Java Speech API
- simon

Modeling Text: Phonetic Dictionary

- Mapping phonemes to words
- Word error rate increases with size
- Pronunciation aided by g2p labeling
- CMU Sphinx has tools to generate dictionaries

```
config.setDictionaryPath("resource:<language>.dict");
```

Modeling Text: Phonetic Dictionary

```
autonomous AO T AA N AH M AH S
autonomously AO T AA N OW M AH S L IY
autonomy AO T AA N AH M IY
autonomy(2) AH T AA N AH M IY
autopacific AO T OW P AH S IH F IH K
autopart AO T OW P AA R T
autoparts AO T OW P AA R T S
autopilot AO T OW P AY L AH T
```

How to train your own language model

- Language model training is easy™ (~100,000 sentences)
- Some tools:
  - Boilerpipe (HTML text extraction)
  - Logios (model generation)
  - lmtool (CMU Sphinx)
  - IRSLM
  - MITLM

Language model

<s> generally cloudy today with scattered outbreaks of rain and drizzle persistent and heavy at times </s>

<s> some dry intervals also with hazy sunshine especially in eastern parts in the morning </s>

<s> highest temperatures nine to thirteen Celsius in a light or moderate mainly east south east breeze </s>

<s> cloudy damp and misty today with spells of rain and drizzle in most places much of this rain will be light and patchy but heavier rain may develop in the west later </s>

Sphinx Knowledge Base Tool -- VERSION 3

This is the new version of the lmtool! **FAQ**

Changes should be transparent (unless you automate, see note below).
Problems? Please help by sending a report to the maintainer.

New! Follow us on `@CMUSpeechGroup` for announcements and status updates.

What it does: Builds a consistent set of lexical and language modeling files for Sphinx (and compatible) decoders.

To use: Create a sentence corpus file, consisting of all sentences you would like the decoder to recognize. The sentences should be one to a line (but do not need to have standard punctuation). You may not need to exhaustively list all possible sentences: the decoder will allow fragments to recombine into new sentences.

Upload a sentence corpus file:

Choose File No file chosen

COMPILE KNOWLEDGE BASE

The **new version of lmtool** has been reorganized internally to make use of the Logios package. This will make lmtool easier to maintain in the future and will allow it to take advantage of ongoing development in Logios. These changes should be transparent to regular users. Please give it a try. If you have any problems, or discover bugs, let the maintainer know. If things look good (i.e., I stop getting bug reports) this will become the standard version.

NOTE: If you have automated the use of this tool you will need to update your code. The main difference is that the name of the target script has changed. The old script will still be available so nothing will break immediately, but it's unlikely to continue to be maintained. Also, file links are no longer tagged in the html. Please let me know if you make use of this feature and I'll find a fix.

Alex Rudnicky

Sphinx knowledge base generator [lmtool.3a]

Your Sphinx knowledge base compilation has been successfully processed!

The base name for this set is **6166**. TAR6166.tgz is the compressed version.
Note that this set of files is internally consistent and is best used together.

IMPORTANT: Please download these files as soon as possible; they will be deleted in approximately a half hour.

```
SESSION 1455690550_15005
[_INFO_] Found corpus: 4 sentences, 85 unique words
[_INFO_] Found 0 words in extras  (0)
[_INFO_] Language model completed  (0)
[_INFO_] Pronounce completed  (0)
[_STAT_] Elapsed time: 0.042 sec
```

Please include these messages in bug reports.

Name	Size	Description
6166.dic	1.2K	*Pronunciation Dictionary*
6166.lm	5.9K	*Language Model*
6166.log_pronounce	955	*Log File*
6166.sent	520	*Corpus (processed)*
6166.vocab	362	*Word List*
TAR6166.tgz	3.0K	**COMPRESSED TARBALL**

Apache/2.2.22 (Ubuntu) Server at www.speech.cs.cmu.edu Port 80

Modeling Speech: Grammar Model

- JSpeech Grammar Format

```
config.setGrammarPath("resource:<grammar>.gram");
```

```
<size> = /10/ small | /2/ medium | /1/ large;
<color> = /0.5/ red | /0.1/ blue | /0.2/ green;
<action> = please (/20/save files |/1/delete files);
<place> = /20/ <city> | /5/ <country>;
public command = <size> | <color> | <action> | <place>
```

Modeling Speech: Grammar Format

```
public <number> = <hundreds> | <tens> | <teens> | <ones>;
      <hundreds> = <ones> hundred
                   [<tens> | <teens> | <ones>];
          <tens> = ( twenty | thirty | forty | fifty |
                     sixty | seventy | eighty | ninety )
                   [<ones>];
         <teens> = ten | eleven | twelve | thirteen |
                   fourteen | fifteen | sixteen |
                   seventeen | eighteen | nineteen;
          <ones> = one | two | three | four | five | six |
                   seven | eight | nine;
```

Configuring Sphinx-4

```
Configuration config = new Configuration();

config.setAcousticModelPath(AM_PATH);
config.setDictionaryPath(DICT_PATH);
config.setLanguageModelPath(LM_PATH);
config.setGrammarPath(GRAMMAR_PATH);
// config.setSampleRate(8000);
```

Live Speech Recognizer

```
LiveSpeechRecognizer recognizer =
    new LiveSpeechRecognizer(config);

recognizer.startRecognition(true);
…
recognizer.stopRecognition();
```

Live Speech Recognizer

```
while (…) {
    // This blocks on a recognition result
    SpeechResult sr = recognizer.getResult();

    String h = sr.getHypothesis();
    Collection<String> hs = sr.getNbest(3);
    …
}
```

Stream Speech Recognizer

```
StreamSpeechRecognizer recognizer = new
StreamSpeechRecognizer(configuration);
recognizer.startRecognition(
        new FileInputStream("speech.wav"));
SpeechResult result = recognizer.getResult();
recognizer.stopRecognition();
```

Improving recognition accuracy

- Using context-dependent cues
- Structuring commands to reduce phonetic similarity
- Disabling the recognizer
- Grammar swapping
- Busy waiting

Grammar Swapping

```
static void swapGrammar(String newGrammarName) throws
PropertyException, InstantiationException, IOException
{
    Linguist linguist =
            (Linguist) cm.lookup("flatLinguist");
    linguist.deallocate();
    cm.setProperty("jsgfGrammar", "grammarName",
                            newGrammarName);
    linguist.allocate();
}
```

MaryTTS: Initializing

```
maryTTS = new LocalMaryInterface();
Locale systemLocale = Locale.getDefault();
if (maryTTS.getAvailableLocales()
        .contains(systemLocale)) {
    voice = Voice.getDefaultVoice(systemLocale);
}

maryTTS.setLocale(voice.getLocale());
maryTTS.setVoice(voice.getName());
```

MaryTTS: Generating Speech

```
try {
    AudioInputStream audio = mary.generateAudio(text);
    AudioPlayer player = new AudioPlayer(audio);
    player.start();
    player.join();
} catch (SynthesisException | InterruptedException e)
{
    …
}
```

Resources

- CMUSphinx, http://cmusphinx.sourceforge.net/wiki/
- MaryTTS, http://mary.dfki.de/
- FreeTTS 1.2, http://freetts.sourceforge.net/
- JSpeech Grammar Format, http://www.w3.org/TR/jsgf/
- LibriSpeech ASR Corpus http://www.openslr.org/12/
- ARPA format for N-gram backoff (Doug Paul) http://www.speech.sri.com/projects/srilm/manpages/ngram-format.5.html
- Language Model Tool http://www.speech.cs.cmu.edu/tools/lmtool.html

Further Research

- Accurate and Compact Large Vocabulary Speech Recognition on Mobile Devices, research.google.com/pubs/archive/41176.pdf
- Comparing Open-Source Speech Recognition Toolkits, http://suendermann.com/su/pdf/oasis2014.pdf
- Tuning Sphinx to Outperform Google's Speech Recognition API, http://suendermann.com/su/pdf/essv2014.pdf
- Deep Neural Networks for Acoustic Modeling in Speech Recognition, research.google.com/pubs/archive/38131.pdf
- Deep Speech: Scaling up end-to-end speech recognition, http://arxiv.org/pdf/1412.5567v2.pdf

Further Research

- WER progress: https://github.com/syhw/wer_are_we
- Kaldi Speech Recognition Library http://kaldi-asr.org/doc/

Special Thanks

- Breandan Considine (@breandan)
- Alexey Kudinkin (@alexeykudinkin)
- Yaroslav Lepenkin (@lepenkinya)
- CMU Sphinx (@cmuspeechgroup)
- http://github.com/breandan/idear