ML for Audio Study Group

Session 1:

December 10, 2021, 5 PM CET

hf.co/join/discord

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Vaibhav (VB) Srivastav

JOIN

Suggested readings before this session

- https://heartbeat.comet.ml/the-3-deep-learning-frameworks-for-end-to-end-speech-recognition-that-power-your-devices-37b891ddc380

Organisation

- **Community-led!**
  - We'll kick off with some basics, but we'll decide collaboratively where we want to focus
  - Anyone can participate!
  - Members of the HF team and other cool collaborators will join.
- Expectation
  - Before each session: **Read/watch related resources**
  - During each session, you can
    - Ask question in the forum
    - Present a short (~10-15mins) presentation on the topic (agree beforehand)
    - Participate a bit more passively (that's also ok and you're welcomed!)
  - Before/after:
    - Keep discussing/asking questions about the topic
    - Share interesting resources

Introduction

Omar Sanseviero (https://twitter.com/osanseviero)

- ML Engineer at Hugging Face
- Previously
  - SWE at Google Assistant
  - Co-founder AI Learners

Vaibhav Srivastav (https://twitter.com/reach_vb)

- MS student @ Uni Stuttgart/ Working Student @ Deloitte Tax
- Previously
  - Strategy @ Deloitte Consulting

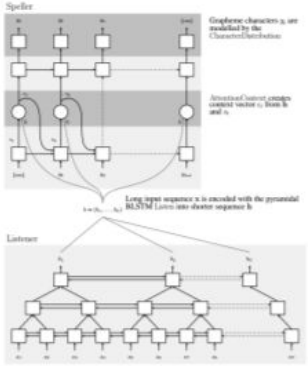
Timeline

- **Dec 14: Kick off session**
- Dec 21: ASR Deep Dive
- Dec 28: TTS Deep Dive
- Jan 11 and forward:
  - Paper discussions
  - Invited speakers
  - Deep dive into a specific task

An exciting time for spoken language processing

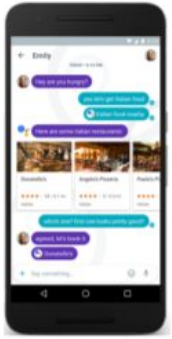
Amazon Alexa +
Alexa Prize
2014

Neural TTS voice cloning
2017



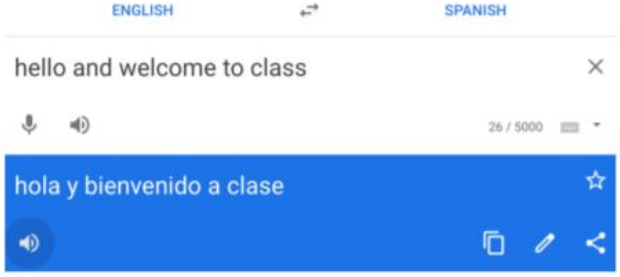
End-to-end neural becomes SOTA
2015 - present

Apple
Siri
2011



Google
Assistant
2016

Microsoft
Cortana
2014



Realtime speech-speech translation
2020

from - stanford CS224S

Dialogue (= Conversational Agents)

- Task-oriented conversations
- Personal Assistants (Alexa, Siri, etc.)
- Design considerations
  - Synchronous or asynchronous tasks
  - Pure speech, pure text, UI hybrids
  - Functionality versus personality

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Dialogue (= Conversational Agents)

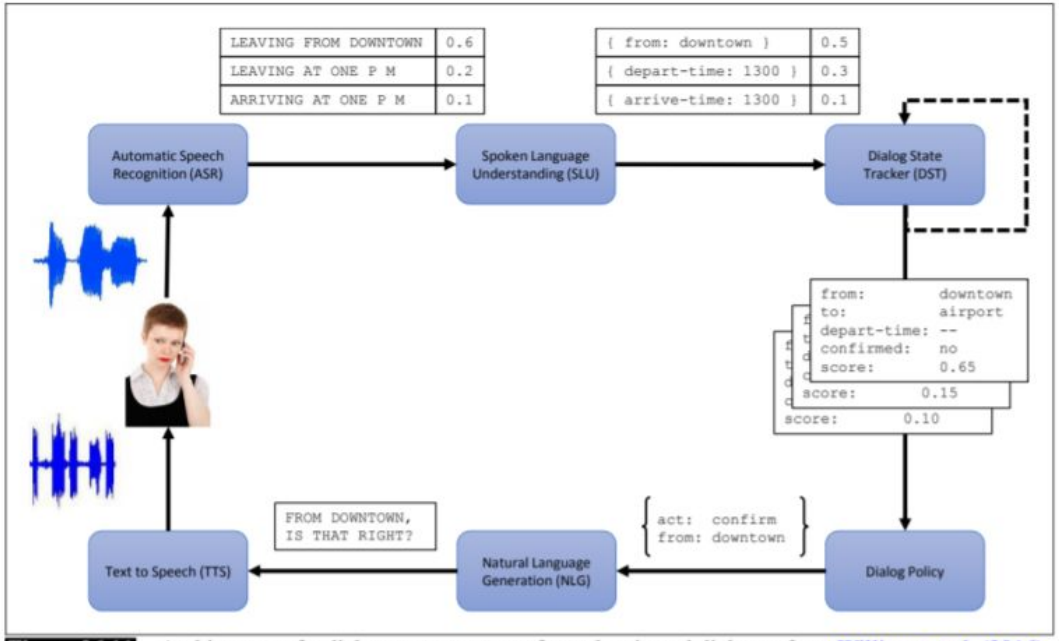


Figure 26.11 Architecture of a dialogue-state system for task-oriented dialogue from Williams et al. (2016).

from - stanford CS224S

Speech Recognition

Speech Recognition

- Large Vocabulary Continuous Speech Recognition (LVCSR)
  - ~64,000 words
  - Speaker independent (vs. speaker-dependent)
  - Continuous speech (vs isolated-word)

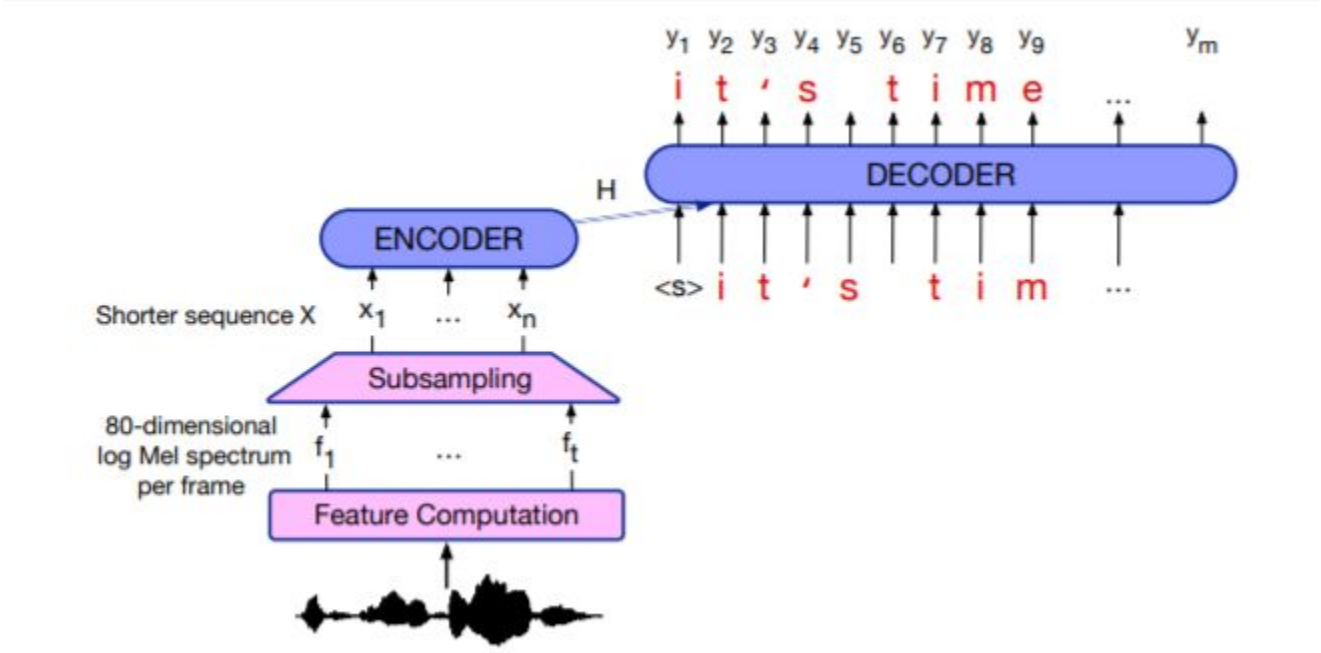
from - stanford CS224S

A slide explaining ASR architecture

It’s time for lunch!

from - SLP3, Ch 26

Basic architecture for ASR



from - SLP3, Ch 26

Current error rates

English Tasks	WER%
LibriSpeech audiobooks 960hour clean	1.4
LibriSpeech audiobooks 960hour other	2.6
Switchboard telephone conversations between strangers	5.8
CALLHOME telephone conversations between family	11.0
Sociolinguistic interviews, CORAAL (AAVE)	27.0
CHiMe5 dinner parties with body-worn microphones	47.9
CHiMe5 dinner parties with distant microphones	81.3
Chinese (Mandarin) Tasks	**CER%**
AISHELL-1 Mandarin read speech corpus	6.7
HKUST Mandarin Chinese telephone conversations	23.5

Figure 27.1 Rough Word Error Rates (WER = % of words misrecognized) reported around 2020 for ASR on various American English recognition tasks, and character error rates (CER) for two Chinese recognition tasks.

from - stanford CS224S

So is speech recognition solved? Why study it vs use some API?

- In the last ~5 years
  - Dramatic reduction in LVCSR error rates (16% to 6%)
  - Human level LVCSR performance on Switchboard
  - New class of recognizers (end to end neural network)
- Understanding how ASR works enables better ASR-enabled systems
  - What types of errors are easy to correct?
  - How can a downstream system make use of uncertain outputs?
  - How much would building our own improve on an API?
- Next generation of ASR challenges as systems go live on phones and in homes

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Speech Synthesis

TTS (= Text-to-Speech) (= Speech Synthesis)

- Produce speech from a text input
- Applications:
  - Personal Assistants
    - Apple SIRI
    - Microsoft Cortana
    - Google Assistant
  - Games
  - Announcements / voice-overs

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TTS Overview

- Collect lots of speech (5-50 hours) from one speaker, transcribe very carefully, all the syllables and phones and whatnot
- Rapid recent progress in neural approaches
- Modern systems are DNN-based, understandable, but not yet emotive

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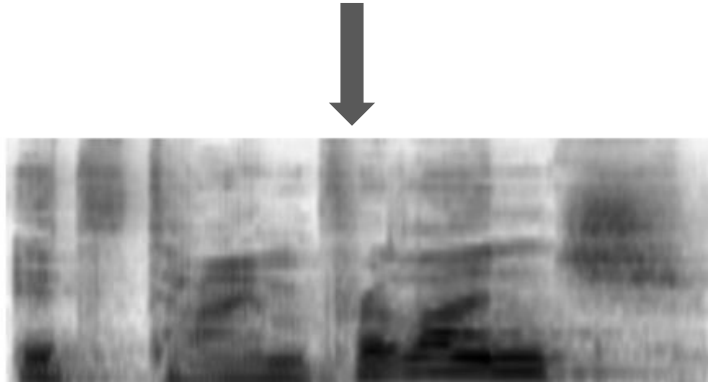
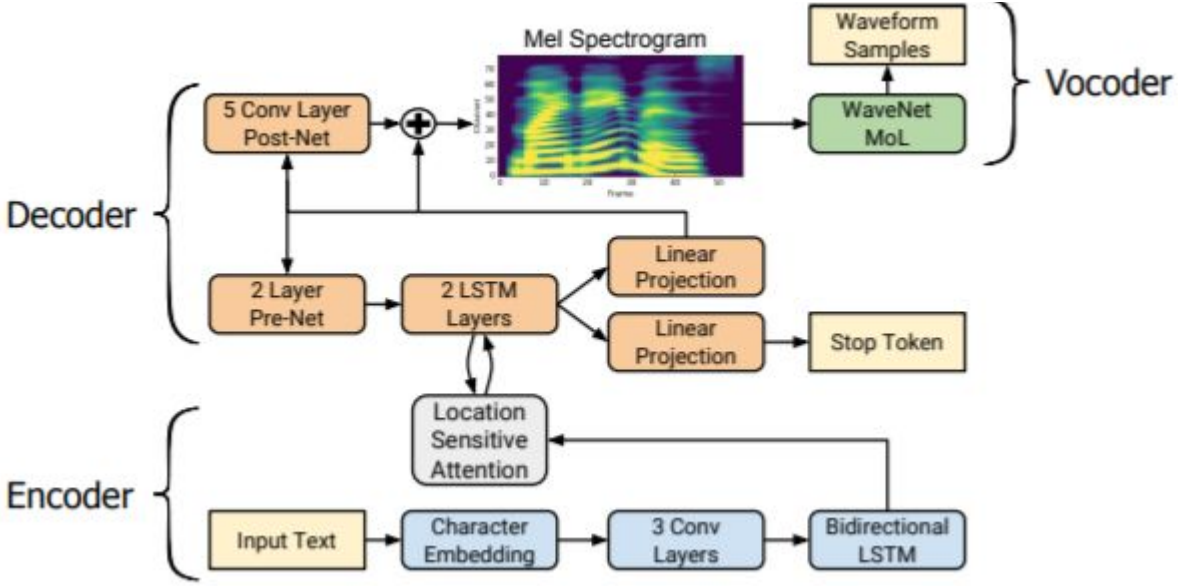
Text to Speech

It’s time for lunch!



from - SLP3, Ch 26

Text to Speech



from - SLP3, Ch 26

Applications

- Machine learning applications
  - Extract information from speech using supervised learning
  - Emotion, speaker ID, flirtation, deception, depression, intoxication
- Dialog system / SLU applications
  - Building systems to solve a problem
  - Medical transcription, reservations via chat

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Other speech related tasks

- **wake word** - to detect a word or short phrase, usually in order to wake up a voice-enable assistant
- **speaker diarization** - determining ‘who spoke when’ in a long speaker diarization multi-speaker audio recording
- **speaker recognition** - task of identifying the speaker
- **language identification** - identify which language is being spoken

Next steps

- Next week: 2 short (10-20min presentations + discussion
  - Presentation 1: Intro to Audio (Omar Sanseviero)
  - Presentation 2: ASR Deep Dive (Vaibhav Srivastav)
- Recommended resources
  - Intro to Audio for FastAI sections 1 and 2
  - SLP Chapter 26.1-26.5

Thanks for tuning in!