Al-assisted analysis of phonological variation in English

Special session on Deep Phonology, AMP 2025 UC Berkeley September 27, 2025

Virginia Partridge, Joe Pater, Parth Bhangla, Ali Nirheche and Brandon Prickett {vcpartridge, pater}@umass.edu

UMassAmherst



Background

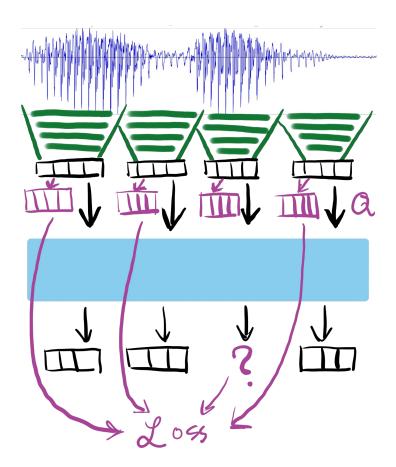
Despite the dramatic recent advances in speech recognition technology, automated phonetic transcription is not widely used in linguistics (as far as we know). This is especially surprising since the paper introducing Wav2Vec 2.0 (<u>Baevski et al. 2020</u>) presents SOTA results on a TIMIT phone recognition benchmark.

Our goal is to increase the usefulness of automated phonetic transcription for the study of phonological variation in English (variation within and across speakers and varieties)

In this paper, we:

- Situate our work within current research on automated transcription
- Present a Wav2Vec 2.0 model fine-tuned on Buckeye (<u>Pitt et al. 2005</u>), and compare it to other models, using TIMIT (<u>Garofolo et al. 1993</u>) as a test set
- Provide a web-based implementation, with Praat textgrid input and output
- Discuss next steps (and also issues with the standard TIMIT benchmark)

Wav2Vec 2.0 Pre-training: Sounds to vectors



Convolutional Layers: Snapshots of what's happening in audio at a particular point in time

Transformer Layer: Take into account relative positions of sounds in context

Masks + Quantization + Loss Function:

Perform *self-supervised learning* by hiding sound representations that must be predicted using a probabilistic function

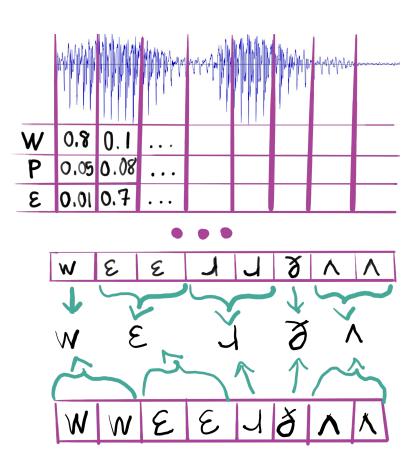
Facebook Research has open sourced many many pre-trained models for others to use.

Fine-tune: Add phone prediction

Bring your own audio with transcriptions.

Using Connectionist Temporal Classification loss, maximize the probability of the correct output by:

- Predicting each symbol's probability at each time segment
- 2) **Merging** repeated symbols across neighboring time segments
- 3) Maximize the marginal probability of alignments that lead to the correct output



Multilingual automated transcription

The primary focus of contemporary research on automated phonetic transcription is the development of universal models, trained on data from multiple languages, and applicable in principle to any language.

This research includes fine-tuned Wav2Vec 2.0 models (Xu et al. 2021, Taguchi et al. 2023) as well as the quite different Allosaurus (Li et al. 2020), which emphasizes the distinction between phoneme and phone recognition.

The latest contribution to this line of research is <u>Zhu et al. (2025)</u>. It advances the SOTA on benchmarks, but also recognizes that "error analysis reveals persistent limitations in modeling socio-phonetic diversity, underscoring challenges for future research"

Zhu et al. illustrate these limitations by testing the model on the Buckeye English corpus. We'll be doing this for multilingual models as well, but not with theirs, since we were unfortunately unable to get it to run.

Grapheme-to-phoneme vs. actual transcriptions

Multilingual transcription leverages the availability of large amounts of orthographically transcribed speech by applying grapheme-to-phoneme conversion (e.g. Epitran; Mortensen et al. 2018) to obtain phonetic transcriptions.

These are similar to dictionary transcriptions, and are of course not equivalent to transcriptions of the individual utterances:

- The pronunciation of individual words can vary across speakers, across utterances by an individual speaker, and across phonetic and phonological contexts
- The pronunciation of phonemes can differ within words. Allosaurus also uses phoneme-to-allophone conversion, but the transcriptions are still relatively abstract

AutoIPA

We call our model "AutoIPA" (Al-assisted in our title is thus ambiguous) It is a Wav2Vec 2.0 pre-trained model (facebook/wav2vec2-large-xlsr-53) fine-tuned on the Buckeye corpus:

- 40 speakers, balanced for gender and age (over 40 vs. under 40)
- White residents of the greater Columbus area (Northern Midland)
- 30 to 60 minutes of conversational speech for each speaker about 20 hours total
- Phonemic transcription + vowel nasalization, flap and glottal/glottalized stop (we adopt the <u>Seyfarth and Garellek 2020</u> revisions), includes syllabic sonorants
- No distinction between stressed/unstressed Λ/ə or 3-/ə- ("perceived quality alone" <u>Kiesling</u> et al. 2006: 18; see relatedly <u>Lindsay 2022</u>).

As far as we know, this is the first time Buckeye has been used in the training of an automated transcriber, though Buckeye and TIMIT have been used in training phone alignment models (e.g. <u>Kreuk et al. 2020</u>), and Zhu et al. use it as a test set.

Processing the Buckeye data

Data from 24 speakers used for training, 8 for development, and 8 for testing

- Each demographic is represented in equal proportion across these data sets
- E.g., 6 younger women, 6 older women, 6 younger men, and 6 older men in training
- Success on test data requires generalization to new speakers
 Training data divided into samples 8-12 seconds in length for Way2Vec 2.0 fine-tuning

Training data divided into samples 8-12 seconds in length for Wav2Vec 2.0 fine-tuning

ARPABET transcriptions converted to IPA; the symbol set with our ARPABET to IPA mappings can be seen here. We used Phonecodes (Hasegawa-Johnson 2019): our GitHub fork, our Python package

The task is phone (rather than feature) prediction, affricates and diphthongs treated as single phones, and nasalization and syllabicity are properties of individual phones

Evaluation Metrics: Phone Error Rate (PER)

What's the **ratio of phones** we get wrong in each test sample?

- 1) Compute the edit distance between your prediction *p* and your reference *r* to find the total number of phone insertions, substitutions or deletions between them.
- 2) Normalize by the number of phones in the reference *r*.

$$per(p,r) = rac{edit_distance(p,r)}{length(r)}$$

```
per([bop], [pop]) = \frac{1}{3}
per([bop], [po]) = 1
per([bip], [po]) = 1.5
```

LOWER IS BETTER!

Evaluation Metrics: Phone Feature Error Rate (PFER)*

How many articulatory features are we getting wrong?

- 1) Compute edit distance to find phone substitutions, deletions and insertions between your prediction *p* and your reference *r*.
- 2) Sum up the total cost of edit distance errors between aligned phones:
 - Phone deletions and insertions each cost 1
 - For substitutions, each feature mismatch costs 1/24 (there are 24 features in the table). This is the normalized Hamming distance between the articulatory features of the phones.

```
pfer([bo], [po]) = 0.04166
pfer([bop], [po]) = 1.04166
pfer([pɛ], [po]) = 0.125
```

LOWER IS BETTER!

^{*} Misleadingly **not really a rate**, but does obey the triangle inequality

Evaluation Metrics: Implementation

We use the <u>PanPhon Python library</u> (<u>Mortensen et al. 2016</u>) to calculate phone and feature edit distances.

For ease of use and reproducibility, we've published <u>a wrapper around PanPhon's</u> <u>distance metrics</u> that is compatible with the <u>HuggingFace evaluate package</u>. This makes them easy to compute when working with standard Python libraries for transformer-based speech recognition models.

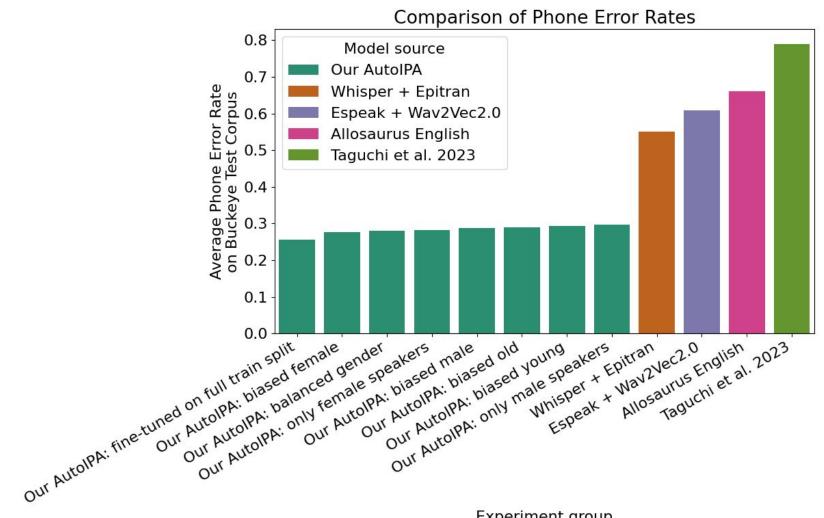
The results we report for models here are average PER and PFER across samples in the test corpus.

A caution on model comparison

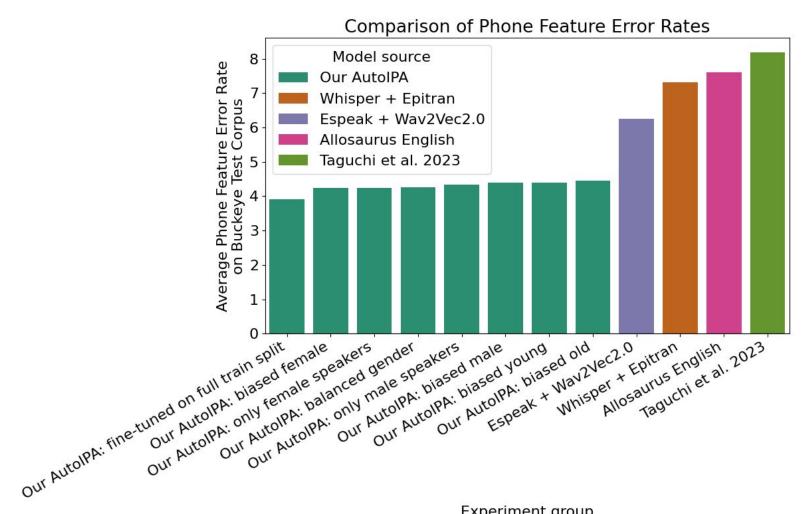
It is completely expected that AutoIPA will outperform the multilingual models!

But we need some way of relativizing the performance of AutoIPA

Future work might take our results as a benchmark to improve on



Experiment group



Some example transcriptions

The following are examples of transcriptions of utterances from the Buckeye test set by the models discussed here. The orthographic and phonetic transcriptions at the top are from Buckeye.

We have inserted spaces between words to increase readability.

s3802a_Utt33 (Older male)



where the trees were mature and they didn't really ruin our development and cut down a lot of the large trees were mature and they didn't really ruin our development and cut down a lot of the large trees were mature and they didn't really ruin our development and cut down a lot of the large trees were mature and they didn't really ruin our development and cut down a lot of the large trees

wει δε tʃıiz wɨ mʌtʃʊɪ εn neɪ dɪɾn ıɛli ɹuʌn aɪ dɪvɛlpmɛʔ n kʌt daʊr λ laɾʌ ðʌ laɹdʒ tʃɹiz

AutoIPA
ðə tɪiz wɨ mət͡ʃʊɪ ænd ðej dɪdnti ɪli ɪuən awɨ dɪvɛləpməntæ kʌt dawn ə lat ʌv ðə laɹðʒ
tɪiz

wει ðə tıiz wɨ məfʃu ænd ðej dɪdnt ıɪl iɹuən awɨ dɪvɛləpmənt ænd kʌt dawn ə lat ʌv ðə laɪd͡ʒ tıiz

Whisper + Epitran

Taguchi
wɜː ðə tuiːz wɜː mətʃuːu æn ðeɪ dɪdən uɪli uuːɪn ɑːu dɨvɛlpməntən kʌt daʊn ɐ lɑːɾʌv ðə
lɑːudʒ tuiːz
ESpeak+Wav2Vec 2.0

or ðe triz wu mitej on ei din wejvi wen aj divewmen ke dan bland do lads tris

s3801b_Utt80 (Older male)



yknow they benefited a lot from it now ji noυ ðei bεηλfirλd λ la? fiλm λt nau

ji rov ðei berafirar a la? flam a? nav Autolpa

ðej bεnəfitid ə lat fıλm naw

Whisper + Epitran

Allosaurus

ðej benəfitid ə lat finm it

i nεν ðεi benεfedid ε aː fam nεw Taguchi

i: na dei peutturi a la:t try usa

ESpeak+Wav2Vec 2.0

s3902a Utt109 (Younger female)



still just sitting down and my computer sits next to it VOCNOISE um stil dans sirin dann n mai kampjudi sits nekst tu i? am

stil dzus sirin daur am mai kmpjiri sits neks tu λ? λm AutolPA

stil danst sitin dawn on maj kompjuti sets neks to it

stil danst sitin dawn on maj kompjuti sits nekst to it

stow: dwi s:erin da:n a maj kimpçirj sets neks te wi am

stil dans sirin dan a mai kəmpilrə sittəs nekstulit am

Allosaurus

Taguchi

Whisper + Epitran

ESpeak+Wav2Vec 2.0

Top edit distance errors on the Buckeye test split

These are the **most common mistakes** we saw from a model fine-tuned on all data in the Buckeye train split (ginic/full_dataset_train_3_wav2vec2-large-xlsr-53-buckeye-ipa).

Substitution	Percentage of all substitutions
$I \to V$	5.27 %
I → E	4.70
$V \to I$	4.02
r~→ n	2.57
$\varepsilon \to V$	2.32
$O \rightarrow V$	2.29
i → I	2.28

Deletion	Percentage of all deletions
I	14.74 %
Ω	9.34
Λ	9.09
ı	5.88
n	5.70
t	4.95
1	3.64

Insertion	Percentage of all insertions
I	9.76 %
t	8.57
Λ	7.68
n	7.57
d	4.37
ε	4.13
σ	3.95

Vowel Error Rates

For vowels v in the vocabulary V, we want to see **which are most challenging** across the entire corpus.

Which vowels is the model getting wrong most frequently?

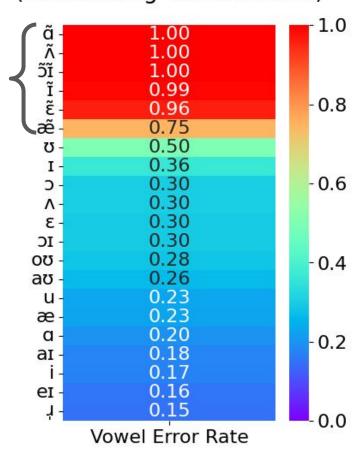
$$count_errors(v) = \sum_{w \in V} count_substitutions(v, w) + count_deletion(v)$$

$$error_rate(v) = \frac{count_errors(v)}{total\ count(v)}$$

AutoIPA Buckeye Test Set Vowel Error Rates (Descending worst to best)

We don't do well on nasalized vowels.

They are relatively rare overall, which makes them more challenging for the model to recognize. They are also apparently not well transcribed by Buckeye's humans.



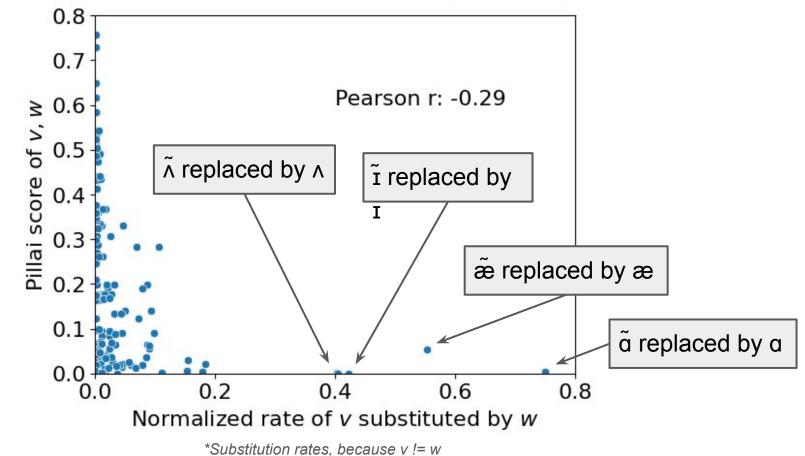
Pillai scores

Pillai scores are a measure of distributional overlap used in sociolinguistics to quantify the degree of vowel merger (<u>Hay et al. 2006</u>, <u>Stanley and Sneller 2023</u>).

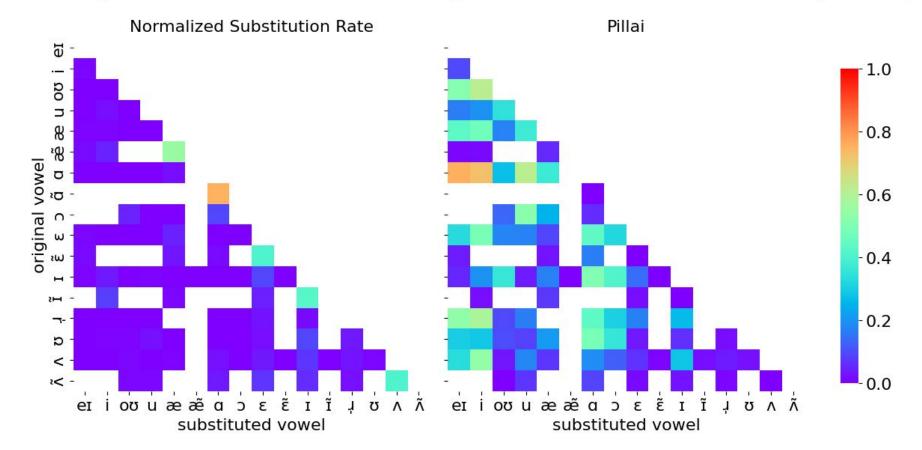
We calculated Pillai scores based on F1 and F2 measurements of the Buckeye (and TIMIT) vowels to see if they correlated with error rates

A high Pillai score indicates a low degree of overlap, or a high degree of contrast

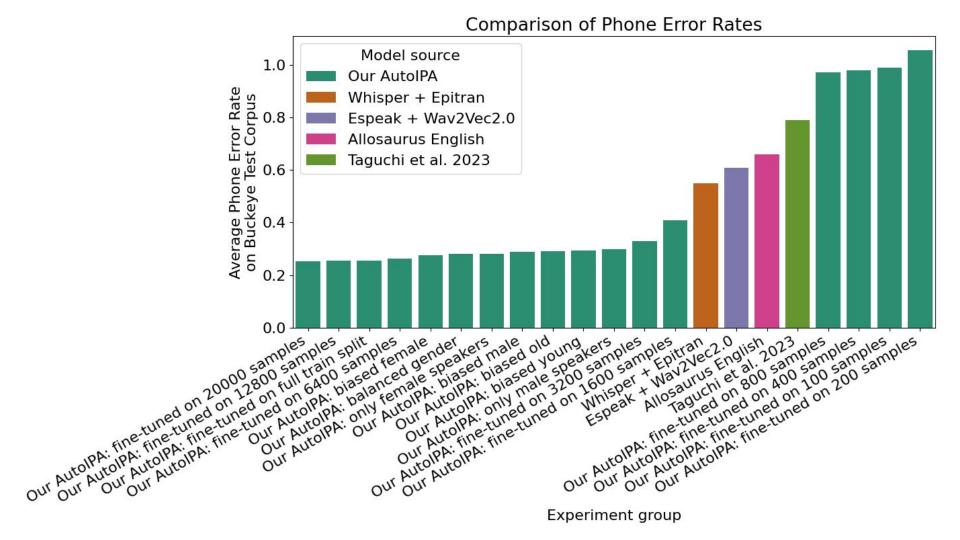
Relationship between Pillai score within Buckeye and substitution rate in the Buckeye test split

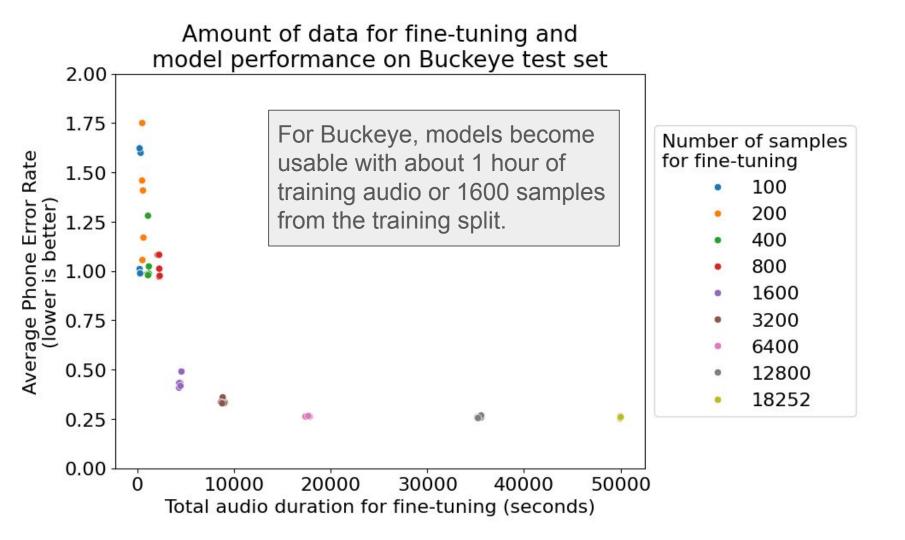


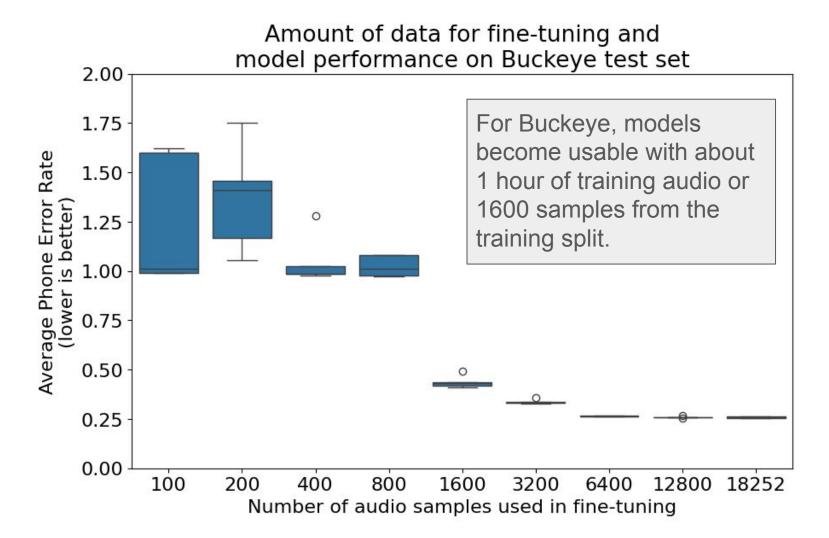
Relationship between Pillai score within Buckeye and substitution rate in the Buckeye test split



Impact of the quantity of fine-tuning data







Amount of fine-tuning data: Practical implications



Speaker S25 in Buckeye test split: "over by Riverside Hospital"

Original Buckeye transcription: [סט ע ש b aı ı ı ע א s aı d h a s p ı ɾ l]

Model fine-tuned on full Buckeye train split	In a shiri as ar q h a s b i u l
12800 sample fine-tuned model	s n i p ar i r n i s ar q p a s b i u j
6400 sample fine-tuned model	s n m i p aı ı ı n i s aı q µ a s b ı u
3200 sample fine-tuned model	אָר זו p <mark>eī או s s aī eī</mark> d h a s b ז ר <mark>סמ</mark>
1600 sample fine-tuned model	<mark>ɪf </mark>
800 sample fine-tuned model	
400 sample fine-tuned model	

Amount of fine-tuning data: Practical implications

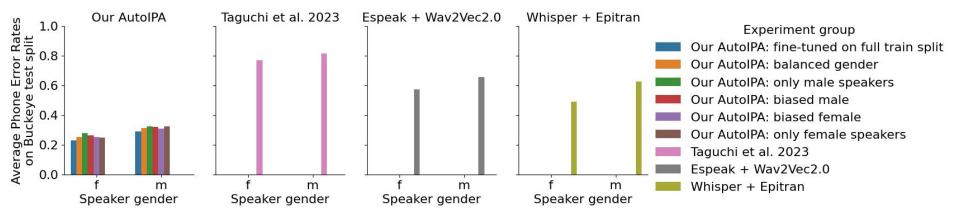
Speaker S38 in Buckeye test split: "oh they are they're killing on it they really are"
Original Buckeye transcription: [ου ð ε α ɹ ð ɹ k ɪ l ɪ ŋ ɔ n ʌ t ð ε ɹ i l i ɔ ɹ]

Model fine-tuned on full Buckeye train split	ου ð <mark>eɪ</mark> α ı ð <mark>ɛ</mark> ı k ɪ l <mark>n</mark>
12800 sample fine-tuned model	ου ð <mark>eɪ</mark> a ı ð <mark>ɛ </mark> k ɪ l <mark>n</mark>
6400 sample fine-tuned model	ου ð <mark>eɪ</mark> a ıð <mark>ε </mark> k ɪ l <mark>n n a m</mark> n M l i a s
3200 sample fine-tuned model	ου ð <mark>eɪ</mark> a ı ð <mark>ɛ l ʔ </mark>
1600 sample fine-tuned model	αðæα Mðεkεl Maλ Ellau
800 sample fine-tuned model	
400 sample fine-tuned model	t

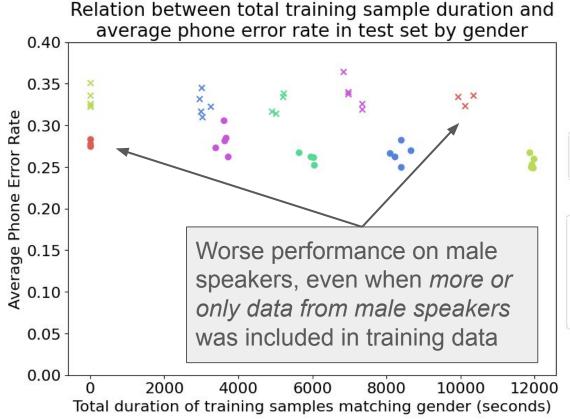
Red = Deletion, Blue = Insertion, Yellow = Substitution

Gender Effects

All models had better performance on female speakers in the Buckeye test data...



...regardless of how fine-tuning data was selected



Test sample speaker

- x male speaker
- female speaker

Fine-tuning data setting

- Balanced gender
- Only male speakers
- Biased with 70% training samples male
- Biased with 70% training samples female
- Only female speakers

Yao et al. 2010: 108 on Buckeye data

Generally speaking, female speakers produce slightly longer vowels than male speakers (p=0.009). As expected, they also have higher formant frequencies compared to male speakers (p<0.001 for F1, p=0.002 for F2). More importantly, as can be seen from Figure 3, on average female speakers have a much larger vowel space than male speakers. This is also consistent with previous findings (Byrd, 1994). Both longer duration and more expanded vowel space are indicators of clear speech (Bradlow et al., 1996), which suggests that female speakers produce clearer speech than male speakers.

Adda-Decker & Lamel 2005: on speech recognizers performing better on female speakers

Results consistently show a lower word error rate on female speech ranging from 0.7 to 7% depending on the condition. An analysis of automatically produced pronunciations in speech training corpora (totaling 4000 hours of speech) revealed that female speakers tend to stick more consistently to standard pronunciations than male speakers. Concerning speech disfluencies, male speakers show larger proportions of filled pauses and repetitions, as compared to females.

TIMIT as a test set

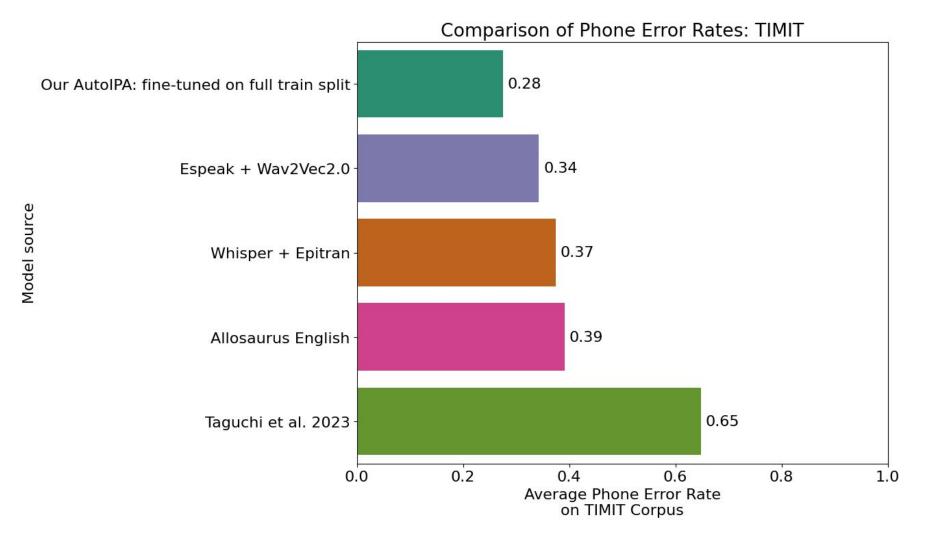
To begin to get a sense of how our model performs on speech from other varieties of English, we tested it on TIMIT

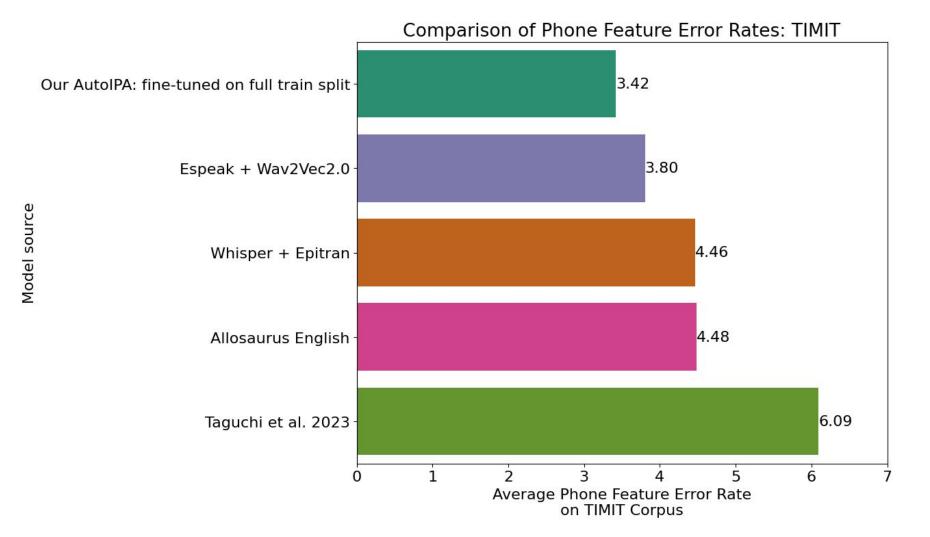
TIMIT consists of transcribed speech from 630 speakers from 8 dialect regions, each reading ten "phonetically rich" sentences. There are 5 hours of speech in total.

The ARPABET to IPA translation was trickier for TIMIT, because stop closure is indicated separately from release. We decided to merge all closure-release sequences for the same phone into a single segment.

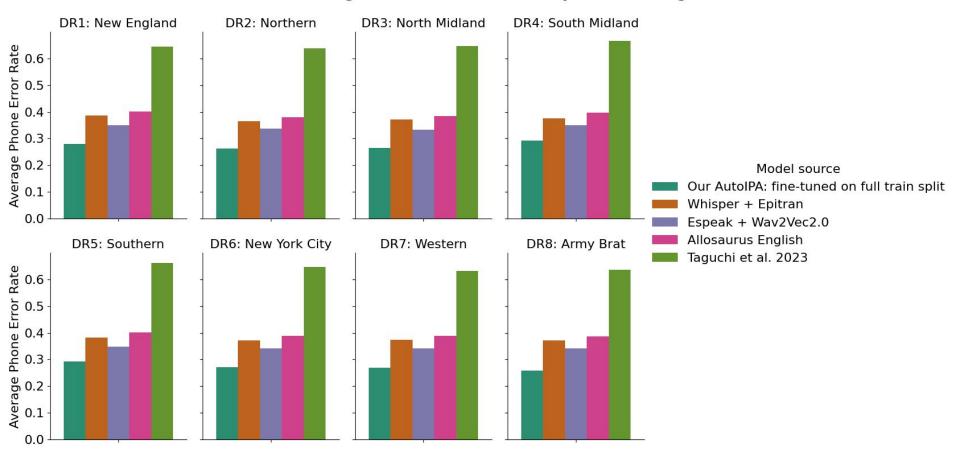
The mappings can be examined <u>here</u> (note that Phonecodes is greedy, so the longer input will be chosen first)

The standard approach to TIMIT in speech recognition (<u>Lee and Hon 1989</u>) is to merge all closures with silence. This means that all coda unreleased stops are lost (!)

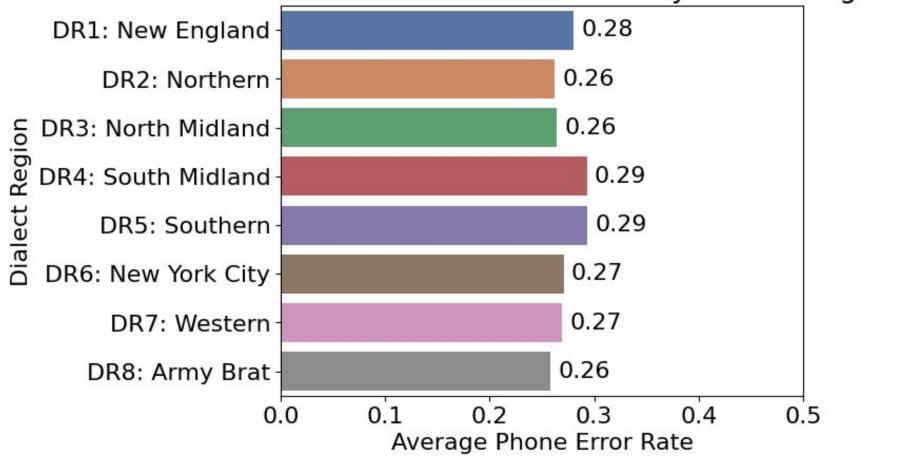




Models' Average Phone Error Rates by Dialect Region



Our AutoIPA's TIMIT Performance by Dialect Region



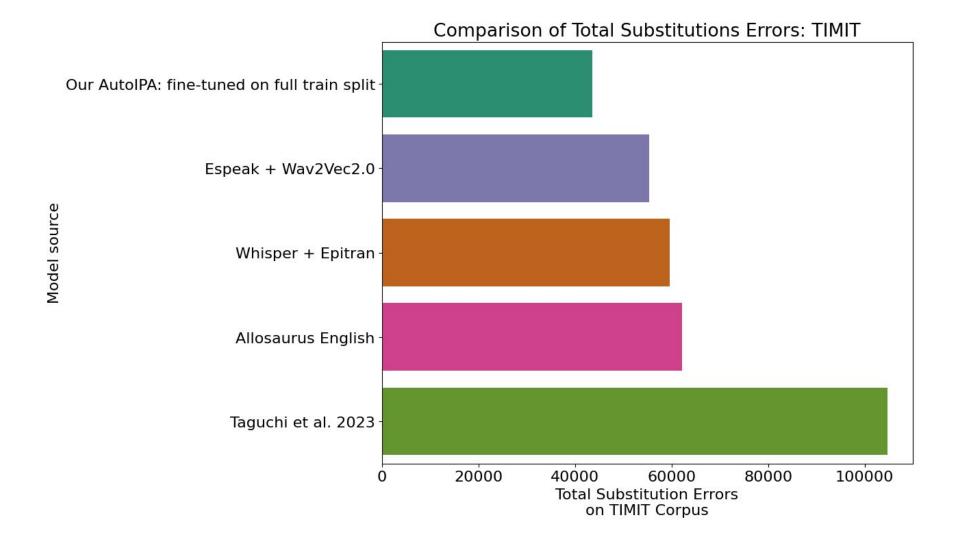
Same sentence, different regions, TIMIT vs. AutoIPA

She had your dark suit and greasy wash water all year

```
New England ("best New England accent so far") (VMH0, F, b. 1960, rec. 1986) 

ʃi hæd jɨ dak sutn gɹisi waʃ warə ʔɔl jɪə ΤΙΜΙΤ
ʃi hæd jɪ dak suʔn gɹisi waʃ warʌ οʊ jɛ AutoIPA
```

TIMIT Error Analysis



Top substitution errors on the TIMIT Corpus

Our AutoIPA

Substitution	Count of error
$i \rightarrow I$	4704
9 → V	3741
$\stackrel{\circ}{\to}^{1}$	3202

Espeak + Wav2Vec2.0

Substitution	Count of error
i → iː	4075
i→I	3502
a → a:	3435

Whisper + Epitran

Substitution	Count of error
ı⇒j	6444
i→ə	4545
$\Omega \to M$	3049

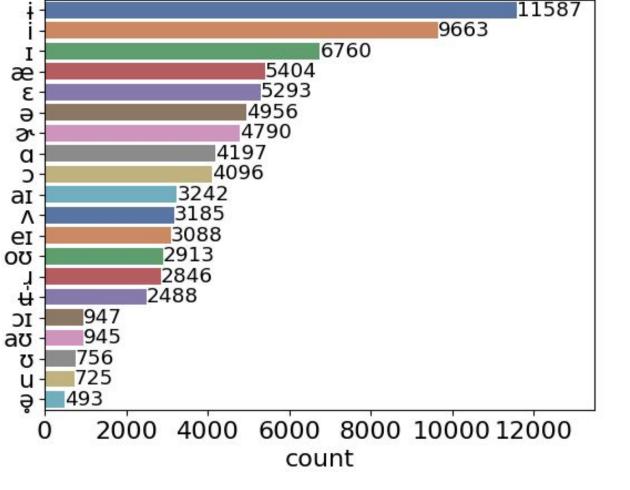
Allosaurus English

Substitution	Count of error
ı → j	6189
i→ə	4515
i → I	3122

Taguchi et al. 2023

Substitution	Count of error
ı → i	2777
ı→r	1948
ı → j	1763

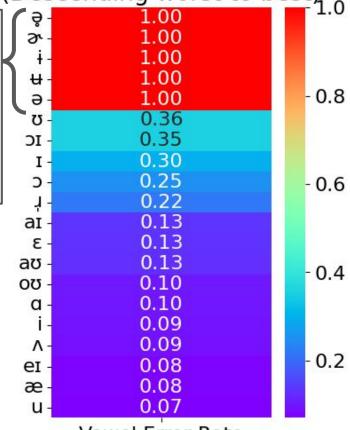
Counts of TIMIT Vowel Occurrences



AutoIPA TIMIT Vowel Error Rates (Descending worst to best)

These errors reflect differences in transcription conventions between corpora.

Because these are not in the Buckeye vocabulary, our model cannot output the symbols.



Vowel Error Rate

Distributions of Errors for a Given Vowel

For a given vowel *v* in the TIMIT vocabulary *V*, we want to understand **how the model** is getting the vowel wrong.

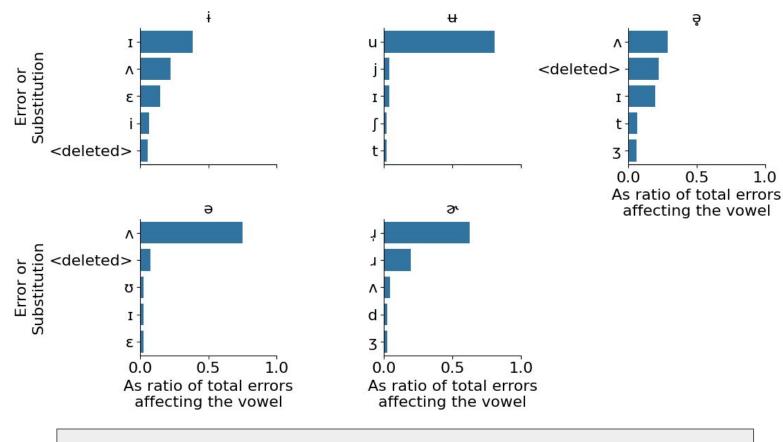
What **kinds of mistakes** and **in what distributions** does the model make for each vowel? Can this tell us anything about the corpus or its language varieties?

For a fixed vowel v and specific type error on v (deletion or substitution by w), normalize by the total number of errors:

$$\frac{count_deletion(v)}{count_errors(v)} \qquad \frac{count_substitutions(v,w)}{count_errors(v)}$$

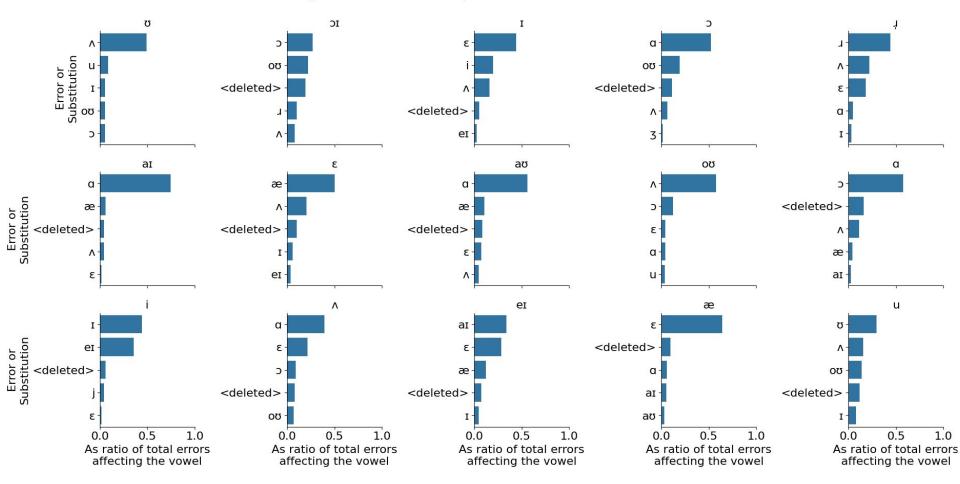
^{*} Conditional probability of each error given that there is some error affecting v

Top 5 errors for vowels AutoIPA always incorrectly transcribes

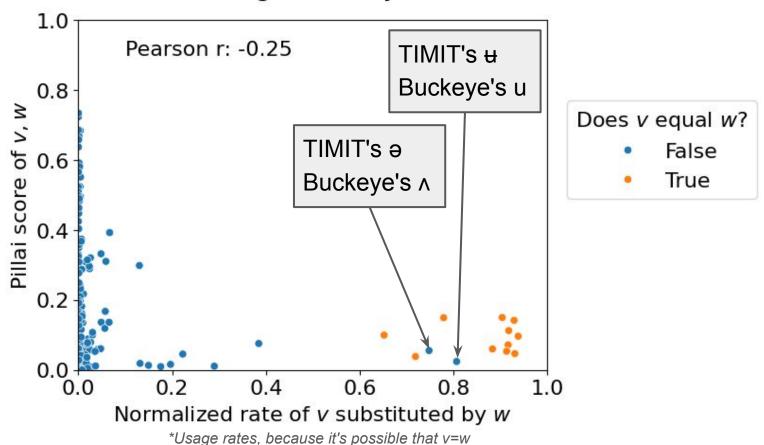


Correctable with simple replacement in pre- or post- processing

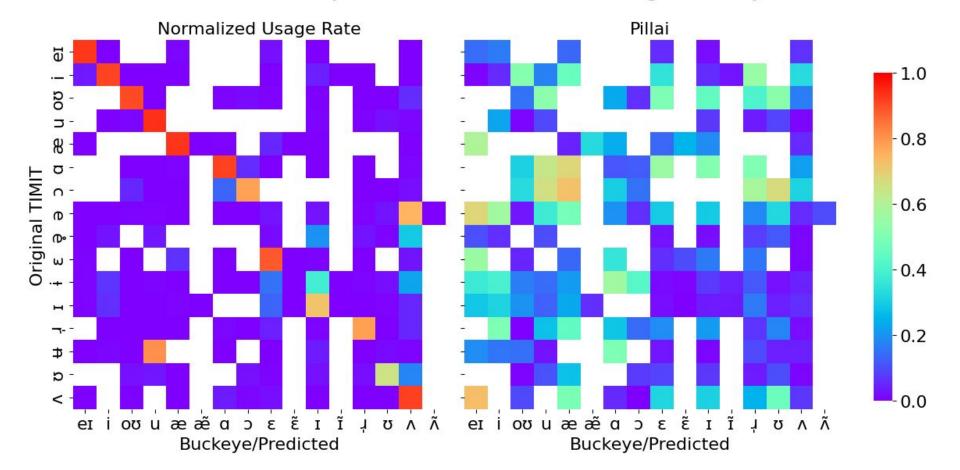
Remaining TIMIT Vowels: Top 5 AutoIPA Errors for each vowel



Pillai scores of TIMIT vs. Buckeye Vowels and normalized usage rates by AutoIPA on TIMIT



Pillai scores of TIMIT vs. Buckeye Vowels and normalized usage rates by AutoIPA on TIMIT



Web-based implementation with text grid support

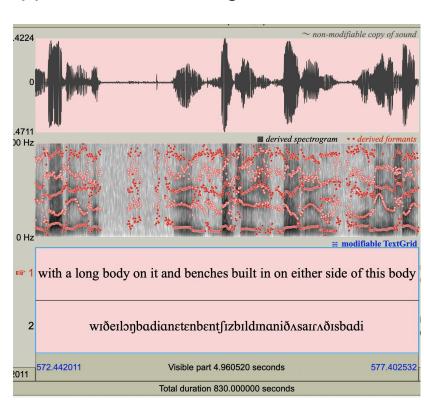
AutoIPA is now available on hugging face, with support for Praat text grids

We are now working on outputting phone alignments

Transcription of JP (quasi-Canadian) saying "I put my cat on a cot. I put my cat on a cot. I put my cot on a cat.":

ar ρυ? mar **ka** anλ **kɔ** λ ρυ? m **ka** an nλ **kɔ?** λ ρυ? mar **kɔ?** an nλ **kap**

It uses [æ] for JP's "ran".



Next steps

Wav2Vec 2.0 has been used as a classifier by <u>Kim et al. 2024</u> for nasalization and by <u>Tanner et al. 2025</u> for stop realizations

AutoIPA will likely be useful as a pretrained model to be fine-tuned for that kind of work

It likely also has many applications out of the box (especially in the study of word-final consonant realizations)

Our next step is in applying it to the study of phonological variation at the "border" of western and eastern New England dialect regions in Western Massachusetts

We foresee an iterative approach, getting first pass transcriptions from AutoIPA, correcting them, and then fine-tuning our model with the new transcriptions

Notes on the standard TIMIT phone reduction protocol

Acknowledgments

We gratefully acknowledge the support of the Center for Data Science and Artificial Intelligence, the HFA/CICS Collaborative Seed Fund, and the Public Interest Technology Initiative, all at UMass Amherst.