Developing an App to collect data for people at risk of Alzheimer's Disease Yuan Shang (yshang42)

Part 1 - Project Topic

Alzheimer's Disease (AD, here and after) is a chronic neurodegenerative disease and the most common cause of dementia. In the United States, more than 6 million people are living with AD, and the number are predicted to triple by 2050, which would be a heavy social burden. The early signs of the disease include forgetting recent events or conversations. Over time, it progresses to serious memory problems and loss of the ability to perform everyday tasks. This whole process may last for over 20 years.

Timing of interventions to the disease are very critical to the clinical outcomes. A lot of phase-3 clinical trials fail due to the interventions are given too late, and much hope has been placed on interventions for patients who are at prodromal and preclinical stages. Thus, we need more accurate early diagnostic methods to identify patients at the early biological stage of the disease for early intervention. Currently, the Clinical Dementia Rating Scale (CDR) is a global assessment tool that can be used to diagnose dementia caused by AD. However, the CDR takes about 90 minutes to administer, and is based on an interview with the patient and a professional caregivers or informant, making it not feasible to identify potential subjects with AD risk. More convenient tools are highly in demand.

Recent years, Linguistic performance has been shown to be useful as an early biomarker of AD in the Framingham Heart Study participants. However, this has not been widely applied due to 1. Lacking tools to collect and store these linguistic data; 2. Lacking a quantitative feature to quantify the linguistic performance that align with AD matrix such as CDR-SB.

To address this question, I plan to develop an APP to collect and quantify linguistic performance. This APP will have two modes: 1. Calibration mode and 2. Record mode. In the mode 1, both audio data and CDR-SB are recorded. This mode will help to calibrate the model, and need to be used by a candidate and a clinician for CDR-SB. In the mode 2, the voice data is recorded and evaluated. This mode could be used by a candidate in daily use.

Part 2 - Technical Design

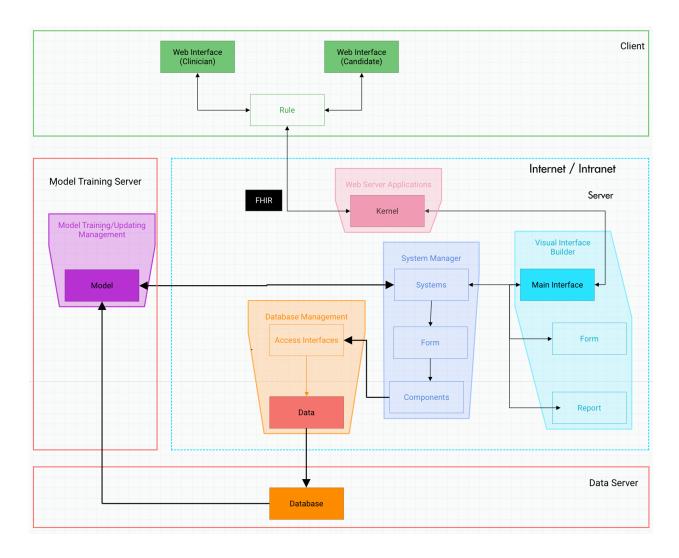
Tools/Technology:

- 1. Python will be the main developing language. Python fhir.resources library will be used if required.
- 2. Postgres will be the database to record candidate's demographics data.
- 3. Python flash will be used for web development.

Datasets and Data Sources

- 1. Based on fhir.resources and OMOP, design relationship DB to store patient information.
- 2. Based on CDR-SB, design test matrix collections webpages to get CDR-SB.
- 3. Based on goolge-news, extract a random paragraph of reasonable length. This text will be provided to the candidate to read through.

Architecture Diagram

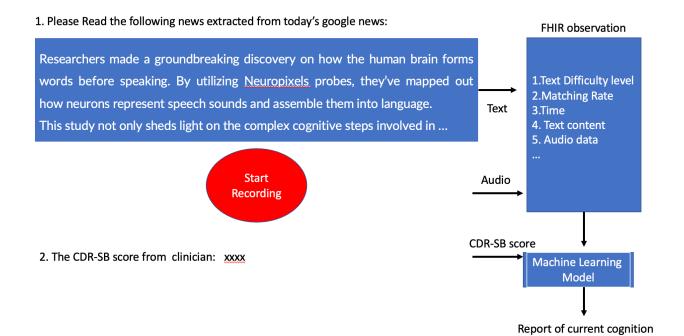


The architecture diagram is generated using online tool at (https://draft.io/2uztsmcfdv6kxwma4hdtuuvf73xae6qxknxsrgb2huhf). The Client server is the browser which showed the interface to interaction with candidates or clinicians.

The data between Client and Internet Server is based on FHIR. The data is parsed. Based on some rules, alignments between audio data and CDR-SB will be updated through Model training server(NOT implemented here as it would take tremendous effort to dealing with audio data, and need a lot of data to training deep learning model). The data will also be sent to data server through database management system. The assessment of current audio data from the current model will be presented to the client as a report.

Screen Mock-up(s)

An diagram of the Screen Mock-up showing the data collections is shown below:



Part 3 - Implementation Plan

3.1 Project Tasks

Task	Time
1. Information Flow	Mar 11th
2. Database schema, tables	Mar 11th
3. Abstract code	Mar 18th
4. Flash webserver deployment	April 4th
5. Data simulation for machine learning models and deployment	April 18th
6. Finetuning and finalization	April 30th

3.2 Needs/Risks

The largest risk for the project is how to deal with the audio data. How to extract useful features from audio data? How to access audio data with the text?

However, I know it is very important to collect these data. Thus, at the current stages, I will use 'artificial machine learning models' and only use features such as time finishing reading, correct rate etc to train simple regression models.

In future, if this APP was widely used and more and more data were collected, we could develop more fancy machine learning models that could match the audio data to the real time cognition status. If this become true, the impact to the field would be huge.