Enhancing Automatic Speech Recognition with Contextual Understanding using Natural Language Processing

An
Industry Oriented Mini Project Report on

Submitted In Partial Fulfillment of the Requirements for the Award of

Bachelor Of Technology in Computer Science and Engineering

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NALLA NARASIMHA REDDY
Education Society's Group of Institutions - Integrated Campus
(UGC AUTONOMOUS INSTITUTION)

School of Engineering Department of Computer Science and Engineering

CERTIFICATE

This is to certify that the project report titled **"ENHANCING AUTOMATIC SPEECH RECOGNITION WITH CONTEXTUAL UNDERSTANDING USING NATURAL LANGUAGE PROCESSING"** is being submitted by **K. Sandhya (217Z1A0586) and P.Nithik Roshan (217Z1A5E2),K.Bala Manikanta (217Z1A587),** in Partial fulfilment for the award of **Bachelor of technology in Computer Science & Engineering** is a record bonfire work carried out by them. The results embodied in this report have not been submitted to any other University for the award of any degree.

Internal Guide **Head of the Department**

(Mrs. Priyanka Pandarinath) (Dr. K. Rameshwaraiah)

Submitted for Viva voce Examination held on……………………….

External Examiner

DECLARATION

We K.Sandhya, P.Nithik Roshan and K. Bala Manikanta, the students of **Bachelor of Technology in Computer Science and Engineering, Nalla Narasimha Reddy Education Bachelor Society's Group Of Institutions,** Hyderabad, Telangana, hereby declare that the work presented in this project work entitled **Enhancing Automatic Speech Recognition with contextual understanding using Natural LanguageProcessing** is the outcome of our own bonafide work and is correct to the best of our knowledge and this work has been undertaken taking care of engineering ethics. It contains no material previously published or written by another person nor materialwhich has been accepted for the award of any other degree or diploma of the university or other institute of higher learning.

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ABSTRACT

Automatic speech recognition (ASR) has advanced from responding to limited sound to fluently understanding natural language. Used in voice search, virtual assistants, and speech-to-text systems to enhance user experience and productivity. Began with basic sound recognition and evolved into comprehensive language comprehension. Despite significant advancements in automatic speech recognition (ASR) technology, existing systems often struggle to accurately transcribe spoken language in context where semantic nuances and contextual cues play a crucial role. The problem arises from the inherent limitations of conventional ASR approaches to comprehensively understand and intercept the contextual information efficiently, resulting in inaccuracies misinterpretations and errors in transcriptions, especially in scenarios involving ambiguous or context dependent speech. Incorporating Natural Language Processing (NLP) techniques into ASR systems presents a promising avenue to address this challenge.

Keywords: *Audio Signal Processing, Feature Extraction, Acoustic Modeling, Language Modeling,Decoding, Post-Processing.*

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CHAPTER ONE
INTRODUCTION

- *Motivation*

Automatic speech recognition (ASR) has advanced from responding to limited sounds to fluently understanding natural language. It has sparked significant interest due to the drive to automate human-machine interaction. Used in voice search, virtual assistants and speech-to-text systems to enhance user experience andproductivity.

- *Problem Statement*

The problem in automatic speech recognition (ASR) lies in accurately converting spoken language into text, especially in challenging environments with background noise, varying accents, speaker variability and ambiguous pronunciations. Achieving high accuracy while minimizing errors in transcription remains a significant challenge, particularly in real-time, divers audio settings.

- *Purpose*

The purpose of Enhancing Automatic Speech Recognition (ASR) with contextual understanding using Natural Language Processing (NLP) to improve theaccuracy of transcriptions by incorporating context. This allows ASR systems to better interpret ambiguous phrases, recognize domain-specific language and adaptto various speaking styles, ultimately leading ton more reliable outputs.

- *Scope*

Enhancing automatic speech recognition by integrating contextual understanding through natural language processing for improved accuracy, relevance, and use intent comprehension.

- *Project Objective*

The main objective of this project for Automatic speech Recognition (ASR) by integrating natural language processing (NLP) for better contextual understanding.

The main objective of this project for farmers is to adapt according to the technology. This enhancement aims to reduce errors by leveraging context to disambiguate similar-sounding words, improve accuracy in noisy environments, and enable ASR systems to provide more human-like.

- *Limitations*

The limitations of enhancing Automatic Speech Recognition (ASR) with contextual understanding using Natural Language Processing (NLP) include processing complexity, increased computational cost, potential errors in contextual interpretation, difficulty handling ambiguous or novel phrases and challenges in adopting to diverse accents, dialects. Additionally, ASR systems may have difficulty understanding slang, idioms, or rare phrases. Privacy concerns arise as contextual models often need access to personal data to provide relevant context. Moreover, maintaining high performance across multiple languages and domains remains a challenge due to the complexity of linguistic nuances.

CHAPTER TWO
LITERATURE SURVEY

➢ *Introduction*

Natural Language Processing (NLP) has become a critical component in various applications, ranging from text auto-completion to conversational AI. However, traditional NLP models often struggle with maintaining contextual relevance and semantic coherence, especially in longer text sequences. Reinforcement Learning (RL), which enables models to learn from interactions and improve over time, offers a promisingsolution to these challenges. By integrating RL with NLP, it is possible to develop systems that not only predict the next words in a sequence but also continuously refine theirpredictions to be more contextually accurate and semantically meaningful. This survey explores the current landscape of NLP and RL integration highlighting the potential forcreating a model that predicts the next three words in a text sequence with enhancedaccuracy.

Automatic Speech Recognition (ASR) has evolved significantly from its early days in the 1960s where systems relied on acoustic-phonetic methods to decode speech. Initial approaches used hand-crafted features and rule-based models, but advancements in statical modeling in the 1980s introduced hidden Markov Models (HMMs).

The 2000s saw the rise of machine learning techniques, particularly the use of neural network, which enhanced ASR performance by learning complex patterns in speech data. Deep learning further revolutionized the field in the 2010s, and long short-term memory networks (LSTMs), which significantly improved recognition accuracy.

Recent advancements include the application of tranformers and self-supervised learning, which leverage vast amounts of unlabelled data to improve ASR systems robustness and adaptability. Technologies like end-to-end ASR models simplify the architecture by combining acoustic and learning modeling into a single framework.

Overall,ASR research has transitioned from rule-based systems to sophisticated neural architectures, leading to more accurate and versatile speech recognitionapplications.

➢ *Existing System*

Despite significant advancements in Automatic Speech Recognition (ASR)technology, existing systems often struggle to accurately transcribe spoken languagein contexts where semantic nuances and contextual cues play a crucial role. Traditional ASR systems primarily rely on acoustic features and language models based on statistical patterns, which may lead to suboptimal performance whenconfronted with complex linguistic structures and varied contextual information.

The problem arises from the inherent limitations of conventional ASR approaches to comprehensively understand and interpret the contextual nuances present in spoken language. Current systems lack the ability to leverage contextual information effectively.

Resulting in inaccuracies, misinterpretations, and errors in transcriptions, especially in scenarios involving ambiguous or context-dependent speech.

➢ *Proposed System*

Traditional Automatic Speech Recognition (ASR) systems, while having revolutionized human-computer interaction, have notable disadvantages that limit their effectiveness. These systems often rely heavily on acoustic models and predefined language models that do not adapt well to diverse accents, dialects, and noisy environments, leading to higher error rates. They typically struggle with understanding context, making them less accurate in recognizing homophones and context-dependent words, which results in incorrect transcriptions. Furthermore, traditional ASR systems lack the capability to learn and improve from user interactions over time, making them static and less responsive to individual user needs and evolving language patterns. Their dependence on large, annotated datasets for training also makes them less scalable and adaptable to new languages or specialized vocabularies without extensive. manual effort,Overall, these limitations hinder their accuracy, adaptability, anduser- friendliness, especially in dynamic, real-world applications.

End-to-end (E2E) automatic speech recognition (ASR) addresses traditional ASR problems by integrating the entire recognition process into a single model, typically based on deep learning architectures like recurrent neural networks (RNNs) or transformers. Unlike traditional ASR, which involves separate modules for feature extraction, acoustic modeling, pronunciation modeling, and language modeling, E2E ASR learns to directly map input audio features to output text sequences. This approach simplifies the architecture, reduces error propagationbetween modules, and often improves overall accuracy by allowing the model to optimize all aspects of speech recognition simultaneously. E2E ASR systems have shown promising results in various domains, benefiting from end-to-end optimization and the ability to capture complex dependencies in speech.

CHAPTER THREE
SYSTEM ANALYSIS

➢ *Functional Requirements*

- *Contextual Correction:*
  The system shall automatically correct misrecognized words based on the context ofthe conversation.

- *User-Specific Adaptation:*
  The system shall learn and adapt to individual users' speech patterns and vocabularyover time.

- *Real-Time Updates:*
  The system shall provide real-time transcription updates as the user speaks.

- *Multi-Language Support:*
  The system shall support recognition for multiple languages and dialects.

- *Interactive Feedback:*
  The system shall provide an interface for users to give feedback onrecognition errors.

- *Noise Handling:*
  The system shall adapt to different noise environments to maintain recognitionaccuracy.

- *Summarization:*
  The system shall provide automatic summarization of long speeches orconversations**.**

➢ *Non-Functional Requirements*

- *Performance*
  Enhancing automatic speech recognition with contextual understanding through NLPimproves accuracy by interpreting context and nuances in spoken language.

- *Security*
  The system shall ensure the confidentiality and integrity of user data, adhering to relevant data protection regulations. User authentication and authorization mechanisms shall be implemented to prevent unauthorized access.

- *Usability*
  Enhance automatic speech recognition with contextual understanding usingnatural language processing improves accuracy by interpreting speech withincontext, making interactions more intuitive and efficient in diverse applications.

- *Reliability*
  Enhancing automatic speech recognition with contextual understanding using NLPboosts accuracy, but requires extensive training data and fine-tuning.

- *Scalability*
  Enhancing ASR with NLP boosts scalability by adapting to diversecontexts, improving accuracy, and handling varying speech patterns efficiently.

- *Maintainability*
  To enhance automatic speech recognition with contextual understanding using natural language processing, maintainability involves regular updates to algorithms,integration of diverse data sources, and adaptive learning to handle evolving language patterns.

➢ *Interface Requirements*

- *User Requirements*
  Python Gradio transformerWishper AI

  Pipe line Hugging face

- *System Requirements:*

✓ *Hardware Requirements:*

System	: MINIMUM i3.
Hard Disk	: 40 GB.
Ram	: 4 GB.

- *Software Requirements:*

Operating System	:Windows 8	Coding Language	: Python 3.7

- *Performance Requirements*

It works fine with moderate internet speed. The connection is secured and user details are stored in a secured manner. The switch from one screen to another is quick and smooth. The inputs from users are taken correctly and response is recorded quickly.

CHAPTER FOUR
SYSTEM DESIGN

➢ *Uml Diagrams*

UML stands for Unified Modeling Language. UML is a standardized general- purpose modeling language in the field of object-oriented software engineering. Thestandard is managed, and was created by, the Object Management Group. The goal is for UML to become a common language for creating models of object oriented computer software. In its current form UML is comprised of two major components: aMeta-model and a notation. In the future, some form of method or process may also be added to; or associated with, UML.

The Unified Modeling Language is a standard language for specifying, Visualization, Constructing and documenting the artifacts of software system, as wellas for business modeling and other non-software systems. The UML represents a collection of best engineering practices that have proven successful in the modeling of large and complex systems.

The UML is a very important part of developing objects oriented software and the software development process. The UML uses mostly graphical notations to express the design of software projects.

- *Goals*:

The Primary goals in the design of the UML are as follows:

✓ Provide users a ready-to-use, expressive visual modeling Language so that theycan develop and exchange meaningful models.
✓ Provide extendibility and specialization mechanisms to extend the core concepts. Be independent of particular programming languages and development process. Provide a formal basis for understanding the modelinglanguage.
✓ Encourage the growth of OO tools market.
✓ Support higher level development concepts such as collaborations,frameworks, patterns and components.

➢ *Use Case Diagram*

A use case diagram in the Unified Modeling Language (UML) is a type of behavioral diagram defined by and created from a Use-case analysis. Its purpose is to present a graphical overview of the functionality provided by a system in terms of actors, their goals (represented as use cases), and any dependencies between those use cases. The main purpose of a use case diagram is to show what system functions are performed for which actor. Roles of the actors in the system can be depicted.

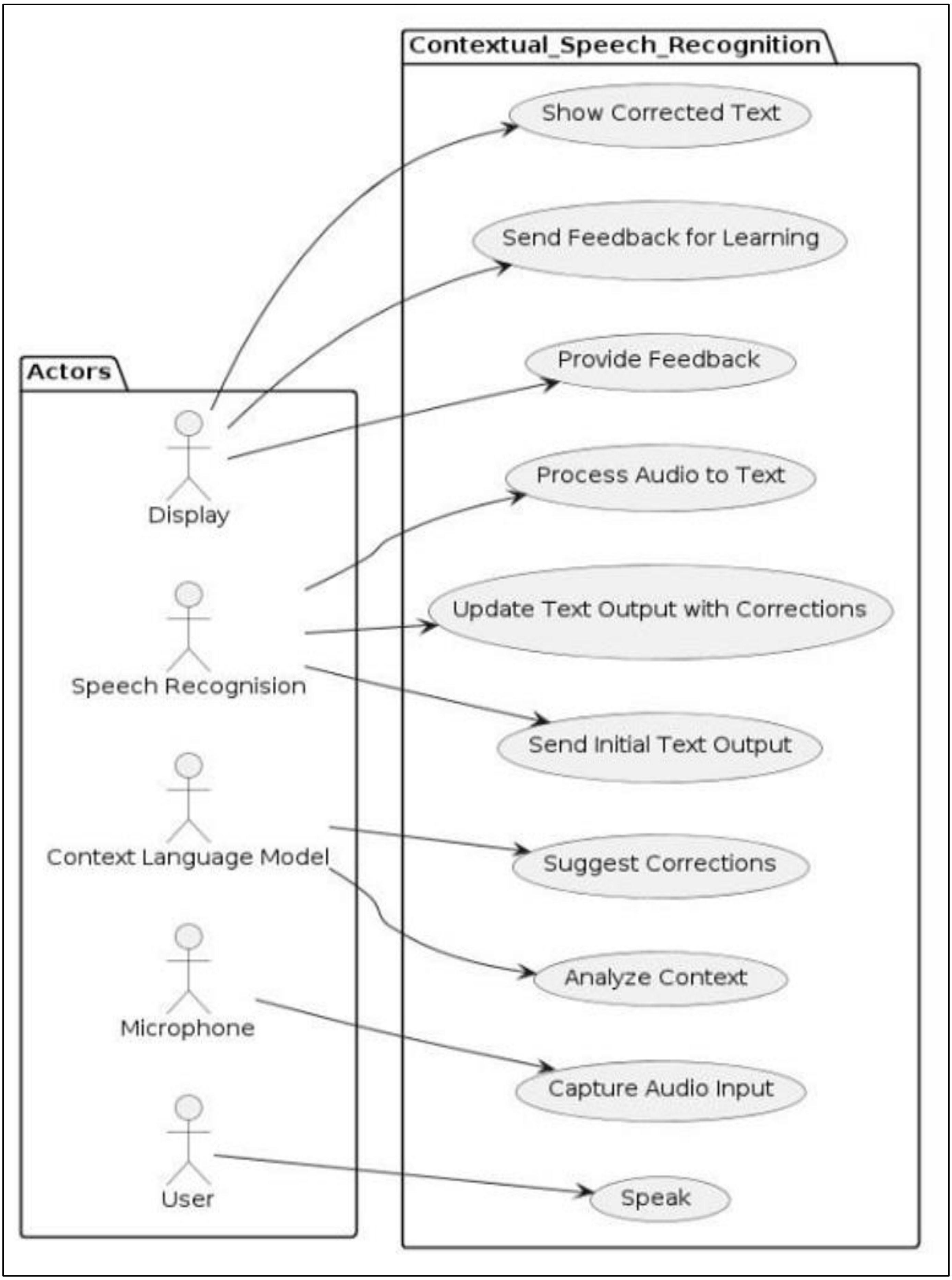


Fig 1 Usecase Diagram

➢ *Sequence Diagram:*

A sequence diagram in Unified Modelling Language (UML) is a kind of interaction diagram that shows how processes operate with one another and in what order. It is a construct of a Message Sequence Chart. Sequence diagrams are sometimes called event diagrams, event scenarios, and timing diagrams.

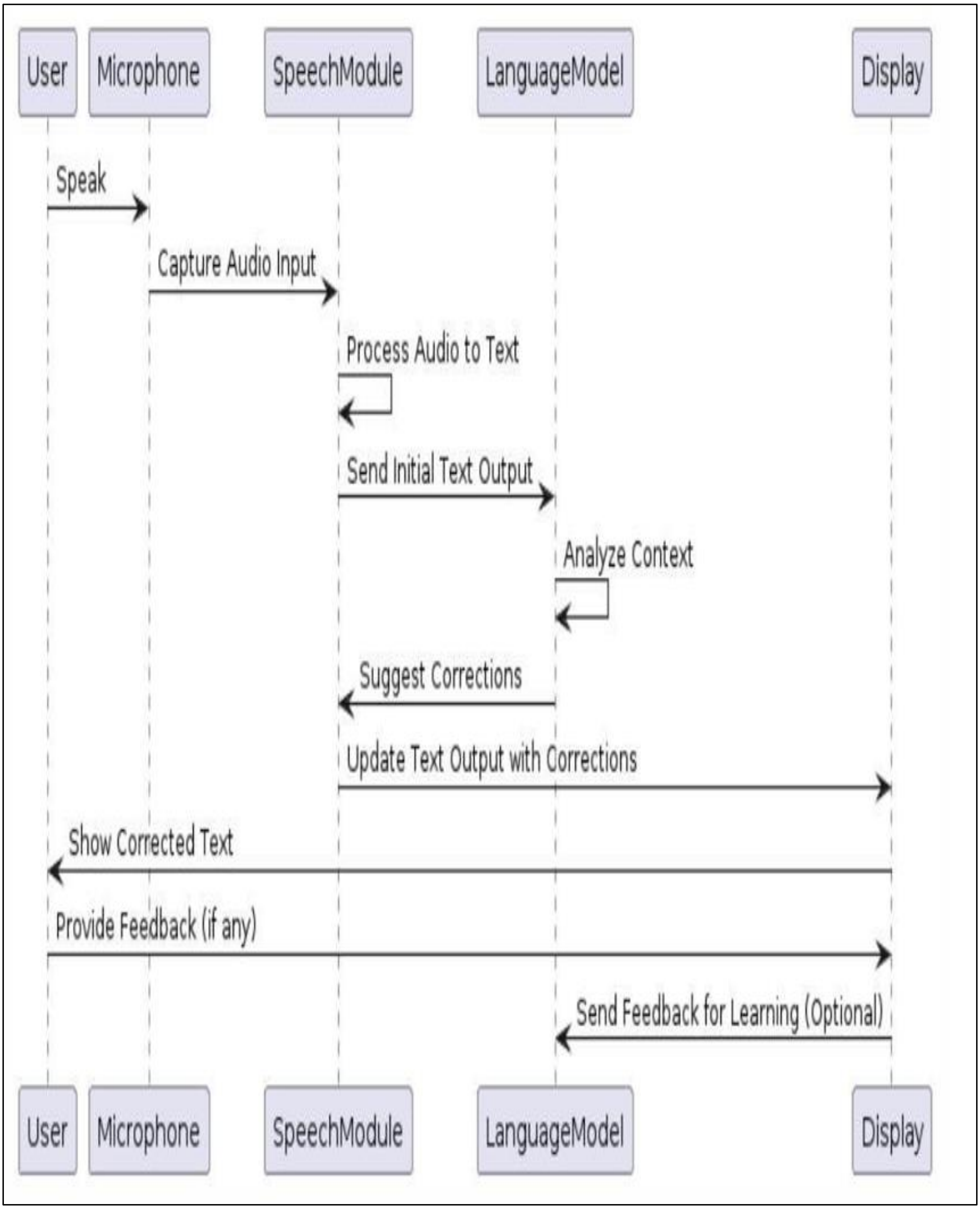


Fig 2 Sequencee Diagram

➢ *Activity Diagram:*

Activity diagrams are graphical representations of workflows of stepwise activities and actions with support for choice, iteration and concurrency. In the Unified Modeling Language, activity diagrams can be used to describe the businessand operational step- by-step workflows of components in a system. An activity diagram shows the overall flow of control.

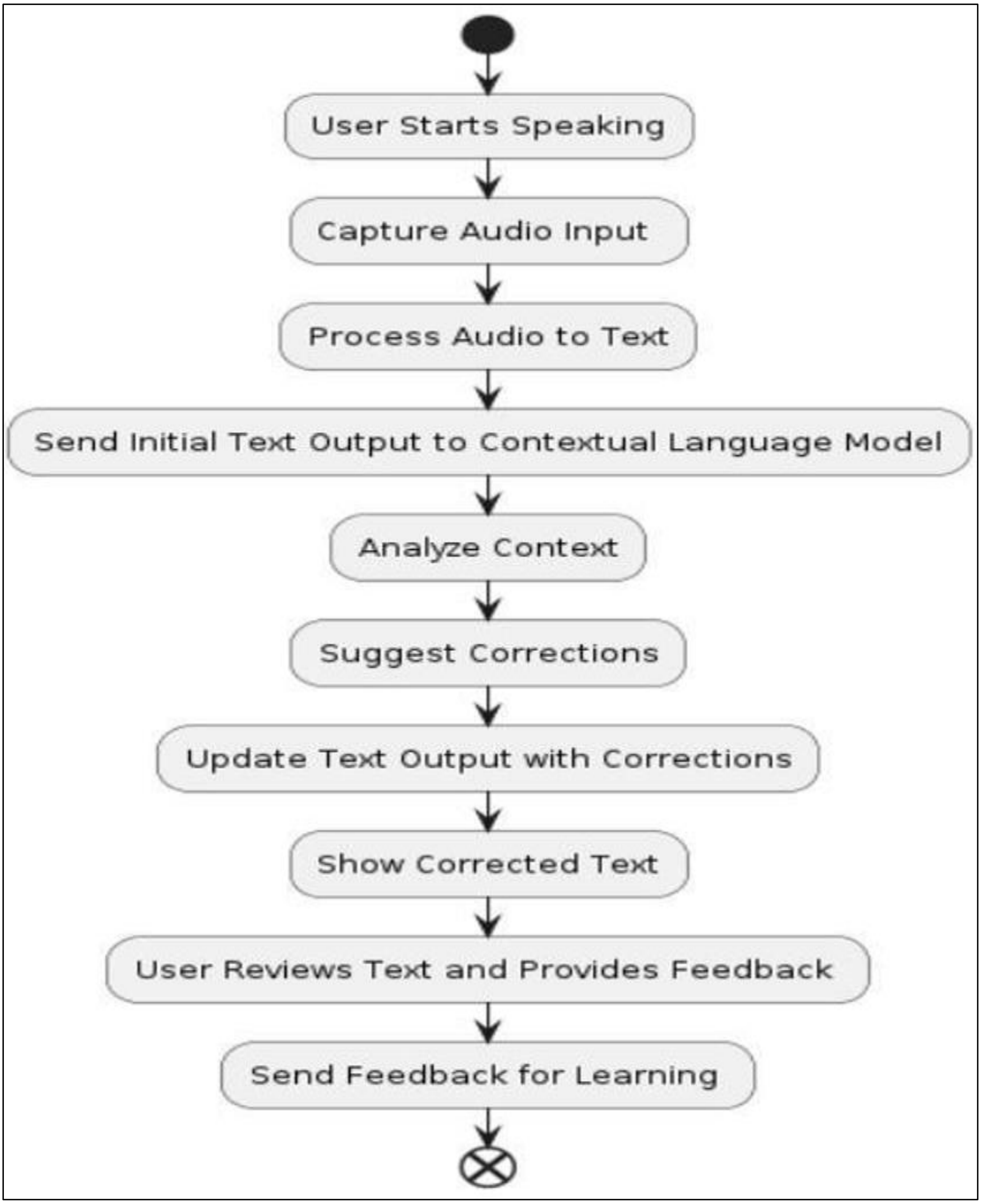

User Starts Speaking
Capture Audio Input
Process Audio to Text
Send Initial Text Output to Contextual Language Model
Analyze Context
Suggest Corrections
Update Text Output with Corrections
Show Corrected Text
User Reviews Text and Provides Feedback
Send Feedback for Learning

Fig 3 Activity Diagram

➢ *Modules:*

- *System Module:*

❖ *Acoustic Model:*

✓ *Purpose*:
Converts raw audio signals into phonetic units.

✓ *Enhancement*:
Deep learning-based models like Convolutional Neural Networks(CNNs) and Recurrent Neural Networks (RNNs) improve the accuracy of recognizing phonemes

❖ *Language Model (LM):*

✓ *Purpose:*
Provides context by predicting word sequences and reducing recognition errors.

✓ *Enhancement:*
Transformer models (like BERT, GPT) or sequence-based models like RNNs can be integrated to better predict words based on previous context.

❖ *Contextual Language Model:*

✓ *Purpose:*
Incorporates specific contextual knowledge like user preferences, location, or recent interactions to enhance understanding.

✓ *Enhancement:*
Fine-tuning pretrained models on domain-specific data or using external knowledge sources (like Knowledge Graphs) for adapting the model to specific contexts.

❖ *Named Entity Recognition (NER):*

✓ *Purpose:*
Identifies proper nouns, locations, and specific entities within thespeech that need more precise recognition.

✓ *Enhancement:*
NER integrated with ASR can boost the accuracy of recognizing specific names and terminologies relevant to the domain

❖ *Text Normalization:*

✓ *Purpose:*
Converts recognized speech into grammatically correct, well- structured sentences.

✓ *Enhancement:*
NLP techniques to handle abbreviations, numbers, dates, and punctuation that might not be easily recognizable in raw ASR output.

❖ *Context-Aware Re-ranking:*

✓ *Purpose:*
Re-ranks ASR hypotheses (potential transcriptions) using additional context from NLP models.

✓ *Enhancement:*
Post-processes ASR output using contextual clues or external knowledge to improve transcription accuracy.

❖ *Dialogue Management System:*

✓ *Purpose:*
Handles multi-turn interactions and maintains context across dialogues.

- ✓ *Enhancement:*
  NLP-based systems like Reinforcement Learning (RL) models or Transformer-based Dialogue Systems track user intents and context during the conversation to improve responses.

❖ *Word Embeddings:*

- ✓ *Purpose:*
  Converts words into vectors for semantic understanding.

- ✓ *Enhancement:*
  Word embedding techniques (like Word2Vec, GloVe, or BERTembeddings) can represent words in a contextually rich manner, allowing ASR systems to better understand meaning beyond surface-level recognition.

❖ *Intent Classification:*

- ✓ *Purpose:*
  Determines the user's intent behind the spoken words.

- ✓ *Enhancement:*
  NLP models can classify user intent from spoken language, helping the system respond appropriately, based on understanding the speech ina goal- oriented manner.

❖ *End-to-End ASR Models with NLP Integration:*

- ✓ *Purpose:*
  Combine acoustic modeling, language modeling, and contextunderstanding in one neural network model.

- ✓ *Enhancement:*
  End-to-end models like Deep Speech or Transformer-based models combine all these tasks and leverage contextual NLP during trainingand inference.

❖ *Context-Aware Decoder:*

- ✓ *Purpose:*
  Improves decoding of speech by integrating contextual information.

- ✓ *Enhancement:*
  Decoder that adapts based on dynamic context, including the subject of conversation or surrounding environment.

❖ *Error Correction Module:*

- ✓ *Purpose:*
  Corrects transcription errors by leveraging context and syntaxanalysis.

- ✓ *Enhancement:*
  Apply NLP techniques like grammatical error correction orsequence tagging to rectify mistakes in the recognized text.

- *End User Module*
  An end-user module enhancing ASR integrates contextual understanding usingNLP to improve transcription accuracy and semantic interpretation

CHAPTER FIVE
IMPLEMENTATION AND RESULTS

➢ *Method of Implementation*

- *What is Python:*

Python is currently the most widely used multi-purpose, high-level programming language.Python allows programming in Object-Oriented and Procedural paradigms. Python programs generally are smaller than other programming languages like Java. Programmers have to type relatively less and indentation requirement of the language, makes them readable all the time.

Python language is being used by almost all tech-giant companies like – Google, Amazon, Facebook, Instagram, Dropbox, Uber… etc. The biggest strength of Python is huge collection of standard library which can beused for the following –GUI Applications (like Kivy, Tkinter, PyQt etc.) Test frameworksMultimedia

- *Python*

Python is an interpreted high-level programming language for general-purposeprogramming. Created by Guido van Rossum and first released in 1991, Python has a design philosophy that emphasizes code readability, notably using significant whitespace.

Python features a dynamic type system and automatic memory management. Itsupports multiple programming paradigms, including object-oriented, imperative, functional and procedural, and has a large and comprehensive standard library.

✓ *Python is Interpreted*

Python is processed at runtime by the interpreter. You do notneed to compile your program before executing it. This is similar to PERL and PHP.

✓ *Python is Interactive*

you can actually sit at a Python prompt and interact with theinterpreter directly to write your programs.

Python also acknowledges that speed of development is important. Readable andterse code is part of this, and so is access to powerful constructs that avoid tedious repetition of code. Maintainability also ties into this may be an all but useless metric,but it does say something about how much code you have to scan, read and/or the ease with which a programmer of other languages can pick up basic Python skills and the huge standard library is key to another area where Python excels. All its tools have been quick to implement, saved a lot of time, and several of them have later been patched and updated by people with no Python background - without breaking.

- *What is Machine Learning :*

Before we take a look at the details of various machine learning methods, let's start by looking at what machine learning is, and what it isn't. Machine learning is often categorized as a subfield of artificial intelligence, but I find that categorization can often be misleading at first brush. The study of machine learning certainly arose from research in this context, but in the data science application of machine learning methods, it's more helpful to think of machine learning as a means of building models of data.

Fundamentally, machine learning involves building mathematical models to help understand data. "Learning" enters the fray when we give these models tunable parameters that can be adapted to observed data; in this way the program can be considered to be "learning" from the data. Once these models have been fit to previously seen data, they can be used to predict and understand aspects of newly observed data. I'll leave to the reader the more philosophical digression regarding theextent to which this type of mathematical, model-based "learning" is similar to the "learning" exhibited by the human brain.

Understanding the problem setting in machine learning is essential to using thesetools effectively, and so we will start with some broad categorizations of the types of approaches we'll discuss here.

- *Modules Used in Project:*

TensorFlow is a free and open-source software library for dataflow and differentiable programming across a range of tasks. It is a symbolic math library, andis also used for machine learning applications such as neural networks. It is used for both research and production at Google. TensorFlow was developed by the Google Brain team for internal Google use. It was released under the Apache 2.0 open-source license on November 9, 2015.

- *Numpy*

Numpy is a general-purpose array-processing package. It provides a high- performance multidimensional array object, and tools for working with these arrays. Itis the fundamental package for scientific computing with Python. It contains various features including these important ones:

- ✓ A powerful N-dimensional array object. Sophisticated (broadcasting) functions.Tools for integrating C/C++ and Fortran code.
- ✓ Useful linear algebra, Fourier transform, and random number capabilities.
- ✓ Besides its obvious scientific uses, Numpy can also be used as an efficient multi- dimensional container of generic data. Arbitrary data-types can be defined using Numpy which allows Numpy to seamlessly and speedily integrate with a wide variety of databases.

- *Pandas*

Pandas is an open-source Python Library providing high-performance data manipulation and analysis tool using its powerful data structures. Python was majorly used for data munging and preparation. It had very little contribution towards data analysis. Pandas solved this problem.

Using Pandas, we can accomplish five typical steps in the processing and analysisof data, regardless of the origin of data load, prepare, manipulate, model, and analyze.

Python with Pandas is used in a wide range of fields including academic and commercial domains including finance, economics, Statistics, analytics, etc.

- *Wishper AI-*

Whisper AI, developed by OpenAI, is an automatic speech recognition (ASR) system that converts spoken language into text. It is particularly adept attranscribing speech from various languages and accents, even in noisy environments. Whisper AI utilizes deep learning techniques, specifically transformer-based neural networks, to achieve high accuracy in speech recognition tasks.

One of its key uses is in applications like transcription services for podcasts, meetings, and lectures, making it easier for users to convert audio content into text. It is also integrated into customer service systems, enabling voice-based interactions and automating tasks such as call transcription or voice-activated commands.

Additionally, Whisper AI is used in accessibility technologies, where learners can practice pronunciation and have their spoken input accurately transcribed and analyzed. Developers integrate Whisper into apps and devices for tasks like voice search, speech-to-text for virtual assistants, and translation services.

By being open-source, Whisper has enabled a broad range of applications, from personal productivity tools to enterprise-level AI solutions for voice processing.

- *Hugging Face-*

Hugging Face is a platform for natural language processing (NLP) and machine learning, widely known for its open-source libraries, especially **Transformers**, which provides pre-trained models for tasks like text classification,translation, and question answering. Developers can easily fine-tune these models or use them for inference. It supports multiple programming languages, making integration into projects simple.

Hugging Face also offers the **Hub**, where users can upload, share, and discover models and datasets. It's used in research and industry for fast, efficient, and scalable deployment.

AI models without needing extensive computational resources.

- *Python is Interpreted –*

Python is processed at runtime by the interpreter. You do notneed to compile your program before executing it. This is similar to PERL and PHP.

- *Python is Interactive –*

you can actually sit at a Python prompt and interact with the interpreter directly to write your programs.

Python also acknowledges that speed of development is important. Readable and terse code is part of this, and so is access to powerful constructs that avoid tedious repetition of code. Maintainability also ties into this may be an all but useless metric, but it does say something about how much code you have to scan, read and/or understand to troubleshoot problems or tweak behaviors.

This speed of development, the ease with which a programmer of other languages can pick up basic Python skills and the huge standard library is key to another area where Python excels. All its tools have been quick to implement, saved a lot of time, andseveral of them have later been patched and updated by people with no Pythonbackground - without breaking.

- *Install Python Step-by-Step in Windows and Mac:*

Python a versatile programming language doesn't come pre-installed on yourcomputer devices. Python was first released in the year 1991 and until today it is a verypopular high- level programming language. Its style philosophy emphasizes code readability with its notable use of great whitespace.

The object-oriented approach and language construct provided by Python enables programmers to write both clear and logical code for projects. This software does not come pre-packaged with Windows.

- *Howto Install Python on Windows and Mac:*

There have been several updates in the Python version over the years. The questionis how to install Python? It might be confusing for the beginner who is willing to start learning Python but this tutorial will solve your query. The latest or the newest version ofPython is version 3.7.4 or in other words, it is Python 3.

- *Note*:

The python version 3.7.4 cannot be used on Windows XP or earlier devices. Before you start with the installation process of Python. First, you need to know aboutyour **System Requirements**. Based on your system type i.e. operating system andbased processor, you must download the python version. My system type is a **Windows64-bit Operating system**.

So, the steps below are to install python version 3.7.4 on Windows 7 device or to install Python 3. Download the Python Cheatsheet here.The steps on how to install Python on Windows 10, 8 and 7 are **divided into 4 parts** to help understand better.

- *Step 1:*

Go to the official site to download and install python using Google Chrome orany other web browser. OR Click on the following link: **https://www.python.org**

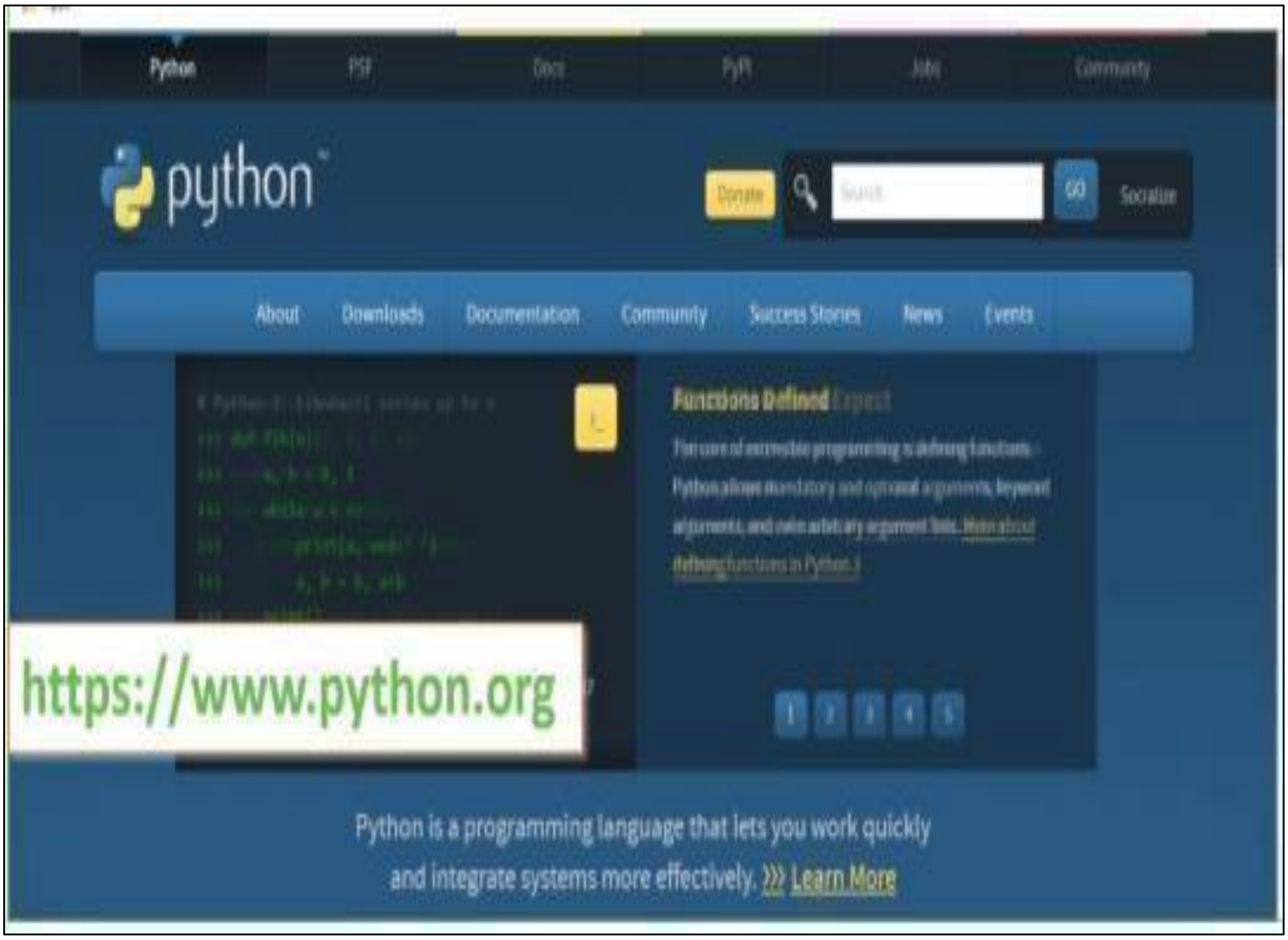
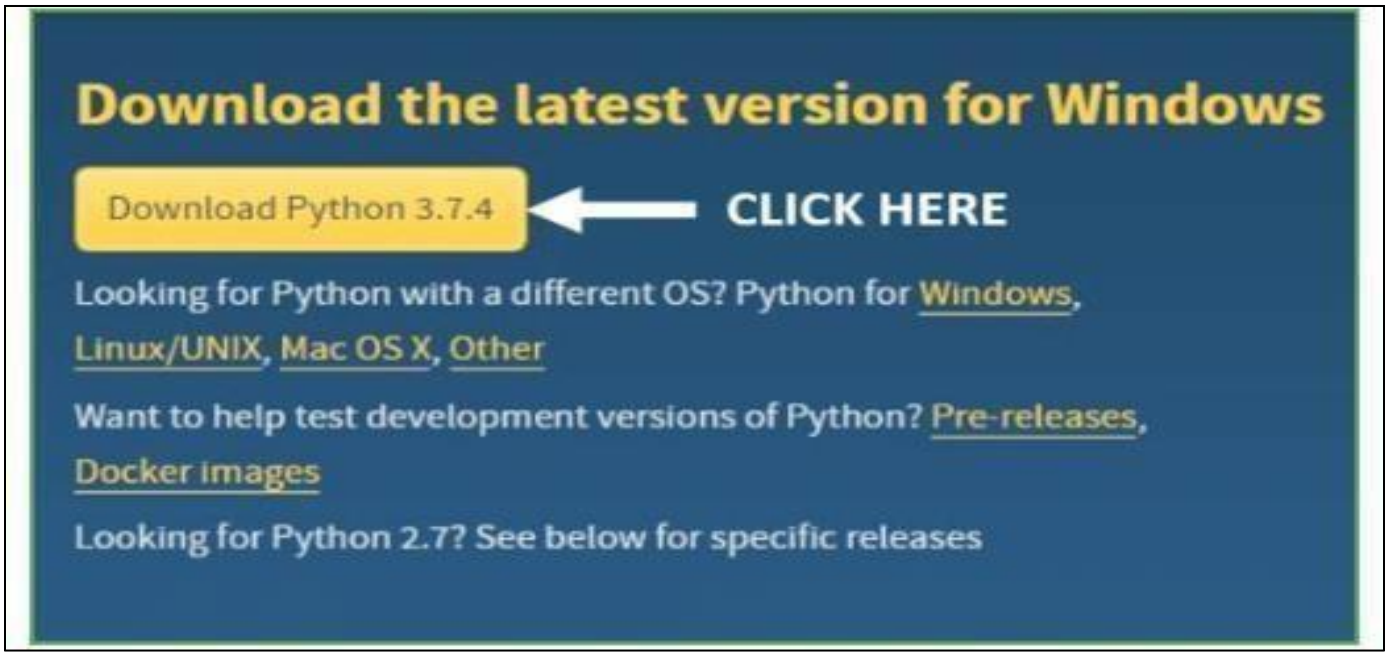


Fig 4 Download Python

Now, check for the latest and the correct version for your operating system.

- *Step 2:*
  Click on the Download Tab.



Download the latest version for Windows

Download Python 3.7.4 CLICK HERE

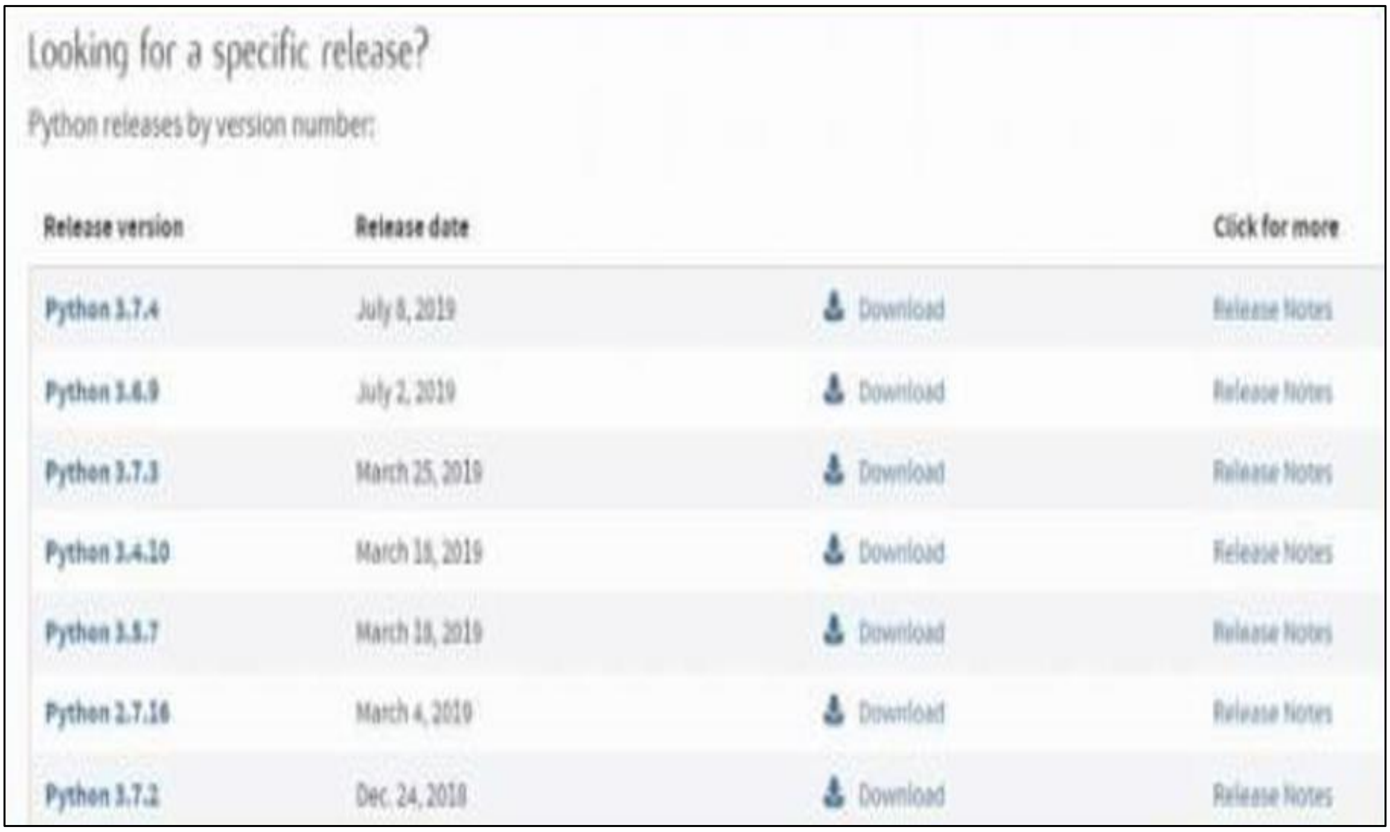
Looking for Python with a different OS? Python for Windows, Linux/UNIX, Mac OS X, Other

Want to help test development versions of Python? Pre-releases, Docker images

Looking for Python 2.7? See below for specific releases

Fig 5 Python

- *Step 3:*
  You can either select the Download Python for windows 3.7.4 button in Yellow Color or you can scroll further down and click on download with respective to their version. Here, we are downloading the most recent python version for windows 3.7.4



Looking for a specific release?

Python releases by version number:

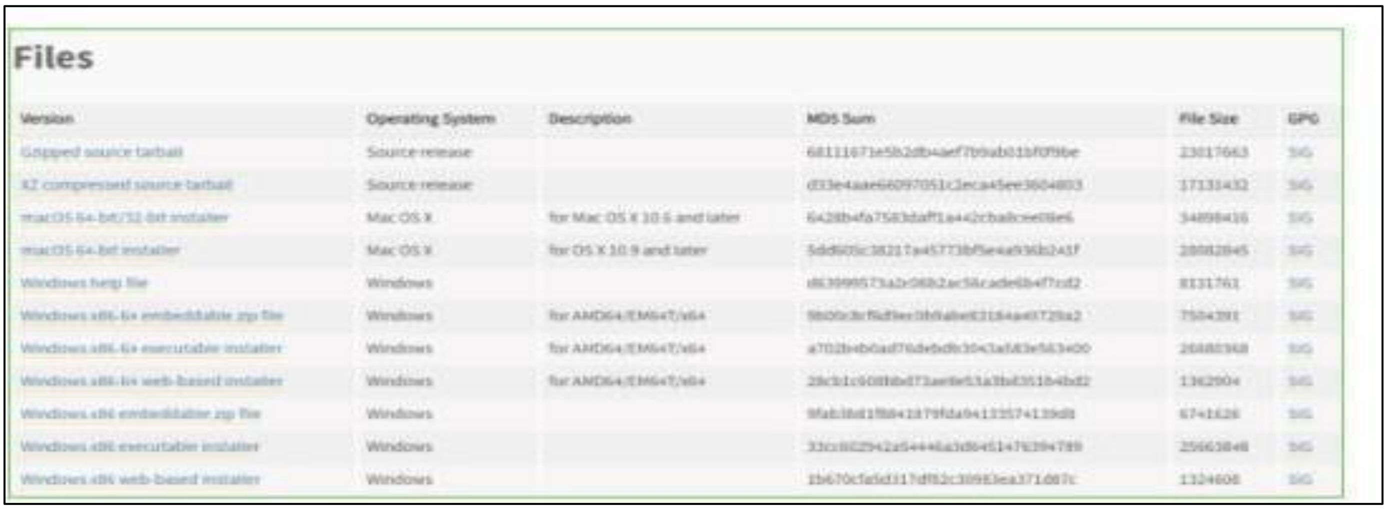
Release version	Release date		Click for more
Python 3.7.4	July 8, 2019	Download	Release Notes
Python 3.6.9	July 2, 2019	Download	Release Notes
Python 3.7.3	March 25, 2019	Download	Release Notes
Python 3.4.10	March 18, 2019	Download	Release Notes
Python 3.5.7	March 18, 2019	Download	Release Notes
Python 2.7.16	March 4, 2019	Download	Release Notes
Python 3.7.2	Dec. 24, 2018	Download	Release Notes

Fig 6 Specific Release

- *Step 4:*

Scroll down the page until you find the Files option.

- *Step 5:*
  Here you see a different version of python along with the operating system.



Files

Version	Operating System	Description	MD5 Sum	File Size	GPG
Gzipped source tarball	Source release		68111671e5b2db4aef7b9ab01bf0f9be	23017663	SIG
XZ compressed source tarball	Source release		d33e4aae66097051c2eca45ee3604803	17131432	SIG
macOS 64-bit/32-bit installer	Mac OS X	for Mac OS X 10.6 and later	6428b4fa7583daff1a442cba8cee08e6	34898416	SIG
macOS 64-bit installer	Mac OS X	for OS X 10.9 and later	5dd605c38217a45773bf5e4a936b241f	28082845	SIG
Windows help file	Windows		d63999573a2c56b2ac56cade6b4f7cd2	8131761	SIG
Windows x86-64 embeddable zip file	Windows	for AMD64/EM64T/x64	9b00c8cf6d9ec0b9abe83184a40729a2	7504391	SIG
Windows x86-64 executable installer	Windows	for AMD64/EM64T/x64	a702b4b0ad76debdb3043a583e563400	26680368	SIG
Windows x86-64 web-based installer	Windows	for AMD64/EM64T/x64	28cb1c60886bd73ae8e53a3bd351b4bd2	1362904	SIG
Windows x86 embeddable zip file	Windows		9fab3b81f8841879fda94133574139d8	6741626	SIG
Windows x86 executable installer	Windows		33cc602942a54446a3d6451476394789	25663848	SIG
Windows x86 web-based installer	Windows		1b670cfa5d317df82c30983ea371d87c	1324608	SIG

Fig 7 Files

To download Windows 32-bit python, you can select any one from the three options: Windows x86 embeddable zip file, Windows x86 executable installer or Windows x86 web-based installer. To download Windows 64-bit python, you can selectany one from the three options: Windows x86-64 embeddable zip file, Windows x86-64executable installer or Windows x86-64 web-based installer.

we will install Windows x86-64 web-based installer. Here your first part regarding which version of python is to be downloaded is completed. Now we move ahead with the second part in installing python i.e. Installation.

- *Note*:
  To know the changes or updates that are made in the version you can click on theRelease Note Option.

  Installation of Python

- *Step 1:*
  Go to Download and Open the downloaded python version to carry out theinstallation process.

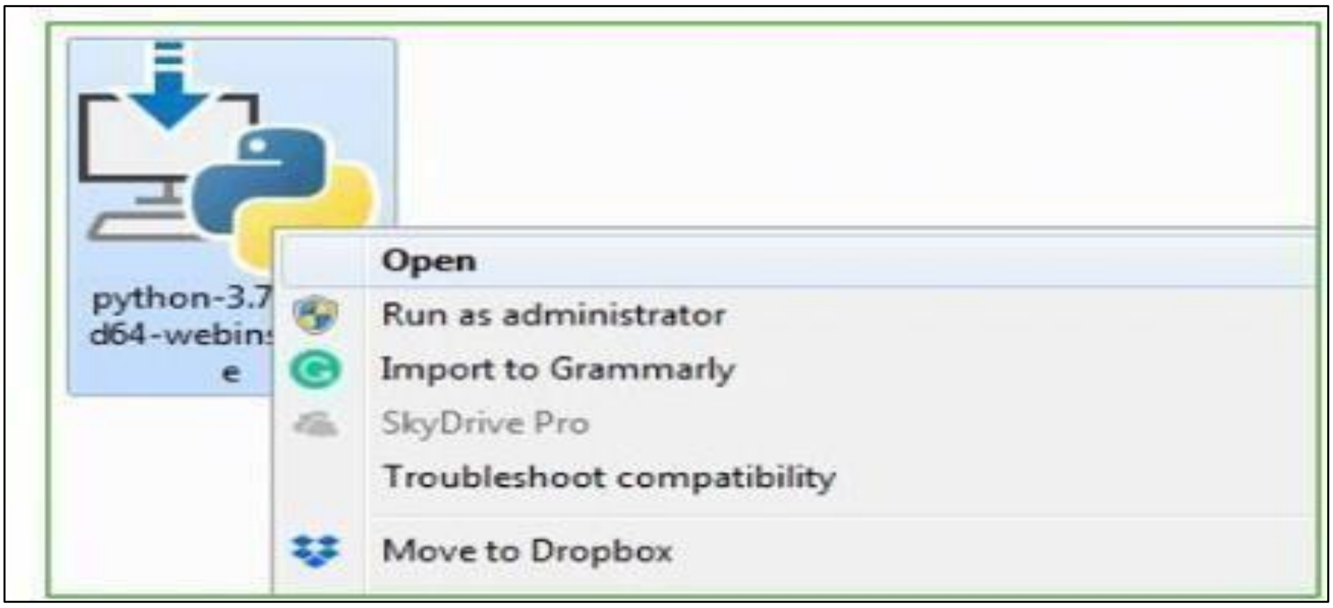
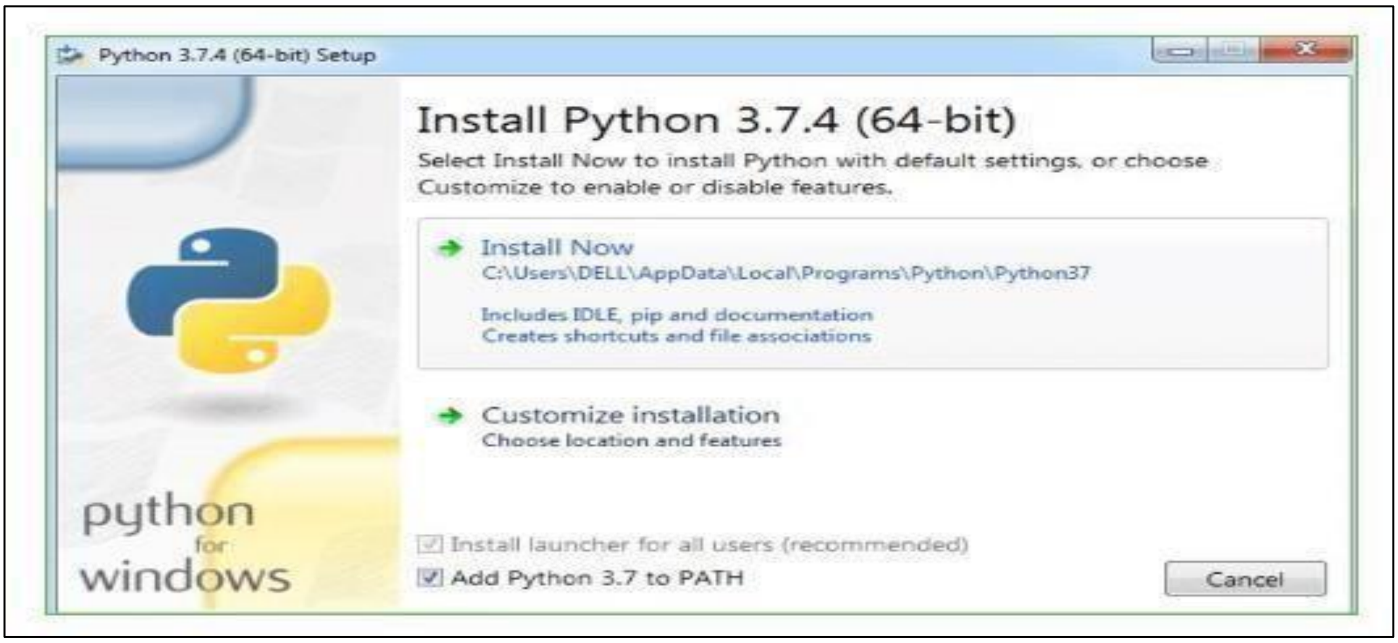


Fig 8 Open Python

- *Step 2:*
  Before you click on Install Now, Make sure to put a tick on Add Python 3.7 toPATH.



Python 3.7.4 (64-bit) Setup

Install Python 3.7.4 (64-bit)

Select Install Now to install Python with default settings, or choose Customize to enable or disable features.

→ Install Now
C:\Users\DELL\AppData\Local\Programs\Python\Python37

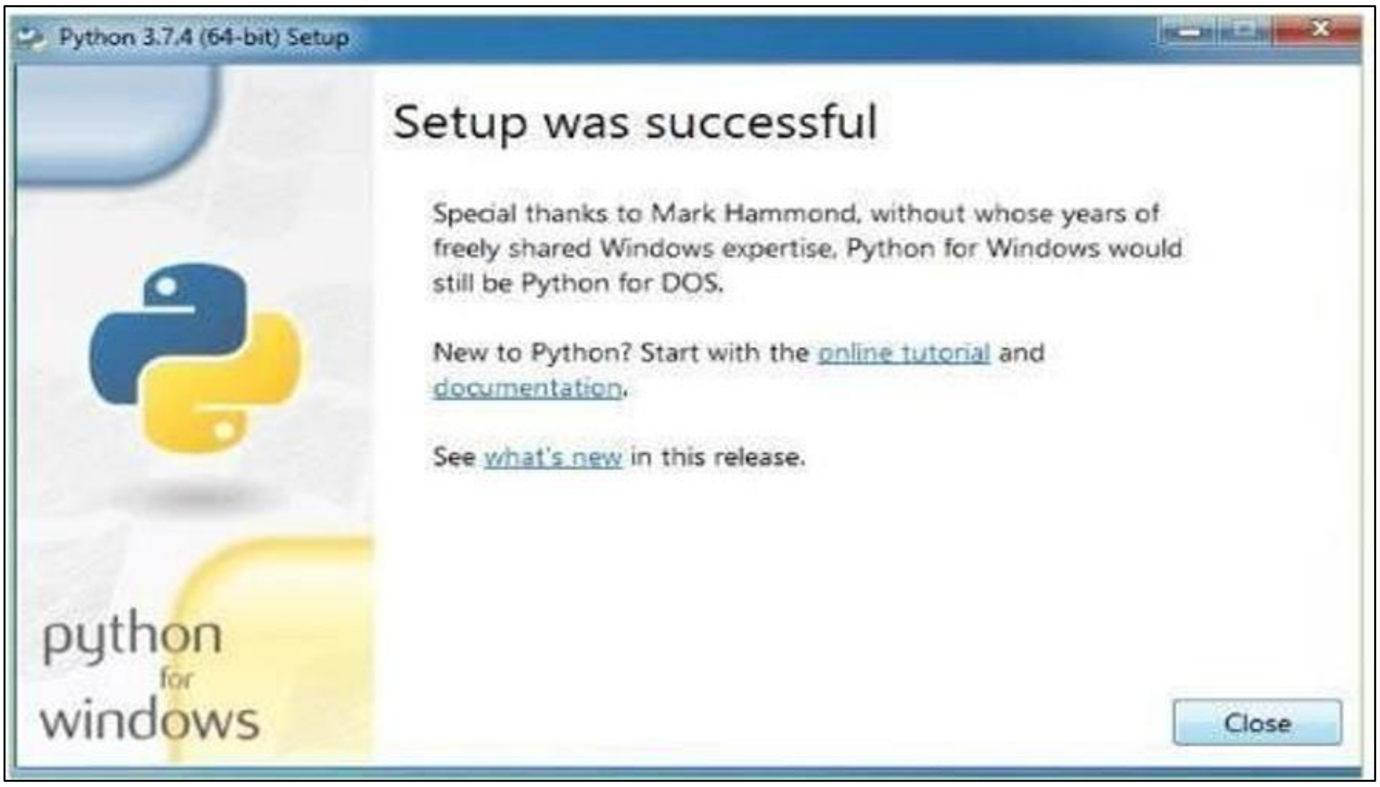
Includes IDLE, pip and documentation
Creates shortcuts and file associations

→ Customize installation
Choose location and features

[✓] Install launcher for all users (recommended)
[✓] Add Python 3.7 to PATH

Fig 9 Install Python

- *Step 3:*
  Click on Install NOW After the installation is successful. Click on Close.



Python 3.7.4 (64-bit) Setup

Setup was successful

Special thanks to Mark Hammond, without whose years of freely shared Windows expertise, Python for Windows would still be Python for DOS.

New to Python? Start with the online tutorial and documentation.

See what's new in this release.

Fig 10 Installed Successfully

With these above threesteps on python installation, you havesuccessfully and correctlyinstalled Python. Now is the time to verify the installation.

- *Note:*
  The installation process might take a couple of minutes.

Verify the Python InstallationStep 1: Click on Start

- *Step 2:*
  In the Windows Run Command, type "cmd".

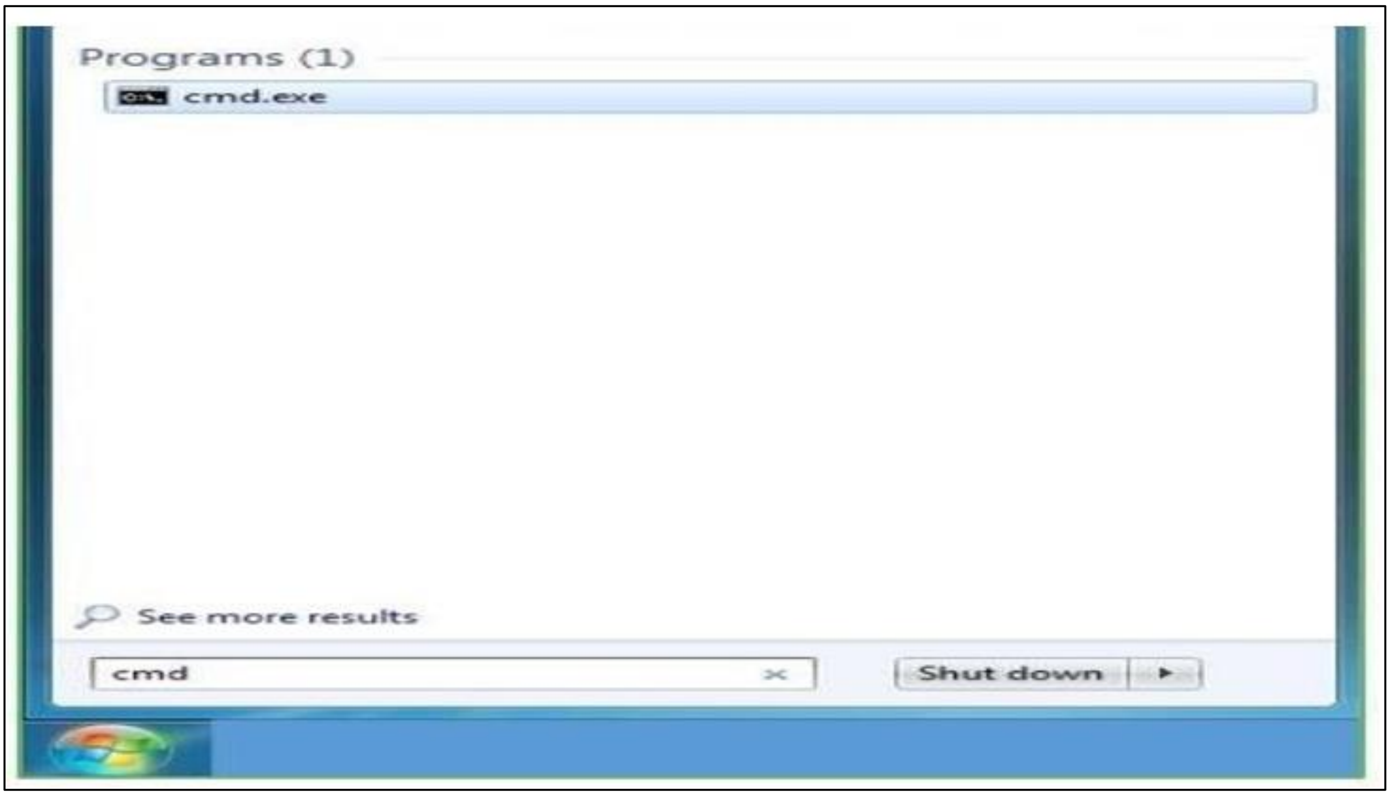
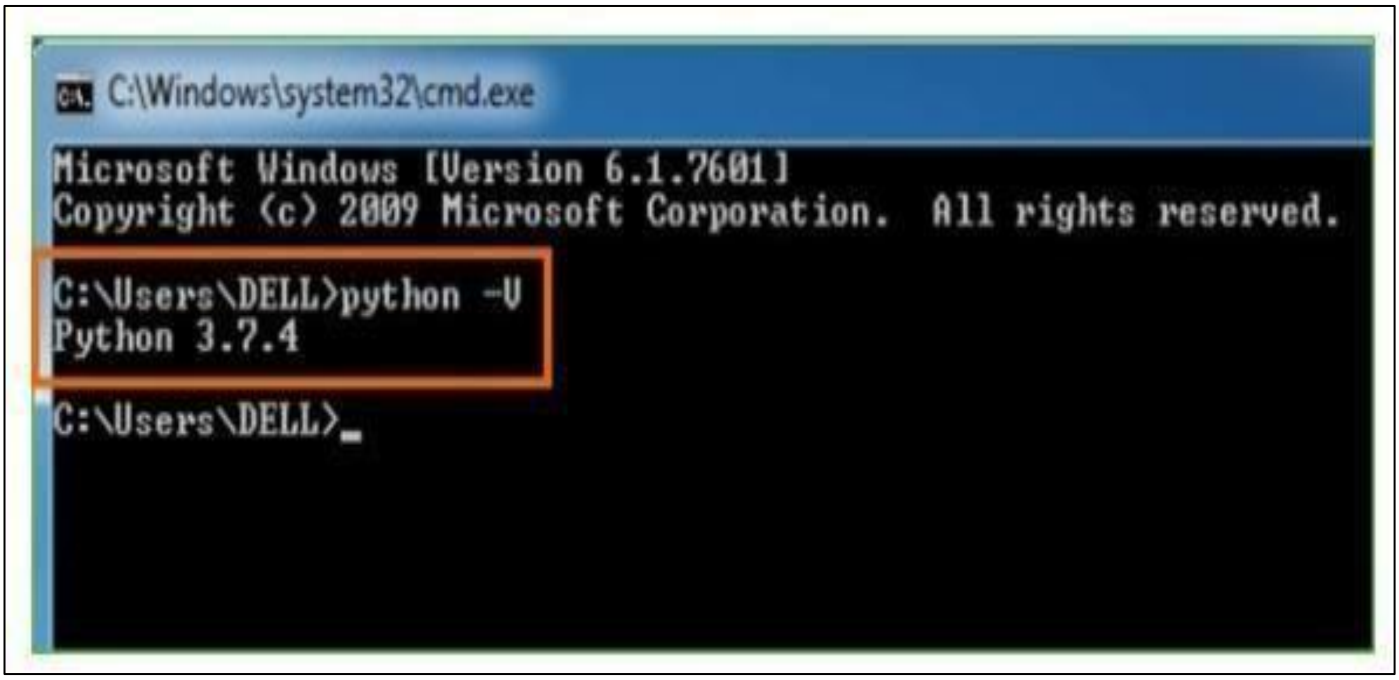


Fig 11 Select Command Prompt

- *Step 3:*
  Open the Command prompt option.

- *Step 4:*
  Let us test whether the python is correctly installed. Type **python –V** and pressEnter.



```
Microsoft Windows [Version 6.1.7601]
Copyright (c) 2009 Microsoft Corporation.  All rights reserved.

C:\Users\DELL>python -V
Python 3.7.4

C:\Users\DELL>_
```

Fig 12 Search Python Version

- *Step 5:*
  You will get the answer as 3.7.4

- *Note:*
  If you have any of the earlier versions of Python already installed. You must firstuninstall the earlier version and then install the new one.

➢ *Check how the Python IDLE works*

- Step 1: Click on Start

- Step 2: In the Windows Run command, type "python idle".

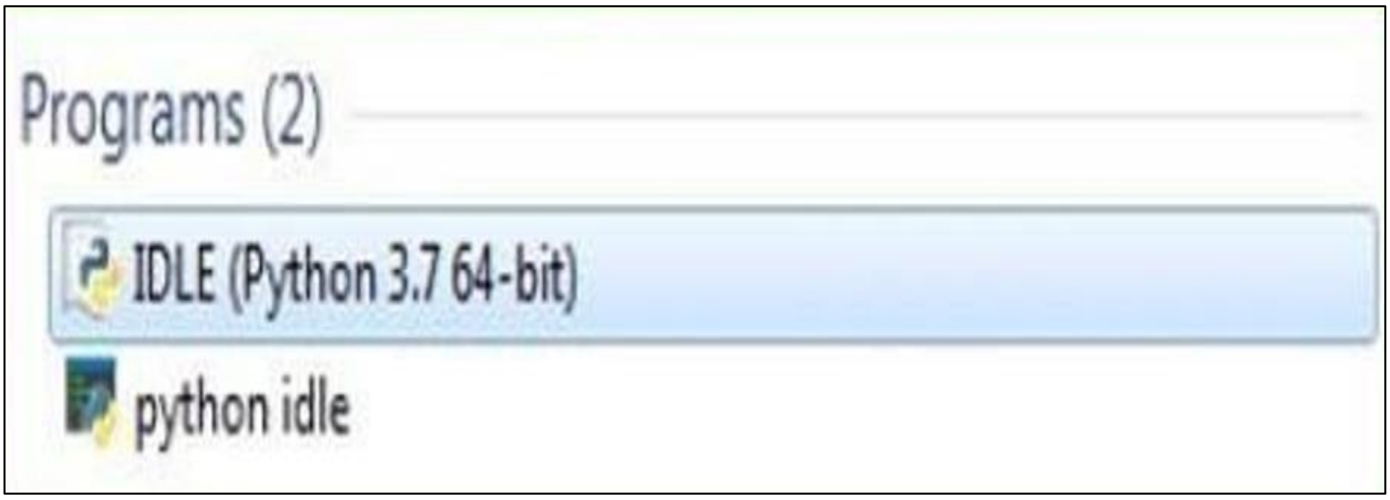
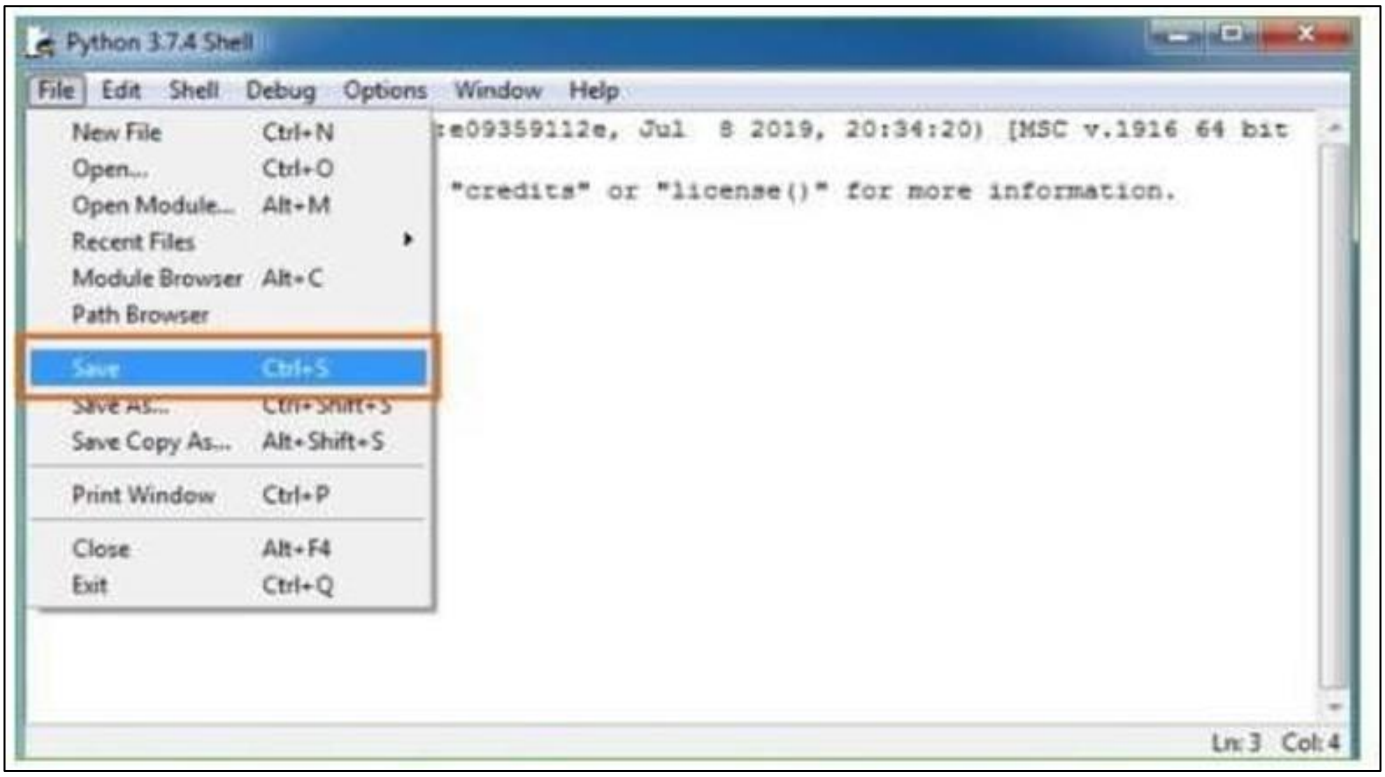


Fig 13 Idle Python

- Step 3: Click on IDLE (Python 3.7 64-bit) and launch the program

- Step 4: To go ahead with working in IDLE you must first save the file. Click on File> Click on Save



```
:e09359112e, Jul  8 2019, 20:34:20) [MSC v.1916 64 bit
 "credits" or "license()" for more information.
```

Fig 14 Save The File

- Step 5: Name the file and save as type should be Python files. Click on SAVE. Here Ihave named the files as Hey World

- Step 6: Now for example.enter print.

➢ Extension of Key Function:

The code we have written performs fine-tuning of the Whisper model from Hugging Face's `transformers` library on a speech-to-text dataset. Here are the key functions and components in the program:

- *Dataset Loading and Preparation*

✓ *Load dataset:*
Loads a custom dataset from Hugging Face Hub(`thitherto/indian_english_extended`) for fine-tuning.

✓ *Train test split:*
Splits the dataset into training and test sets.

✓ *Audio*:
Casts the audio column to the appropriate format and ensures audio isresampled to 16kHz for Whisper model compatibility

- *Feature Extraction and Tokenization*

✓ *Whisper Feature Extractor:*
Extracts the log-mel spectrogram features from audio samples. Whisper Tokenizer: Tokenizes the transcriptions (text) into input sequences of integers.

✓ *Prepare dataset:*
A function that prepares each batch by:

Converting audio into log-Mel spectrogram features.

Encoding transcriptions into token sequences (labels).

- *Data Collator*

✓ Data Collator Speech Seq-Seq With Padding:
Handles the padding of both the inputfeatures and target labels during batching, ensuring consistent input sizes. It also adds attention masks and processes labels for training.

- *Model Initialization*

✓ Whisper For Conditional Generation:
Loads the Whisper model for speech-to-text tasks. processor: Combines both the tokenizer and feature extractor.

- *Evaluation*

✓ Evaluate:
Uses the `wer` (Word Error Rate) metric to evaluate the model'sperformance.

The function `compute metrics` calculates WER on the predictions.

- *Training Arguments*

Seq-Seq Training Arguments:
Defines training configurations like learning rate,batch size, number of steps, evaluation strategy, etc.

- *Trainer Setup*

✓ *Seq-Seq Trainer:*
The core Hugging Face trainer that:

Fine-tunes the Whisper model on the training data.

Evaluates the model on the test set.

Uses the provided data collator and metrics for training.

- *Model Saving and Pushing to Hugging Face Hub*

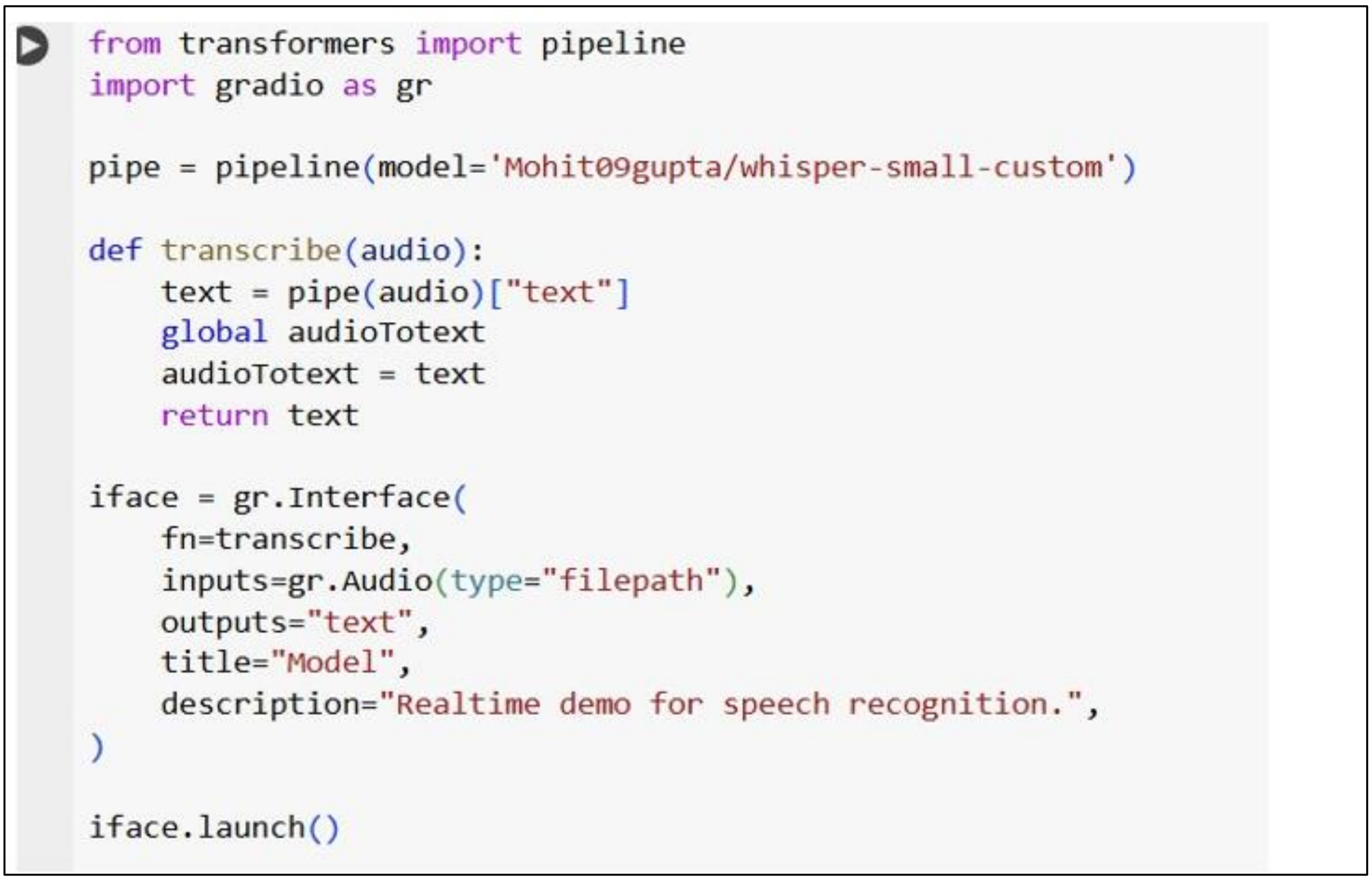
✓ *Processor Save Pretrained:*
Saves the feature extractor and tokenizer for later use. trainer push to hub: Pushes the trained model and configuration to Hugging FaceModel Hub.

- *Optional Monitoring*

✓ Tensor board:
Logs progress for visualization on Tensor Board.

These functions form the backbone of the fine-tuning pipeline for a speech recognitiontask with the Whisper model on Indian English.

➢ *Evaluation:*



```
from transformers import pipeline
import gradio as gr

pipe = pipeline(model='Mohit09gupta/whisper-small-custom')

def transcribe(audio):
    text = pipe(audio)["text"]
    global audioTotext
    audioTotext = text
    return text

iface = gr.Interface(
    fn=transcribe,
    inputs=gr.Audio(type="filepath"),
    outputs="text",
    title="Model",
    description="Realtime demo for speech recognition.",
)

iface.launch()
```

Fig 15 Evaluation

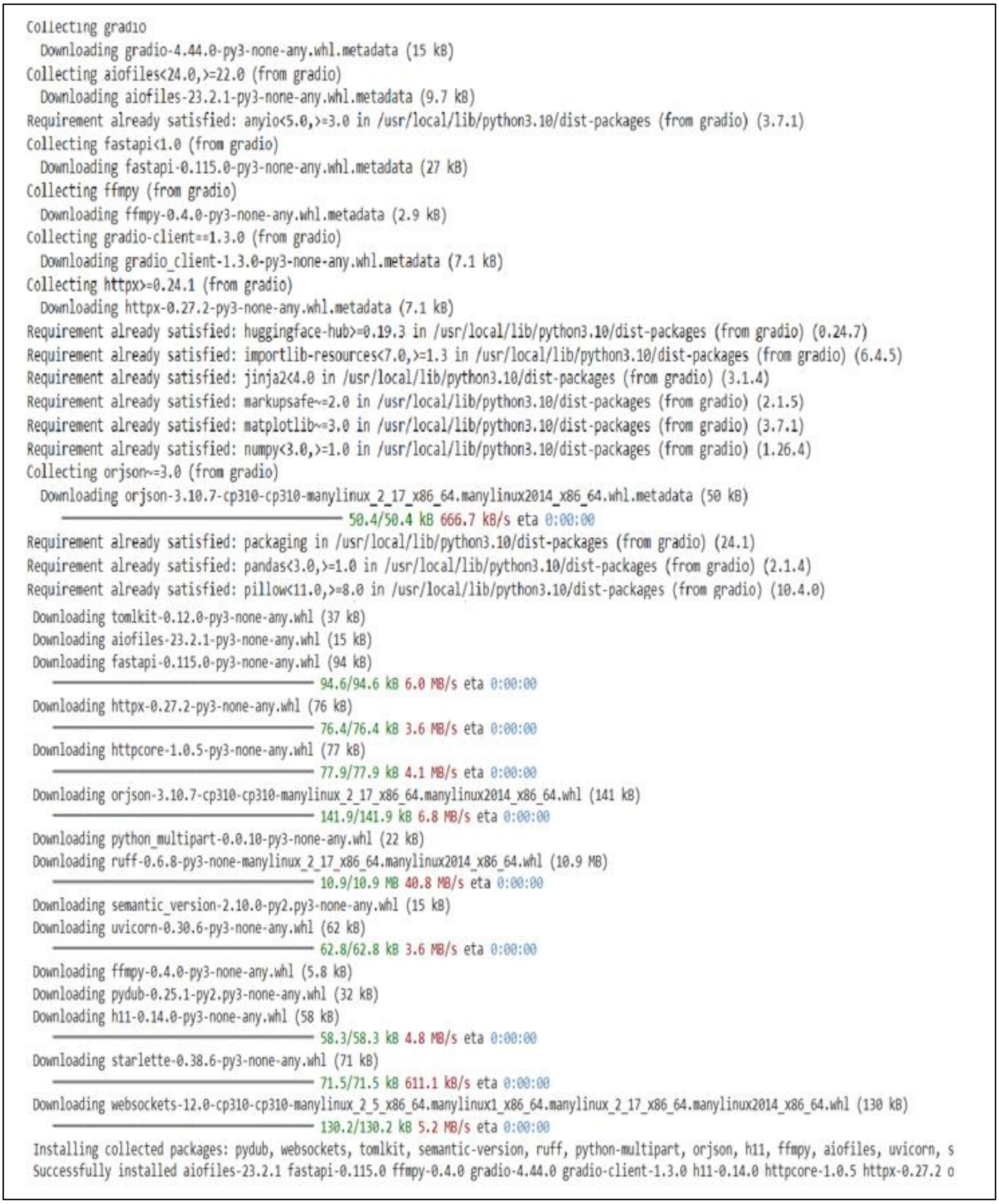
- *Pipeline Setup:*
You set up a Hugging Face `pipeline` for speech recognition using the model `'Mohit09gupta/whisper-small-custom'`.

✓ *Transcription Function:*
The `transcribe` function takes an audio file, runs it through the pipeline, and returns the transcribed text.

✓ *Gradio Interface:*

The Gradio `Interface` is created to enable real-time demos of the ASR system. It takes audio input, processes it, and returns the as output.

➢ *Output Screens:*



```
Collecting gradio
  Downloading gradio-4.44.0-py3-none-any.whl.metadata (15 kB)
Collecting aiofiles<24.0,>=22.0 (from gradio)
  Downloading aiofiles-23.2.1-py3-none-any.whl.metadata (9.7 kB)
Requirement already satisfied: anyio<5.0,>=3.0 in /usr/local/lib/python3.10/dist-packages (from gradio) (3.7.1)
Collecting fastapi<1.0 (from gradio)
  Downloading fastapi-0.115.0-py3-none-any.whl.metadata (27 kB)
Collecting ffmpy (from gradio)
  Downloading ffmpy-0.4.0-py3-none-any.whl.metadata (2.9 kB)
Collecting gradio-client==1.3.0 (from gradio)
  Downloading gradio_client-1.3.0-py3-none-any.whl.metadata (7.1 kB)
Collecting httpx>=0.24.1 (from gradio)
  Downloading httpx-0.27.2-py3-none-any.whl.metadata (7.1 kB)
Requirement already satisfied: huggingface-hub>=0.19.3 in /usr/local/lib/python3.10/dist-packages (from gradio) (0.24.7)
Requirement already satisfied: importlib-resources<7.0,>=1.3 in /usr/local/lib/python3.10/dist-packages (from gradio) (6.4.5)
Requirement already satisfied: jinja2<4.0 in /usr/local/lib/python3.10/dist-packages (from gradio) (3.1.4)
Requirement already satisfied: markupsafe~=2.0 in /usr/local/lib/python3.10/dist-packages (from gradio) (2.1.5)
Requirement already satisfied: matplotlib~=3.0 in /usr/local/lib/python3.10/dist-packages (from gradio) (3.7.1)
Requirement already satisfied: numpy<3.0,>=1.0 in /usr/local/lib/python3.10/dist-packages (from gradio) (1.26.4)
Collecting orjson~=3.0 (from gradio)
  Downloading orjson-3.10.7-cp310-cp310-manylinux_2_17_x86_64.manylinux2014_x86_64.whl.metadata (50 kB)
     ━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━ 50.4/50.4 kB 666.7 kB/s eta 0:00:00
Requirement already satisfied: packaging in /usr/local/lib/python3.10/dist-packages (from gradio) (24.1)
Requirement already satisfied: pandas<3.0,>=1.0 in /usr/local/lib/python3.10/dist-packages (from gradio) (2.1.4)
Requirement already satisfied: pillow<11.0,>=8.0 in /usr/local/lib/python3.10/dist-packages (from gradio) (10.4.0)
Downloading tomlkit-0.12.0-py3-none-any.whl (37 kB)
Downloading aiofiles-23.2.1-py3-none-any.whl (15 kB)
Downloading fastapi-0.115.0-py3-none-any.whl (94 kB)
   ━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━ 94.6/94.6 kB 6.0 MB/s eta 0:00:00
Downloading httpx-0.27.2-py3-none-any.whl (76 kB)
   ━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━ 76.4/76.4 kB 3.6 MB/s eta 0:00:00
Downloading httpcore-1.0.5-py3-none-any.whl (77 kB)
   ━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━ 77.9/77.9 kB 4.1 MB/s eta 0:00:00
Downloading orjson-3.10.7-cp310-cp310-manylinux_2_17_x86_64.manylinux2014_x86_64.whl (141 kB)
   ━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━ 141.9/141.9 kB 6.8 MB/s eta 0:00:00
Downloading python_multipart-0.0.10-py3-none-any.whl (22 kB)
Downloading ruff-0.6.8-py3-none-manylinux_2_17_x86_64.manylinux2014_x86_64.whl (10.9 MB)
   ━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━ 10.9/10.9 MB 40.8 MB/s eta 0:00:00
Downloading semantic_version-2.10.0-py2.py3-none-any.whl (15 kB)
Downloading uvicorn-0.30.6-py3-none-any.whl (62 kB)
   ━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━ 62.8/62.8 kB 3.6 MB/s eta 0:00:00
Downloading ffmpy-0.4.0-py3-none-any.whl (5.8 kB)
Downloading pydub-0.25.1-py2.py3-none-any.whl (32 kB)
Downloading h11-0.14.0-py3-none-any.whl (58 kB)
   ━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━ 58.3/58.3 kB 4.8 MB/s eta 0:00:00
Downloading starlette-0.38.6-py3-none-any.whl (71 kB)
   ━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━ 71.5/71.5 kB 611.1 kB/s eta 0:00:00
Downloading websockets-12.0-cp310-cp310-manylinux_2_5_x86_64.manylinux1_x86_64.manylinux_2_17_x86_64.manylinux2014_x86_64.whl (130 kB)
   ━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━━ 130.2/130.2 kB 5.2 MB/s eta 0:00:00
Installing collected packages: pydub, websockets, tomlkit, semantic-version, ruff, python-multipart, orjson, h11, ffmpy, aiofiles, uvicorn, s
Successfully installed aiofiles-23.2.1 fastapi-0.115.0 ffmpy-0.4.0 gradio-4.44.0 gradio-client-1.3.0 h11-0.14.0 httpcore-1.0.5 httpx-0.27.2 o
```

Fig 16 Output Screen

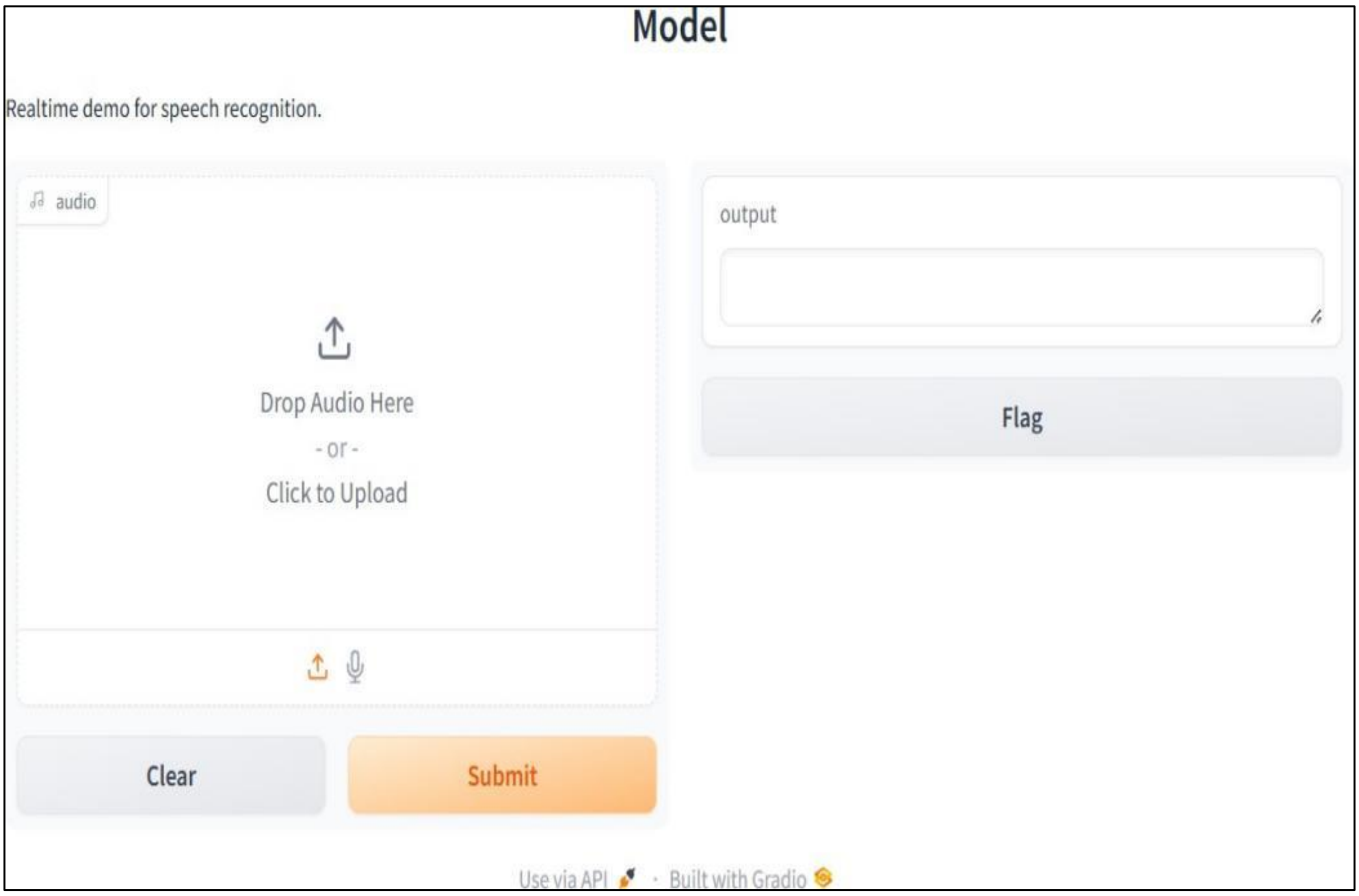

Model
Realtime demo for speech recognition.
audio
Drop Audio Here
- or -
Click to Upload
output
Flag
Clear
Submit
Use via API · Built with Gradio

Fig 17 Output Screen

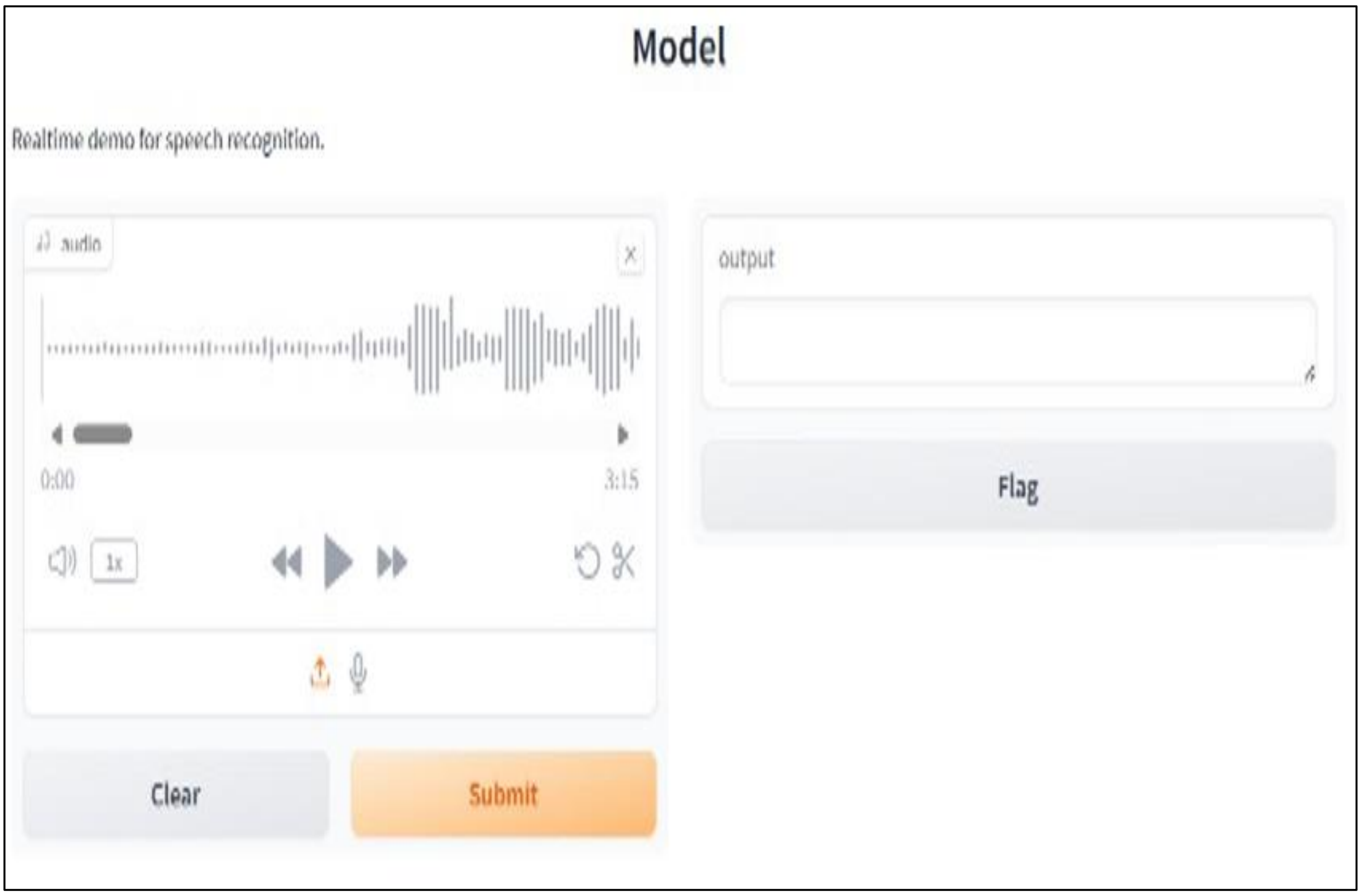
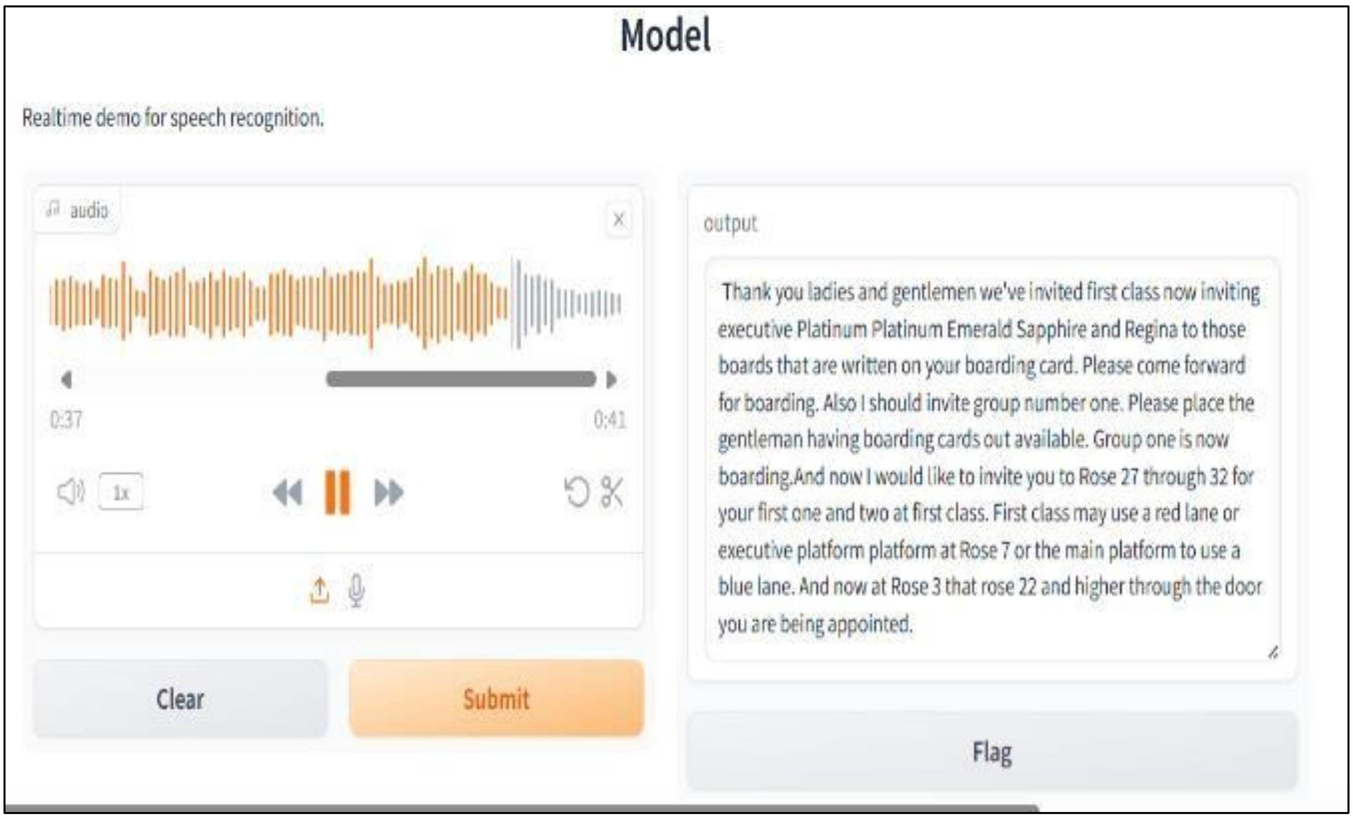
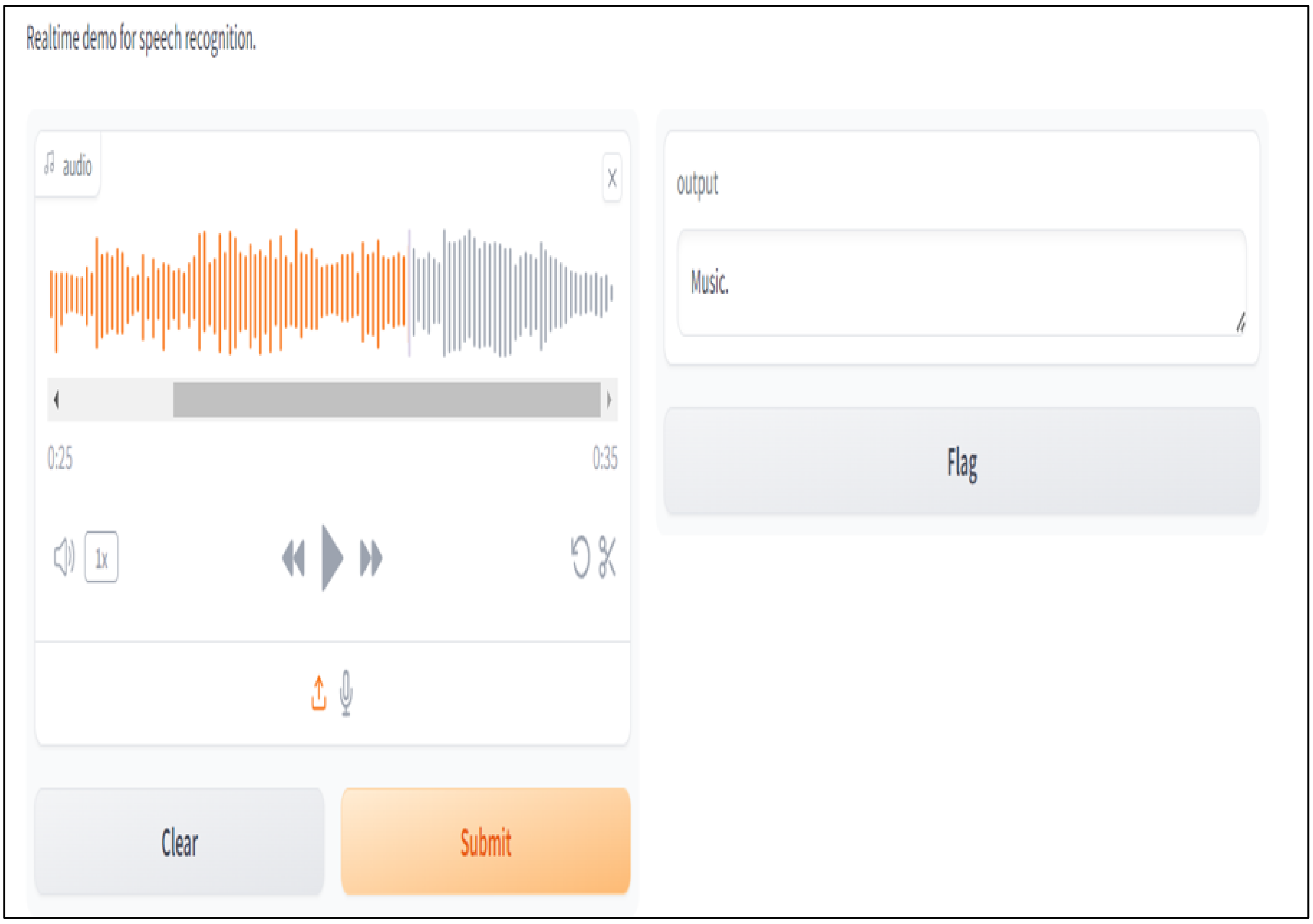


Fig 18 Output Screen



Thank you ladies and gentlemen we've invited first class now inviting executive Platinum Platinum Emerald Sapphire and Regina to those boards that are written on your boarding card. Please come forward for boarding. Also I should invite group number one. Please place the gentleman having boarding cards out available. Group one is now boarding.And now I would like to invite you to Rose 27 through 32 for your first one and two at first class. First class may use a red lane or executive platform platform at Rose 7 or the main platform to use a blue lane. And now at Rose 3 that rose 22 and higher through the door you are being appointed.

Fig 19 Output Screen



Music.

Fig 20 Realtime Demo for Speech Recognition

CHAPTER SIX
SYSTEM TESTING

➢ *Introduction For Testing*

To test a system that enhances automatic speech recognition (ASR) with contextual understanding using natural language processing (NLP), a systematic approach should be designed to evaluate the system's performance in real-world conditions. Here's a step-by-step guide for conducting such a system test:

- *Test Objectives*

✓ *Accuracy Improvement:*
Measure the improvement in transcription accuracywhen context is considered.

✓ *Contextual Understanding:*
Evaluate how well the system understands anduses context to resolve ambiguities.

✓ *Adaptability*:
Assess how the system handles various speakers, accents,environments, and noisy conditions.

- *Test Setup*

✓ *Baseline ASR System:*
Have a baseline ASR system without contextualenhancement for comparison.

✓ *Enhanced ASR System:*
The ASR system integrated with NLP techniques forcontextual understanding.

- *Test Dataset*

✓ *Audio Samples:*
Collect or use pre-existing datasets with a variety of speechsamples, including:

▪ Different speakers (age, gender, accents)
▪ Various acoustic environments (quiet, noisy, reverb)
▪ Context-dependent phrases (e.g., homophones, abbreviations, jargon)

✓ *Contextual Metadata:*
Provide contextual clues such as conversation history,speaker profiles, location data, or domain-specific lexicons.

- *Evaluation Metrics*

✓ *Word Error Rate (WER):*
Calculate the WER for both the baseline andenhanced systems. A lower WER indicates better transcription accuracy.

✓ *Contextual Error Rate (CER):*
Focus on context-dependent errors, such asresolving homophones or identifying named entities.

✓ *Latency:*
Measure the processing time to ensure that adding contextualunderstanding does not significantly delay transcription.

✓ *Adaptability:*
Test how well the system adapts to different contexts,including varying speakers, background noise, and jargon.

- *Test Scenarios*

✓ *Controlled Environment:*
Test in a lab setting with clean audio data andclear context.

✓ *Noisy Environment:*
Test in noisy conditions (e.g., streets, public places)with background noise that challenges speech recognition.

- ✓ *Accent/Dialect Variability:*
  Use speech samples from speakers withdifferent accents or dialects and assess whether context helps improve recognition accuracy.

- ✓ *Context Switching:*
  Evaluate how well the system adapts when the topic or context changes mid-conversation (e.g., moving from discussing sports to finance).

- ✓ *Homophones & Ambiguities:*
  Test cases with homophones (e.g., "sea" vs. "see"), abbreviations, or polysemous words to see if the system can resolve them correctly using context.

- *Testing Tools*

- ✓ *Speech Corpus:*
  Use a diverse set of corpora like LibriSpeech, TED-LIUM,or real-world recorded conversations.

- ✓ *Evaluation Software:*
  Use tools like Kaldi, ASR test suites, or custom scriptsto automatically calculate WER and CER.

- *Testing Procedure*

- ✓ *Initial Testing:*
  Run the baseline ASR system on the test dataset and collectmetrics (WER, CER, etc.).

- ✓ *Contextual Enhancement Activation:*
  Activate the enhanced ASR system with contextual understanding powered by NLP.

- ✓ *Comparison:*
  Compare the results (accuracy, error rates, adaptability) of thebaseline system and the enhanced system across various test scenarios.

- ✓ *Edge Cases:*
  Include edge cases where context plays a crucial role, such as domain-specific terminology or phrases that depend heavily on prior conversation history.

- *Analysis and Reporting*

- ✓ *Quantitative Analysis:*
  Compare WER and CER improvements acrossdifferent environments and speakers.

- ✓ *Qualitative Analysis:*
  Analyze specific cases where context improved orfailed to improve recognition, and identify areas for further refinement.

- ✓ *Latency Trade-off:*
  Examine if enhanced understanding adds latency to therecognition process and if this is acceptable for the application domain.

- *System Optimization*
  Based on test results, fine-tune:

- ✓ *Language Models:*
  Incorporate domain-specific datasets to improve contextaccuracy.

- ✓ *NLP Pipelines:*
  Optimize the NLP components for faster processing withoutsacrificing understanding.

- ✓ *Adaptive Algorithms:*
  Improve adaptability to different speakers, accents,and noisy environments.

- *Real-World Testing*
  After lab-based testing, deploy the system in real-world environments withreal users to gather performance data under

everyday conditions.

➢ *Various Testcase Scenarios*

Table 1 Test Cases

Testcase ID	Description	Test CaseSteps	Test data	Expected result	Actual result	Status
TC01	Contextual Disambiguation: Recognize the meaning of "bank" based onfinancial context.	1. Speak "I need to go tothe bank." 2. Engage in conversationabout financial matters.	"Bank" in financial context.	ASR recognizes "bank" as a financial institution	ASR recognizes "bank" as a financial institution	Success
TC02	Domain- Specific Vocabulary: Recognize specialized medical terms	1. Speak medical terms like "angioplasty"and "cardiologist.	Medical terms like "angioplasty," "cardiologist."	ASR correctly recognizes medical terms.	(To be filled aftertesting)	Success
TC03	Speech Repair:Handle self- corrections in speech.	1. Say "I went to the...no, I stayed home."	Spoken self-correction.	ASR transcribes "Istayed home."	(To be filled aftertesting)	Success
TC04	Named Entity Recognition: Identify named entities like locations and dates.	Say "Book a flight to New York on December 5."	"New York" as location, "December 5" as date.	ASR recognizes "New York" as a location and "December 5" as adate.	(To be filled after testing)	Success
TC05	Pronoun Resolution: Correctly resolve pronouns basedon earlier context.	1. Mention "John" earlier in the conversation. Say "He will arrive at 5 PM."	Mention of "John" in earlier context.	ASR resolves "he" asreferring to "John."	(To be filled aftertesting)	Success
TC06	Code- Switching: Recognize multilingualspeech correctly.	2. 1. Speak partof a sentencein English and switch toSpanish mid-sentence. E.g., "Can you send thefile mañana?"	Mixed- language sentence.	ASR transcribes bothEnglish and Spanish accurately.	(To be filled aftertesting)	Success
TC07	Adaptation toAccents: Handle different accents correctly.	3. 1. Speak witha strong regional accent, e.g., Indian or Scottish accent	Speech witha regional accent.	ASR adjusts to regional variationsand transcribes correctly.	(To be filled aftertesting)	Success
TC08	Emotion Detection: Recognize the emotional toneof speech.	4. 1. Speak withan excited tone, e.g., "I'm so happy today!"	Speech with emotional tone (e.g., happy, sad).	ASR identifies theemotion (e.g., excitement) alongwith the correct transcription.	(To be filled aftertesting)	Success
TC09	Filler Word Detection: Remove fillerwords like "um," "uh."	5. 1. Say "Um, I think it's... uh, maybe tomorrow?"	Speech withfiller words.	ASR removes fillerwords and transcribes "I thinkit's maybe tomorrow."	(To be filled aftertesting)	Success
TC10	Context-Aware Summarization: Summarize a long speech.	6. 1. Speak for an extended period on a topic. 2. Request ASRfor a summary.	Extended conversation.	ASR provides a concise summary ofthe conversation	(To be filled aftertesting)	Success

CHAPTER SEVEN
CONCLUSION AND FUTURE ENHANCEMENT

➢ *Project Conclusion:*

Enhancing ASR with contextual understanding through NLP significantly improves speech recognition accuracy by incorporating domain knowledge, user intent, and context. This integration resolves ambiguities, improves handling of homophones, and adapts to various environments, ultimately making ASR systems more reliable and efficient in real-world applications across diversedomains.

➢ *Future Enhancement:*

Future enhancements in ASR with contextual understanding using NLP could include deep integration of AI-driven contextual models that adapt dynamically tothe speaker's intent, environment, and conversation history. These models will leverage domain-specific knowledge, personalized user data, and advanced semantic analysis to drastically improve transcription accuracy in real-time.

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