Natural Language Processing

1.

1.1 Project Title: Text Summarization from Real Time Speech Recognition

1.2 Team Members:

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1.3 GitHub Link:

https://github.com/PavaniMangugari/NLPProject-9

2. Goals and Objectives:

2.1 Motivation:

Information retrieval is one of the most widely needed use cases these days. The process of extracting the summary of a video lecture while remembering it and manually writing a summary is a time taking and intensive task for humans. There are plenty of text and video material available on the Internet. The process of developing a model for the machine to automatically convert the speech in the video to text would help the students and lectures.

Speech recognition would mean converting the speech to text in the real time scenario i.e., exactly converting the words with means of pronunciation. Text summarization is defined as the process of finding the meanings of difficult words and gets summarized to the shorter version where we can find them easily. So, we plan to develop a model that would take the lecture and interpret and summarize it. Additionally, we are using SR with text summarization to get Question and Answers that would help the students review their knowledge regarding the lecture and automatically check the correctness. We also use Machine Translation to translate the document into another native language.

2.2 Significance:

In this age where remote education and online videos are becoming hugely popular, the need for a model that can convert speech to text is necessary. The model would not only help a lot of users, having problems with understanding the language, it would also help the users with disabilities too. The whole text and lecture are hard to memorize, so having a summary that would brief about the lecture would help the users while revising or just to know the undergoing of the lecture.

With the implementation of additional functionality, it will allow the users to view a few questions based on the lecture so that they can test their understanding in the lecture. This project aims to ease online learning and help users who are struggling whilst listening to these classes and get concise overview of the lecture in their native language.

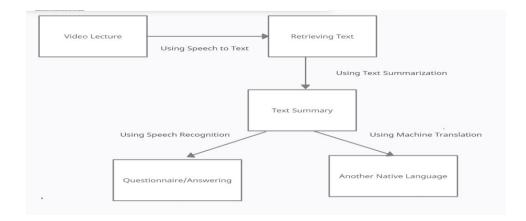
2.3 Objectives:

We are going to implement this project in 3 steps.

- ➤ Speech to Text
- ➤ Text Summarization
- ➤ Q&A module.



Fig 1



We are planning to do the project using the speech to text, text summarization and other training models. We will use the NLTK, SPACY, Genism, and Hugging Face tools to accomplish our project. We use machine translation for translating the text to another language.

We are at the end of the project planning to achieve an interface that would take the video input that converts it to text summary. In turn it uses this summary to build a Q/A model and Machine Translator for translating it to another language (Spanish and German). The goal of our project is to develop a model that will be accurate and precise in the summarization and processing that would help the users.

2.4 Features:

The video speech is converted into text using the python modules that would allow for the speech to turn into the text. We retrieve the text, perform the text cleaning and processing. We then tokenize the words, perform summarization techniques to get a video summary.

Using the machine translation, we convert the text into German and Spanish. We get the questions and answers from the original text. The user would be allowed to click for a correct answer. The answer would then be evaluated by the machine and returns if the user is right or wrong.

We will be implementing the feature that the user will be selecting an appropriate answer so that the system can check if the answer is correct or not.

3 Related Work (Background)

Paper-1: They have considered that the system should consider the word by word from multiple speakers at a time so that it can identify the number of speakers. They have used some profiling techniques to reduce the signal-noise ratio by means of normalization and splitting the words so that they can get the ratio of 1:1. They got the accuracy of 71.7%.

Paper-2: In this they have taken the isolated word recognition and continuous phoneme recognition in a cued speech. They have concatenated the lip movement and hand shape for feature fusion and HMM based recognition has been implemented. They got 94% and 89% as accuracy for normal hearing and deaf people hearing.

Paper-3: In this paper they have used Devanagari script for classifying their classes based on phonemes. It converts the speech to phonemes by means of simple operations like zero-crossing and FTT. They did not get satisfactory results as the vowels and other consonants overlap in the same group. They got the accuracy of 75%, which is not satisfactory.

Paper-4: In this they have converted the normal speech to whispered speech by means of signal processing and voice conversion techniques. They have used gaussian mixture model (GMM) and deep neural networks (DNN) to map the features. They have also evaluated the naturalness and

the speaker similarity after converting them to whispered speech. However, the model has failed in detecting the gender based on the speaker.

Paper-5: They have taken the double handed Indian sign language and converted them into patterns and converts speech to speech less. For this they have used the clustering techniques which may be useful for classifying the text and minimum eigen value algorithm for conversion. They have used bare hands for sign language rather than gloves. They have converted sign language into both text and speech.

Paper-6: they have taken both speech and object detection for speech recognition. They have taken the cognitive model for object detection based on speech to text recognition. In This they have developed a speech command and given it as input and by that the object is detected. The object is detected within a square box. For speech they have used google speech recognition and for object detection they have used YOLO V3 algorithm for that.

Paper-7: Here they are converting text to speech by mapping the source features. They have proposed a novel architecture used in Deep Learning. By considering the linear constraints, it prevents the input from converting features into unrealistic features. It is done by network-based conversion in a time variant way. It is suitable for multiple frames features.

4. Dataset

In the process of developing the application, we send audio data in the wav format. The application can handle data of a long duration. We tested the dataset of a university lecture about food processing. It is one hour long and has a lecture where there is one speaker.

Using the wave, we find the features of our dataset. We can see that we have 2 channels, with a sample width of 2. The frame rate is 44100 and the number of frames is 175584384.

```
import wave
obj = wave.open('food.wav','r')
print( "Number of channels",obj.getnchannels())
print ( "Sample width",obj.getsampwidth())
print ( "Frame rate.",obj.getframerate())
print ("Number of frames",obj.getframes())
print ( "parameters:",obj.getparams())
obj.close()

| Number of channels 2
Sample width 2
Frame rate. 44100
Number of frames 175584384
Number of frames 175584384
```

Fig.3

5. Detail design of Features

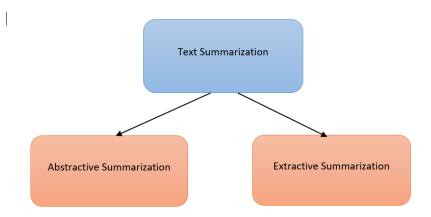


Fig.4

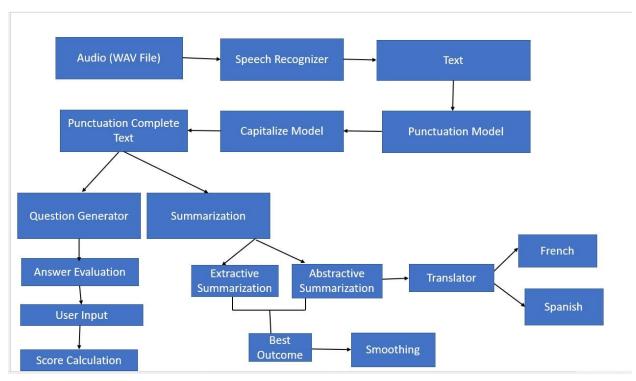


Fig.5 Architecture Diagram.

Our application first takes an audio input as a wav file The audio file is first processed using the wave libraries, the matplotlib and get graphs regarding the audio frequency vs time. Later, we take the text and pass it through a speech recognizer where the speech is converted into the text The processed text is passed through a punctuationModel() where the information that we get from the recognizer. The information that we get from the punctuation model is then sent and we capitalize words using capitalize () method. Now, we get the text from the reformed model and perform text summarization using two different methods We use the bigram model on the summarized text and apply Laplace and interpolation smoothening techniques and find accuracy. Using the original text file, we send the text into an t5 based question-answer based model The model generates question and answer inputs based on a given value of questions. The model also takes an evaluation factor that takes the scores and gives the highest rank answer that is correct by itself. The user is displayed the questions and asked to enter the correct option The system checks if the answer is correct or not and then calculates the score and finally displays the final output score.

The extractive text is taken which we generate from the model and translate it into German and Spanish language.

Workflow diagram with explanation

6. Analysis

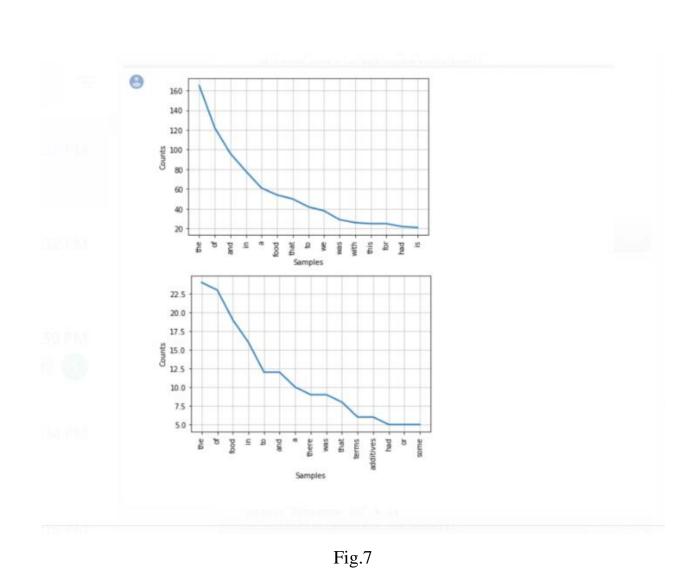
In the analysis phase, we try to take the audio input that we have taken and try to analyze it. Later, we also try to analyze the differences in the word count in the original vs the summarized text in the summarizers that we have used.

```
[41] import wave
obj = wave.open('food.wav','r')
print( "Number of channels",obj.getnchannels())
print ( "Sample width",obj.getsampwidth())
print ( "Frame rate.",obj.getfnamerate())
print ("Number of frames",obj.getfnames())
print ( "parameters:",obj.getparams())
obj.close()

Number of channels 2
Sample width 2
Frame rate. 44100
Number of frames 175584384
parameters: _wave_params(nchannels=2, sampwidth=2, framerate=44100, nframes=175584384, comptype='NONE', compname='not compressed')
```

Fig.6

We first analyze the audio dataset and can see the features of the dataset.



Difference between the counts in the abstractive summarizer vs the original text

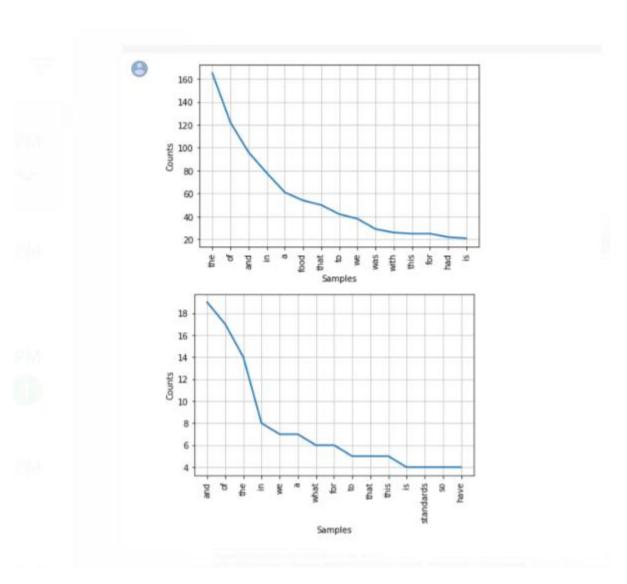


Fig.8

Difference between the counts in the extractive summarizer vs the original text.

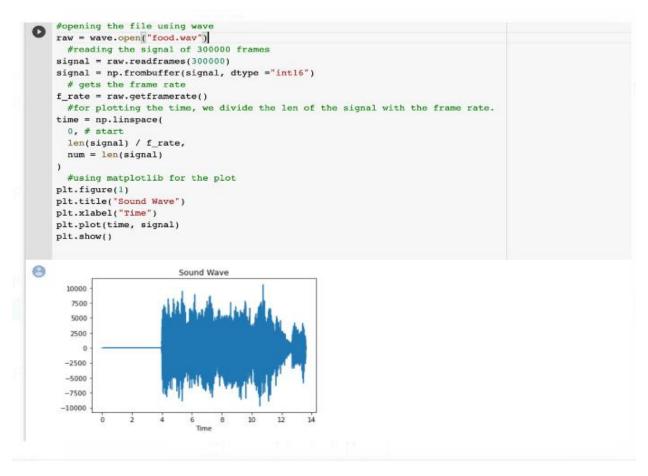


Fig. 9

Drawing a graph between the time and signal of the audio source.

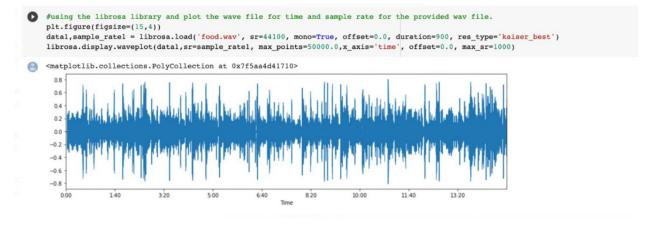


Fig. 10

A graph between the time duration and the energy involved which is the sampling rate and the signal.

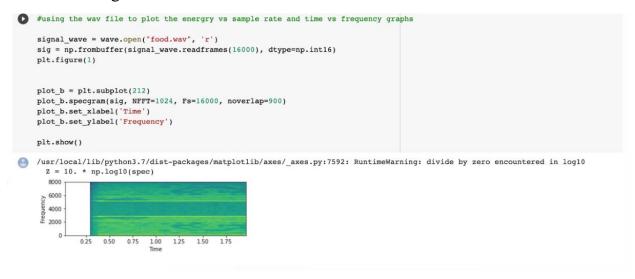


Fig. 11

The analysis between the time and the frequency of the audio wav file.

7. Implementation

Fig. 12

Using the speech recognizer module for reading the audio file.

- Speech to Text

```
#intializing text to empty first
finalText=
  he intial offset is 0
offset=0
#i in the range of for the duration offset, we repeat the conversion for i in range(10):
  #using the re
                   niser to find the text from the audio
  r = sr.Recognizer()
  audio = sr.AudioFile('food.wav')
  with audio as source:
    audio = r.record(source, duration=300,offset=offset)
  # set offset as 300
  offset=offset+300
  text=r.recognize google(audio)
  finalText=finalText+text ## recognizing the speech and converting into text
print(finalText)
hello my name is food toxicology would like to welcome back once again today the subject of the lecture is going to be the history of us food regulation and
```

Fig. 13

Using the speech recognition model, we get converted text



Fig. 14

Using the word punctuation model, we get the converted text with punctuations



Fig. 15

Using the capitalizing text, we capitalize each sentence.

Fig. 16

We summarize text using the abstractive summarization

Text Summarization(Extractive)

Fig. 17

Using extractive summarization, we get the summarized text.

- Bigram model and smoothening

```
from collections import defaultdict
from collections import Counter
from numpy.random import choice
from tqdm import tqdm
class Bigram():
    def __init__(self):
        self.bigram_counts = defaultdict(Counter)
        self.unigram_counts = Counter()
        self.context = defaultdict(Counter)
        self.start count = 0
        self.token_count = 0
        self.vocab_count = 0
    def convert_sentence(self, sentence):
        return ["<s>"] + [w.lower() for w in sentence] + ["</s>"]
    def get_counts(self, sentences):
         # collect unigram counts
         for sentence in sentences:
            sentence = self.convert sentence(sentence)
            for word in sentence[1:]: # from 1, because we don't need the <s> token
                 self.unigram_counts[word] += 1
            self.start_count += 1
```

Fig. 18

```
# collect bigram counts
    for sentence in sentences:
        sentence = self.convert_sentence(sentence)
       bigram_list = zip(sentence[:-1], sentence[1:])
        for bigram in bigram list:
            self.bigram_counts[bigram[0]][bigram[1]] += 1
            self.context[bigram[1]][bigram[0]] += 1
    self.token_count = sum(self.unigram_counts.values())
    self.vocab_count = len(self.unigram_counts.keys())
def generate_sentence(self):
   current_word = "<s>"
   sentence = [current_word]
   while current_word != "</s>":
       prev_word = current_word
       prev_word_counts = self.bigram_counts[prev_word]
        # obtain bigram probability distribution given the previous word
       bigram_probs = []
        total_counts = float(sum(prev_word_counts.values()))
        for word in prev_word_counts:
            bigram_probs.append(prev_word_counts[word] / total_counts)
        # sample the next word
        current word = choice(list(prev word counts.keys()), p=bigram probs)
        sentence.append(current word)
    sentence = " ".join(sentence[1:-1])
    return sentence
```

Fig. 19

```
import nltk
from nltk.tokenize import sent_tokenize
##from nltk.corpus import brown
nltk.download('punkt')
s=sent_tokenize(result)
l=[]
for i in s:
    l1=list(i.split(' '))
    l.append(l1)

print(l)
bigram = Bigram()
bigram.get_counts(l)
for i in range(len(l)):
    print("Sentence %d" % i)
    print(bigram.generate_sentence())
```

Fig. 20

We used the bigram model to generate the sentence from the summarized text.

Fig. 21

Installing the required libraries.

```
| Import en_core_web_sm
| import jacm
| import jacm
| import process
| import re-
| import process
| import re-
| import process
| import re-
| Autonomic process
| import re-
| Autonomic process| import (
| import any import any int, import any interests)
| import any import any int, import any interests'
| import any import any int, import any important any import any important any import any imp
```

Fig.22

Fig.23

The question evaluator file which takes the score and tells the correct answer for a question.

Translating the summarized text



Fig.24

Taking the summarized text and installing the required libraries.

```
from deep_translator import GoogleTranslator translated = GoogleTranslator(source='english', target='spanish').translate_file('summarized_text.txt')
print(translated)

hola, mi nombre es toxicología alimentaria me gustaría darle la bienvenida una vez más. una de las respuestas, una de las respuestas es establecer estándares

from deep_translator import GoogleTranslator
translated = GoogleTranslator(source='english', target='german').translate_file('summarized_text.txt')
print(translated)

hallo, mein name ist lebensmitteltoxikologie möchte mich wieder einmal herzlich willkommen heißen. Eine der Antworten, eine der Antworten ist, Verhaltensstan
```

Fig.25

The summarized text in Spanish and English languages.



Fig.26

While training the translator model, we use 10 epochs and get the final predictions.

8. Results

```
#intializing text to empty first
finalText=""
#the intial offset is 0
offset=0
#i in the range of for the duration offset, we repeat the conversion
for i in range(10):
    #using the recognizer()
    audio = sr.Audiofile('food.wav')
    with audio as source:
        audio = r.record(source, duration=300,offset=offset)
    # set offset as 300
    offset=offset+300
    text=-r.recognize_google(audio)
    finalText=finalText+text ## recognizing the speech and converting into text
    print(finalText)

hello my name is food toxicology would like to welcome back once again today the subject of the lecture is going to be the history of us food regulation and
```

Fig. 27

First, we get the text from converting the speech which is way file.

```
[ ] print(result)

hello, my name is food toxicology would like to welcome back once again. today. the subject of the lecture is going to be the history of us food regulation,
```

Fig. 28

Second, we add punctuation to the text.

```
[ ] capitalizedres=capitalizng_sentence(result)
    print(capitalizedres)

Hello, my name is food toxicology would like to welcome back once again. Today. The subject of the lecture is going to be the history of us food regulation,
None
```

Fig. 29

Then, we capitalize each sentence after adding punctuation.

```
def interpolation(text2, bigram, lambdas):
       bigram_lambda = lambdas[0]
unigram_lambda = lambdas[1]
        zerogram_lambda = 1 - lambdas[0] - lambdas[1]
        sentence = bigram.convert_sentence(text2)
       bigram_list = zip(sentence[:-1], sentence[1:])
       prob = 0
        for prev_word, word in bigram_list:
            # bigram probability
            sm_bigram_counts = bigram.bigram_counts[prev_word][word]
           if sm_bigram_counts == 0: interp_bigram_counts = 0
           else:
               if prev_word == "<s>": u_counts = bigram.start_count
                else: u_counts = bigram.unigram_counts[prev_word]
                interp_bigram_counts = sm_bigram_counts / float(u_counts) * bigram_lambda
            # unigram probability
           interp_unigram_counts = (bigram.unigram_counts[word] / bigram.token_count) * unigram_lambda
           # "zerogram" probability: this is to account for out-of-vocabulary words, this is just 1 / |V| vocab_size = len(bigram.unigram_counts)
           interp_zerogram_counts = (1 / float(vocab_size)) * zerogram_lambda
           prob += math.log(interp_bigram_counts + interp_unigram_counts + interp_zerogram_counts)
        return prob
   bigram_interpolation = Bigram()
   bigram_interpolation.get_counts(training_set)
   plex_interpolation = calculate_perplexity(test_set, bigram_interpolation, interpolation, (0.8, 0.19))
   print(plex_interpolation)
```

Fig. 30

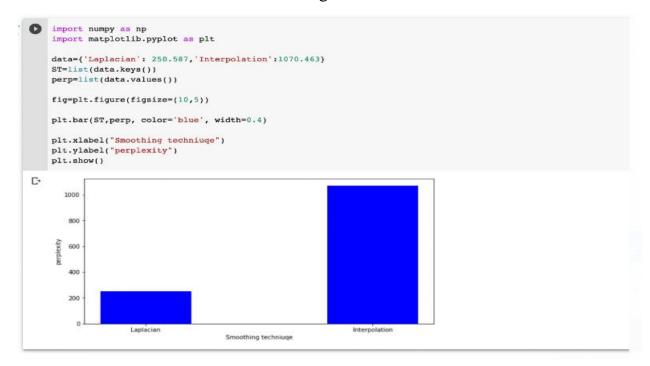


Fig. 31

Using the two models i.e., Laplacian and interpolation, we got the perplexities as shown below.

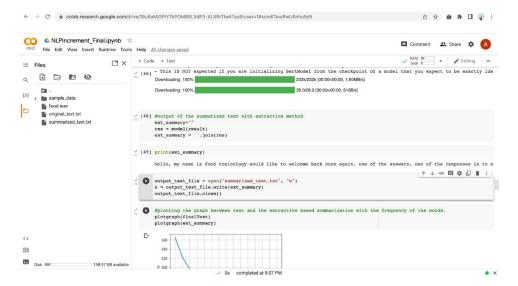


Fig. 32

The two files of the original and summarized text are downloaded.



Fig.33

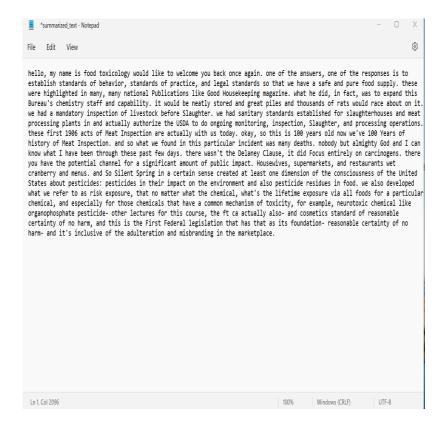


Fig. 34

The two files of the original and summarized text are shown.

```
| Consumer control position of the control position of
```

Fig.35

The Command line code where the user is given some questions from the summary, and he can choose the correct answer. From this, we can see that the user can select options and based on a evaluator it will automatically generate if the answer is right or not and finally, the user is given a score.

```
import matplotlib.pyplot as plt
epochValues = [i+1 for i in range(4)]
epoch_accuracies = [0.5431,0.5904,0.6206,0.6371]
epoch_loses = [1.3149,1.1628,1.0816,1.0450]
plt.plot(epochValues, epoch_accuracies)
plt.title('Accuracy in epochs')
plt.xlabel('epochs')
plt.ylabel('accuracy')
plt.show()
```

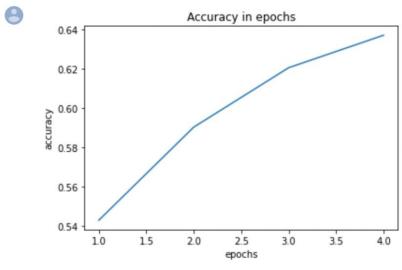


Fig.36

When we are translating the text from English to Spanish we plot the accuracy vs epochs graph.

```
import matplotlib.pyplot as plt
epochValues = [i+1 for i in range(4)]
epoch_accuracies = [0.5431,0.5904,0.6206,0.6371]
epoch_loses = [1.3149,1.1628,1.0816,1.0450]
plt.plot(epochValues, epoch_loses)
plt.title('losses in epochs')
plt.xlabel('epochs')
plt.ylabel('losses')
plt.show()
```

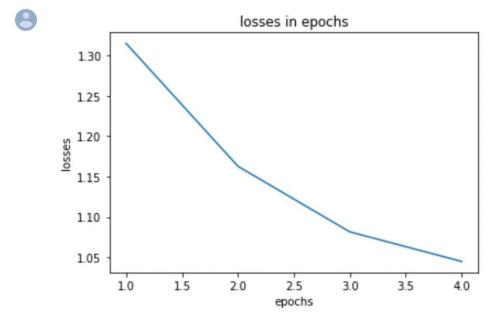


Fig.37

When we are translating the text from English to Spanish, we plot the losses vs epochs graph. We can see the accuracies of the model and the language losses and how they decrease with time. The accuracy increases after each epoch.

9. Project Management

Getting the best outcomes as well as directing the project with all the available resources is a crucial task. We have incorporated users' and participants' perspectives into this application's data gathering and analysis.

The objective of the project is to deliver a finished product that meets the requirements of the outcome. The project's objective is frequently to modify that to be more effectively achieve the project's goals.

9.1 Implementation Status Report

9.1.1 Work Completed:

We have worked on the speech to text, Q/A module, text translation and the smoothening techniques.

• Responsibility and contributions:

S.no	Person	Task description	Contributions
1.	Achala Samudrala	Converted Speech-text recognizer. Q&A module	24%
2.	Pavani Mangugari	Applied Smoothing techniques. Q&A module	26%
3.	Tharun Puri	Added punctuation to the recognized text. Text translation	26%
4.	Venkata Sai Preetham B	Summarized text from recognized text. Text translation	24%

• Issues/Concerns:

Difficulty in overcoming the errors faced by the summarizer and speech to text recognition and Q/A module. We went through lot of online sources and worked on resolving errors.

It became hard for us to communicate as we have different class schedules. However, we have managed the schedules so that we can communicate among ourselves.

10. References/Bibliography:

https://ieeexplore.ieee.org/document/9622156

https://paperswithcode.com/paper/predicting-subjective-features-from-questions

https://paperswithcode.com/paper/iit-uhh-at-semeval-2017-task-3-exploring

https://paperswithcode.com/paper/qdee-question-difficulty-and-expertise

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https://ieeexplore.ieee.org/document/8465680

https://ieeexplore.ieee.org/document/5898904

https://ieeexplore.ieee.org/document/9315985

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