Assignment Objectives

Create a RESTful service. This service allows its users to upload audio files and specify the language of audio (it could be English or Hindi). The service extracts the following pieces of information from the audio file:

- 1. How many speakers were present in the audio. There can be up to two speakers.
- 2. What word was said at what time and by whom within the audio.
- 3. Dominant sentiment in the audio.

Since the audio processing can take a while, the API works asynchronously. When the user submits an audio file, the API returns a job-id. The user can query that job-id to find the status of audio processing. If the audio processing is still in progress, the query returns a response indicating that the file is still being processed. However, if the audio processing is over, the query returns the information extracted from the audio file.

Finally, write a document that describes your approach for solving the problem, what open source tools and cloud services you used, what audio files did you use for testing the code and the instructions for someone else to run your code.

Extra credit for packaging the service in a docker image.

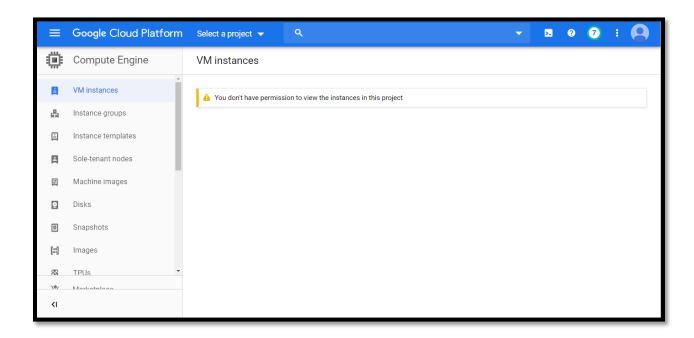
Approach:

To achieve the project objectives followed the below approach:

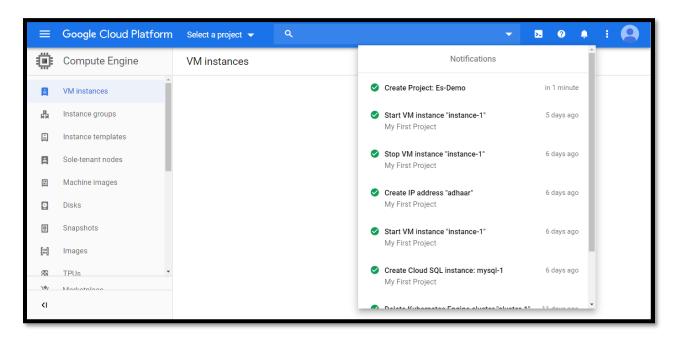
- 1) Used Google Speech APIs for Speech-to text conversion and Speaker Diarization (Distinguishing Speakers in an audio file)
- 2) GCP Platform for development purposes GCP Virtual Machines (Debian 10), Google Cloud Storage (for intermediate storage of data for asynchronous API requests) and Google Secret Manager for storing application access details.
- 3) Used Flask python based framework for APIs development.
- 4) RQ python library that uses Redis for queueing jobs and process them in background with workers
- 5) Postman to initiate requests and test REST API calls responses.
- 6) Created 2 simple audio files in .wav format. One each in English and Hindi languages
- 7) Packaged as Docker Image (work in progress)

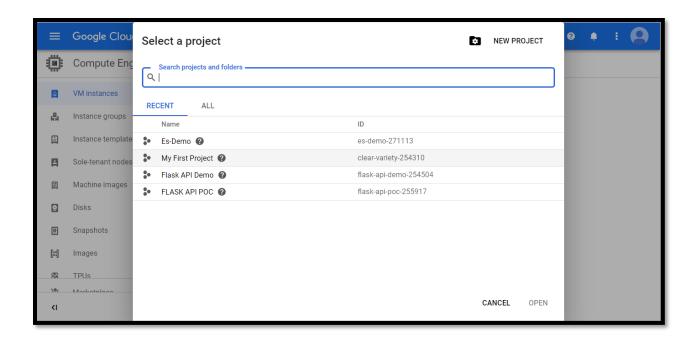
Instructions to run the code:

- 1) Install postman. This will be used to fire REST API requests.
- 2) Login to your google cloud account and create a new project.



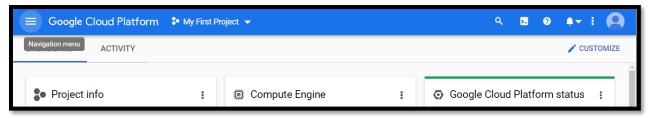
3) Once your project is created, select your project



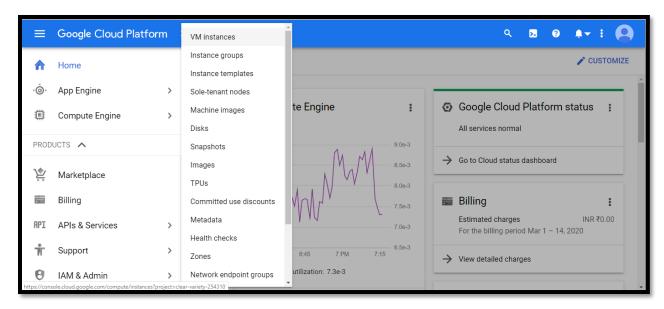


4) Now we require a VM, which can host our application. Here we are going to use compute engine for our application. To create VM follow below mentioned steps:

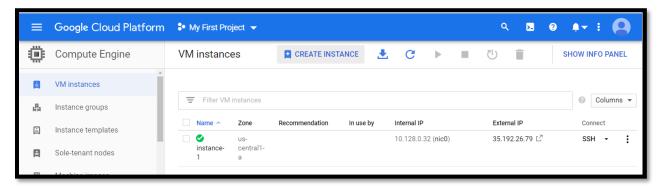
First go to Navigation menu



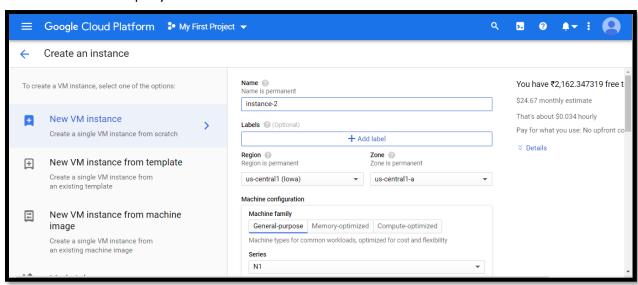
Hover over compute engine and select VM instances



Now select **CREATE INSTANCE**



Provide VM Name as per your need.

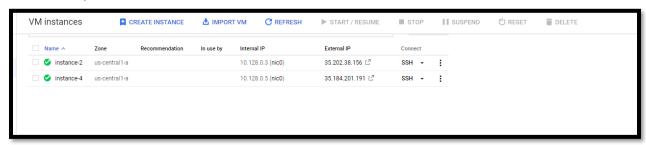


Please make sure to check below while creating VM instance in GCP.



Cloud API access scopes
Allow full access to all Cloud APIs

You can see your instance once created



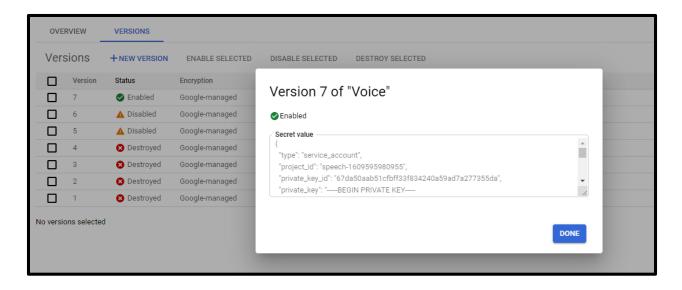
Click on SSH to open a terminal

```
Connected, host fingerprint: ssh-rsa 0 72:C0:8F:2B:DE:OD:D0:E2:9B:DE:C0:8A:79:72
:9E:34:01:1F:3D:54:00:2B:17:D7:9F:9F:B5:A9:7A:5A:F9:24
Linux instance-2 4.19.0-13-cloud-amd64 #1 SMP Debian 4.19.160-2 (2020-11-28) x86
_64

The programs included with the Debian GNU/Linux system are free software;
the exact distribution terms for each program are described in the
individual files in /usr/share/doc/*/copyright.

Debian GNU/Linux comes with ABSOLUTELY NO WARRANTY, to the extent
permitted by applicable law.
Last login: Sat Jan 9 02:35:04 2021 from 173.194.93.97
asheeshg018instance-2:~$ []
```

5) Please load Speech-67da50aab51c.json in GCP secret Manager. Please make sure to tailor the parameters like bucketname etc. based on your GCP storage bucket where you want to store your output files.



- 6) Install below libraries in VM
 - a. Flask
 - b. Flask-RESTful
 - c. pydub
 - d. wave
 - e. config
 - f. rq
 - g. google-cloud-language
 - h. google-cloud-secret-manager
 - i. google-cloud-storage
 - j. google-cloud-speech
 - k. google-cloud

7) Install Redis Server on VM

```
sudo apt-get update
sudo apt-get upgrade
sudo apt-get install redis-server
sudo service redis-server restart
```

Once, the Redis server is installed and running (default port 6379), we can use below command to test if Redis is running. The output will be PONG.

redis-cli ping

8) Pip Install python 3.7 and Redis in VM

9) Store files Project.py, config.py, Speech-67da50aab51c.json and worker.py in VM home directory. Create Directory under VM home by the name input. All the code has been uploaded in Github at below link

https://github.com/Asheeshg/Speech-to-text/upload

10) Run project app by firing the below command

Python3.7 Project.py

```
Asheeshg01@instance-2:~$ python3.7 Project.py
//home/asheeshg01/.local/lib/python3.7/site-packages/pydub/utils.py:170: RuntimeWarning: Couldn't find ffmpeg or avc
onv - defaulting to ffmpeg, but may not work
warn("Couldn't find ffmpeg or avconv - defaulting to ffmpeg, but may not work", RuntimeWarning)
* Serving Flask app "Project" (lazy loading)
* Environment: production
MASNING: This is a development server: Bo not use it in a production deployment.
Use a production WSGI server instead.
* Debug mode: on
* Running on http://o.o.o.0:8087/ (Press CTRL+C to quit)
* Restarting with stat
/ home/asheeshg01/.local/lib/python3.7/site-packages/pydub/utils.py:170: RuntimeWarning: Couldn't find ffmpeg or avc
onv - defaulting to ffmpeg, but may not work
warn("Couldn't find ffmpeg or avconv - defaulting to ffmpeg, but may not work", RuntimeWarning)
* Debugger is active!

1 * Debugger PIN: 968-419-224
```

11) Fire up below command in another VM terminal window. This will listen for a queue called default and established a connection to Redis server on a localhost:6379

```
asheeshg01@instance-2:~$ python3.7 worker.py
19:24:30 Worker rq:worker:19353ca9dd214f1e8d0f112cec2be9c4: started, version 1.7.0
19:24:30 Subscribing to channel rq:pubsub:19353ca9dd214f1e8d0f112cec2be9c4
19:24:30 *** Listening on default...
19:24:30 Cleaning registries for queue: default
```

12) Through Postman, fire a POST request at <a href="http://<external IP of VM>:8087/app/v1/file-upload">http://<external IP of VM>:8087/app/v1/file-upload. In body, please enter file as KEY and select file in KEY cell dropdown. In VALUE cell, click on 'Select Files' button and select the input audio file. Please note that audio file should be in .wav format and should be either English or Hindi.

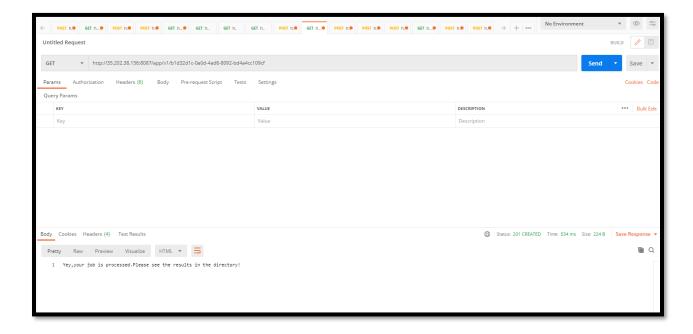


13) Click on Send button in Postman and should be able to see output like below

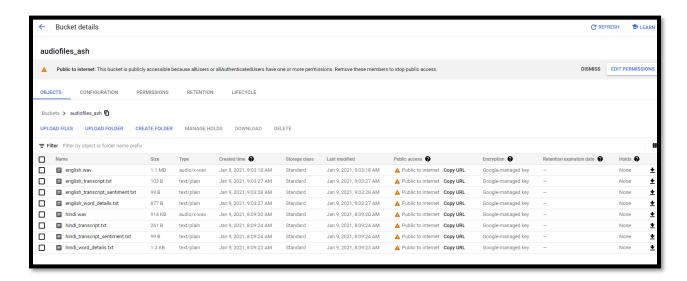


As can be seen above, Job id is returned after firing the asynchronous API. Please note down this Job id as this will be used in subsequent steps to check whether Job is completed or not.

14) Please fire the below GET request in postman to retrieve the status of the job <a href="http://<external IP of VM>:8087/app/v1/<job-id>"http://<external IP of VM>:8087/app/v1/<job-id>"http://external IP of VM>:8087/app/v1/

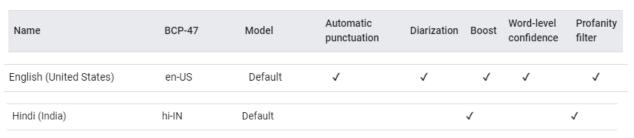


If message received is "Yey, your job is processed. Please see the results in the directory!" then this means that job is completed and all the outputs can be checked in Google Global storage in audiofiles_ash bucket. Please note that bucket name can be changed by modifying the same in json stored in Google Secret Manager.

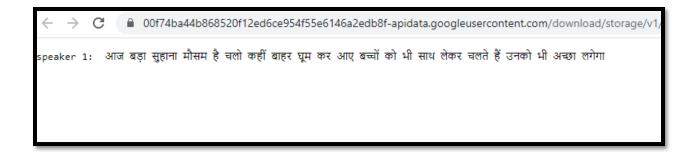


As can be seen above the input audio file hindi.wav has three output files:

a) <audio_filename>_transcript.txt : This file contains the converted audio to txt transcript. Please note that speakers will be distinguished for English language as Diarization is supported by Google Speech-to-Text API, however for Hindi language Diarization is not supported by Google APIs.



User should be able to see the transcript after downloading the file. Note- Please make sure that UTF-8 is enabled on the browser to see Devanagari fonts.



```
← → C © 00f74ba44baa735ce857716ee94d2d5a5a4b39b171-apidata.go

speaker 1:
speaker 2: hello how are you I am fine thank you
speaker 1: what is the format of the spine
```

b) <audio_filename>_word_details.txt : This file contains the word details like who said which word at what start time and end time. User should be able to see output like below.



Nord hells: start_time: 1.1: end_time: 1.6: speaker 1 Nord: how: start_time: 1.0: end_time: 1.0: speaker 1 Nord: how: start_time: 2.0: speaker 1 Nord: how: start_time: 2.0: end_time: 0.1: speaker 1 Nord: how: start_time: 0.0: end_time: 0.0: 0

c) <audio_filename>_transcript_sentiment.txt : This file will provide the sentiment score and magnitude of the overall transcript. The score of a document's sentiment indicates the overall emotion of a document. The magnitude of a document's sentiment indicates how much emotional content is present within the document, and this value is often proportional to the length of the document. Please see details on how to interpret sentiment score and magnitude at below link.

https://cloud.google.com/naturallanguage/docs/basics#interpreting_sentiment_analysis_values



Will load docker image and instructions in Glthub later.