

Class 11: Audio Recognition and Alignment

Theme: Audio

Computational Analysis of Text, Audio, and Images, Fall 2023

Aarhus University

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Aarhus University

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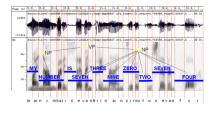
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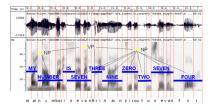
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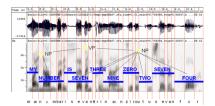
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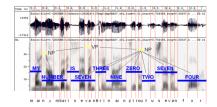
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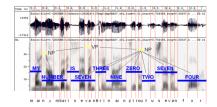
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- Direct text analysis (e.g. sentiment or topics)
- Indirect text analysis (e.g. hostile rhetoric removed in official transcripts)

Open-Source Tools

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	Whisper	Wav2Vec 2.0
Learning	weak-supervision	self-supervision
Input	log-mel spectrograms	raw waveforms
Languages	> 100	> 1,400
Architecture	encoder/decoder	encoder
Output	processed	raw
Timestamps	segment-level	character-level
Error	5 - 30%	20 - 50%
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Other (paid) systems as well, but Whisper and Wav2Vec 2.0 are the best options:

- Google Cloud Platform
- Amazon
- Assembly AI

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- WER is a generalization of Levenshtein's distance to the word-level rather than the character-level

Exercise

Consider the following two sentences:

- Target: "The cat is sleeping on the mat"
- ASR: "The a cat is sweeping on mat"
- 1. Compute the WER
- 2. Discuss weaknesses of the WER metric and potential alternative metrics

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- Audio transcription a preprocessing step for ASR
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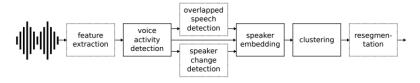
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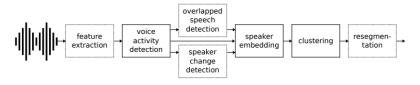
- Audio transcription a preprocessing step for ASR
- Speaker recognition a preprocessing step for speaker recognition
- → decomposes audio to the speaker-level

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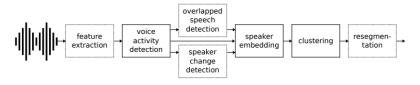


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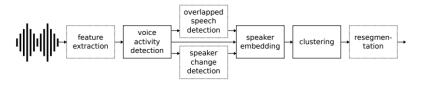
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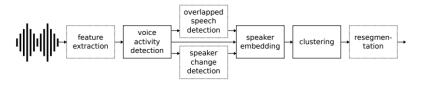
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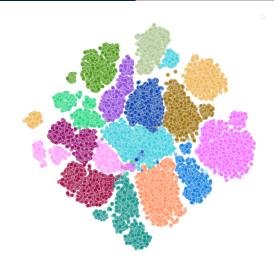


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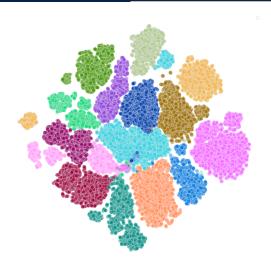
- Comparable to word embeddings: a fixed-length vector representation of a speaker's unique vocal traits, speaking style, and speech-related information
- Encoded with neural networks pretrained models work surprisingly well

Two-Dimensional Visualization of Speaker Embeddings

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 \leadsto Embeddings are fixed-length x-vectors (Snyder et al., 2018) computed on diarized speech segments and then reduced using t-SNE

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 - Generally requires a supervised setup, but a weakly supervised setup is possible if we have auxiliary targets

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Speaker diarization functions as a preprocessing step that simplifies the subsequent tasks – when combined, you have a powerful annotation pipeline (Rask, 2023)

When speech-level transcripts are available for the recording we want to annotate, we can combine speaker diarization and ASR to perform weakly-supervised speaker identification using fuzzy string matching:

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- → Joint speaker diarization, ASR, and speaker recognition using unsupervised and weakly-supervised learning

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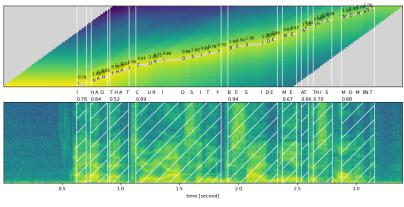
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 - The level of alignment depends on where we think the variation is in each modality

Exercise

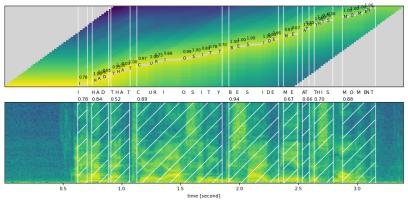
- 1. Discuss the difference between audio measurement when using the speech-level compared to the word-level.
- 2. Compare audio measures with text measures (e.g. using a dictionary) when we use speeches as the unit of analysis.

We can also align each audio and text at the level of each word:

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- → Combines wav2vec2.0 with a phoneme model as overhead to perform character-level ASR
- → Alternative approach: Faster Whisper or WhisperX
- → Alignment is non-destructive we can go back and forth between levels

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See you next week!

Theme: Audio

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References i

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