# StenoPhone

Team D6: Cambrea Earley, Ellen Seeser, and Mitchell Yang

#### **Use Case**

- Increase of distributed workplaces
- Web conference tech like zoom designed for individual participants
- Our project: a team meeting solution
  - Microphone/speaker hardware and web application
  - Audio conferencing
  - Automatic transcript generation featuring speaker identification

- ECE areas
  - Software (networking, web server, machine learning integration)
  - Signals (audio processing)

## Requirements: Audio Conferencing

- Setting Specifications: Conference room
- Audio transmission latency: "mouth to ear delay" from mic to second speaker less than 150 ms
- Audio quality: fewer than 5% of audio packets should be lost
- Battery life: two hours of continuous use

## Requirements: Transcripting

- Transcript latency: average word delay of less than three seconds
- Transcript accuracy: word error rate of less than 25%
  - Word errors: insertion, deletion, substitution of words
- Speaker identification accuracy: identification error of less than 25%
- Transcript formatting error for multi-room meetings: 5%
  - Formatting errors: incorrect chronology, missing speaker identification

## Key Challenges

- Networking: streaming audio with metadata
  - Risks: Latency, dropped packets
  - Mitigations: UDP connection
- Overlapping audio from multiple users in one room
  - o Risks: Can affect transcription
  - Mitigations: Protocol for speaker selection, beamforming
- Identification of moving speakers
  - Risks: speaker movement near the beginning of the meeting when the ML model has very little voice data for the speaker
  - Mitigations: merge ML findings with direction-of-arrival

## Solution Approach: Hardware

- Microphone array: ReSpeaker Mic Array v2.0 with attached Speaker
- Local networking and processing: Raspberry Pi 3
- AWS server with both UDP and HTTP/TCP Ports
- Lithium Ion Battery

## Solution Approach: Software and ML

Audio Processing: Audacity python scripting

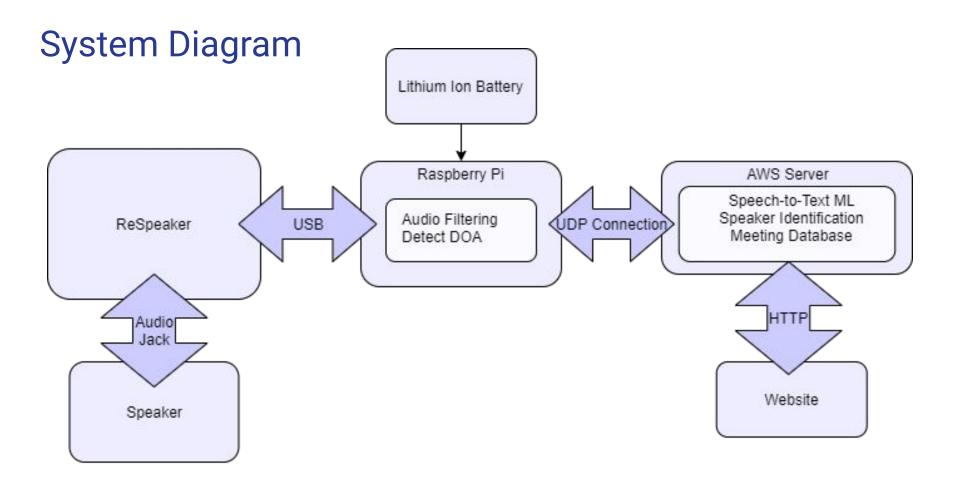
Website: Django, SQLite database

#### Speaker Identification & Diarization:

Solution	Advertised DER
pyannote audio	25% (2020)
руВК	12% (2016)
Hitachi Speech EEND	15% (2019)
BUTSpeechFIT VBx	22% (2020)

#### Speech to Text Solution:

Solution	Advertised WER
Mozilla DeepSpeech	7.5% (2019)
CMU PocketSphinx	10% (2006)
PyPi SpeechRecognition	Variety of engines included



## Testing, Verification, and Metrics

Requirement	Metric	Test
Audio Transmission Latency	Mouth-to-Ear Latency (ms) < 150 ms	Capture timestamp of sound snippet input and final sound output after routing through server
Audio Quality	Dropped packets (%) < 5%	Count original and final number of packets after transmitting an audio stream
Battery Life	Hours of continuous use > 2hrs	Run device under heavy load for a set time to find battery usage

## Testing, Verification, and Metrics

Transcript Latency	Average Word Delay (s) < 3s	Capture timestamp of audio captured by mic and timestamp of packet arrival in browser
Transcript Accuracy	Word Error Rate (%) < 25%	Check transcript for word error (substitution, deletion, and insertion) after speaking a known text
Speaker Identification Accuracy	Speaker Identification Error (%) < 25%	Check transcript for identification error after conducting a conversation with speaker switches
Formatting Accuracy (chronology and speaker ID tags)	Formatting Error Rate (%) < 5%	Check transcript for formatting error instances after conducting a conversation with known contents

#### Tasks and Division of Labor

#### Cambrea

- Hardware setup and integration
- Networking
- Audio streaming

#### Mitchell

- Audio processing
- Web app infrastructure
- Database

#### Ellen

- Machine learning integration
- Speaker identification system
- Meeting setup flow

### Schedule

