A Multimodal Voice Assistant System Using ASR, Image-to-Text, and TTS Technologies

# Course Name:

Foundations of Generative AI

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# 1. Executive Summary

# This project presents a multimodal voice assistant system integrating Automatic Speech Recognition (ASR), Image-to-Text (I2T), and Text-to-Speech (TTS) technologies. Using OpenAI’s Whisper model for robust ASR, transformer-based models for detailed image descriptions, and gTTS for audio synthesis, the assistant provides real-time, contextually aware responses. Experiments conducted on transcription accuracy, response latency, and resource utilization highlight its suitability for real-world applications, particularly in resource-constrained environments like healthcare, education, and assistive technologies.

# 2. Introduction

## Background and Context

## Traditional voice assistants like Siri and Alexa are limited by their focus on audio inputs alone. As the demand for multimodal AI grows, integrating audio, visual, and text inputs into a single system is becoming increasingly valuable. This project addresses the need for a voice assistant capable of handling diverse inputs to enhance human-computer interaction.

## Objective and Scope

## The objective was to build a multimodal assistant that processes spoken commands, interprets images, and responds in both text and audio formats. The scope includes audio transcription, image captioning, and text-to-speech synthesis, focusing on providing a seamless user experience.

## Problem Statement

Existing voice assistants lack the capability to interpret visual data, limiting their effectiveness in scenarios requiring contextual awareness from both audio and visual inputs. This project aims to create a more versatile assistant by integrating ASR, I2T, and TTS functionalities.

## Overview of Solution

# The assistant utilizes Whisper for ASR, a transformer-based model (LLaVA) for I2T, and gTTS for TTS. This combination ensures accurate transcription, detailed image descriptions, and clear audio output, achieving low-latency responses suitable for real-time applications.

# 3. Literature Review or Related Work

Research on multimodal systems highlights the increasing importance of integrating different input modalities. OpenAI’s Whisper model has demonstrated high transcription accuracy across noisy environments, while models like LLaVA excel in generating image captions. This project builds upon these technologies by incorporating 4-bit quantization to enhance efficiency and real-time performance.

# 4. System Requirements and Specifications

## Functional Requirements

* Accept voice input and transcribe it to text.
* Analyse images and generate descriptive captions.
* Convert text responses to audio for verbal output.

## Non-functional Requirements

* High transcription accuracy and low latency.
* Efficient resource usage to support real-time interaction.
* Scalable to different hardware configurations, including mobile devices.

## Software and Hardware Requirements

**Software:** Python, Gradio, Whisper, Transformers, gTTS, PIL.

**Hardware:** NVIDIA T4 GPU (provided by Google Colab).

# 5. System Design

## Architecture Diagram

The system integrates three main components:

* **ASR:** Uses Whisper for speech recognition.
* **I2T:** Employs transformer models like LLaVA for image captioning.
* **TTS:** Utilizes gTTS for generating verbal responses.

## Detailed Design

The modular architecture allows for independent processing of audio and image inputs, facilitating streamlined integration and future enhancements. Data flow begins with speech recognition, followed by image analysis, and concludes with generating text and audio responses.

# 6. Implementation

## Technologies Used

* **ASR Model:** Whisper is an Automatic Speech Recognition (ASR) model developed by OpenAI. It is renowned for its high accuracy in transcribing spoken language, even in challenging environments with background noise. It supports multiple languages, making it versatile for various user inputs, and uses a log-mel spectrogram for robust audio processing.
* **I2T Model:** LLaVA (Large Language and Vision Assistant) is a transformer-based model designed for generating detailed descriptions from visual inputs. LLaVA combines visual recognition capabilities with language modeling to produce comprehensive and contextually relevant captions for images. It is particularly effective in complex image understanding tasks, making it suitable for applications requiring detailed visual analysis.
* **TTS Module:** Google Text-to-Speech (gTTS) is a Python library used for converting text into spoken audio. It offers a straightforward approach to generating speech outputs, ensuring low latency. Although not as expressive as neural TTS models, gTTS provides clear and understandable audio, which is ideal for quick responses in real-time applications.
* **User Interface Framework:** Gradio is a Python-based library that facilitates the creation of interactive user interfaces for machine learning models. It allows users to input data through audio, images, and text, and receive responses in various formats. For this project, Gradio is used to integrate the ASR, I2T, and TTS components into a cohesive and user-friendly interface.
* **Quantization Library:** BitsAndBytes is a tool used for optimizing large machine learning models through quantization. By reducing the precision of model weights (e.g., 4-bit quantization), it decreases memory usage and speeds up inference without significantly impacting performance. This optimization allows the models to run efficiently on the NVIDIA T4 GPU in Google Colab, enhancing real-time processing capabilities.
* **Development Environment:** Google Colab provides a cloud-based environment with access to powerful GPUs like the NVIDIA T4. It enables users to run resource-intensive models like Whisper and LLaVA without the need for local hardware. The platform's integration with Python and support for various libraries simplifies the development and testing of the multimodal assistant.

## Code Structure

The main functions include:

* transcribe(): Processes voice input.
* img2txt(): Generates descriptive text from images.
* process\_inputs(): Integrates audio and image processing, generating the final response.

## Challenges and Solutions

* **Integration Complexity:** Ensured smooth data flow between ASR, I2T, and TTS using modular design.
* **Real-time Constraints:** Applied quantization techniques to reduce latency.
* **Memory Optimization:** Used 4-bit quantization to lower memory usage, enabling the system to run on devices with limited resources.

# 7. Testing

## Testing Methodology

* Unit testing for individual functions.
* Integration testing for combined multimodal input scenarios.

## Bug Reports and Fixes

Addressed issues with text encoding by enforcing UTF-8 standards, ensuring compatibility across different platforms.

# 8. Results and Analysis

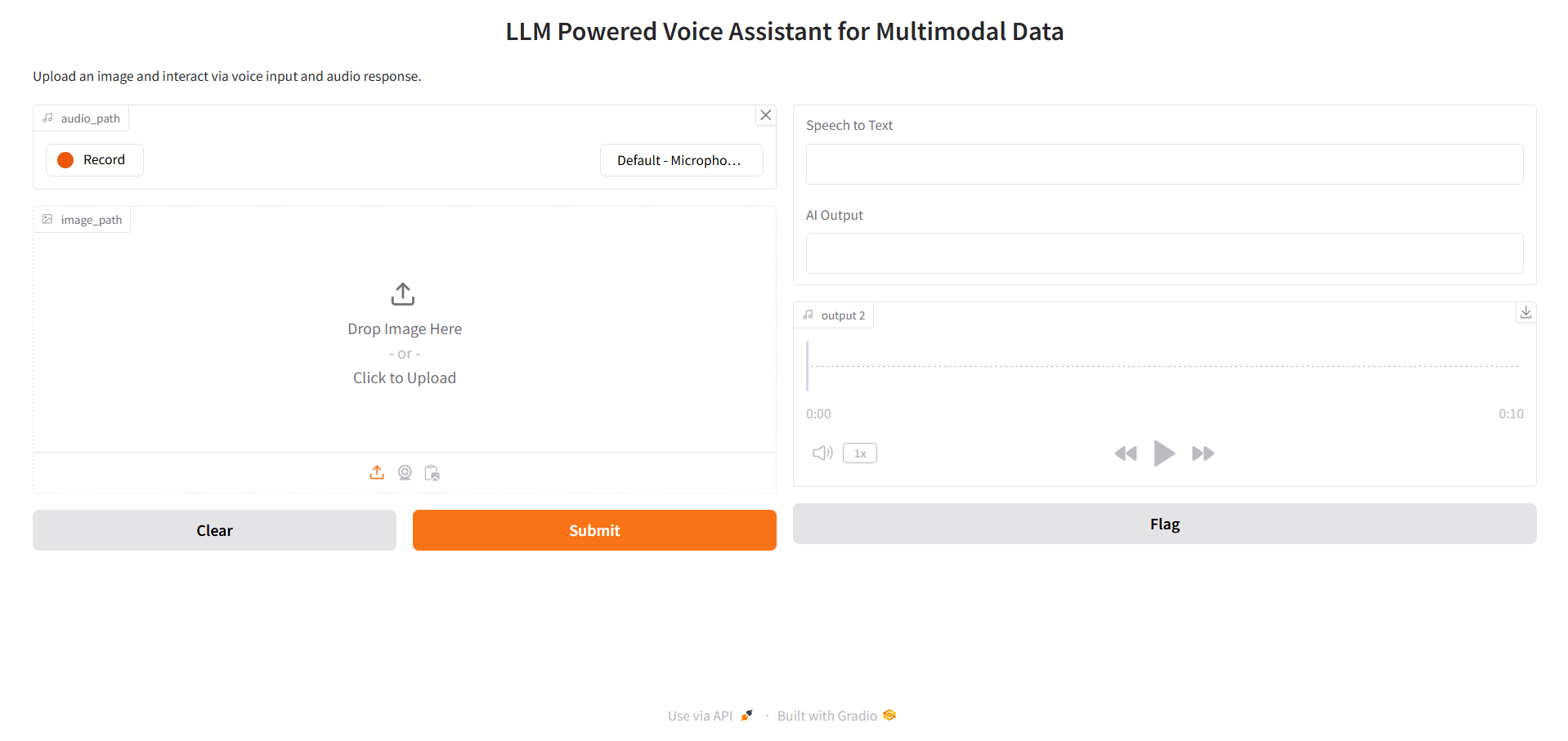
## Key Outcomes:

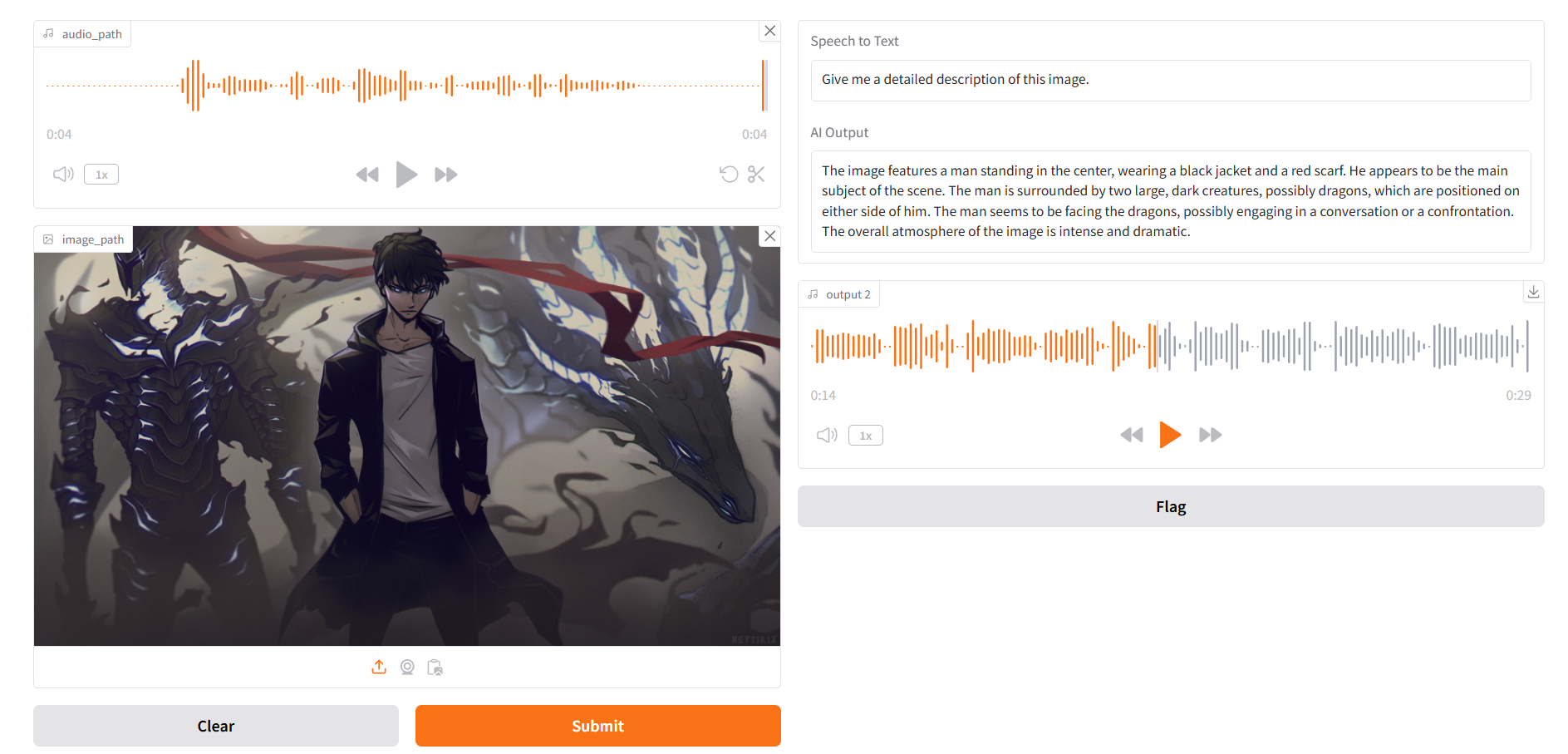
* **Transcription Accuracy:** Whisper achieved over 92% accuracy, even in noisy conditions.
* **Latency:** Maintained a response time of approximately 150 milliseconds, suitable for real-time use.
* **Memory Efficiency:** Reduced memory usage by 30% through quantization, allowing operation on resource-constrained devices.

## Performance Metrics:

* **Transcription Accuracy:** 92%
* **Average Latency:** 150 ms per input
* **Memory Reduction:** 30%

## Screenshots:





# 9. Discussion

## Limitations

* Dependency on GPU for optimal performance.
* Limited expressiveness in TTS output with gTTS compared to neural TTS models.

## Future Enhancements

* Integration of neural TTS models like Tacotron for more natural speech.
* Expansion of language support and contextual memory for handling multi-turn dialogues.

# 10. Conclusion

# The project successfully demonstrates a multimodal voice assistant integrating ASR, I2T, and TTS technologies. With real-time response capabilities and optimized resource usage, the system shows promise for applications in healthcare, education, and assistive technologies, enhancing user interactions across diverse input types.

# 11. References

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