Lecture 13: Speech Data

LING 1340/2340: Data Science for Linguists

Na-Rae Han

Objectives

- Speech data
  - Conversion: TextGrid, WAV, etc.
  - Command-line tools, conversion
  - Forced alignment demo: Montreal Forced Aligner
  - ASR theory

TextGrid

- Praat was able to parse TIMIT's PHN file format (phone tier)
- Saving it out to a proper TextGrid file →
- However, Praat couldn't handle:
  - SA1.TXT (utterance tier)
  - SA1.WRD (word tier)

  ← How to get them into TextGrid?

There's a python library (or two) for that!

praat-textgrids 1.3.1

```
pip install praat-textgrids
```

[ˈpɒɹ.səlˌmaʊθ]

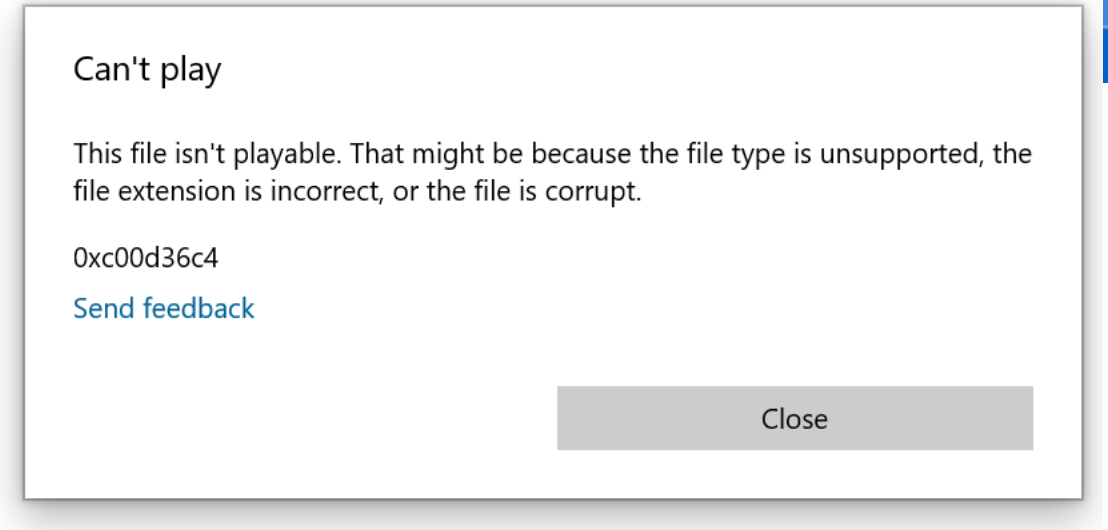
Parselmouth – Praat in Python, the Pythonic way

```
File type = "ooTextFile"
Object class = "TextGrid"

xmin = 0
xmax = 2.92
tiers? <exists>
size = 2
item []:
    item [1]:
        class = "IntervalTier"
        name = "phn"
        xmin = 0
        xmax = 2.92
        intervals: size = 37
        intervals [1]:
            xmin = 0
            xmax = 0.19062500000000002
            text = "h#"
        intervals [2]:
            xmin = 0.19062500000000002
            xmax = 0.2849375
            text = "sh"
        intervals [3]:
            xmin = 0.2849375
            xmax = 0.3576875
            text = "ix"
        intervals [4]:
            xmin = 0.3576875
            xmax = 0.415125
            text = "hv"
        intervals [5]:
            xmin = 0.415125
            xmax = 0.54825
            text = "eh"
        intervals [6]:
```

.WAV format?

- Also, even though PRAAT was able to open the .WAV files, Windows 10 cannot…



- These files are not really .WAV…
  - **SPHERE format**, normally with .SPH extension.
- How to convert to WAV?

Solution 1: Praat script

- Write a praat script
  - (Or, grab someone else's…)

```
# prep_audio_mfa.praat
# Written by E. Chodroff
# Oct 23 2018
# extract left channel and resample to 16 kHz for all wav files in a directory

### CHANGE ME!
# don't forget the slash at the end of the path
dir$ = "/Users/Eleanor/Desktop/align_input/"
###

Create Strings as file list: "files", dir$ + "*.wav"
nFiles = Get number of strings

for i from 1 to nFiles
	# read in WAV file
	selectObject: "Strings files"
	filename$ = Get string: i
	Read from file: dir$ + filename$

	# extract left channel
	Extract one channel: 1

	# resample to 16kHz with 50 point precision (default)
	Resample: 16000, 50

	# save WAV file
	Save as WAV file: dir$ + filename$

	# clean up
	select all
	minusObject: "Strings files"
	Remove
endfor
```

Solution 2:
SoX + bash shell

```
sox <input-file> -b 16 -t wav <output-file>
```

```
for x in *.WAV
do
sox $x -b 16 0t wav true_wav/$x
echo $x finished
done
```

Declared as x, subsequent references as $x

```
narae@T480s MINGW64 ~/Desktop/FCJF0
$ alias sox="/d/util/sox-14.4.2/sox.exe"

narae@T480s MINGW64 ~/Desktop/FCJF0
$ ls
SA1.PHN  SA2.WAV     SI1657.PHN  SI648.WAV  SX217.PHN  SX307.WAV  SX397.PHN
SA1.TXT  SA2.WRD     SI1657.TXT  SI648.WRD  SX217.TXT  SX307.WRD  SX397.TXT
SA1.WAV  SI1027.PHN  SI1657.WAV  SX127.PHN  SX217.WAV  SX37.PHN   SX397.WAV
SA1.WRD  SI1027.TXT  SI1657.WRD  SX127.TXT  SX217.WRD  SX37.TXT   SX397.WRD
SA2.PHN  SI1027.WAV  SI648.PHN   SX127.WAV  SX307.PHN  SX37.WAV   true_wav/
SA2.TXT  SI1027.WRD  SI648.TXT   SX127.WRD  SX307.TXT  SX37.WRD

narae@T480s MINGW64 ~/Desktop/FCJF0
$ sox SA1.WAV -b 16 -t wav true_wav/SA1.wav

narae@T480s MINGW64 ~/Desktop/FCJF0
$ ls true_wav/
SA1.wav
```

converting a single file

```
narae@T480s MINGW64 ~/Desktop/FCJF0
$ for x in *WAV
> do
> sox $x -b 16 -t wav true_wav/$x
> echo $x finished
> done
SA1.WAV finished
SA2.WAV finished
SI1027.WAV finished
SI1657.WAV finished
SI648.WAV finished
SX127.WAV finished
SX217.WAV finished
SX307.WAV finished
SX37.WAV finished
SX397.WAV finished

narae@T480s MINGW64 ~/Desktop/FCJF0
$ ls true_wav/
SA1.wav  SI1027.WAV  SI648.WAV  SX217.WAV  SX37.WAV
SA2.WAV  SI1657.WAV  SX127.WAV  SX307.WAV  SX397.WAV
```

for loop in bash!

General-purpose audio/video manipulation software

- Audacity
  - Open-source audio software
- SoX
  - Sound eXchange; audio format conversion tool
- FFmpeg
  - For recording and converting audio/video data

Powerful command-line tools!!

https://musicinformationretrieval.com/sox_and_ffmpeg.html

Popular speech data analysis tools for linguists (1)

- Praat (Boersma & Weenink, 2021)
- Klatt formant synthesizer (Klatt 1975, 1984)
- Forced aligners
  - Penn Phonetics Lab Forced Aligner (Yuan & Liberman 2009) → legacy, became FAVE-align
  - FAVE-align (Rosenfelder et al. 2011)
  - Montreal Forced Aligner (McAuliffe et al. 2017)
  - EasyAlign (Goldman 2011 -- Windows only)
- ELAN multimodal annotator (Wittenberg et al. 2006)
  - Audio as well as video!

Popular speech data analysis tools for linguists (2)

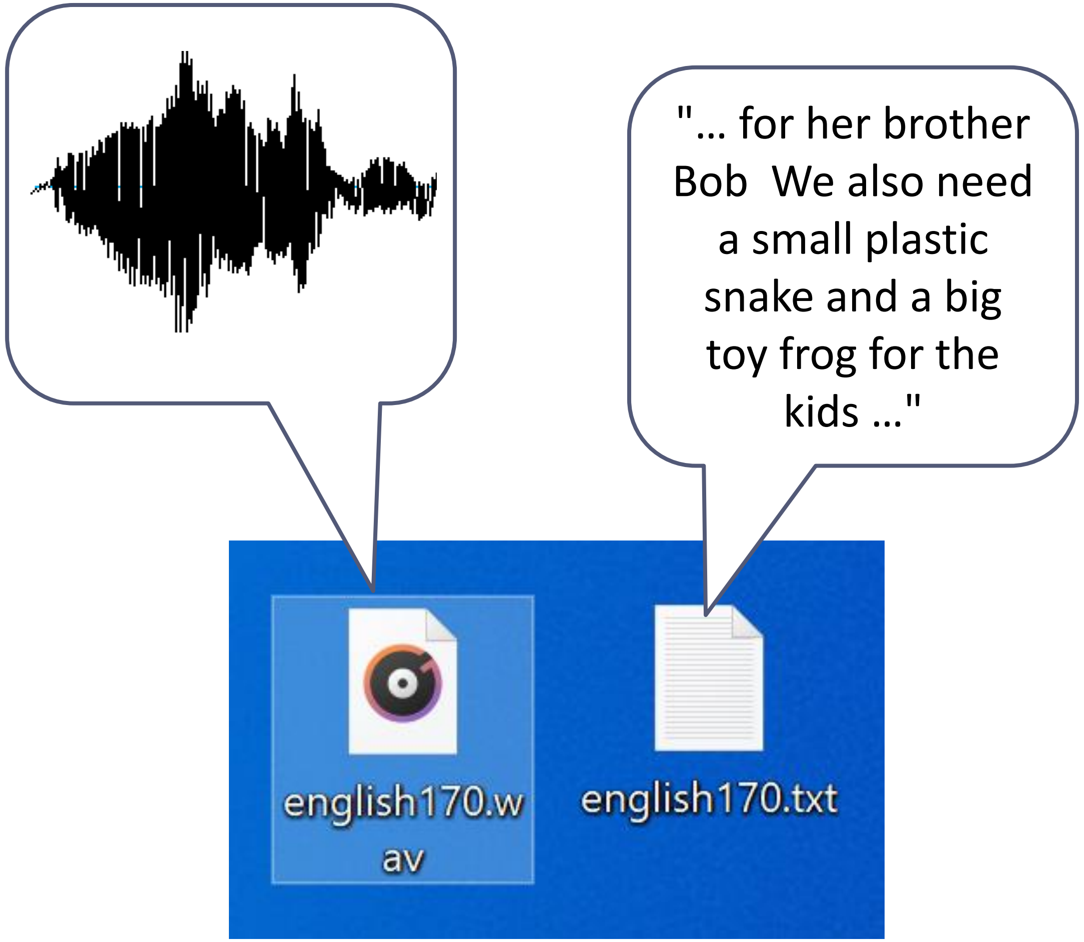
Some tools are online:

- NORM: the Vowel Normalization and Plotting Suite
- DARLA: Dartmouth Linguistic Automation

←You upload an audio file and a transcript file, the site will process them and email you the results, etc!

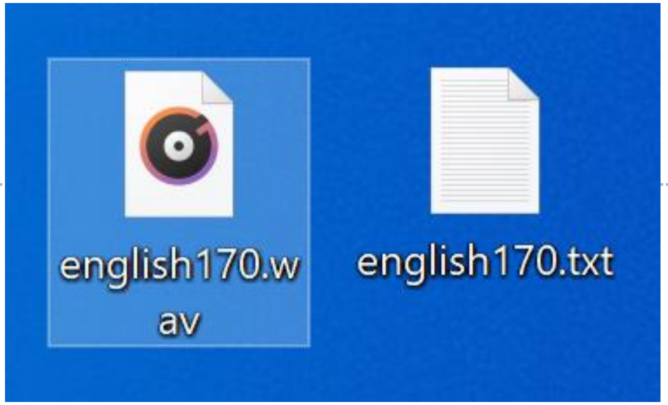
Forced alignment

- "**Forced alignment**": automatic synchronization of a sequence of phones with an audio file.
- Purpose: speed up manual segmentation and annotation
  - Rather than doing everything manually from scratch, correct output from forced aligner
  - Makes life easier for linguists doing speech-focused research!

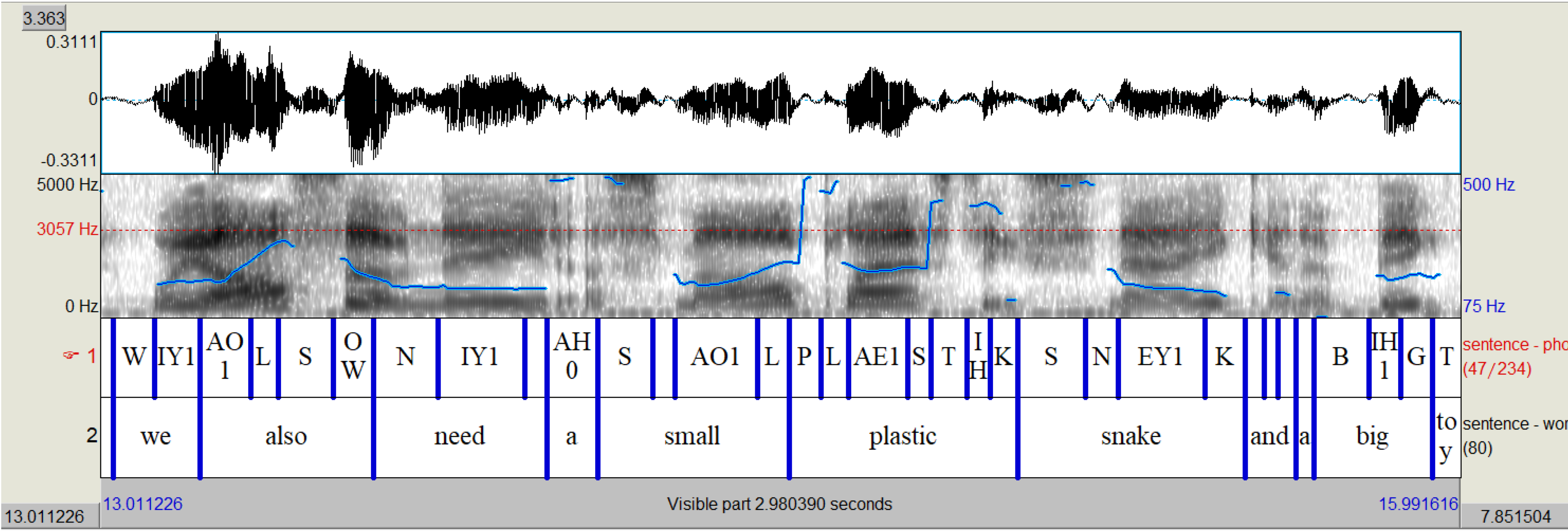


Forced alignment

- You have: a speech file (.wav), a transcript file (.txt) →

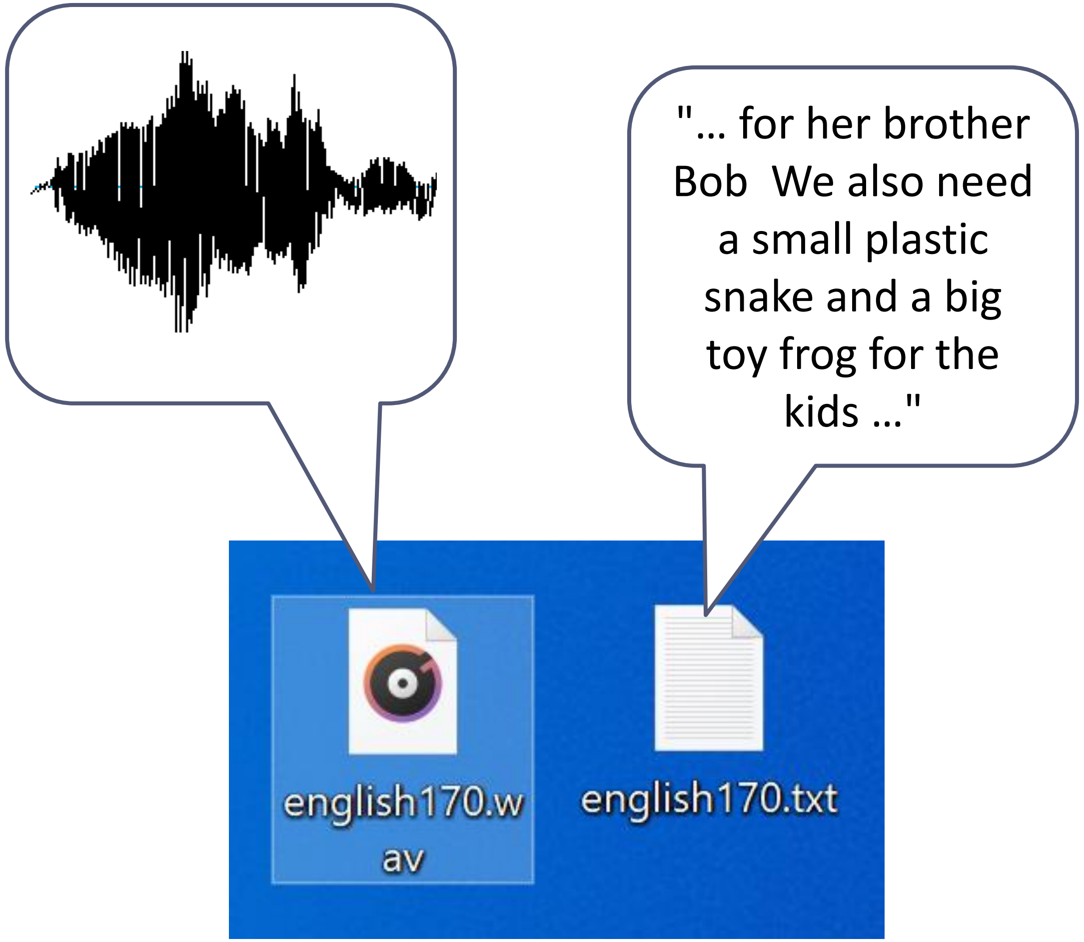


- You want:



Sound wave, words, phones

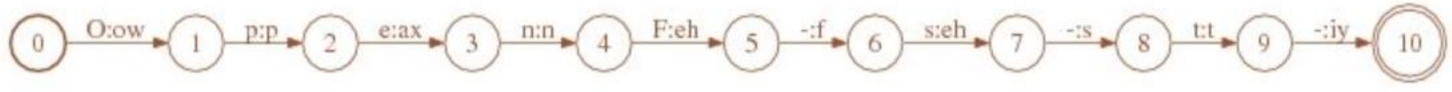
- What additional linguistic information is needed?
  - Pronunciation dictionary
    - Phonemic representations for "brother", "we", "also"…
    - More broadly: orthography → phone (G2P, "grapheme-to-phoneme")
  - Acoustic model
    - How phonemic representation relates to sound wave



Demo: Montreal Forced Aligner

- Home page:
  - https://montreal-forced-aligner.readthedocs.io/en/latest/introduction.html#what-is-forced-alignment
- GitHub project page:
  - https://github.com/MontrealCorpusTools/Montreal-Forced-Aligner

- Builds on popular/standard libraries:
  - **Kaldi** ASR toolkit
    - [home] [GitHub repo]
  - which builds on **OpenFST**
    - [home]



Steps (latest MFA version 2.0)

- Install Kaldi, MFA
  - Windows users: For ver 2.0, you need WSL (**W**indows **S**ubsystem for **L**inux, essentially Linux on Windows!) to use full G2P functionality. Alternatively: install older ver 1.0.1 available here, which is Windows-native.
- Prepare data to align
  - Speech files (WAV format, single-channel)
  - Transcript files (.lab or .txt format; no punctuation)

We'll use TIMIT data for demo (pretend it came with audio files and .TXT transcripts only)

- Download language models (pre-trained, MFA offers many)
  - A pronunciation dictionary for the language
    - If not available: produce one by running language-specific G2P (grapheme-to-phoneme) on your transcript files
  - An acoustic model for the language
- Run:
  - `mfa align <input-dir> <pron-dict> <acoustic-model> <output-dir>`
- New TextGrid files in the output dir! Examine.

Cleaning transcript files

MINGW64:/c/Users/narae/Desktop/true_wav

```
narae@T480s MINGW64 ~/Desktop/FCJF0
$ cat *TXT
0 46797 She had your dark suit in greasy wash water all year.
0 34509 Don't ask me to carry an oily rag like that.
0 49460 Even then, if she took one step forward he could catch her.
0 45466 Or borrow some money from someone and go home by bus?
0 57856 A sailboat may have a bone in her teeth one minute and lie becalmed the next.
0 24679 The emperor had a mean temper.
0 27751 How permanent are their records?
0 23143 The meeting is now adjourned.
0 36250 Critical equipment needs proper maintenance.
0 39220 Tim takes Sheila to see movies twice a week.
```

Initial digits and punctuation need to go

```
narae@T480s MINGW64 ~/Desktop/FCJF0
$ perl -npe 's/^\d \d+ //' SA1.TXT
She had your dark suit in greasy wash water all year.

narae@T480s MINGW64 ~/Desktop/FCJF0
$ perl -npe 's/^\d \d+ //; s/\.//g;' SA1.TXT
She had your dark suit in greasy wash water all year
```

Perl + regular expressions to clean up

MINGW64:/c/Users/narae/Desktop/true_wav

```
narae@T480s MINGW64 ~/Desktop/FCJF0
$ perl -npe 's/^\d \d+ //; s/[\.,\?]//g;' *.TXT
She had your dark suit in greasy wash water all year
Don't ask me to carry an oily rag like that
Even then if she took one step forward he could catch her
Or borrow some money from someone and go home by bus
A sailboat may have a bone in her teeth one minute and lie becalmed the next
The emperor had a mean temper
How permanent are their records
The meeting is now adjourned
Critical equipment needs proper maintenance
Tim takes Sheila to see movies twice a week

narae@T480s MINGW64 ~/Desktop/FCJF0
$ for x in *TXT
> do
> perl -npe 's/^\d \d+ //; s/[\.,\?]//g;' $x > ../true_wav/$x
> echo $x completed
> done
SA1.TXT completed
SA2.TXT completed
SI1027.TXT completed
SI1657.TXT completed
SI648.TXT completed
SX127.TXT completed
SX217.TXT completed
SX307.TXT completed
SX37.TXT completed
SX397.TXT completed

narae@T480s MINGW64 ~/Desktop/FCJF0
$ cd ../true_wav/

narae@T480s MINGW64 ~/Desktop/true_wav
$ ls
SA1.TXT  SA2.TXT  SI1027.TXT  SI1657.TXT  SI648.TXT  SX127.TXT  SX217.TXT  SX307.TXT  SX37.TXT  SX397.TXT
SA1.WAV  SA2.WAV  SI1027.WAV  SI1657.WAV  SI648.WAV  SX127.WAV  SX217.WAV  SX307.WAV  SX37.WAV  SX397.WAV
```

Use bash for-loop to create cleaned-up version of all .TXT files

.WAV and .TXT files are now ready…

Download language models

- MFA's pre-trained models:
  - https://montreal-forced-aligner.readthedocs.io/en/latest/pretrained_models.html

Pretrained acoustic models

As part of using the Montreal Forced Aligner in our own research, we have trained acoustic models for a number of languages. If you would like to use them, please download them below. Please note the dictionary that they were trained with to see more information about the phone set. When using these with a pronunciation dictionary, the phone sets must be compatible. If the orthography of the language is transparent, it is likely that we have a G2P model that can be used to generate the necessary pronunciation dictionary.

Any of the following acoustic models can be downloaded with the command `mfa download acoustic <language_id>`. You can get a full list of the currently available acoustic models via `mfa download acoustic`. New models contributed by users will be periodically added. If you would like to contribute your trained models, please contact Michael McAuliffe at michael.e.mcauliffe@gmail.com.

Language	Link	Corpus	Number of speakers	Audio (hours)	Phone set
Arabic	Arabic acoustic model	GlobalPhone	80	19.0	GlobalPhone
Bulgarian	Bulgarian acoustic model	GlobalPhone	79	21.4	GlobalPhone
Croatian	Croatian acoustic model	GlobalPhone	94	15.9	GlobalPhone
Czech	Czech acoustic model	GlobalPhone	102	31.7	GlobalPhone
English	English acoustic model	LibriSpeech	2484	982.3	Arpabet (stressed)
French (FR)	French (FR) acoustic model	GlobalPhone	100	26.9	GlobalPhone

Available pronunciation dictionaries

Any of the following pronunciation dictionaries can be downloaded with the command `mfa download dictionary <language_id>`. You can get a full list of the currently available dictionaries via `mfa download dictionary`. New dictionaries contributed by users will be periodically added. If you would like to contribute your dictionaries, please contact Michael McAuliffe at michael.e.mcauliffe@gmail.com.

Language	Link	Orthography system	Phone set
English	English pronunciation dictionary	Latin	Arpabet (stressed)
French	French Prosodylab dictionary	Latin	Prosodylab French
German	German Prosodylab dictionary	Latin	Prosodylab German

CMU pronouncing dictionary

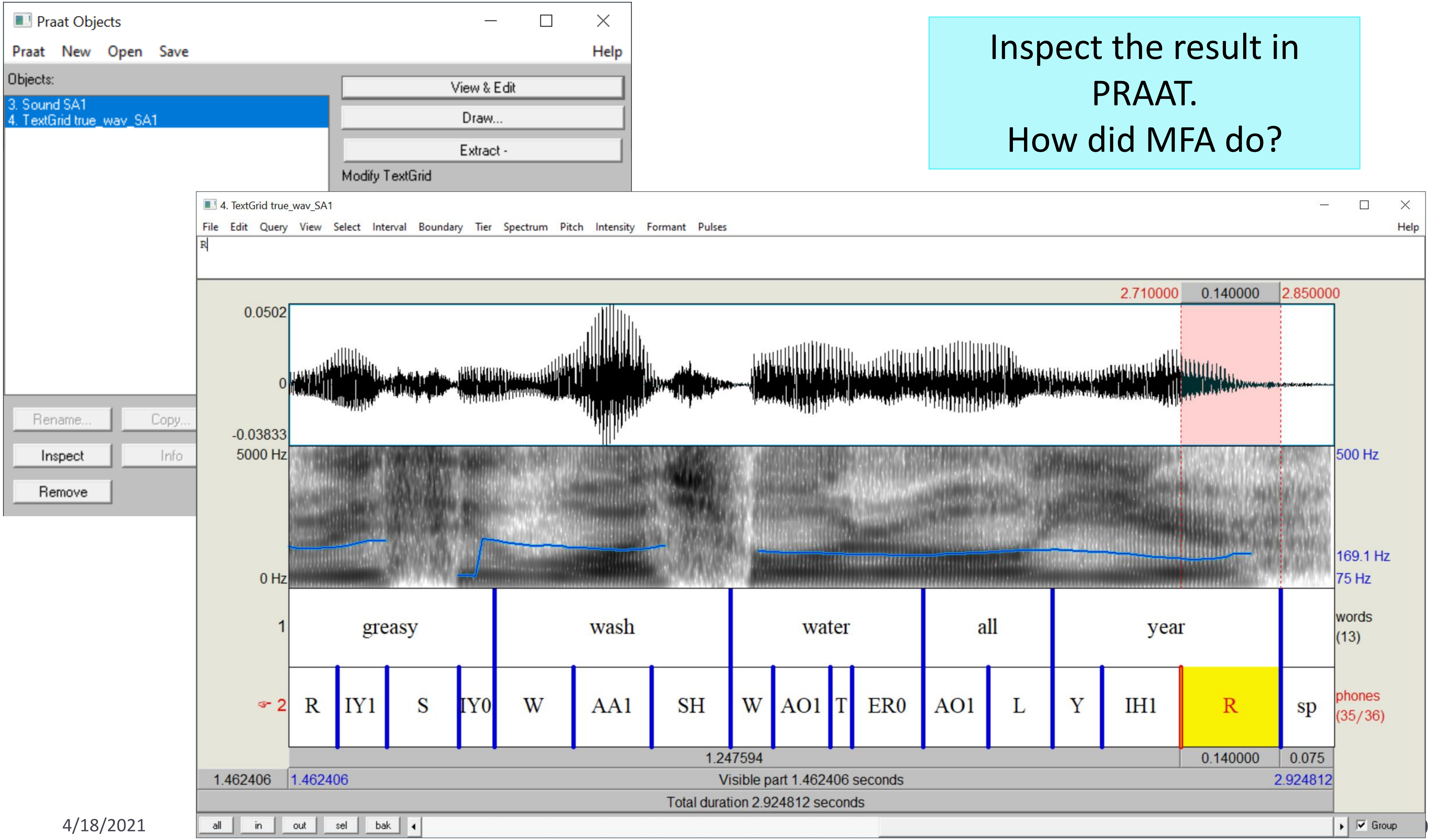
MFA is installed on **WSL**, need to bring out **Ubuntu console**

narae@T480s: /mnt/c/Users/narae/Desktop

```
(base) narae@T480s:~$ conda activate aligner
(aligner) narae@T480s:~$ cd /mnt/c/Users/narae/Desktop/
(aligner) narae@T480s:/mnt/c/Users/narae/Desktop$ ls
FCJF0  desktop.ini  english.dict  english.zip  tidy  true_wav
(aligner) narae@T480s:/mnt/c/Users/narae/Desktop$ mkdir mfa out
(aligner) narae@T480s:/mnt/c/Users/narae/Desktop$ mfa align true_wav/ english.dict english.zip mfa_out/
All required kaldi binaries were found!
/home/narae/Documents/MFA/true_wav/align.log
INFO - Setting up corpus information...
loading from source
INFO - Number of speakers in corpus: 1, average number of utterances per speaker: 10.0
INFO - Number of speakers in corpus: 1, average number of utterances per speaker: 10.0
INFO - Parsing dictionary without pronunciation probabilties without silence probabilties
INFO - Creating dictionary information...
INFO - Setting up training data...
INFO - Generating base features (mfcc)...
INFO - Calculating CMVN...
INFO - Done with setup!
INFO - Performing first-pass alignment...
INFO - Calculating fMLLR for speaker adaptation...
INFO - Performing second-pass alignment...
INFO - All done!
(aligner) narae@T480s:/mnt/c/Users/narae/Desktop$ ls mfa_out/
true_wav_SA1.TextGrid     true_wav_SI1657.TextGrid  true_wav_SX217.TextGrid  true_wav_SX397.TextGrid
true_wav_SA2.TextGrid     true_wav_SI648.TextGrid   true_wav_SX307.TextGrid
true_wav_SI1027.TextGrid  true_wav_SX127.TextGrid   true_wav_SX37.TextGrid
(aligner) narae@T480s:/mnt/c/Users/narae/Desktop$
```

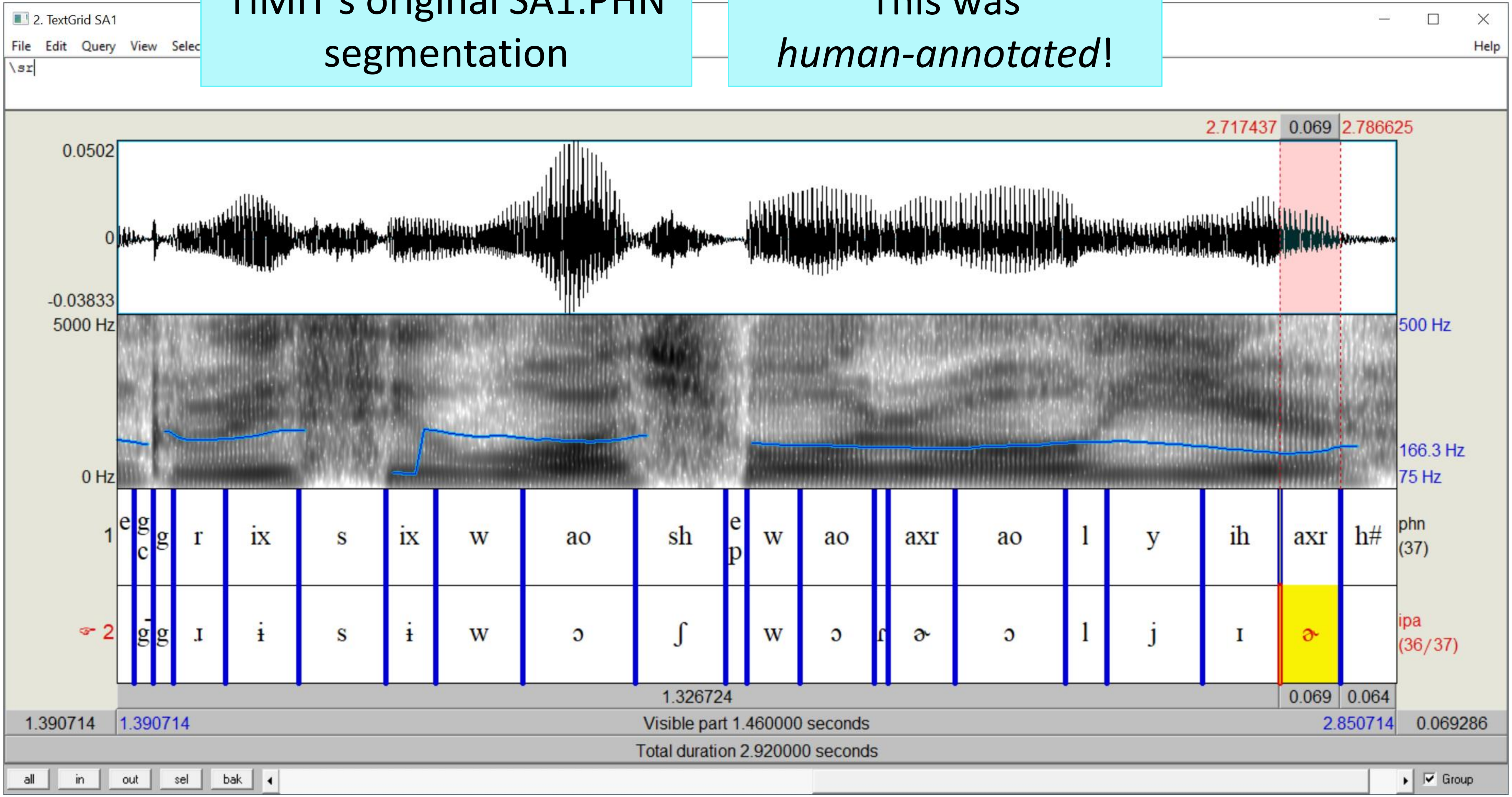
SUCCESS!
New crop of TextGrid files

Inspect the result in PRAAT.
How did MFA do?



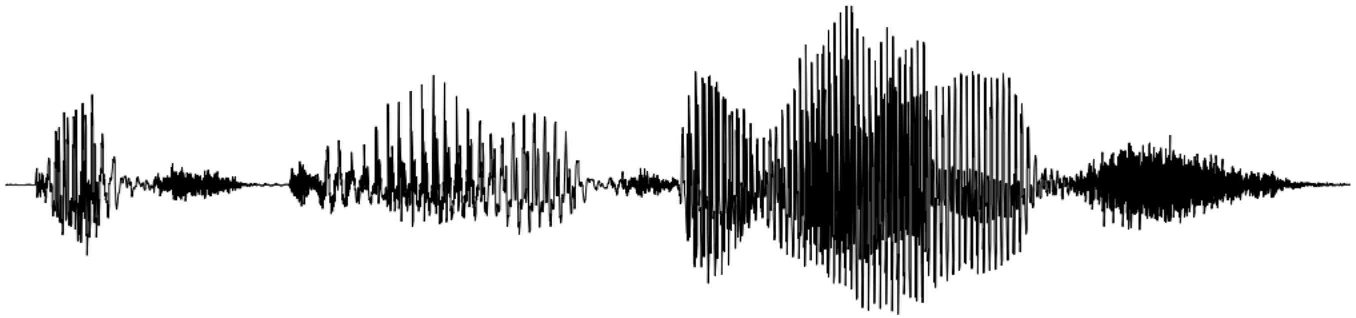
Compare with
TIMIT's original SA1.PHN
segmentation

This was
human-annotated!



Backing up: ASR

- Forced alignment is based on ASR technology.
- This is NOT an NLP class, but we should at least have some sense of how ASR works…



It's time for lunch

Is **processing speech** going to be entirely different from **text processing technologies**?

In Which We Skim Through Blog Articles (Again) In Lieu of Proper Academic Textbook

- Proper academic textbook chapter on ASR/TTS:
  - Jurafsky & Martin (2020) *Speech and Language Processing* Ch. 26 Automatic Speech Recognition and Text-to-Speech

- More accessible:
  - Speech Recognition – ASR Model Training (by Jonathan Hui)
  - Introduction to ASR (by Maël Fabien, with IPA!!)

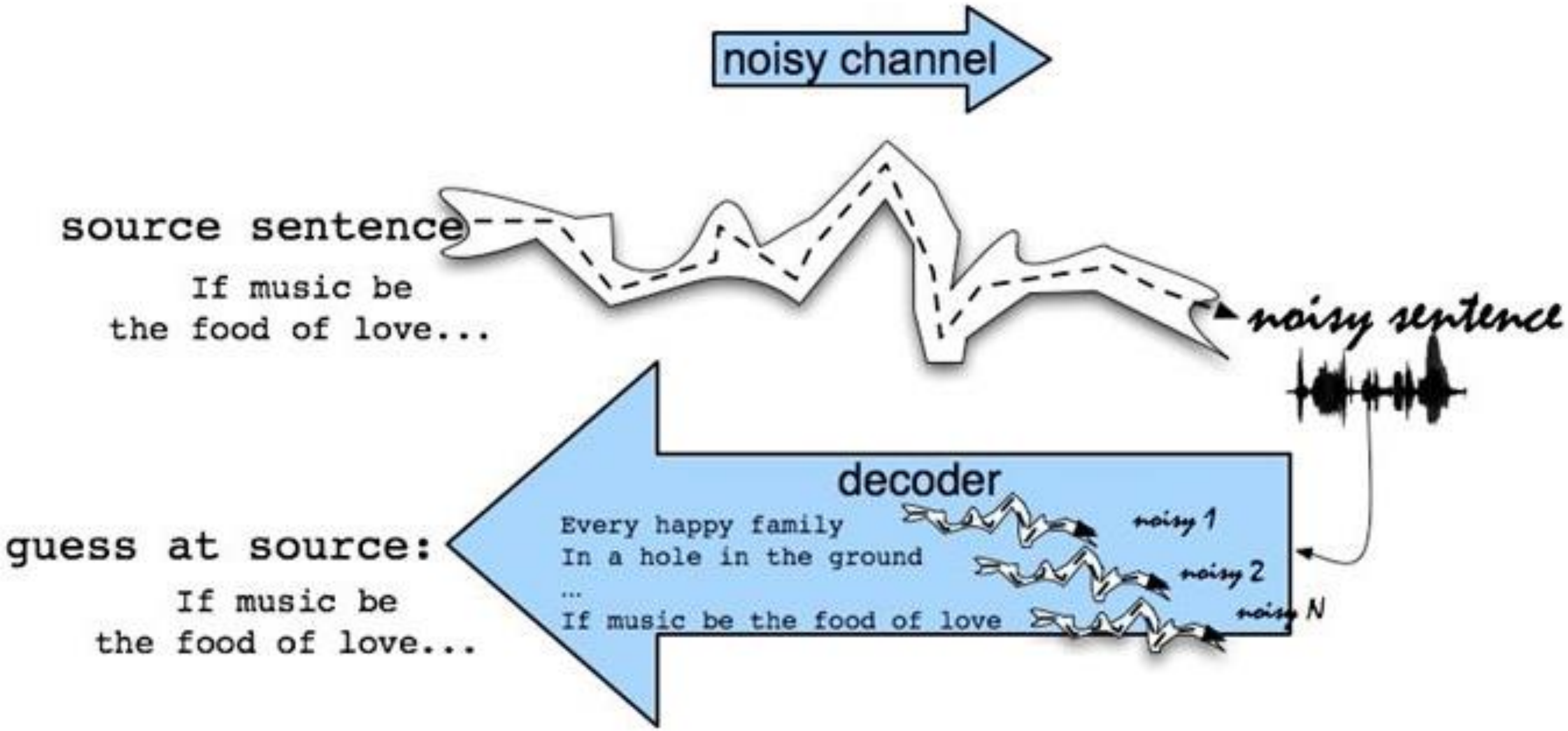
All the building blocks…

- English:
  - ARPAbet
  - CMU Pronouncing Dictionary
- World languages:
  - G2P (grapheme-to-phoneme)
- HMM (Hidden Markov Model), HTK (HMM ToolKit)
- Kaldi (ASR toolkit, built on HTK)
- Finite-State Transducer (OpenFST)
- N-gram language models

Many of them look familiar…
from LING 1330
Intro to CompLing!

The Noisy Channel Model

SLP, Jurafsky & Martin
https://web.stanford.edu/~jurafsky/slp3/B.pdf



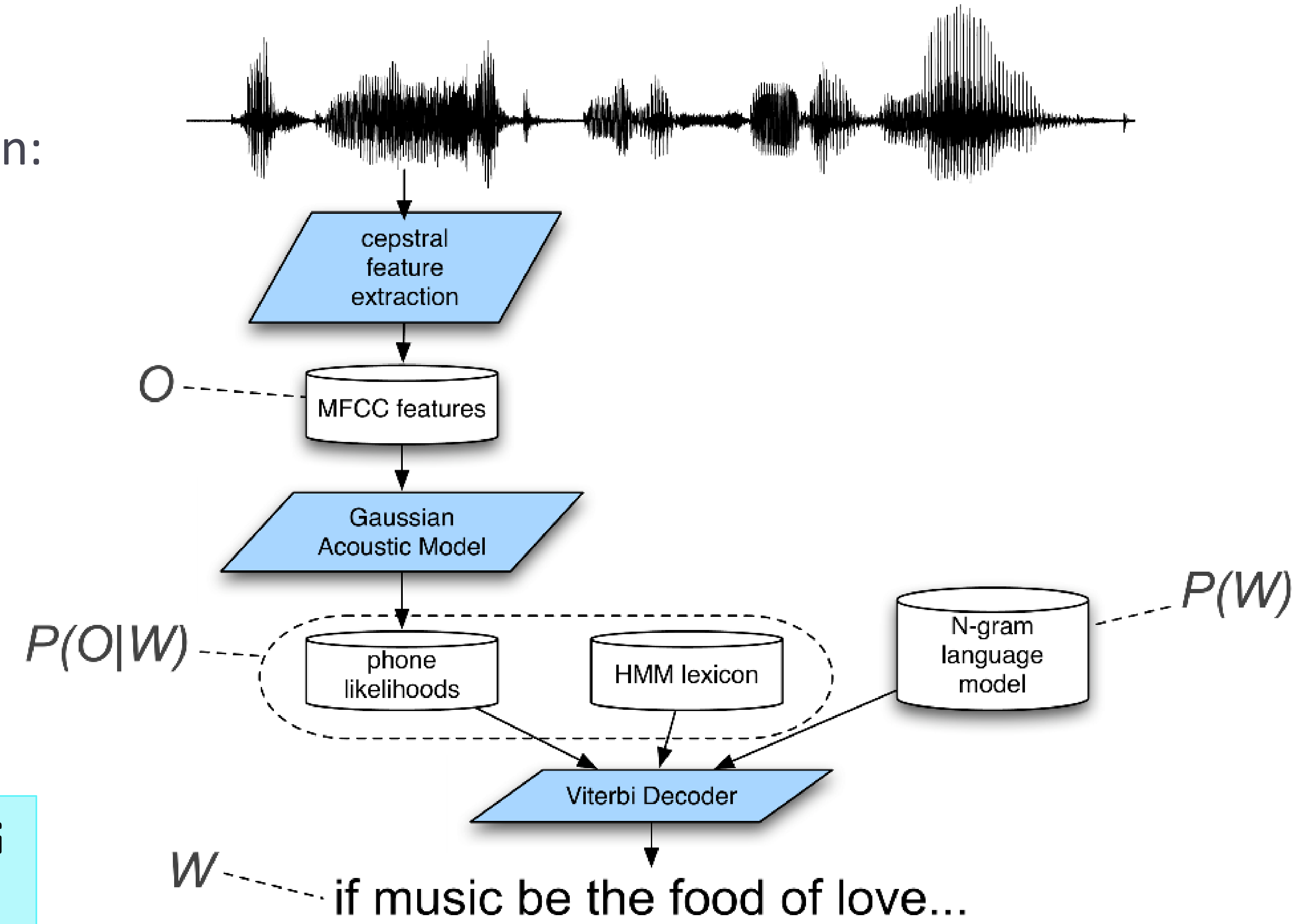
Speech recognition architecture (classic)

SLP, Jurafsky & Martin

- ASR components
  - Lexicons and pronunciation:
    - Hidden Markov Models
  - Feature extraction
  - Acoustic modeling
  - Decoding
  - Language modeling:
    - N-gram models

- But: why "classic"?

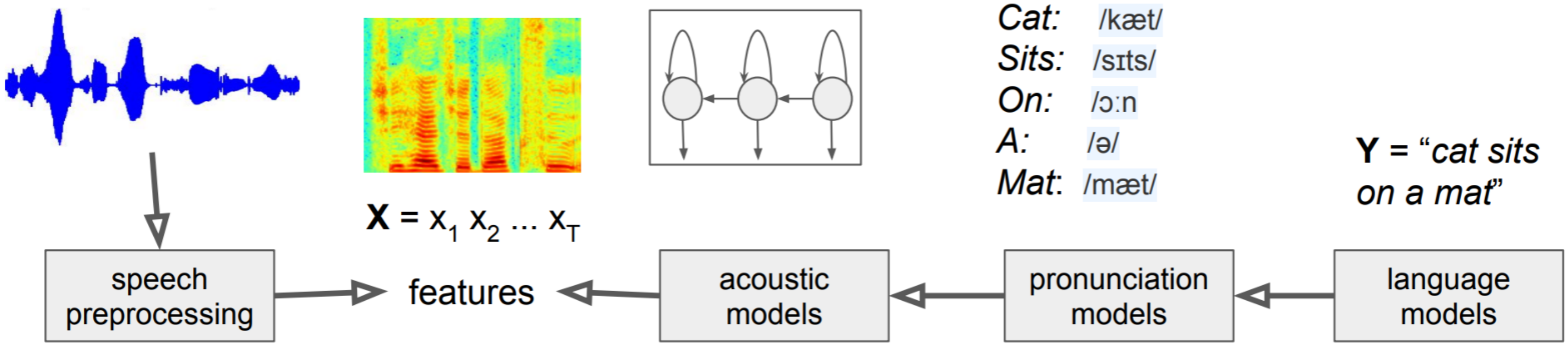
Because **DEEP LEARNING**
(what else?)



Speech recognition architecture (classic)

SLP, Jurafsky & Martin

- Inference: Given audio features $\mathbf{X} = x_1x_2\ldots x_T$ infer most likely text sequence $\mathbf{Y}^* = y_1y_2\ldots y_L$ that caused the audio features

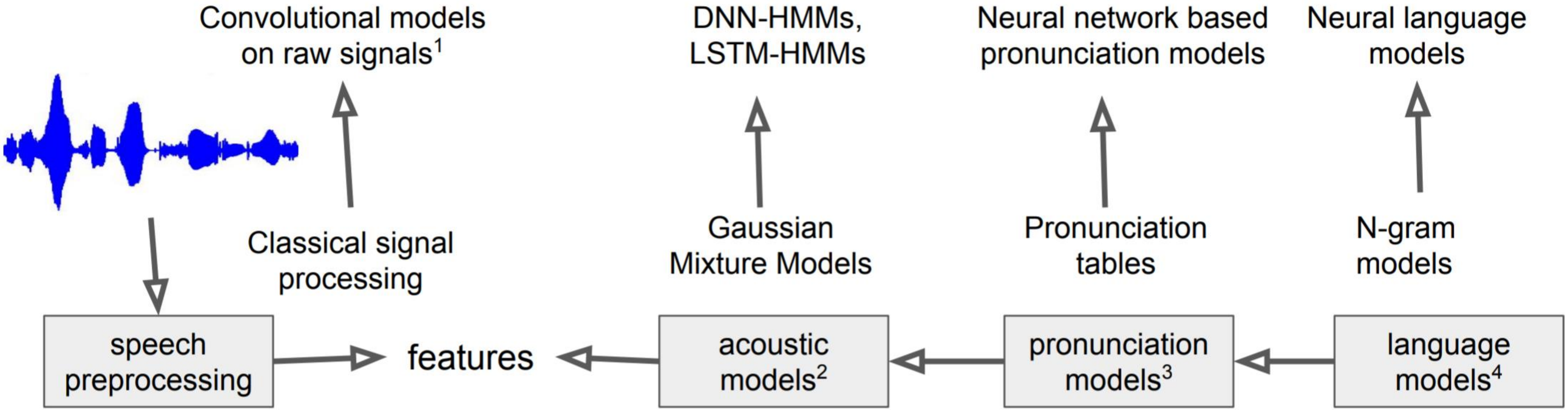


$$\mathbf{Y}^* = \arg\max_{\mathbf{Y}} p(\mathbf{X}|\mathbf{Y})\, p(\mathbf{Y})$$

Speech recognition architecture (neural net)

SLP, Jurafsky & Martin

- Each of the components seems to be better off with a neural network



Wrapping up

- Next class:
  - ELAN
  - Quick survey: speech data processing in Python

  - Project presentations: SC, EM