

Automatic Speech Recognition: Introduction

Steve Renals & Hiroshi Shimodaira

Automatic Speech Recognition— ASR Lecture 1
15 January 2018

Course details

- **Lectures:** About 18 lectures, plus a couple of extra lectures on basic introduction to neural networks
- **Labs:** Weekly lab sessions – using Kaldi (kaldi-asr.org) to build speech recognition systems.
 - Lab sessions in AT-4.12: Tuesdays 10:00, Wednesdays 10:00, Wednesdays 15:10, start week 2 (23/24 January)
 - Select one lab session at <https://doodle.com/poll/gxmh9kwp3a8espwx>
- **Assessment:**
 - Exam in April or May (worth 70%)
 - Coursework (worth 30%, building on the lab sessions): out on Monday 12 February; in by Wednesday 14 March
- **People:**
 - Lecturers: Steve Renals and Hiroshi Shimodaira
 - TAs: Joachim Fainberg and Ondrej Klejch

Your background

If you have taken:

- Speech Processing *and* either of (MLPR or MLP)
 - Perfect!
- either of (MLPR or MLP) *but not* Speech Processing
 - You'll require some speech background:
 - A couple of the lectures will cover material that was in Speech Processing
 - Some additional background study (including material from Speech Processing)
- Speech Processing *but neither of* (MLPR or MLP)
 - You'll require some machine learning background (especially neural networks)
 - A couple of introductory lectures on neural networks
 - Some additional background study

- Series of weekly labs using Kaldi.
- Labs start week 2 (next week)
- **Note:** Training speech recognisers can take time
 - ASR training in some labs will not finish in an hour...
 - Give yourself plenty of time to complete the coursework, don't leave it until the last couple of days

What is speech recognition?

Speech-to-text transcription

- Transform recorded audio into a sequence of words
- Just the words, no meaning.... But do need to deal with acoustic ambiguity: “Recognise speech?” or “Wreck a nice beach?”
- Speaker diarization: Who spoke when?
- Speech recognition: what did they say?
- Paralinguistic aspects: how did they say it? (timing, intonation, voice quality)
- Speech understanding: what does it mean?

Why is speech recognition difficult?

Variability in speech recognition

Several sources of variation

Size Number of word types in vocabulary, perplexity

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Acoustic environment Noise, competing speakers, channel conditions (microphone, phone line, room acoustics)

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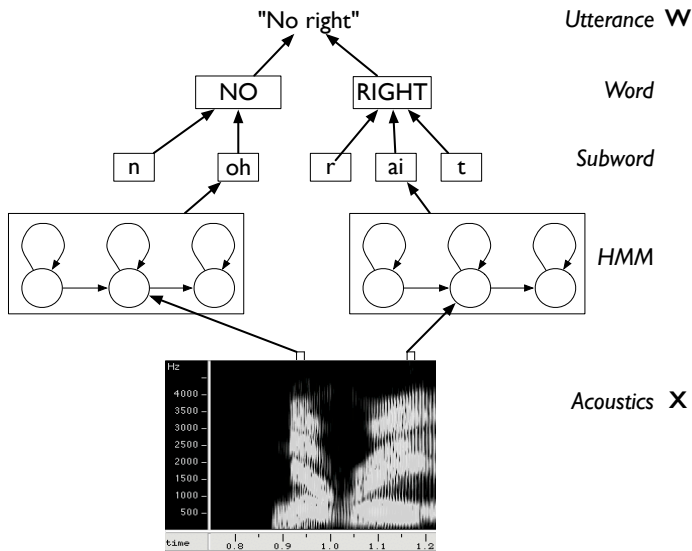
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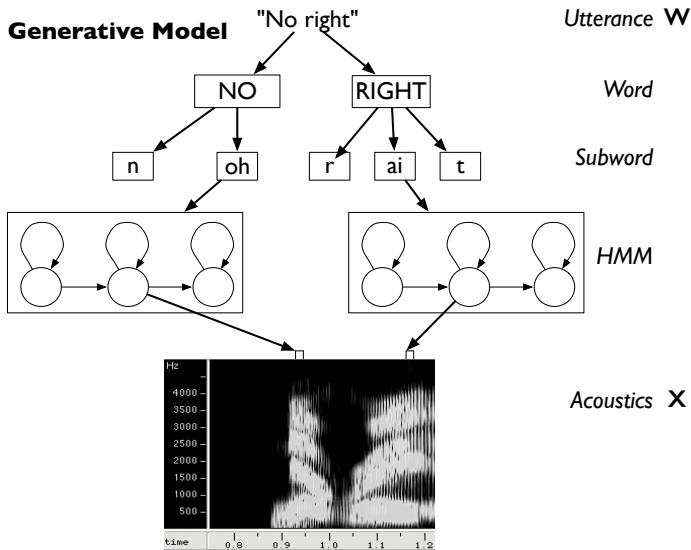
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Style Continuously spoken or isolated? Planned monologue or spontaneous conversation?

Hierarchical modelling of speech



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“Fundamental Equation of Statistical Speech Recognition”

If \mathbf{X} is the sequence of acoustic feature vectors (observations) and \mathbf{W} denotes a word sequence, the most likely word sequence \mathbf{W}^* is given by

$$\mathbf{W}^* = \arg \max_{\mathbf{W}} P(\mathbf{W} | \mathbf{X})$$

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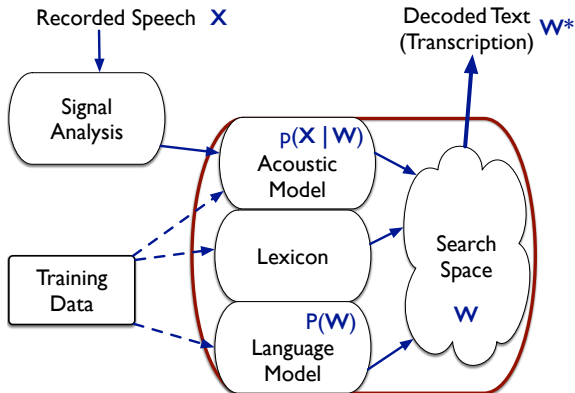
Applying Bayes' Theorem:

$$\begin{aligned} P(\mathbf{W} | \mathbf{X}) &= \frac{p(\mathbf{X} | \mathbf{W})P(\mathbf{W})}{p(\mathbf{X})} \\ &\propto p(\mathbf{X} | \mathbf{W})P(\mathbf{W}) \\ \mathbf{W}^* &= \arg \max_{\mathbf{W}} \underbrace{p(\mathbf{X} | \mathbf{W})}_{\text{Acoustic model}} \underbrace{P(\mathbf{W})}_{\text{Language model}} \end{aligned}$$

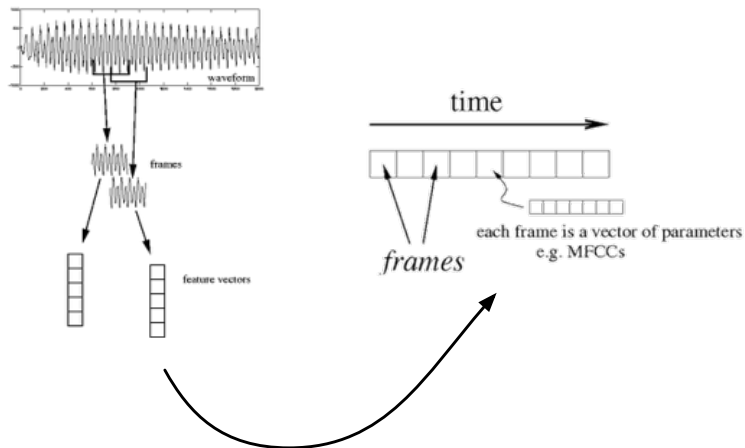
Speech Recognition Components

$$\mathbf{W}^* = \arg \max_{\mathbf{W}} p(\mathbf{X} | \mathbf{W})P(\mathbf{W})$$

Use an acoustic model, language model, and lexicon to obtain the most probable word sequence \mathbf{W}^* given the observed acoustics \mathbf{X}



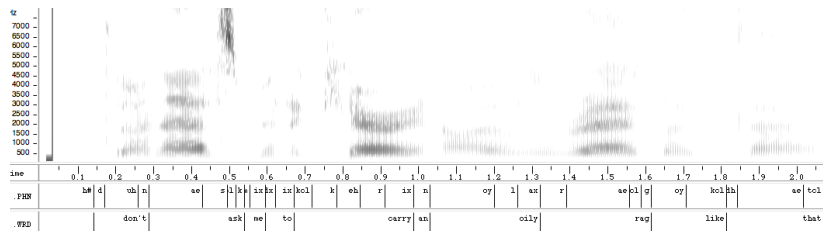
Representing recorded speech (X)



Represent a recorded utterance as a sequence of *feature vectors*

Reading: Jurafsky & Martin section 9.3

Labelling speech (W)



Labels may be at different levels: words, phones, etc.

Labels may be *time-aligned* – i.e. the start and end times of an acoustic segment corresponding to a label are known

Reading: Jurafsky & Martin chapter 7 (especially sections 7.4, 7.5)

- **Phonemes**

- abstract unit defined by linguists based on contrastive role in word meanings (eg “cat” vs “bat”)
- 40–50 phonemes in English

- **Phones**

- speech sounds defined by the acoustics
- many *allophones* of the same phoneme (eg /p/ in “pit” and “spit”)
- limitless in number
- Phones are usually used in speech recognition – but no conclusive evidence that they are the basic units in speech recognition
- Possible alternatives: syllables, automatically derived units, ...

(Slide taken from Martin Cooke from long ago)

Example: TIMIT Corpus

- TIMIT corpus (1986)—first widely used corpus, still in use
 - Utterances from 630 North American speakers
 - Phonetically transcribed, time-aligned
 - Standard training and test sets, agreed evaluation metric (phone error rate)
- TIMIT phone recognition - label the audio of a recorded utterance using a sequence of phone symbols
 - Frame classification – attach a phone label to each frame data
 - Phone classification – given a segmentation of the audio, attach a phone label to each (multi-frame) segment
 - Phone recognition – supply the sequence of labels corresponding to the recorded utterance

Basic speech recognition on TIMIT

- Train a classifier of some sort to associate each feature vector with its corresponding label. Classifier could be
 - Neural network
 - Gaussian mixture model
 - ...

The at test time, a label is assigned to each frame

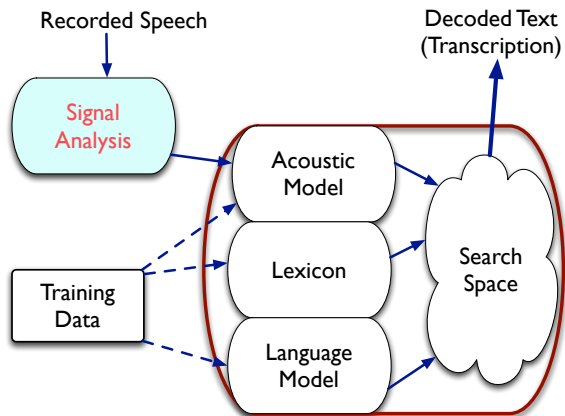
- Questions
 - What's good about this approach?
 - What the limitations? How might we address them?

- How accurate is a speech recognizer?
- String edit distance
 - Use dynamic programming to align the ASR output with a reference transcription
 - Three type of error: insertion, deletion, substitutions
- Word error rate (WER) sums the three types of error. If there are N words in the reference transcript, and the ASR output has S substitutions, D deletions and I insertions, then:

$$\text{WER} = 100 \cdot \frac{S + D + I}{N} \% \quad \text{Accuracy} = 100 - \text{WER}\%$$

- For TIMIT, define phone error error rate analagously to word error rate
- Speech recognition evaluations: common training and development data, release of new test sets on which different systems may be evaluated using word error rate

Next Lecture



- Jurafsky and Martin (2008). *Speech and Language Processing* (2nd ed.): Chapter 7 (esp 7.4, 7.5) and Section 9.3.
- General interest: *The Economist Technology Quarterly*, “Language: Finding a Voice”, Jan 2017.

<http://www.economist.com/technology-quarterly/2017-05-01/language>