# Enjoyable - a multi-contexts and real-time audio description method on YouTube videos for the Visual Impaired

By

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# **ABSTRACT**

Video streaming platforms often lack sufficient accessibility features for visually impaired users, as generating audio descriptions (AD) manually is timeintensive and resource-heavy. This project introduces "Enjoyable," an online platform with an automated AD system. Instead of relying on external databases, the system enables content creators to label clustered faces directly within videos, improving character recognition across diverse genres. A structured script-based approach integrates image captions and dialogue, forming a comprehensive movie script. This script is processed by LLM-based Single-Prompt Multiturn Multi-Agent Reasoning System (SMARS), comprising agents—Investigator agent, Visual Validator agent, Context Historian agent, Integrator agent, Audio Describers agent, Syntax Fixers agent, Language Flow Expert agent, Word Count Checker agent, Fact Checker agent, Messenger agent, Target Audience agent—who collaboratively generate personalized ADs. This method enhances accessibility by streamlining visual-auditory data interaction and addressing limitations of existing AD systems.

Area of Study: Humanistic AI

Keywords: Automated Audio Description Generation System, LLM-based Multi Agent System, Multilingual Narration, Single-Prompt Multiturn Multi-Agent Reasoning System, Blind and Visually Impaired People

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# LIST OF ABBREVIATIONS

AD Audio Description

LLM Large Language Model

SSIM Structural Similarity Index Measure

ORB Oriented FAST and rotated BRIEF

TTS Text-to-Speech

STT Speech-to-Text

MAS Multi-Agent System

SMARS Single-Prompt Multiturn Multi-Agent Reasoning System

#### **CHAPTER 1**

# **Project Background**

## 1.1 Introduction

Video streaming platforms have transformed media consumption, granting audiences unparalleled access to diverse content. Despite their immense popularity, these platforms frequently fall short of accommodating blind and visually impaired users, who depend on audio descriptions to interpret visual material. Audio descriptions are verbal narrations that convey crucial visual elements such as actions, settings, and non-verbal cues, thereby enabling visually impaired users to fully immerse themselves in the viewing experience [1].

Currently, prominent platforms like YouTube do not offer native support for audio descriptions, excluding a substantial segment of their audience from fully experiencing the content [2]. Consequently, blind and visually impaired users often find themselves marginalized in accessing mainstream media, educational content, and other videos available on these platforms.

Netflix, a leading subscription-based streaming service, has made some advancements by providing audio descriptions for select content [3]. However, these descriptions are usually limited to one language, restricting accessibility for non-English speakers. Although Netflix's initiatives mark a step forward, the overall availability and comprehensiveness of audio descriptions remain inadequate to effectively serve the global blind and visually impaired community.

To address these shortcomings, independent initiatives like YouDescribe have been developed. YouDescribe is a platform that allows volunteers to generate and share audio descriptions for YouTube videos, creating a valuable resource for visually impaired users [4]. Nevertheless, the platform's reliance on volunteers results in variable coverage and quality, and its dependency on external applications and tools complicates the user experience, as it is not directly integrated into the mainstream streaming services that users typically use.

#### 1.2 Problem Statement and Motivation

The creation of audio descriptions for video content is an intricate, labor-intensive process that typically necessitates manual scripting, narration, and synchronization with the visual aspects of the video. This manual process is time-consuming and incurs significant costs, with prices ranging from \$15 to \$75 per minute of video [5]. Such high costs create a considerable barrier to the widespread adoption of audio descriptions, particularly for smaller content creators and platforms with extensive video libraries.

Due to these challenges, the availability of audio descriptions across most video streaming platforms remains limited. Even on platforms like Netflix, where efforts have been made to include audio descriptions, coverage is usually incomplete; many videos remain undescribed. Furthermore, these descriptions are generally offered in only one language, typically English, which limits accessibility for non-English-speaking users.

The current reliance on a manual approach is not scalable, resulting in inefficient production and distribution of audio descriptions. These inefficiencies contribute to the limited availability of audio-described content, ultimately excluding blind and visually impaired users from fully accessing and enjoying video content. Additionally, the predominance of a single language in the available descriptions, as seen on platforms like Netflix, further exacerbates the problem by not catering to the needs of a diverse, global audience.

These limitations indicate the need for an automated and scalable solution to deliver high-quality audio descriptions across multiple languages, making video content accessible to all users, regardless of their visual or linguistic abilities.

The motivation for this project arises from the escalating need for accessible video content driven by a rapidly increasing demand from blind and visually impaired users. This substantial audience continues to expand, indicative of a broader societal movement toward inclusivity and accessibility within digital media. As video content becomes increasingly prevalent across the internet, it is vital to ensure it remains accessible to all users, including those with visual impairments, making this a critical issue to address.

In Malaysia, approximately 1.2% of the population is blind or visually impaired, highlighting a significant group that requires accessible content to engage in the digital era fully [6]. This underserved demographic not only represents a clear need but also

offers a compelling market opportunity for platforms and services that can effectively cater to their requirements. The significance of the audio description market extends beyond Malaysia to a global scale, as more content creators and platforms realize the importance of catering to a diverse audience.

Additionally, rising awareness and evolving legal mandates concerning accessibility in digital media are accelerating the demand for scalable and cost-effective solutions within the audio description market. Content creators, streaming platforms, and broadcasters actively seek ways to deliver high-quality audio descriptions without the associated high costs. This project aims to address these demands by developing an automated system for generating audio descriptions, providing a solution that is both cost-effective and inclusive. The combination of the opportunity to serve a rapidly growing audience and the societal impact of enhancing accessibility makes this project both timely and compelling.

# 1.3 Project Objectives

The primary objective of this project is to develop a sophisticated, automated system capable of generating accurate and personalized audio descriptions for video content, thereby improving accessibility for users with visual impairments and catering to diverse audience needs.

To achieve this primary objective, the project is divided into the following subobjectives:

- 1. Develop an Automated Audio Description Generation Pipeline: Create a complete system that automates the generation of audio descriptions from video content. This includes integrating modules for I-frame extraction, voice activity detection, face clustering, and detailed image captioning.
- 2. To develop contextually accurate and audience-tailored audio descriptions through a Multi-Agent Personalization Framework, which incorporates audience persona modeling to generate descriptions that are both factually precise and stylistically aligned with the preferences and cognitive needs of the target user group.
- 3. To design a systematic evaluation system to methodically measures accessibility impact to the visual impaired communities.

# 1.4 Project Scopes and Direction

The scope of this project encompasses the development, implementation, and evaluation of an automated and personalized audio description system aimed at enhancing the accessibility of video content for users, particularly those with visual impairments. The system is designed to be integrated into existing online video platforms, offering content creators a sophisticated tool to generate detailed, context-aware audio descriptions for their videos.

The project involves the creation of multiple interrelated modules, including video frame extraction, face detection, speech-to-text conversion, face clustering, and audio description generation, all of which form a comprehensive pipeline for producing audio descriptions. Additionally, the project will implement a multi-agent system powered by advanced large language models like Gemini, designed to tailor the generated audio descriptions to distinct user personas.

This project is scoped to not only cover the technical development and integration of these tools but also to test and refine the system to ensure its effectiveness across diverse video content types. The project will culminate in a system that can be broadly applied across different content platforms, significantly improving accessibility for a wide range of users.

#### 1.5 Contributions

This project offers several innovative contributions to the field of automated audio description (AD) generation for video content:

- 1. Enhanced Video Recognition through Clustering and Labelling: We introduce a novel method involving the collaboration of content creators to label clustered facial data, thereby improving video recognition capabilities. This approach broadens the system's applicability, enabling it to process a more diverse video types, including lesser-known and unique content, beyond the scope of pre-existing datasets typically sourced from IMDb. Our system is thus versatile and adaptable to various video genres beyond mainstream media.
- 2. Multilingual Support for Global Accessibility: Our solution addresses a significant limitation in existing AD systems by offering support for multiple languages. This feature ensures that audio descriptions can reach a broader global audience, enhancing accessibility for non-English-speaking viewers and expanding the inclusion of diverse audiences.
- 3. Structured Script-Based Description Generation: Unlike traditional methods that generate audio descriptions directly from video frames, our system first constructs a comprehensive movie script. This script includes detailed image captions for each frame and dialogues with character names, serving as a robust foundation for generating more accurate and contextually appropriate audio descriptions.
- 4. Single Prompt, Multi-Turn, Multi-Agent Reasoning System for Tailored Audio Descriptions: We transform traditional multi-agent systems by enabling a collaborative, dynamic, and efficient workflow within a single prompt stream. By integrating multiple specialized agents in a continuous interaction, the system fosters real-time refinement, ensuring the final output is of high quality and coherence. Whether used for dialogue refinement, content generation, or problem-solving, SMARS offers a more fluid and efficient approach to multiagent reasoning compared to traditional methods, streamlining processes while maintaining flexibility and precision.

# 1.6 Report Organization

The report is organized into several chapters to ensure a coherent and systematic presentation of the project. Chapter 1 introduces the background, objectives, scope, and significance of the project. Chapter 2 provides a comprehensive review of related technologies, tools, and existing systems relevant to the development process. Chapter 3 outlines the system design and architecture, detailing how different components interact and function within the overall framework. Chapter 4 focuses on the implementation of key modules, describing the methods and techniques applied to build the system. Chapter 5 presents system testing and evaluation, including both quantitative metrics and qualitative analysis to assess performance and user satisfaction. Chapter 6 discusses the results obtained, along with insightful interpretations and reflections on system effectiveness. Finally, Chapter 7 concludes the report by summarizing key achievements, challenges encountered, and suggesting directions for future work. Supplemental sections such as references and appendices provide additional support and documentation relevant to the project.

#### **CHAPTER 2**

# **Literature Review**

# 2.1 Review of the Technologes

This section outlines the various hardware and software technologies utilized throughout the development of the "Enjoyable" platform — a real-time, AI-powered audio description system designed to enhance video accessibility for visually impaired users. The review includes the hardware platform, operating environment, programming stack, algorithms, and data management choices made to ensure optimal

system performance and integration.

#### 2.1.1 Hardware Platform

The development, training, and testing processes were all executed on a highperformance consumer-grade laptop — HP Victus by HP Laptop 16-e0122AX . The specifications are as follows:

**Processor**: AMD Ryzen 5 5600H with Radeon Graphics

Graphics: NVIDIA GeForce RTX 3050 Laptop GPU, AMD Radeon(TM)

Graphics

**Memory**: 16GB DDR4 RAM

Storage: 512GB NVMe SSD + 1TB TEAM SSD

**Operating System**: Windows 11

The RTX 3050 GPU provided CUDA acceleration for deep learning tasks such as face detection, frame analysis, and model inference. This setup was sufficient to perform real-time processing and batch computations, especially during video segmentation and inference-based captioning tasks.

# 2.1.2 Firmware/OS

The system was built and operated on Windows 11. This modern operating system offers a stable environment for Python development, supports GPU drivers needed for machine learning libraries (e.g., CUDA, cuDNN), and ensures compatibility with key development tools. Driver support for NVIDIA GPU and general USB/HDMI

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video/audio I/O was fully integrated, supporting testing and potential extension to assistive devices.

#### 2.1.3 Database

Due to the non-relational and lightweight nature of the data handled — such as intermediate JSON files, frame metadata, and transcription data — the system did not use a full-scale relational DBMS like MySQL or PostgreSQL. Instead, formats such as:

- **JSON**: For storing semantic metadata (e.g., speaker timestamps, visual descriptions).
- Local Storage : For I-frame images and pre-processed video segments.

This choice was guided by the system's focus on streaming and file-based batch processing rather than structured transactional data operations. For future expansion involving user data, a NoSQL solution such as MongoDB may be considered.

# 2.1.4 Programming Language

The primary programming language used in the system development is **Python 3.10**. Python was chosen for its simplicity, vast ecosystem, and top-tier support for AI, video processing, and computer vision libraries. The following major Python libraries and tools were used:

- **OpenCV**: For video frame extraction and processing
- **PyTorch**: For handling LLM model integration and image captioning
- Whisper by OpenAI : For high-accuracy Speech-to-Text (STT)
- **dlib and mediapipe**: For facial detection and tracking
- **Transformers** (**HuggingFace**): For prompt handling and calling general-purpose language models (e.g., GPT-4 vision APIs)

Python also enabled easy integration between different modules and experimentations with various machine learning workflows.

## 2.1.5 Algorithm

The system leverages a composition of image, audio, and language AI algorithms to form an intelligent multi-modal processing pipeline. Key algorithms include:

- **I-Frame Extraction**: Using video stream indexing to isolate keyframes for visual captioning.
- Face Detection & Recognition: Via CNN-based algorithms (e.g., dlib / deep metric learning) to track speaker continuity and eliminate redundancy.
- Voice Activity Detection (VAD): To detect silent intervals for inserting descriptions.
- **Speech-to-Text** (**STT**) : OpenAI's Whisper model, used for high-accuracy transcription of the spoken content.
- Active Speaker Detection: Lip movement alignment with speech segments to associate dialogue with visually identified faces.
- Image Captioning with GPT-4 Vision: Multi-modal LLM processes key visual scenes and generates descriptions.
- SMARS (Single Prompt Multi-Turn Multi-Agent Reasoning System): An internal multi-agent collaborative reasoning framework simulated within a single prompt using Gemini. Agents perform roles like investigation, rewriting, flow checking, and caption length control per temporal budget.

These algorithms together enable the system to create automated, high-quality, and timed audio descriptions for any video input.

# 2.1.6 Summary of the Technologies Review

In summary, the "Enjoyable" platform utilizes a strong combination of modern hardware, flexible software tooling, and advanced AI algorithms. The selection of Python and state-of-the-art pre-trained models offers the flexibility needed to rapidly iterate and integrate both vision and language tasks. The combination of lightweight data storage formats and GPU-accelerated model execution ensures that the system can perform audio captioning effectively on standard consumer hardware, making it practical for deployment in both institutional and personal use cases. Most critically,

# **CHAPTER 2**

the SMARS reasoning framework gives the system an intelligent, human-like captioning capability without requiring a distributed backend or complex orchestration, simplifying the implementation while enhancing the quality of output.

## 2.2 Previous Works on Audio Description

Audio description is a narrative service specifically created to enhance the accessibility of visual media, including movies, TV shows, live events, and other visual content, for persons who are blind or visually impaired. Narrative commentary is the process of delivering verbal explanations that describe significant visual components, including movements, facial expressions, scenery, and other crucial aspects, during natural pauses in the dialogue. The inclusion of this additional level of description facilitates users in acquiring a deeper understanding of the provided material.

# 2.2.1 Toward Automatic Audio Description Generation for Accessible Videos

The Tiresias system represents a notable advancement in the field of automated audio description (AD) generation, offering a comprehensive and innovative approach to enhancing video content accessibility for visually impaired users. The system is meticulously designed to tackle the various challenges associated with producing high-quality audio descriptions. It is structured around three core modules: Insertion Time Prediction, Audio Description Generation, and Audio Description Optimization [7]. Each of these modules plays an integral role in ensuring that the final audio descriptions are not only precise and contextually relevant but are also seamlessly integrated into the existing audio track of a video.



Figure 2.1 Tiresias consists of three core modules: Insertion Time Prediction, Audio Description Generation, and Audio Description Optimization [7].

This Time Prediction module's primary function is to analyze the video content and precisely determine the optimal moments for inserting audio descriptions. The key objective is to ensure that these insertions do not interfere with the original audio, such as dialogues, significant sound effects, or important background music. Through this intelligent insertion, the system ensures that the accessibility of the video is enhanced without detracting from the overall viewing experience.

To achieve this, the Tiresias system employs sophisticated tools like Katna for keyframe extraction. These tools are adept at detecting and capturing significant visual changes within the video, often signaling important events or transitions that would benefit from an accompanying audio description. By identifying these keyframes, the system effectively segments the video into manageable clips that can be individually analyzed for the optimal placement of audio descriptions.

The system further strengthens its analysis by integrating a deep learning network, which includes two sub-networks dedicated to visual and audio features, respectively. The deep learning model aims to identify any inconsistencies or alignments between these two modalities, ensuring that the system maintains a high level of accuracy in preserving audio-visual consistency. This robust approach allows the system to identify when a visual event is sufficiently significant to warrant a description while also ensuring that the description does not overlap with any crucial audio elements. The result is a refined segmentation process that sets the foundation for generating accurate and contextually aware audio descriptions.

Following the identification of the appropriate insertion points, the next critical task is the creation of the audio descriptions themselves. This responsibility falls on the Audio Description Generation module. The core objective of this module is to produce textual descriptions that accurately and succinctly convey the visual content of the video segments, focusing on essential details such as character actions, significant objects, and scene transitions.

The Tiresias system utilizes an advanced attention-based model to accomplish this task. Central to this model are two critical components: the Sentence Localizer and the Caption Generator. The Sentence Localizer is responsible for pinpointing the most crucial visual events within the segmented video clips, determining what aspects of a scene require description. Rather than approaching this task as simple image captioning, the model employs a cross-attention framework that integrates both video and textual features. The Caption Generator then creates the textual description that aligns with the identified visual events, ensuring that the generated text is both accurate and contextually appropriate.

Training this model necessitates a large dataset rich with detailed event descriptions that enable the system to learn how to align visual cues with textual descriptions properly. The Tiresias system leverages the ActivityNet Captions dataset, a resource specifically curated to provide an expansive set of event descriptions across a broad range of videos. This extensive dataset is crucial to the system's ability to generalize its audio description generation capabilities across different types of video content, from films and documentaries to educational videos and beyond. The combination of an attention-based model with comprehensive training data empowers the system to produce high-quality audio descriptions that are both informative and engaging.

The final module in the Tiresias system is the Audio Description Optimization module, which is essential in refining the generated descriptions to meet the highest standards of relevance, diversity, and grammatical accuracy. This step is indispensable, as even the most accurately generated descriptions may require further refinement to ensure they are as effective and user-friendly as possible.

The optimization process is guided by applying various cost functions, each designed to address a specific aspect of the generated descriptions. For instance, the irrelevance cost function penalizes descriptions that contain unnecessary or redundant information, directing the system toward providing only the most pertinent details. The diversity cost function encourages various language use, preventing repetitive or monotonous descriptions that could reduce listener engagement. Lastly, the perplexity cost function ensures that the descriptions are fluent and grammatically correct, enhancing their readability and comprehensibility.

These cost functions are synthesized into a total cost function, which the system aims to minimize using dynamic programming techniques. Dynamic programming permits the system to efficiently explore several combinations of description options, ultimately selecting the set that best balances relevance, diversity, and fluency. Through this approach, the Tiresias system guarantees that the final audio descriptions are accurate and contextually appropriate and polished and engaging for the listener.

## 2.2.2 AutoAD: Movie Description in Context

The paper titled "AutoAD: Movie Description in Context" marks a transformative milestone in the field of automatic audio description (AD) generation, fundamentally improving the way descriptions are crafted for movies. Recognizing that the intricate narrative arcs and complex character interactions within films require more than just basic frame-by-frame analysis, AutoAD implemented a sophisticated strategy of contextual integration that combines multiple layers of information—such as temporal sequences, dialogue, and subtitles—to generate audio descriptions that are not only seamless and coherent but also well-aligned with the overall narrative framework of the film.

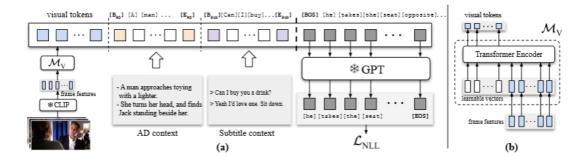


Figure 2.2 (a) Overview of AutoAD (b) Detail of the visual mapping network [8].

The contextual Integration technique is designed to align the generated audio descriptions with the unfolding narrative of the movie. The model effectively leverages numerous types of context to ensure that ADs not only reflect the visual content but also the underlying story that evolves throughout the film.

AutoAD recognizes the importance of remembering and referencing past events, characters, and descriptions generated in earlier scenes. Films often involve intricate narratives where elements introduced early on reappear or gain significance later in the story. AutoAD captures this complexity by maintaining a temporal memory of previously generated descriptions, which it can then reference when producing new ones. For instance, if a character introduced as a minor player early on becomes central to the plot later, AutoAD will recall this character's past and ensure that new descriptions acknowledge their growing importance. This continuity enriches the viewing experience for visually impaired audiences, allowing them to follow the story more holistically.

Understanding dialogue is crucial in filmmaking, as it often conveys key narrative elements, reveals character motivations, and builds emotional tension. AutoAD incorporates dialogue context into its AD generation pipeline by extracting and analyzing dialogue from subtitles or transcriptions. This enables the model to generate character-aware descriptions relevant to ongoing spoken interactions. For example, if a character makes an ominous remark, AutoAD might generate a description reflecting the following dark mood, perhaps noting the sudden dimming of a room or a close-up of a character's face. By integrating dialogue into the descriptive process, AutoAD enhances the story's coherence and immersion.

Subtitles provide critical cues about dialogue and other non-verbal audio elements—such as door slams, footsteps, or shifts in background music—essential for advancing the plot. AutoAD leverages these cues to synchronize ADs with the film's narrative flow, ensuring that descriptions enhance rather than disrupt critical audio moments. For instance, the model can predict a significant event, like the buildup of tense music before a dramatic scene, and produce a description that aligns with this anticipation. This careful timing allows the viewer to experience the film's pacing and mood as if they were visually following it.

One of the perennial challenges in AD generation is the scarcity of extensive, annotated datasets that are specifically tailored for this task. To circumvent this, AutoAD adopts an innovative Data Augmentation strategy that involves pretraining the model on large-scale datasets, which, while not directly related to movie AD, provide essential context and narrative data.

AutoAD leverages text-only datasets, which consist of rich narrative descriptions that lack corresponding visual frames. These datasets, including items like screenplay plots, summaries, or closed captions, are invaluable for training the text generation component of AutoAD—typically, a GPT-based model. Although these datasets don't contain visual data, they offer a broad understanding of narrative structures and linguistic conventions, enabling the model to produce ADs that seamlessly integrate with the movie's dialogue and narrative flow.

On the visual side, the model is pretrained using visual captioning datasets that match images or short video clips with descriptive captions. While these datasets may

not possess the complex temporal dynamics found in films, they are instrumental in teaching the model how to associate visual cues with descriptive language. This visual-textual alignment is later finely tuned when the model is trained on more context-rich movie data, ensuring that AutoAD generates descriptions that are both visually and contextually appropriate.

In addition to data and context integration, AutoAD introduces another significant innovation through Model Enhancement via lightweight adapters [8]. These adapters serve as intermediary modules that connect the outputs of pretrained foundation models—such as GPT for text and CLIP for vision—with the specific contextual needs of AD generation. These adapters refine the generated descriptions by embedding relevant context, such as character information or narrative progression, directly into the text. For instance, if a scene involves a character whose importance has recently increased, the adapter can ensure that this shift is reflected accurately in the AD, preserving narrative consistency and enhancing viewer comprehension.

These adapters also play a critical role in merging various forms of context—temporal, dialogue, and subtitle—into the AD process. This ensures that generated descriptions are well-informed by the ongoing narrative and character development, rather than being isolated observations. For example, if a character's emotional state has shifted significantly, the adapter can adjust the AD to reflect this, thus producing a more coherent and impactful narrative for the viewer.

"AutoAD: Movie Description in Context" represents a significant leap in the evolution of automatic audio description technology. Through its focus on Contextual Integration, Data Augmentation, and Model Enhancement, AutoAD addresses the limitations of earlier models and establishes a new benchmark for AD quality in cinematic contexts.

# 2.2.3 AutoAD II: The Sequel – Who, When, and What in Movie Audio Description

AutoAD II is thoughtfully engineered to tackle three essential challenges in movie AD: accurately identifying characters (Who), determining the optimal moments for description insertion without interrupting dialogue (When), and generating contextually rich and meaningful descriptions (What) [9].

AutoAD II is built upon the understanding that successful audio description for films demands a profound grasp of the visual content, auditory cues, and the underlying narrative structure. Creating automatic AD for movies presents a unique challenge as it requires intricate management of narrative complexities. These include accurately identifying characters, deciding the precise timing for inserting descriptions, and ensuring that those descriptions capture the essence of visual content without disrupting dialogue or other critical audio elements.

One of the most crucial components in movie audio description is the precise identification and naming of characters. For visually impaired viewers, knowing the names of characters or understanding their roles in various scenes is vital for following the narrative and tracking who is performing what actions.

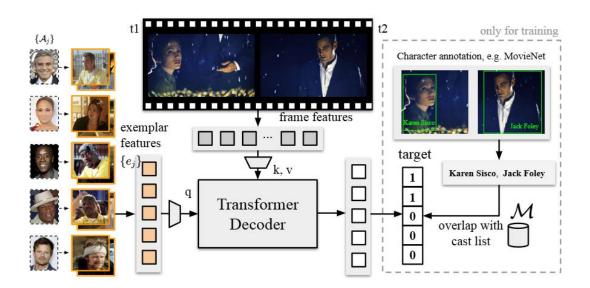


Figure 2.3 Character Recognition Module [9].

AutoAD II addresses this by introducing a Character Bank, specifically designed to link character names with their respective visual and auditory representations within the film. This bank is meticulously constructed for the principal cast in each movie and includes:

- Character Names: The actual names of characters that are essential for accuracy in audio descriptions.
- Character Faces: Visual features derived from CLIP (Contrastive Language-Image Pretraining), which can differentiate and recognize characters based on their appearances across scenes.
- **Actor Information:** Distinctive facial features and voice identifiers associated with the actor portraying the character.

Through this Character Bank, AutoAD II ensures accurate referencing of characters throughout the film. As scenes evolve, the model uses the Character Bank to correctly label individuals, thereby preserving narrative continuity and improving the viewer's comprehension of the story.

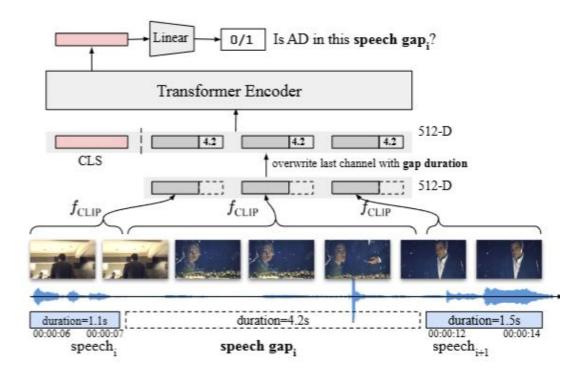


Figure 2.4 Architecture for AD temporal proposal classification [9].

The timing of audio descriptions is as critical as the content itself. Insertions should occur during moments of low or no dialogue to prevent confusion or dilution of the impact of both the description and the dialogue. However, the rapid pace and dense

dialogue sequences in movies often make it challenging to find appropriate gaps for insertion. AutoAD II employs an advanced temporal model that assesses the auditory landscape of each scene. Through Automatic Speech Recognition (ASR), the system identifies exact intervals in scenes where no dialogue is present or where the dialogue is minimal and non-essential. These intervals are classified as opportunities for optimal AD insertion.

The system does not stop at identifying available time slots; it also evaluates visual importance during these intervals. If a significant visual event occurs during a dialogue-free moment, that time slot is prioritized for AD insertion. This process is managed by a transformer-based model capable of considering long-range dependencies and contextual cues from the movie.

A binary classification model, informed by visual features from CLIP and text derived from ASR, is applied to each temporal interval [9]. This model predicts whether an AD should be generated based on the visual content and available time slots. Furthermore, a sliding window technique is used to progressively evaluate the entire timeline of the movie. This ensures that descriptions are inserted when most appropriate, thereby maintaining the rhythm and coherence of the narrative.

The final crucial pillar of AutoAD II is deciding what exactly should be described. Knowing when and who is not enough; the system must also determine what details are important enough for the description. AutoAD II utilizes a vision-language model augmented with cross-attention mechanisms to generate high-quality descriptions. Cross-attention permits the model to incorporate both visual cues from the movie and contextual insights derived from previous scenes, ensuring that the descriptions are not only accurate but also significant to the overall narrative.

AutoAD II: The Sequel represents a significant leap in the generation of audio descriptions for movies, offering comprehensive solutions to the Who, When, and What of AD. By leveraging state-of-the-art machine learning, natural language processing, and computer vision technologies, AutoAD II not only accurately identifies and names characters but also precisely times and tailors the content of descriptions, ensuring seamless integration into films.

## 2.2.4 AutoAD III: The Prequel – Back to the Pixels

AutoAD III introduces meticulously curated datasets designed to bridge the existing gap between visual content and corresponding audio descriptions, thereby addressing long-standing challenges such as the lack of high-quality paired video-AD data and the difficulties in temporal alignment [10].

The CMD-AD (Character-Movie-Description) dataset emerges as a pivotal development. By combining audio descriptions from the AudioVault repository with video clips from the Combined Movie Dataset (CMD), AutoAD III does not simply pair descriptions with video clips—it demands precise temporal alignment between the ADs and the exact scenes they describe. Temporal alignment is particularly challenging due to the differences in timing and pacing found across various movies. AutoAD III overcomes this hurdle by employing advanced synchronization algorithms that ensure the ADs are accurately coordinated with the relevant movie events. Unlike earlier datasets that provided only frame-level features, the CMD-AD dataset offers a more holistic perspective by encapsulating entire scenes and integrating character information directly into the dataset.

In addition to CMD-AD, the project introduces the HowTo-AD dataset, which repurposes the extensive HowTo100M video dataset. Although HowTo100M is originally composed of narrated instructional videos rather than designed specifically for AD tasks, it contains a vast amount of spoken and visual information that can be harnessed for AD-like applications. By employing language models, AutoAD III converts these instructional narratives into pseudo-audio descriptions. This innovative repurposing effectively addresses the challenge of data scarcity by providing a rich source of training data that includes both temporal sequences and narrative context. The HowTo-AD dataset is particularly valuable as a pseudo-ground-truth label generator, enabling the model to learn from a broad base of examples even when manually annotated ADs are unavailable. These pseudo-labels, while imperfect, offer foundational insights into describing actions, objects, and sequences within a time-dependent context, and these insights are subsequently refined during the fine-tuning process using more accurately labeled data.

Beyond the advancements made in data, the model architecture in AutoAD III also represents a significant leap forward. The architecture is centered around a Q-

former-based design, building upon methods informed by the BLIP-2 (Bootstrapping Language-Image Pretraining) approach. The primary function of the Q-former is to enhance the integration between visual and textual information, ensuring that the resulting audio descriptions are context-sensitive and deeply relevant.

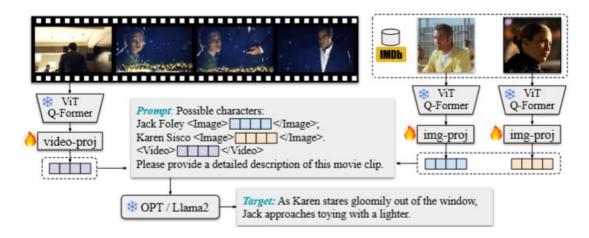


Figure 2.5 Architecture of AutoAD III [10].

One of the key innovations is the incorporation of multi-frame video clips into the AD generation process. Traditional methods often struggle with understanding video content that evolves over time because they typically focus on single-frame analysis. However, AutoAD III introduces a multi-frame approach that allows the model to analyze multiple frames simultaneously. This enables the system to capture the temporal dynamics of scenes—how actions develop, how emotions shift, and how the focus transitions from one character to another. By reflecting the temporal evolution of events, the descriptions generated by AutoAD III are richer and more faithful to the dynamic nature of visual storytelling in movies.

Another innovative aspect is the integration of a Character Bank into the model's architecture. The Character Bank serves as a repository of character-specific information, including names, visual features extracted from CLIP, and textual associations. Each time a character appears in a scene, the Q-former leverages the Character Bank to ensure that the description accurately identifies the character by name. Additionally, it can provide further information such as their role in the scene or their relationship to other characters. This character-awareness adds depth to the audio descriptions, enabling visually impaired viewers to maintain a clearer understanding of the narrative as it unfolds.

The Q-former mechanism itself acts as a vital bridge between the vision and language models. It effectively queries the visual model with textual prompts derived from the language model's understanding of the narrative context and refines the output to align with what is visually present in the scene. This approach addresses the traditional gap between visual recognition and language generation, resulting in more precise and contextually relevant audio descriptions. The Q-former mechanism thus ensures that the generated descriptions not only accurately capture what is happening but also fit seamlessly within the narrative, preserving the film's pacing and emotional tone.

Recognizing the limitations of traditional evaluation methods, AutoAD III introduces a new set of evaluation metrics aimed at better assessing the quality of generated ADs, with particular attention to character identification and semantic appropriateness.

A standout metric introduced in AutoAD III is the Character Reference Informed Textual Information Content (CRITIC) metric. CRITIC focuses on the accuracy of character identification, which is crucial in narratives where understanding "who is doing what" is essential to the storyline. This metric evaluates how well the descriptions integrate character names and relational information, ensuring that the ADs are not only descriptive but also narratively coherent. This is especially important in films where characters and their actions are central to the plot.

In addition, AutoAD III incorporates LLM-based (Large Language Model-based) evaluations that utilize advanced language models to assess the semantic coherence and overall quality of the generated text. Given that these large language models have been trained on vast amounts of textual data, they can act as a proxy for human judgment, evaluating whether the ADs accurately reflect the nuances of the scenes and align with expected language use. The LLM-based evaluation examines factors such as grammar, fluency, and contextual alignment, providing a more nuanced and human-aligned measure of AD quality. The combination of these metrics not only enhances the reliability of the evaluation process but also ensures that the model is fine-tuned to produce high-quality, contextually appropriate descriptions that resonate with visually impaired audiences.

In conclusion, "AutoAD III: The Prequel – Back to the Pixels" stands as a significant milestone in the progression of automatic audio description systems. It offers sophisticated solutions to some of the field's most persistent challenges through its introduction of new datasets like CMD-AD and HowTo-AD, its Q-former-based model architecture, and its implementation of advanced evaluation metrics such as CRITIC and LLM-based assessments.

## 2.3 Limitations of Existing Works

The development of automatic audio description (AD) systems has seen notable progress, yet several limitations persist, particularly when it comes to contextual generalization, character recognition, and traditional frame-by-frame generation methods. These challenges highlight the complexities involved in creating effective and accurate descriptions that cater to a wide variety of films, genres, and cultural contexts, while providing a seamless viewing experience for visually impaired audiences.

One of the most pressing limitations of current AD systems is their struggle to effectively generalize across diverse genres, directorial styles, and cultural contexts. While these models often perform adequately for mainstream genres—such as action or romance films, where visual cues and narrative structures align with established cinematic conventions—they tend to falter when faced with more niche or unconventional content, such as avant-garde or experimental cinema, as well as animated films. Take, for instance, avant-garde films, which prioritize visual symbolism and non-linear narrative structures. These require a deeper understanding of subtle visual motifs and abstract storytelling techniques, which standard AD models, often trained on more conventional content, may fail to capture. As a result, these systems may produce audio descriptions that do not align with the director's intent or comprehend the thematic elements underlying the visual content.

Cultural context further complicates the task, as the cultural background of a film plays a significant role in its visual language, symbolism, and narrative pacing. For instance, films created outside of Western or English-centric traditions might employ unique visual forms or symbolism that AD models, primarily trained on such Western datasets, fail to recognize or convey accurately. This cultural myopia can lead to descriptions that overlook critical narrative elements or, worse, misrepresent the cultural subtleties embedded within the story.

Moreover, directorial style is another hurdle. Directors with distinctive stylistic signatures, such as Quentin Tarantino's non-linear storytelling, Wes Anderson's symmetrical framing, or Terrence Malick's poetic imagery, present unique challenges for AD systems. These styles rely heavily on specific visual motifs and narrative techniques. Without fine-tuning to recognize and adapt to these unique approaches, AD models risk producing generic descriptions, thereby obscuring the stylistic intentions

that are fundamental to the film's narrative. For example, a description that fails to recognize the significance of visual repetition in a narrative might weaken the viewer's understanding of the film's deeper meanings.

Character recognition and consistent identification within an audio description are essential for maintaining narrative clarity, particularly in films with large casts, complex character dynamics, or characters undergoing transformation. Traditional AD systems like AutoAD, AutoAD II, and AutoAD III rely on Character Banks sourced from IMDb—a repository linking character names to their visual and auditory characteristics [8]-[10]. While this represents a step forward, it introduces new challenges related to the creation and maintenance of these datasets. The effectiveness of a Character Bank hinges entirely on the quality, accuracy, and comprehensiveness of the data it contains. Since these systems depend on IMDb, they cannot recognize characters if the videos are not films or TV shows listed in that database. If the dataset is incomplete or inaccurate, or if the content falls outside of IMDb's scope, the system might fail to correctly identify characters, leading to confusion or incorrect descriptions.

Traditional AD generation methods rely heavily on a frame-by-frame analysis, which can introduce several significant limitations that affect the quality and coherence of the produced descriptions. One major drawback is the lack of scene-wide context. By focusing on isolated segments of visual content, frame-by-frame AD generation often ignores the broader context within the scene or the movie as a whole, leading to descriptions that are accurate at a micro-level but fail to capture the larger narrative or thematic elements that give the story its depth and meaning.

For instance, a frame might depict a character smiling, but without understanding the surrounding context—whether the smile is sinister or affectionate—the resulting description could be overly simplistic, misleading, or even contradictory to the storyline. Furthermore, films are constructed through sequences of connected scenes designed to convey a coherent story. Actions or objects in one frame are often directly connected to events in previous or forthcoming frames, meaning that ADs generated in isolation may miss these crucial narrative links, leading to fragmented or disjointed descriptions. This disconnection is particularly detrimental in critical scenes, where the buildup of tension or the emotional significance of a moment spans multiple frames or shots.

In addition, the use of visual motifs, symbolic imagery, and recurring elements that contribute to the thematic depth of a story can also be missed when relying solely on frame-by-frame analysis. Such motifs often only reveal their significance when viewed in sequence with other frames, meaning that a description limited to a single frame may overlook essential thematic elements, resulting in a shallower and less immersive viewing experience for the audience.

Lastly, traditional frame-by-frame methods may fall short in aligning the timing of descriptions with the film's pacing and rhythm. In action sequences, for instance, where rapid-fire ADs are necessary to match the fast-moving visuals, frame-by-frame analysis may lead to delayed or disjointed narration, thereby detracting from the tension and excitement of the scene.

Table 2.1 Summarization of Existing Works.

Paper	<b>Key Modules / Features</b>	Strengths	Limitations
Toward	1.Insertion Time	- Intelligent time	- Primarily trained
Automatic	<b>Prediction</b> : Determines	prediction avoids	on conventional
Audio	optimal moments for	overlap with	datasets like
Description	insertion	critical audio.	ActivityNet, might
Generation	2.Audio Description	- Advanced	struggle with non-
for	Generation: Uses an	attention-based	conventional
Accessible	attention-based model	model for	visual content.
Videos [7]	(Sentence Localizer &	sentence	- Generalization
	Caption Generator).	generation.	across different
	3. Audio Description	- Cost function	cultural or
	Optimization: Uses cost	optimizes fluency	directorial styles
	functions to refine AD for	and relevance	not addressed
	relevance, diversity, and		extensively.
	fluency.		
AutoAD:	1. Contextual	- Strong	- Data
Movie	Integration: Leverages	integration of	augmentation with
Description	temporal sequences,	multi-modal	non-movie-
	dialogues, and subtitles.		specific datasets

in Context	2. Data Augmentation:	contexts (visual,	may compromise
[8]	Pretrained on visual	dialogue).	quality for specific
	captioning datasets and	- Pretraining on	film genres.
	narratives like screenplays.	large datasets	- Context
	3. Model Enhancement	improves	sensitivity may
	via Adapters: Adapters	generalization.	still struggle with
	merge temporal, dialogue,	- Dialogue and	avant-garde or
	and subtitle contexts.	temporal	culturally unique
		continuity help	films.
		narrative	
		coherence	
AutoAD II:	1. Character Recognition:	- Accurate and	- Character Bank
Who,	Character Bank (faces,	coherent character	depends on IMDb
When,	names, actor info) to ensure	identification.	data, limiting
<b>What [9]</b>	correct identification of	- Transformer-	generalizability to
	cast.	based model for	non-film/video
	2. Temporal Insertion	optimal	content.
	Module: ASR-based	description	- May struggle
	dialogue-free interval	timing.	with scenes
	analysis and importance-	- Cross-attention	without easily
	based visual event		classifiable
	prioritization.	contextual	characters or
	3. What to Describe:	accuracy	specialized visual
	Vision-language model		styles
	using cross-attention for		
	description generation		
AutoAD	1. Datasets (CMD-AD,	- Incremental	- Existing dataset
III: Back to	HowTo-AD): Temporal	dataset	sources (like
the Pixels	alignment between ADs	innovations	HowTo-AD) may
[10]	and video content.	(CMD-AD,	not reflect movie-
	2. <b>Q-former-based</b>	HowTo-AD)	specific
	Model: Multi-frame video		complexities.

clip analysis to capture	improve AD-	- Heavier reliance
dynamic narrative	video alignment.	on temporal
evolution.	- Q-former allows	alignment
3. Character Bank:	multi-frame	mechanisms,
Integrated into architecture.	temporal	sensitive to pacing
4. CRITIC Metrics: LLM-	dynamics,	mismatches.
based evaluations and new	enhancing	
metrics for character	narrative	
identification quality	coherence.	
	- CRITIC metric	
	evaluates	
	character	
	accuracy	

#### 2.4 Solution to the Weaknesses

The proposed solution to address the weaknesses identified in existing automatic audio description (AD) systems centers on developing a comprehensive and contextually aware approach to AD generation. This involves enhancing the accuracy of character identification, improving contextual alignment in description generation, and refining the personalization of audio descriptions.

A critical aspect of the proposed solution is the integration of advanced modules for Active Speaker Detection (ASD), face detection, face clustering, and recognition. These components work collaboratively to overcome the limitations associated with character recognition and continuity in narrative comprehension, particularly in complex scenarios featuring multiple characters. The ASD module functions by synchronizing speech timestamps with the corresponding faces detected in the video, ensuring accurate character identification at specific moments. This is further optimized by leveraging tools such as Dlib for face detection and applying clustering algorithms like the Chinese Whispers Algorithm. These algorithms allow the system to differentiate between characters, even in scenes crowded with participants. By accurately identifying, clustering, and labeling faces, the system facilitates seamless

character tracking and annotation throughout the video, addressing inconsistencies and improving descriptive specificity.

To tackle the challenges related to generating contextually relevant descriptions and narrative continuity, the system employs GPT-40 for detailed image captioning. This model translates visual content into rich and meaningful descriptions that consider not just the surface-level visual elements but also the narrative context, character roles, and emotional tones within the scenes. By doing so, GPT-40 goes beyond mere labeling, encapsulating the broader essence of each scene in its captions. These detailed captions, in conjunction with extracted dialogues and contextual annotations, are then compiled into a cohesive and structured movie script by the Movie Script Generation Module. The integration of visual and auditory elements in a single, harmonized narrative ensures that descriptions do not fragment the story but rather enhance comprehension and engagement, thereby addressing the limitations of frame-by-frame AD generation methods.

Recognizing the importance of personalization in enhancing the viewing experience for diverse audiences, the proposed solution incorporates a Multi-Agent System (MAS) driven by large language models (LLMs) like Gemini. This system personalizes content based on detailed audience personas, ensuring that audio descriptions resonate with users' unique cultural, emotional, and demographic backgrounds. Specialized agents within the MAS collaborate to tailor the narrative style, refine linguistic elements, and align content with the specific needs of different user groups. The result is an audio description that not only aligns contextually with the video but also feels personalized and relevant to the intended audience, making the content more accessible and relatable.

#### **CHAPTER 3**

# **System Methodology/Approach**

The outcome of this project is an Automated Audio Description System named "Enjoyable". This system includes a series of advanced modules, such as I-Frame Extraction, Frame Filtering, Voice Activity Detection (VAD), Speech-to-Text (STT), Active Speaker Detection (ASD), Face Detection, Face Clustering, Face Labelling, Character Face Recognition and Annotation, Detailed Image Captioning, Movie Script Generation, and an LLM-Based Multi-Agent System.

# 3.1 System Use Case

The system use case revolves around a video streaming platform integrated with an automated audio description system that enhances video accessibility for visually impaired users. The platform allows users to register, log in, and perform actions such as adding, updating, deleting, and playing video content. Once a video is uploaded, the system initiates its automated processing pipeline. This involves the extraction of I-Frames (keyframes) from the video, followed by frame filtering to eliminate irrelevant or poor-quality frames. Subsequently, the system performs face detection to identify individuals in the video and groups similar faces using face clustering. Detected faces are further labeled, recognized, and annotated for accurate characterization.

To enrich the accessibility features, the system incorporates advanced audio processing techniques. It detects voice activity (VAD) to segregate speech from background noise and converts spoken content into text via speech-to-text (STT) conversion. Active speaker detection identifies which individual is speaking at any moment in the video. The transcribed text is then processed in conjunction with image captions and contextual descriptions to develop a comprehensive narrative through the movie script generation module. The LLM-based Multi-Agent System plays a pivotal role, aligning these tasks to ensure accurate, context-aware audio descriptions. Finally, the generated audio descriptions are converted back into speech and synchronized with the original video during the audio insertion phase, ensuring seamless playback for users.

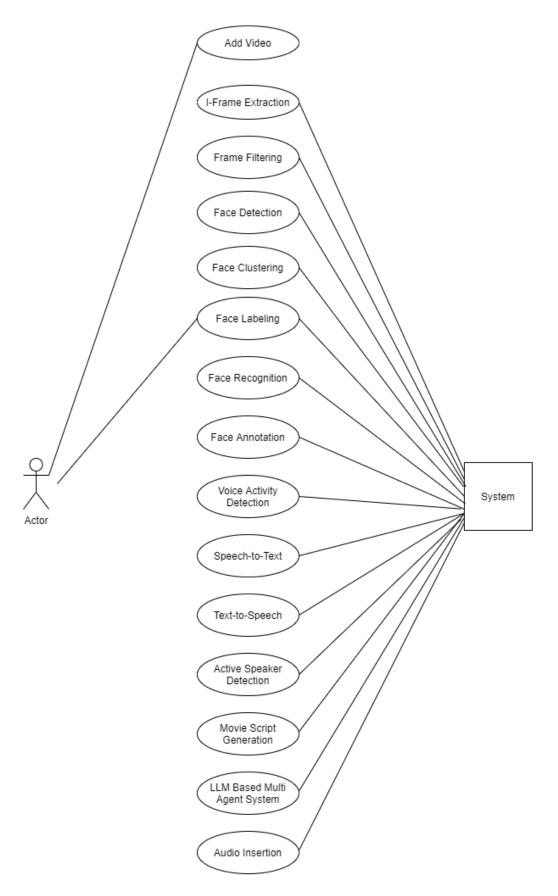


Figure 3.1 System Use Case Diagram.

### 3.2 System Architecture of Automated Audio Description System

The Automated Audio Description System uses Python to automate the creation of detailed audio descriptions for video content. The system is composed of interconnected modules that collaboratively process and analyze video frames and audio, ultimately generating comprehensive and contextually accurate narratives.

Python's libraries and frameworks, such as dlib, OpenCV, and FFmpeg, facilitate key tasks like frame extraction, face detection, face extraction, face recognition, and the integration of descriptive audio. These tools are essential for efficiently handling the complex processes required to produce high-quality audio descriptions.

By utilizing these advanced technologies, the system ensures that the audio descriptions are accurate and accessible, effectively enhancing the viewing experience for individuals with visual impairments.

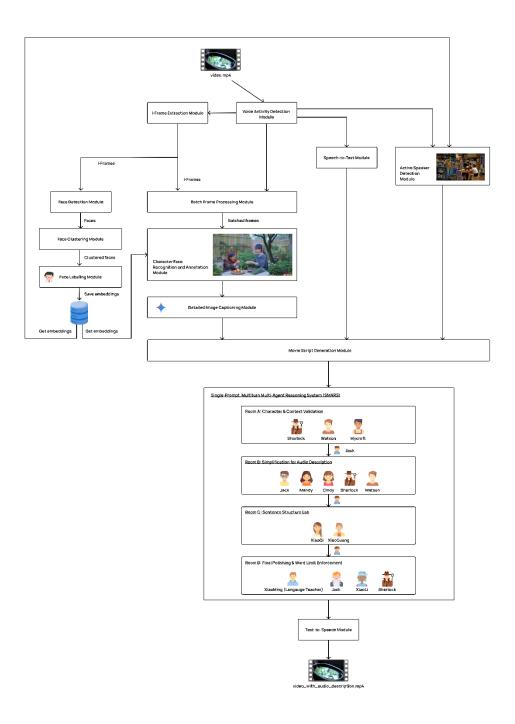


Figure 3.2 Architecture of Automated Audio Description System.

# 3.3 Activity Diagram

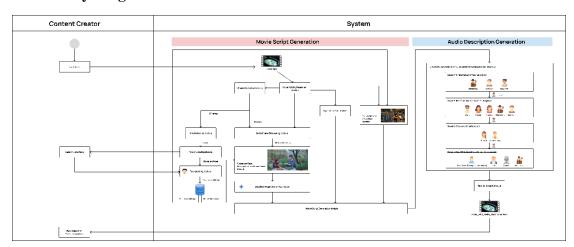


Figure 3.3 Activity Diagram of Enjoyable

# **CHAPTER 4**

# **System Design**

#### **4.1 I-Frame Extraction Module**

Videos are composed of different types of frames—Intra-frames (I-frames), Predictive frames (P-frames), and Bidirectional frames (B-frames) [14]. Each frame type serves a specific function in video compression standards such as H.264. I-frames are the keyframes that contain the complete image data, while P-frames and B-frames store only the changes from one frame to another.

This module focuses on I-frame extraction, strategically targeting I-frames to optimize computational efficiency and reduce costs while preserving the quality of visual data. By isolating P-frames and B-frames, the computational load is significantly reduced, allowing the analysis to concentrate on frames that contain complete information, thereby improving the effectiveness and efficiency of subsequent analytical processes.

#### 4.1.1 Workflow of I-Frame Extraction

The I-Frame Extraction Module utilizes advanced video processing libraries, FFmpeg, to efficiently identify and extract key frames from video streams. Within this module, FFmpeg is employed for two primary functions.

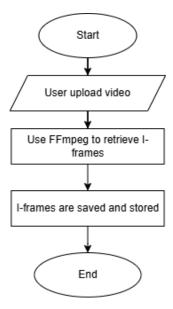


Figure 4.1 I-Frame Extraction Flowchart.

First, the system conducts frame detection by scanning the uploaded video to identify the occurrence of I-frames. This is accomplished through video parsing techniques that accurately detect coding sequences characteristic of I-frames within the video stream.

Following detection, the extraction process is initiated, where the identified I-frames are isolated and stored as individual image files.

### 4.2 Voice Activity Detection (VAD) Module

The Voice Activity Detection (VAD) Module is designed to identify specific segments within a video where speech occurs. These detected speech segments serve as the foundation for several crucial downstream processes, including speaker recognition, dialogue mapping, and the ultimate generation of synchronized audio descriptions that accurately complement the original dialogue within the video.

# 4.2.1 Workflow of Voice Activity Detection

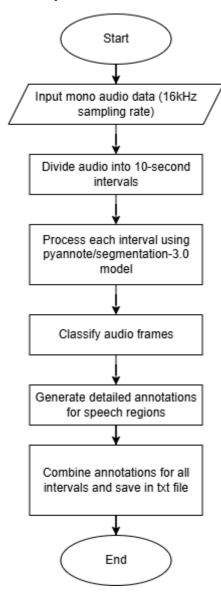


Figure 4.2 Voice Activity Detection Flowchart.

The Voice Activity Detection process is powered by the pyannote/segmentation-3.0 model from Hugging Face, a model highly optimized for

audio segmentation tasks. This model has been trained on a diverse collection of datasets, including AISHELL, AliMeeting, AMI, AVA-AVD, DIHARD, Ego4D, MSDWild, REPERE, and VoxConverse, curated by Séverin Baroudi. The model excels not only in identifying speech segments but is also finely tuned for related tasks such as speaker diarization, speaker change detection, and overlapped speech detection [17].

In its operation, the pyannote/segmentation-3.0 model processes audio data in 10-second intervals using mono audio sampled at a rate of 16kHz. The model classifies audio frames into categories such as non-speech and various speaker combinations. Through multi-class encoding, the model precisely identifies the presence of speech and generates detailed annotations to delineate speech regions meticulously.

# 4.3 Speech-to-Text (STT) Module

The Speech-to-Text (STT) Module is responsible for transcribing spoken dialogue in videos into written text. This step is crucial for our platform as it generates a textual representation of spoken content, which forms the basis for creating the movie script. Accurate transcription ensures that the generated script faithfully mirrors the original dialogue, making it indispensable for subsequent processes such as generating audio descriptions.

# 4.3.1 Workflow of Speech-to-Text

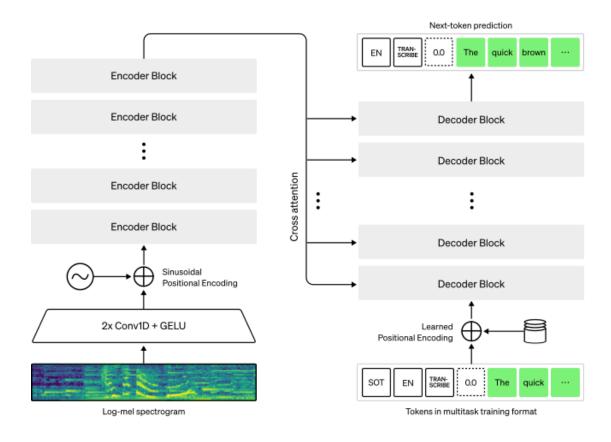


Figure 4.3 Architecture of Whisper [18].

The Whisper model, specifically the whisper-large-v3 variant, is chosen for this task due to its superior performance in recognizing speech across multiple languages and under various acoustic conditions. Whisper is a transformer-based model employing an encoder-decoder architecture, or sequence-to-sequence model, which

excels in translating spoken language into text. It is used to transcribes the segmented audio with state-of-the-art accuracy, closely capturing the nuances of the spoken word.

One of Whisper's standout features is its ability to handle multiple languages and accents, making it an excellent fit for global video content platforms that feature diverse user-generated content. Additionally, Whisper is highly resilient to background noise, music, and other common audio disruptions, maintaining clarity and accuracy in the transcriptions even in less-than-ideal acoustic environments.

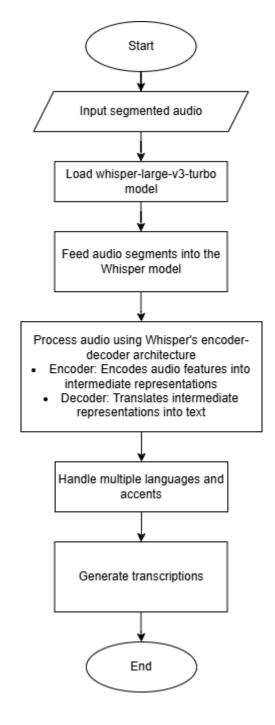


Figure 4.4 Speech-to-Text Flowchart.

The Speech-to-Text process begins with the segmented audio generated from the Voice Activity Detection (VAD) module. These audio segments are then input into the Whisper-large-v3 model, a transformer-based sequence-to-sequence architecture. The model processes the audio by first encoding its features into intermediate representations through its encoder. These representations are subsequently passed to the decoder, which translates them into text. Whisper's ability to handle multiple languages and accents ensures accurate transcription for diverse user-generated content, adapting the transcription process based on the detected language. Additionally, the model demonstrates resilience to common audio disruptions such as background noise and music, maintaining high clarity in the transcriptions. The transcriptions for each segment are then combined to produce a complete and cohesive text output, representing the spoken content with state-of-the-art accuracy.

# 4.4 Active Speaker Detection (ASD) Module

The Active Speaker Detection (ASD) Module is a crucial component in identifying the speaking characters within specific video timestamps. This identification ensures that the dialogue is synchronized with the name of the active speaker, thereby enhancing the overall completeness of the movie script.

### 4.4.1 Workflow of Active Speaker Detection

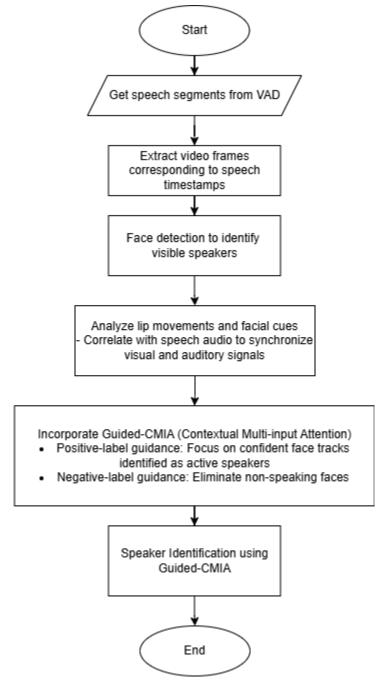


Figure 4.5 Active Speaker Identification Flowchart.

The Active Speaker Detection (ASD) module starts by utilizing speech segments identified by the Voice Activity Detection (VAD) module, which timestamp the precise moments when speech occurs within a video. Following this, the ASD process involves several key steps. First, timestamp-based segment extraction isolates the video frames corresponding to these speech segments, focusing on relevant portions of the video for analysis. Next, visual processing and face detection are performed on these extracted segments, to identify potential speakers visible during the speech. The system then analyzes lip movements and other facial cues from the detected faces to correlate with the audio, synchronizing these visual cues with the speech timestamps to identify the speaker. Finally, the system attributes the speech to the corresponding face, ensuring accurate speaker identification.

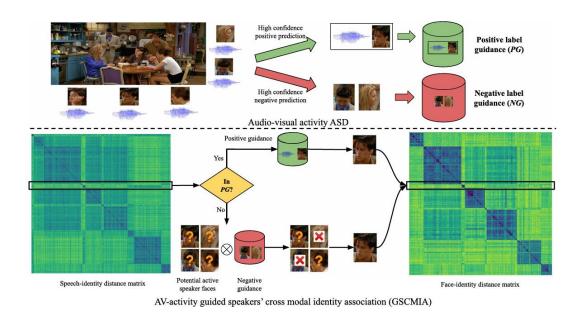


Figure 4.6 An overview of the guided speaker identity association across modalities (GSCMIA) audio-visual activity [19].

To further enhance the reliability of speaker detection, the system incorporates the Guided-CMIA (Contextual Multi-input Attention) approach [19]. This advanced technique integrates audio-visual activity information with speaker identity data, refining the process of speaker identification. Guided-CMIA uses high-confidence audio-visual (AV) activity predictions to form positive and negative guidance sets. Positive-label guidance focuses on face tracks that are most confidently identified as active speakers, while negative-label guidance eliminates non-speaking faces from

# **CHAPTER 4**

consideration. By synthesizing these insights, Guided-CMIA significantly improves the ASD module's accuracy and robustness, particularly in complex video scenarios where multiple speakers and significant background noise may be present.

#### **4.5 Face Detection Module**

Face detection involves identifying and locating human faces in digital images or video sequences. This module is responsible for systematically scanning each frame, detecting and cropping out all faces of the frames. This module underpins the accuracy of subsequent tasks, such as active speaker identification and face recognition.

#### 4.5.1 Workflow of Face Detection

The Face Detection Module utilizes the Dlib library, known for its robust and reliable performance in face detection tasks. Dlib, a modern C++ toolkit that integrates advanced machine learning algorithms, is particularly suited for high-performance real-time applications. The module begins its process by loading each video frame, which has been filtered before.

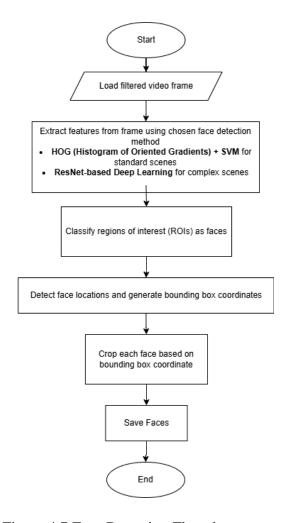


Figure 4.7 Face Detection Flowchart.

The core of the face detection process is executed using Dlib's Histogram of Oriented Gradients (HOG) combined with a linear Support Vector Machine (SVM) [20]. This method is highly effective at detecting faces, even under challenging conditions such as variations in lighting, scale, or angle. The combination of HOG for feature extraction and SVM for classification allows for precise detection across a wide range of video content.

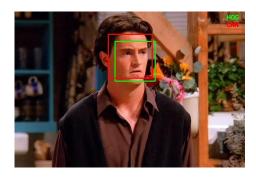


Figure 4.8 Two face detection functions are available in the dlib library. One is a deep learning MMOD CNN face detector, and the other is a HOG + Linear SVM face detector [20].

In scenarios where complex scenes are encountered—for instance, when faces are partially occluded—the system escalates its detection capabilities by employing Dlib's deep learning-based face detector, which utilizes a ResNet architecture. This approach offers enhanced accuracy by tapping into the power of deep neural networks, making it particularly effective in handling difficult cases where traditional detection methods might struggle.

Once faces are detected in a frame, the module proceeds to individually crop each face using the bounding box coordinates provided by the detection algorithm.

### **4.6 Face Clustering Module**

Face clustering is the process of grouping similar face images so that each cluster corresponds to a distinct individual within a video. The Face Clustering Module is tasked with organizing the detected and cropped face images into distinct clusters, where ideally, each cluster matches a unique individual or character seen throughout the video content.

# 4.6.1 Workflow of Face Clustering

The workflow of the Face Clustering Module begins with the application of the Chinese Whispers Algorithm, a highly efficient and robust graph-clustering technique well-suited for unsupervised learning scenarios, which is particularly advantageous for the task of face clustering [21].

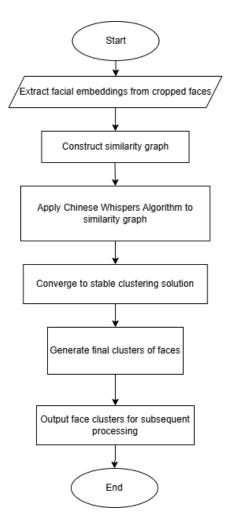


Figure 4.9 Face Clustering Flowchart.

The process initiates with feature extraction, whereby the detected faces—previously isolated and cropped—are converted into vector representations known as facial embeddings. These facial embeddings capture crucial facial features, including elements such as the eyes, nose, mouth, and overall facial geometry. The essence of these embeddings lies in their ability to extract distinctive features that enable the system to differentiate between different individuals.

Once the facial embeddings have been generated, the system constructs a similarity graph. In this graph, each face is depicted as a node, and the relationships between them are represented by edges that are weighted based on the degree of similarity between the corresponding facial embeddings. This graph-based representation is instrumental in enabling the system to make a comparative and relational analysis of all detected faces within the video segment.

Next, the Chinese Whispers algorithm is applied to this similarity graph. The algorithm functions by iteratively propagating labels across the network of nodes, with these labels being adjusted according to the weighted connections between nodes. As the algorithm iterates, it gradually converges toward a stable clustering solution, wherein nodes that represent highly similar facial features are grouped under the same label. The convergence of the algorithm results in distinct clusters, with each cluster ideally consisting of multiple instances of the same individual's face across different scenes in the video.



Figure 4.10 Example of clustered faces.

The final outcome of this workflow is a set of clusters, each representing a unique character or individual throughout various segments of the video. These clusters serve as the foundational data for subsequent processes, such as character identification and dialogue mapping.

#### **4.7 Face Labelling Module**

The Face Labeling Module is a user-assisted process, where the content creator or uploader plays an active role in labeling the clustered faces identified in the video. This module is crucial in ensuring that each cluster of faces corresponds accurately to distinct characters.

### 4.7.1 Workflow of Face Labelling

The Face Labelling process within the module is designed to be both intuitive and efficient, allowing content creators or uploaders to easily label the face clusters generated by the preceding Face Clustering Module. The system initiates this process by presenting the uploader with clusters of faces—these clusters represent various instances of the same individual appearing across multiple scenes in the video. To assist in proper identification, the system also provides supplementary metadata, such as the specific frames or timestamps where these faces were detected, along with any additional contextual information, like associated dialogues when available. This metadata helps the uploader gain a deeper understanding of the context around each cluster, ensuring more accurate labelling.

The labelling process is conducted through a user-friendly interface designed to make the task of identifying and labelling characters straightforward. This interface typically includes features such as the ability to view the corresponding video segments related to each face cluster, which aids the uploader in precisely identifying the character depicted. Additional functionalities, like dropdown menus or search tools, are provided to further streamline the process, allowing for the quick and consistent input of character names, roles, or other relevant identifiers.



Figure 4.11 Example of clustered faces labelled with names.

# **CHAPTER 4**

Once the uploader has labelled each cluster, the system finalizes the labeling process by storing the labels in association with the respective face clusters. These labels are then utilized in subsequent modules, including face recognition, active speaker detection, and dialogue mapping.

# 4.8 Character Face Recognition and Annotation Module

The Character Face Recognition and Annotation Module plays a crucial role in the final identification and annotation of characters of video frames. This module uses the face embeddings generated by the Face Embedding Module to accurately identify specific characters. Once identified, the module visually distinguishes these characters by annotating the video with colored bounding boxes around their faces.

# 4.8.1 Workflow of of Character Face Recognition and Annotation

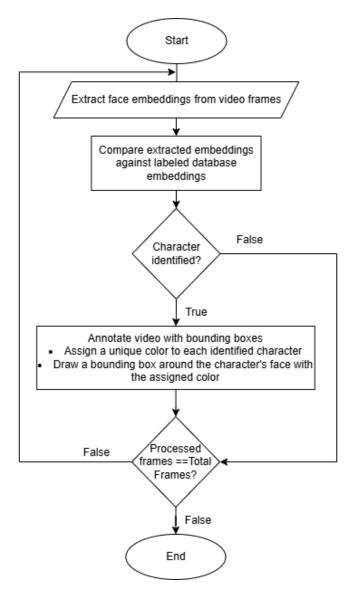


Figure 4.12 Face Recognition and Annotation Flowchart.

The process of Character Face Recognition begins by comparing the face embeddings extracted from the video frames against the labeled embeddings stored in the system's database. The primary objective of this comparison is to ensure accurate identification and differentiation between characters, maintaining consistency in character recognition throughout the video.

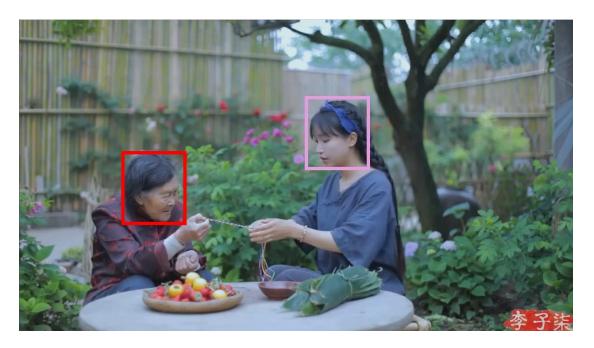


Figure 4.13 Example of different characters in the frames, each labeled with a different color bounding box.

Once a character is successfully identified, the system proceeds to the annotation phase. In this step, the video is marked with colored bounding boxes over each recognized character's face. Each character is assigned a unique color, providing a clear and immediate visual cue that helps users easily track and differentiate between different characters as they appear on screen.

#### 4.9 Image Captioning Module

The Image Captioning Module plays a significant role in generating comprehensive and descriptive narratives for the visual scenes within video content. This module operates on video frames that have already undergone processing through the Character Face Recognition and Annotation Module. By harnessing the advanced capabilities of Gemini and GPT-40 [22], the platform is able to produce detailed, context-aware captions that able to enhance the quality of the audio description generated in the LLM-Based Multi-Agent System Module.

### 4.9.1 Workflow of Image Captioning

The primary objective of the Detailed Image Captioning Module is to translate visual content into meaningful text that elevates the accessibility and understanding of videos for all users, particularly through the creation of audio descriptions. The workflow begins by organizing video frames—after they have been labeled and annotated with character-specific information, such as color-coded bounding boxes and names—into batches. Each batch represents a coherent sequence of frames that corresponds to a specific scene or segment within the video.

These color-annotated frames are then processed by GPT-40 to generate detailed captions. GPT-40 takes a holistic approach, analyzing the entire context of each scene by considering both the visual elements and the character annotations. This ensures that the captions generated do not merely describe isolated objects or characters, but also capture the broader essence of the scene, including actions, interactions, and emotional undertones.

The process continues with GPT-40 integrating character recognition data and the specified colors of each character into the captions. For example, suppose a character is highlighted with a red bounding box. In that case, GPT-40 might initially generate a caption like " <pink>is dressed in a traditional blue outfit with a blue headband, engaging in the activity of stringing colorful beads onto a thread. <red>, highlighted within a red rectangle, is attentively observing or participating in the task." Leveraging the character recognition data, this caption is then refined to be more specific, such as " Li Zi Qi is dressed in a traditional blue outfit with a blue headband,

# **CHAPTER 4**

engaging in the activity of stringing colorful beads onto a thread. Grandma, highlighted within a red rectangle, is attentively observing or participating in the task."

### 4.10 Movie Script Generation Module

The Movie Script Generation Module is designed to integrate various streams of textual data into a cohesive and structured document that reflects the overall narrative of the video. This script encapsulates the essence of each scene by combining captions, dialogues, character annotations, and scene descriptions, ensuring that both visual and auditory elements are seamlessly woven together.

#### 4.10.1 Workflow of Movie Script Generation

The workflow of the Movie Script Generation Module begins with the careful compilation of dialogue and image captions. To achieve this, the system first collates the detailed image captions produced by the Image Captioning Module with the dialogues transcribed by the Speech-to-Text (STT) Module. To enhance accuracy and context, these dialogues are further enriched with speaker identifications provided by the Active Speaker Detection Module, ensuring that the dialogue is associated with the correct characters.

Once all textual elements have been gathered, the system organizes these components chronologically based on their timestamps. This approach is critical for maintaining the narrative flow of the video, as it aligns the visual descriptions precisely with the corresponding dialogues. The result is a structured script that mirrors the video's timeline, presenting a seamless and coherent narrative that faithfully represents both the visual and auditory aspects of the content.

#### 4.11 Single-Prompt Multiturn Multi-Agent Reasoning System (SMARS)

To efficiently transform complex video dialogue into concise, well-structured, and multilingual audio descriptions for blind users, this project introduced a novel modular framework called the Single Prompt Multiturn Multi-Agent Reasoning System (SMARS). Unlike traditional distributed multi-agent systems that rely on external APIs and orchestration frameworks (e.g., LangChain, CrewAI, AutoGen), SMARS operates entirely within a single LLM prompt session, simulating multiple agents with clear roles and responsibilities executing one after another in a deterministic, highly controlled reasoning loop.

Built using structured prompt engineering, the system organizes the workflow across four distinct stages—Room A through Room D —each simulating a collaborative environment where specific tasks are performed by distinct agent personas. These agents, such as Sherlock (investigator), Jolin (rewriter), Xiaoli (language flow expert), and Xiaoming (character counter), emulate real-world editorial roles to refine raw video dialogues into linguistically and semantically optimized audio descriptions. This approach is notably language-agnostic, allowing content to be described in any target language (e.g., Chinese, Malay, English), expanding accessibility globally.

### 4.11.1 Workflow of LLM-Based Multi-Agent System

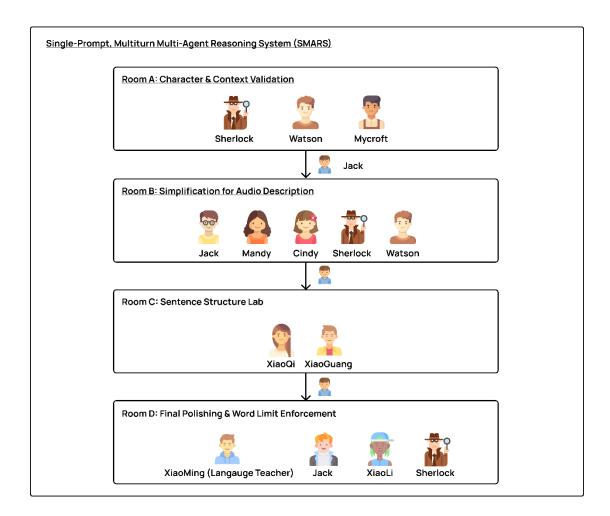


Figure 4.14 LLM-Based Multi-Agent System of Automated Audio Description System.

This system simulates a structured editing and reasoning workflow designed to transform raw visual dialogue into polished, concise, and accessibility-friendly audio descriptions for blind users. The process unfolds across four conceptual "rooms," each staffed by specialized virtual agents that represent different reasoning or editing roles. This compartmentalized approach mirrors the structure of an editorial or accessibility task force, where each team handles a specific phase of the transformation pipeline.

In the first phase, known as Room A (Novel Consistency Room), the focus is on context verification and character resolution. The agents in this room ensure that the dialogue is factually aligned with the novel version of the movie, eliminating contradictions, resolving unknown characters, and correcting any internal

inconsistencies. Sherlock, the primary investigator, probes all unknown elements, unlabelled individuals, or mismatches between character appearances and dialogue context, drawing from a library of known character traits to provide evidence-based identifications. Mycroft, the visual validator, meticulously checks that facial features, clothing, and accessories align with established character traits, challenging Sherlock's conclusions when necessary. Watson, the context historian, supplements their investigation with historical context from previous dialogues, highlighting recurring characters or already-clarified scenes. Finally, Jolin, the rewriter, synthesizes the results and rewrites the dialogue, ensuring character labels are correct and the dialogue is logically cohesive. The output from this room is a validated, preprocessed version of the dialogue, ready for the next step.

Room B (Detail Reduction Room) focuses on semantic simplification and prioritization. The agents in this room refine the verified dialogue by removing redundancies and unnecessary visual details, ensuring that only essential narrative actions are preserved for audio storytelling. Jack, Mandy, and Cindy, the audio describers, work together to condense complex visual scenes into clear, action-focused summaries. They cut verbose or repeated descriptions, especially those involving clothing, facial expressions, or gestures unless they are critical to the plot. Watson acts as a repetitive filter, cross-checking recent outputs to identify and flag echoed descriptions, offering alternative sentence structures or more varied language. Sherlock oversees this reduction process to ensure coherence and that no critical details are lost. The result is a concise, vivid snapshot that balances engagement with brevity.

In Room C (Subject Clarity Room), the agents focus on ensuring that each sentence has a single, unambiguous grammatical subject, preventing confusion, especially in audio form. Xiaoguang and Xiaoqi specialize in restructuring sentences with multiple subjects so that each maintains a clear logical subject. For example, if two characters are performing different actions in the same sentence, they will determine which action should be the main clause and place the other as background context or a supporting phrase. This restructuring eliminates ambiguity, ensuring that listeners can easily follow the actions in the scene without confusion about who is doing what. The output from this room is grammatically coherent and ensures clarity in terms of actions.

Finally, in Room D (Final Polishing Room), the focus is on language fluency, word count compliance, and qualitative verification. Xiaoli, the language fluency specialist, refines clunky or overly literal sentence fragments, rephrasing them to match natural narrative rhythms. Xiaoming, the word count checker, verifies that each sentence adheres to the predefined character count limit. If the limit is exceeded, Xiaoli trims non-essential clauses, ensuring that only key content remains while maintaining narrative accuracy. Holmes, the narrative fact verifier, ensures that no essential plot elements are omitted during this trimming process. This room also checks the vocabulary to ensure it aligns with the intended blind user's profile, avoiding unfamiliar or inaccessible terms. If the sentence meets all criteria, it is finalized; otherwise, it may be sent back for adjustments. If the dialogue lacks meaningful content, such as generic conversation, Room D will return a null code AD404, indicating the dialogue is not worth including.

Throughout the entire process, John, the logical messenger, acts as the intermediary examiner. He ensures that the dialogue transitions smoothly from one room to the next, maintaining consistency and delivering the finalized output for compilation. His role guarantees that the entire multi-agent pipeline runs predictably and transparently, ensuring a successful and high-quality result.

### **CHAPTER 5**

### **System Implementation**

# 5.1 Hardware Setup

The hardware used for developing, processing, and testing the online video streaming platform and the automated audio description system is outlined below. This laptop is the central hub for all developmental activities, ensuring the systems run smoothly and efficiently.

Table 5.1 Specifications of Laptop.

Description	Specifications		
Model	HP Victus by HP Laptop 16-e0122AX		
Processor	AMD Ryzen 5 5600H with Radeon Graphics		
Operating System	Windows 11		
Graphic	NVIDIA GeForce RTX 3050 Laptop GPU, AMD		
	Radeon(TM) Graphics		
Memory	16GB DDR4 RAM		
Storage	512GB SSD (MTFDHBA512TDV-1AZ1AABHA), 1TB		
	SSD (TEAM TM8FPD001T)		

### **5.2 Software Setup**

The software required for this project was installed and configured on the development system to build a robust environment. The major software components used were:

1. Python 3.10:

This served as the primary programming language, alongside essential libraries like OpenCV, dlib, and PyTorch for image and video processing.

### 2. Whisper by OpenAI:

Deployed for Speech-to-Text (STT) processing, Whisper provided state-of-theart transcription accuracy, helps for generating textual representations of spoken content.

#### 3. **GPT-40:**

Employed for detailed image captioning, GPT-40 processed batches of annotated video frames to generate comprehensive and contextually aware image descriptions.

## 4. Gemini 2.0 Flash:

Multilingual Large Language Model (LLM) used in the LLM-Based Multi-Agent System module. This model tailored generated audio descriptions to various user personas, ensuring inclusivity and adaptability across diverse content.

#### 5. CUDA Toolkit 11.8:

Integrated into the system to enable GPU acceleration for computationally intensive workflows such as I-frame extraction, face detection, and active speaker detection.

## 6. **FFmpeg:**

Utilized for video manipulation tasks, including I-frame extraction.

#### 5.3 Settings and Configuration

This section outlines the system settings, library dependencies, environment configurations, and model parameters required to set up and run the "Enjoyable" automated audio description platform efficiently on a local machine. Proper configuration ensures that the system modules — from video processing to multi-agent reasoning — operate cohesively.

## **5.3.1 Environment Setup**

The system was developed and tested on a **Windows 11 64-bit** environment with NVIDIA RTX GPU support. The preferred environment manager is **Anaconda**, with Python 3.10. The following key Python libraries and frameworks are installed:

- 1. opency-python
- 2. torch
- 3. transformers
- 4. openai-whisper
- 5. mediapipe

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- 6. moviepy
- 7. librosa
- 8. numpy
- 9. pandas
- 10. scikit-learn
- 11. ffmpeg

#### **5.4 System Operation**

The Voice Activity Detection (VAD) module segments the audio from a video into speech and non-speech intervals. The audio from the video file was extracted and resampled into a mono format at 16kHz using FFmpeg. The resampled audio was then passed through the pretrained PyAnnote pipeline, which processed it as a waveform sample. From this analysis, only segments containing speech were identified and segmented, with relevant metadata, including timings. This segmentation forms the basis for identifying speakers and text transcription.

```
[1.752219-3.152844] SPEAKER 01:
[8.755344-9.970344] SPEAKER 01:
[12.062844-14.020344] SPEAKER_01:
[16.382844-17.445969] SPEAKER 00:
[17.783469-19.420344] SPEAKER_01:
[20.095344-29.950344] SPEAKER_01:
[31.620969-32.937219] SPEAKER 01:
[36.447219-37.594719] SPEAKER 01:
[44.429094-46.234719] SPEAKER 01:
[46.572219-48.023469] SPEAKER 01:
[48.529719-49.221594] SPEAKER 01:
[49.626594-53.035344] SPEAKER 01:
[53.102844-54.925344] SPEAKER 01:
[57.355344-58.536594] SPEAKER 01:
[65.235969-65.961594] SPEAKER 01:
[72.171594-73.403469] SPEAKER 01:
[79.022844-79.950969] SPEAKER 01:
[89.046594-90.582219] SPEAKER 01:
[96.927219-98.057844] SPEAKER 01:
[114.224094-116.215344] SPEAKER 01:
[121.159719-122.037219] SPEAKER 01:
[124.214094-125.074719] SPEAKER 01:
[125.850969-127.032219] SPEAKER 01:
[129.698469-131.942844] SPEAKER 01:
[136.752219-137.595969] SPEAKER 01:
[137.967219-139.249719] SPEAKER 01:
[142.709094-144.869094] SPEAKER 01:
[145.932219-147.450969] SPEAKER 02:
[147.923469-148.868469] SPEAKER 02:
[150.977844-153.914094] SPEAKER 01:
[256.024719-257.054094] SPEAKER 01:
[257.847219-259.214094] SPEAKER 01:
```

Figure 5.1 Result of VAD.

The system begins by extracting Intra-coded frames (I-frames) using a video decoding tool such as FFmpeg within non-speech segments. I-frames are key frames that contain the complete image data for a frame, making them ideal for scene analysis. By selecting only I-frames, this module ensures efficient frame sampling by skipping redundant intermediate frames, thereby optimizing the processing speed and reducing computational load throughout the pipeline.

In the pipeline of the processing speed and reducing computational load throughout the pipeline.

Figure 5.2 Result of I-Frame Extraction.

The filtered frames are then passed through the Face Detection Module, which identifies and temporarily excludes frames that consist primarily of faces.

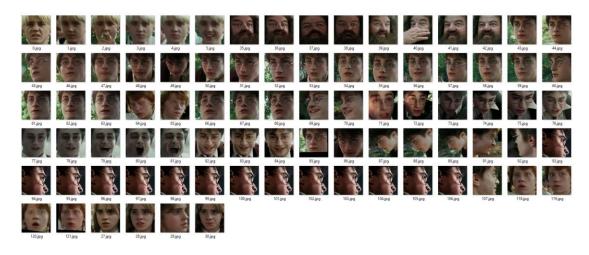


Figure 5.3 Result of Face Detection.

Chinese Whisper Algorithm is used to module groups similar faces across frames

<u> </u>	9/5/2025 7:18	AM File folder
<u> </u>	9/5/2025 7:18	AM File folder
<u></u> 2	9/5/2025 7:18	AM File folder
<u> </u>	9/5/2025 7:18	AM File folder
<u> </u>	9/5/2025 7:18	AM File folder
43 pg 43 pg 45 pg 45 pg 45 pg 45 pg 47 pg	41,jeg 42,jeg 50,jeg 51,jeg 52,jeg	\$3,jpg \$4,jpg \$3,jpg \$3,jpg \$3,jpg
9 jeg 60 jeg 61 jeg 62 jeg 60 jeg	64,jpg 63,jpg 63,jpg 63,jpg	6) jo 70 jog 71 jog 72 jog 73 jog 74 jog
73.jeg 78.jeg 77.jeg 78.jeg 79.jeg	80,pg 81,pg 82,pg 83,pg 84,pg	\$3,jpg \$51,jpg \$1,jpg
\$2.prg \$1.prg \$4.prg \$5.prg \$9.prg	97,pp 943,pp 100,pp 100,pp	102,jpg 102,jpg 104,jpg 103,jpg 106,jpg 107,jpg
T T A T T		

Figure 5.4 Result of Face Clustering.

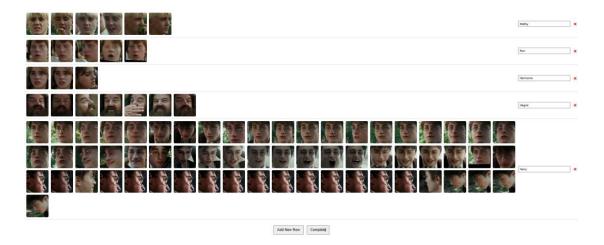


Figure 5.5 UI for Users to label clustered faces.



Figure 5.6 Result of Face Recognition and Annotation.

The Detailed Image Captioning module was implemented and tested to generate detailed and accurate descriptions of visual elements in video frames, with a particular focus on people and objects. The large language model was employed as a "detective" to observe and analyze the intricate details of the frames more effectively. This approach aimed to enhance the precision and comprehensiveness of the generated captions. Below is the prompt used to guide the large language model in producing detailed image captions:

```
Generate detailed descriptions for an audio descriptor to create audio descriptions based on visual
elements of given frames. If people are framed by a <u>colored</u> bounding box, replace the mention of "people" with the corresponding <u>color</u>.
Observe and describe all visual elements of the given frames in detail, as if you are Sherlock Holmes.
Focus on characteristics of people and other visual elements with precision.
- **People**: Describe their physical traits, expressions, attire in very detailed fashion, must include
clothing patterns and color, hairstyles, eye color, posture, and any notable features. Use the color of
the bounding box to refer to them when applicable.
- **Objects**: Detail colors, shapes, sizes, textures, and positions.
# Steps
1. Examine each frame closely.
2. Identify notable people, objects, and settings.
3. Describe each element systematically, ensuring thoroughness.
4. Substitute "people" references with bounding box colors when necessary.
Provide a detailed description in paragraph form, covering all sections: People and Objects
comprehensively.
- **Original Description**: "A girl with long, wavy brown hair and bright green eyes wears a red polka dot dress and is kicking a ball."
  - **Updated Description**: "<blue> with long, wavy brown hair and bright green eyes wears a red polka
dot dress and is kicking a ball." (assuming the girl is within a blue bounding box)
The ID of the images corresponds to the frame number. Take a deep breath and think, If you believe some
frames can be combined, then combine them, if there are separate actions, then do not combine them.
Output the result in the following format:
[ID Num] XXXXXXX
[ID Num - ID Num] XXXX
[ID Num] XXXX.
```

Figure 5.7 Prompt used to generate image captions.

```
[1.0.1.6] The terms it dark and allesst completely obscured, that appears to be a dark those stands vagedly against a backdrop of indistinct vertical lines.

[1.0.1.5] Soutlight streams through the dense ecopy of a forest, illustrating the scene with a soft, degular light. A thosy benedict and with long start stands and the trees, warring a dark vext own a gang garment. Attached to the vext lines of the scene.

[1.0.2.5.1.3] a group of students is standing in wait appears to be a forest clearing. In the solids for, forest, ye a temps good with a solid place, decreve the book with a keen expression. Solidary, the air bulleton and the solid place of the solid place of the solid with a long was a solid place of the solid place. A solid place of the solid place of th
```

Figure 5.8 Result of Detailed Image Captioning.

The Speech-to-Text (STT) module was developed to transcribe spoken dialogue from video audio streams into highly accurate and contextually rich text. The OpenAI whisper-large-v3-turbo model was employed for transcription, offering multilingual support and high accuracy across diverse acoustic environments. The following section outlines the results from the preliminary testing phase associated with the result of voice activity detection and speaker diarization.

```
[1.752219-3.152844] SPEAKER_81: In't he beautiful?
[2.75214-3.978044] SPEAKER_81: In't he beautiful?
[2.65244-1.66243] SPEAKER_81: In't he beautiful?
[2.65244-1.66243] SPEAKER_81: In't he beautiful?
[2.75219-3.978045] SPEAKER_81: In't he beautiful?
[2.75219-3.978045] SPEAKER_81: In't he beautiful?
[2.75219-3.978045] SPEAKER_81: In't have beautiful?
[2.75219-3.978045] SPEAKER_81:
```

Figure 5.9 Result of Speech-to-Text.

The ASD (Active Speaker Detection) module processed audio and video segments from earlier stages to identify active speakers within predefined timestamps. The system tracked visible lip movements and evaluated facial dynamics to detect speaking activity. Lip synchronization was achieved by analyzing localized motion extracted from detected lip regions and aligning these movements with corresponding audio patterns. Additionally, facial dynamics, including subtle head and mouth movements, were assessed to identify speech onset or conclusion. Once the active speakers were identified, the system proceeded to perform face recognition, matching the detected faces with a pre-established Character Bank.

Figure 5.10 Result of Active Speaker Identification.

By combining the data collected from the Active Speaker Detection (ASD) and Face Recognition, Voice Activity Detection, Speaker Diarization, Speech-to-Text, Detailed Image Captioning modules, a comprehensive movie script is generated. This script is constructed by correlating the identified active speakers with their corresponding dialogues, timestamps, and visual cues. The process integrates speaker attribution, character identification, and detailed annotations of the visual and audio elements to create a coherent and accurate script that reflects both the spoken dialogue and the actions of the characters.

```
13.73.12.73.20 Frames Descriptions: In a sun-diapplied frost Classing, a large, sized allike creative with grown as largest facility as if first seasoning the old man. To the left, partially obscured by Versa, a kingle imode structure with on the ground. A low stone wall ross along the edge of the classing, separating if from the dense modes. The smallpirt filters through the trees, creating a sort, diffraced light across the scene.

17.73.13.13.00 Frames benchrothers for the common of the common of the common of the control of the creative for an experiment of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the creative for an experiment of the control of the control of the creative for an experiment of the control of t
```

Figure 5.11 Result of Movie Script Generatikon.

The movie script will go through the **SMARS** module — a simulated multiagent reasoning environment embedded within a single prompt of a large language model. The agents iterate collaboratively inside the same prompt, refining the output until it is optimized in terms of content clarity, style, and time alignment. This allows rich and elegant descriptions that do not exceed their narration time budget. Below is the prompt used for SMARS and its output:

There are 4 audio description writers and 1 English teacher in a company. There are 4 discussion rooms.

Here is the dialogue they need to discuss: "{custom dialogue}". They must discuss it in Chinese.

John is the helper here, and he will pass the discussed content across the meeting rooms.

The dialogue will first go through the first meeting room.

Room A (中文讨论区, 在这里只能使用中文。Identify All Unknown)

The rule in this room is that the novel version of the movie is always correct. If there is a conflict between the dialogue and the novel version of the movie, then change the dialogue to make it match! The novel version of the movie is always correct. If an unknown character is not mentioned in the novel version of the movie, they are extremely unimportant and should be deleted.

In this room, there are Sherlock, a detective, his brother Mycroft, Watson, and Jolin.

After that, Sherlock, as a detective, will examine the dialogue and always ask a lot of questions about 1. Unknown Things 2. People without names in the dialogue. He will then answer those questions by looking at the novel version of the movie or the information he has investigated to find the answers and provide them to others. This is the character name with traits he investigated: '''{traits}'''. For unknown people, he will say something like: "Who is the xxxx? Let me refer back to the Character Name with Traits. She has the xxx, and xxx, and xxx, which matches xxxx, so she is XXXX." He must state the proof or evidence and cannot just guess. He will always end with "The proof is \_\_\_\_\_/ There is no evidence, so ignore me." Note that the script won't have typos. For unknown people, they must discuss further. If there is truly no character name, discuss their role, like a worker, xxx's friend, xxxx, etc.

For Mycroft, he will always discuss with Sherlock. He will ensure the clothing details, hairstyle, and accessories are completely the same before considering it as strong evidence; otherwise, he will argue with Sherlock. Remember, everything must be exactly the same!

For Watson, he helps others by providing information from the previous 20 dialogues, and he will tell the others what he knows to help identify the unknown.

Furthermore, if a character looks to the left or looks to where, you need to mention what they are looking at. If you can't determine this, don't mention the detail, as it could confuse a blind person.

Jolin listens to Sherlock and Mycroft's discussion carefully. After that, she rewrites the sentence by integrating the key points they discussed. She will only pass it along for further discussion after incorporating the latest information. When passing it on, she will say one of the following: "Alright, now let's discuss this dialogue integrated with the latest information: [updated dialogue with name]."

If the Character Name with Traits does not state that the person has certain traits, then it means he or she is not the person.

The dialogue passed to John cannot contain the words "likely" or "maybe." Yes means yes, and no means no.

Room B (中文讨论区,在这里只能使用中文。Delete Unimportant Things)
Jack, Mandy, and Cindy, who are the audio descriptors, and Watson and
Sherlock will begin to review the dialogue passed by John.
Here's how the discussion will flow now:

Delete all clothing details.

If there are fewer than 5 words, delete the surrounding details and focus only on action and feeling.

Remove any text shown on the screen as subtitles.

Remove all spoken dialogue from the audio description. [Remove all sentences like "xxx says:" or "xxx talks."]

If the setting details, such as the place or surroundings, have been mentioned before, directly delete those details.

Remove all unnecessary details and retain only the most important events. Follow these steps: a) Find the primary action b) Check whether the primary action is described before c) If yes, describe it in more detail or describe another event d) If no, describe the primary action in detail.

Watson will check if the dialogue contains details similar to any of the previous 20 dialogues. He will check them one by one. If there are similar details, he will identify the repeated parts and ensure they are no longer used.

For example:

Watson: Repetition detected in "He kneels, blowing on a grass pile in the forest."

Issue: "He kneels" was already mentioned in 267.988-278.900 ("Gang Dan kneels, using a rock and fibrous material to create fire").

Issue: "Blowing" repeats the fire-making action from earlier.

Changed it to:

"Lowering himself, he breathes gently onto a small grass pile, urging the embers to catch."

Changes made:

Replaced "He kneels" with "Lowering himself" to avoid repetition.

Made the action more vivid by using "breathes gently" instead of "blowing."

Clarified intent with "urging the embers to catch."

Therefore, if noticing such a similar case, Watson will notify others and make changes to the dialogue by:

Varying Sentence Structure:

Before:

Shirtless, Gang Dan ascends the palm, reaching for coconuts high above. Bare-chested man climbs coconut tree, reaching for trunks.

After:

Sunlight filters through the fronds as Gang Dan perches near the top, steadying himself. Fingers curl around the coconut's husk-his climb has paid off.

#### After

With practiced ease, he climbs higher, the wind rustling the leaves around him.

 Instead of simply mentioning "barefoot" and "scaling," we add wind rustling the leaves, making the scene more immersive.

#### Reduced Redundant Descriptions:

#### Before:

Bare-chested, Gang Dan climbs for water in the tall palm forest. Shirtless Gang Dan climbs palm, grabbing for coconuts.

#### After:

Gang Dan grips the rough bark, his muscles tensed as he ascends.

 No need to repeat "bare-chested" or "shirtless" because the action already implies exertion. Instead, focusing on his grip and tensed muscles gives a stronger visual.

Natural Flow (Progression Toward the Goal):

#### Before:

High in the palm tree, Gang Dan grips the trunk.

Bare-chested man climbs coconut tree, reaching for trunks.

#### After (shows progression):

Gang Dan grips the rough bark, his muscles tensed as he ascends. With practiced ease, he climbs higher, the wind rustling the leaves around him. Bare feet press firmly against the trunk as he scales toward the canopy. He stretches one hand upward, eyes locked on the ripe coconuts swaying above. Sunlight filters through the fronds as Gang Dan perches near the top, steadying himself. High in the palm, he carefully reaches for the prize, balancing against the breeze. Fingers curl around the coconut's husk-his climb has paid off.

• This sequence gradually builds the scene from climbing to reaching the goal, instead of repeating similar phrases in different ways.

Please help us, Watson. If the blind always hear the repeating words or sentences with similar meanings, they will feel bored. We need your help to filter them. Let's make their world better.

Pass the improved dialogue to John.

After the discussion is complete, it will pass to Room C via John.

#### Room C (转换成合适的主体)

小光和小齐会一起讨论, 以下是例子:

小光: 这句话"鹰头马身兽站立,哈利先伸手示好。"你有没有觉得哪里怪怪的?

小齐: 嗯,好像有点。是两个动作吧,一个是"站立",一个是"伸手示好",可分别是两个不同的主语。

小光: 对!"鹰头马身兽"在"站立","哈利"在"伸手示好",这其实让句子有两个主体了。

小齐: 那要怎么改, 才能让句子只有一个主体呢?

小光:可以把其中一个动作变成修饰成分。比如,如果我们想以"哈利"为主角,可以说:"哈利在鹰头马身兽站立时先伸手示好。"

小齐:这样句子的主语就只有"哈利"了,另一个动作只是背景描述。

小光: 对。如果我们想让"鹰头马身兽"当主语呢?

小齐: 那可以说: "鹰头马身兽站立着,望着哈利,见他先伸手示好。"这样"鹰头马身兽"成了主语。

小光: 没错。关键就是让动作都围绕一个主语展开。

之后, John 会把小光和小齐讨论后的答案传给Room D。

Room D (中文讨论区,在这里只能使用中文。Make it under the word limit and improve the sentence, becoming more like audio description. VERY IMPORTANT: You guys are tasked to rephrase the sentences, not make each sentence short and combine them, so only include sentences when they are important.) 注意,中文和英文语法有很大的不同,请以中文的思想进行讨论,包括语法等。

注意,全部从房间C传来的句子的句子结构有些不太通顺,可能是语序或逻辑关系出了问题。他们表达有些生硬,可以稍微调整,使其更流畅自然。这是因为他们都是英文生成的句子直接翻译过来的,所以可能会有些不符合中文的表达习惯。你可以对这些句子进行适当的调整,使它们更符合中文的自然表达方式,同时保持原意不变。

他们需要检查总结句子是否存在误解。例如,"霍格沃茨的学生手里拿着书"总结成"学生读书"是错误的,因为它只说明他们携带了书,而没有提到他们实际在做什么。

参与者有小丽 (负责音频描述) 、小明 (负责精确计算字数的英语教师) 、福尔摩斯、韦斯莱和杰克。

此外,还有一位用户—— $\{blind.user\}$ ,因此小丽每次都会询问她词汇是否合适。每次结束之前都需要询问她。

如果他们已经讨论过了,最终的结论只是像"XXX和XXX在说话"这种非常笼统的信息,那他们就应该立刻停止讨论,并将代码 AD404 (代表"没有值得汇报的信息") 传达给约翰。

请不要任意揣测剧情/演员的意图,如果需要更改,请询问福尔摩斯,然后他必须查看novel version of the movie,然后给出你们可以更改的证据,不然,不允许添加不相关的东西。

请确保最终的句子只有一个主体。

对于小明,他会确保最终的句子在{custom\_max\_word\_ch}字数以内——不多。他会检查字数是否超出限制并告诉大家是否符合字数要求。

下面是小明的计算方式(逐个字计算,并得出总字数): 例子: 对于这一段,"我们看到一片孤独的灰色沙滩。木材散落一地." 小明: "让我检查字数,确保不超过限制。"1. 我" "2. 们" "3. 看" "4. 到" "5. 一" "6. 片" "7. 孤" "8. 独" "9. 的" "10. 灰" "11. 色" "12. 沙" "13. 滩" "14. 木" "15. 材" "16. 散" "17. 落" "18. 一" "19. 地"。这样总共是19个字,符合要求。"

对于小丽,她也不会逐句拆解,因为这被视为作弊,教授要求她做的是确保整个描述的字数符合限制,而不是逐句调整。如果字数超过限制两次,小丽会先删除一个部分,如果还是超出字数限制,再删除另一个部分......但每次删除之前,她都会向福尔摩斯征求意见,了解删除的部分是否对故事情节很重要。

请注意,如果已经符合要求,请停止讨论并把答案直接传给约翰,请不要再尝试把句子变得更精 简。

Figure 5.12 Prompt of SMARS.

#### **CHAPTER 5**

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Okay, let's start with Room A.
**Room A (中文讨论区)**
**Matson:** 好的,我们先来确定这段对话对应小谈中的哪个部分。这段膨胀对应小说第三篇(The Dance of Rospect)和第四篇(A Flight of Fancy)的交界处,具体能够了能和与巴克比克互动之后,准备额上巴克比克 化行动场景。
**原養内容: **
Chapter 3: "Hagrid's eyes twinkled with mischief. 'Think he might let you ride him now, Harry.
Harry's eyes widened. "What? Really?"
Hagrid chuckled, lifting Harry onto Buckbeak's back. 'Hold on tight now! Get to the weak joint'
Before Harry could fully process what was happening, Buckbeak's powerful legs propelled them forward. A rush of wind filled Harry's ears as they soared into the sky, the Forbidden Forest shrinking below. Harry screamed with delight, a primal joy enupting from deep within him."
**lolin:** 根据Watson的描述,我们可以调整对话如下:
 "88.001-84.727 旁白:唐头马身曾站在树林中,它有着唐一样的头和马一样的身体,覆盖着灰白色的羽毛。它的翅膀半张开,露出黑色的尖端。高大的树木环绕着它,一些树木被石墙围住。
1. **應头易者: ** 税价需要権以它的具体外根据法。在电影和小场中保温保持一致,小规则提到它的羽毛是**shimering silver-gcay*,这里说是左白色,颜色上确微有点出入。但电影中视觉效率可能更偏灰白,可以暂时保留。
2. ***石香: ** 这弦石能的操作用是什么了 是完全围住树木,还是只是一部分? 小峽中建設的是**a low stone wall** 这是一个姊妹的石鬼,作用是围住一块空地。
3. **未以礼 ***。以我用影明也人物,看起朱只有鹰头马身鹰。
4. **桃木木***,以木是百匙大
**的ycroft;*** 规则享5herlack的分析,关于每头马身做的羽毛颜色,小说中的**shimmering silver-gray*在不同光线下可能会呈现出坎白色的视觉效果,因此可以暂时倒留坎白色。但是,需要确保电影中每头马身做的羽毛颜色在垂体上是确中很大色的,如果电影电型电白色,那就要传统。 关于石塘,小说强强了在"156",所以理理这个点。
**lolin:** 好的,我将Sherlock和Mycroft的分析整合一下,修改后的对话如下:
"88.881-84.727 旁白:魔头马身兽站在树林中,它有着鹰一样的头和马一样的身体,覆盖着灰白色的羽毛,它的翅膀半张开,露出黑色的尖端。高大的树木环绕着它,一些树木被建石墙围住。"
**Sherlock:** 很好,现在我们来深入研究一下细节。

    **・"海头马角梅站在树林中": ** 它的边接是怎样的? 臺雪標地环極回南, 还是她松地站在那里? 这可以补充一些信息, 电影中是雪橋地环極回南, The proof is novel_do not have such detail.
    **・"它的翅膀半张开, 雷出周色的分辨": ** 翅膀半张开的角度是怎样的? 是烟筒张开, 还是完全展开? 这可以更精緻地搬送翅膀的状态。电影里是橄榄张开。 The proof is novel_do not have such detail.

**Mycroft;** 我同意Sherlock的观点,补充细节可以使描述更加生动。唐头马券售的站姿和翅膀的状态是重要的视觉信息,可以更精确地描述。
**lolin:** 好的,我根据Sherlock和Mycroft的分析,进一步完善对话:
 **lolin:** 好的,我根据Sherlock和Mycroft的分析,进一步完善对话
 580.001-84.727 旁白:扁头马身臀智惕地站在树林中,它有着扁一样的头和马一样的身体,覆盖着灰白色的羽毛。它的翅膀微微张开,露出黑色的尖遍。高大的树木环绕着它,一些树木被赌石墙围住。"
 ***Jolie: ** Airlight, now let's discuss this dialogue integrated with the latest information: 89.801-84.727 旁白: 魔头马身看管撒地站在树林中,它有着每一样的头和马一样的身体,重量看对白色的羽毛,它的翅膀母腹张汗,重出黑色的尖端,高大的树木环绕着它,一些树木碗袋和撒性。
 **Room B (中立対论区)**
 **lack: ** 好的,我们现在来精简lohn传来的对话。
 **Mandy: ** 第一步,删除所有服装细节。这段描述里没有服装细节。
 **Cindy:** 第二步,如果句子少于5个字,删除角色周围的事物,只关注动作和感受。这句旁白显然超过5个字。
 **Jack:** 第三步,移除屏幕上显示的任何文字 (字幕) 。这段描述没有字幕。
 **Mandy: ** 第四步,移除所有口头对话的音频描述。这段描述是旁白,没有对话。
 **Cindy: ** 第五步,删除所有不必要的细节,只保留最重要的事件。

    a) 找出主要动作: 腐头马身曾站在树林中。
    b) 检查主要动作是否之前描述过: 没有。
    c) 详细描述主要动作。

 **Watson: ** 好的,让我来检查这段对话是否包含与之前28段对话相似的细节。
 Watson: No repetition details detected.
 **Cindy:** 那么,我们来精简一下,突出魔头马身兽的特点和周围环境的重点。
 "80,001-84,727 旁白:魔头马身兽管惧地站在树林中,灰白色羽毛覆盖全身,翅膀微微张开。周围是高大的树木和燧石塘。"
 **Jack: ** 这样更加简洁明了,突出了鹰头马身兽的特征,并且保留了环境的描述。
 **Mandy: ** 同意。
 **Sherlock:** 有一点可以讨论。 "警惕地" 这个词是否必要? 盲人用户主要通过所觉效取信息。他们能"听"出唐头马身悬是否警惕吗? 还是说。这个词是用来描述它的姿态,而这种姿态可以用其他方式来描述?
**Cindy:** 这是一个好问题。我们可以用更具象的描述来代替"警惕地"。比如,我们可以说"鹰头马身兽竖起耳朵站在树林中",这样更具体,也更容易让盲人用户理解。
 **lolis:** 好的,我修改一下:
*80.001-84.727 旁白: 唐头马寿曾竖起耳朵,站在树林中,灰白色羽毛翼盖全身,扭膀阁微微开。周围是高大的树木和矮石墙。
 **John: ** "80.001-84,727 旁白: 鹰头马身曾竖起耳朵,站在树林中,灰白色羽毛覆盖全身,延膀微微张开。周围是高大的树木和矮石境。"
 Room C (转换成合适的主体)
 约翰会把小光和小齐讨论后的答案传去Room D
 **John:** 旁白: 鹰头马身兽竖起耳朵站在树林中,灰白色羽毛覆盖全身,翅膀微微张开。周围是高大的树木和矮石墙
 **Room D (##****************
**小丽:** 我们来优化这句话,让它更适合音频描述。
 **小丽:** 字数超了好多!我们需要翻减一些内容。福尔摩斯,你觉得哪些信息对剧情来说不是特别重要?
 ***福沙寧斯·** 让我们回路一下小说,环境能可混放重要,但"周围是高大的时术如装石墙"可以隔阂。首人用户可以通过之前的场景能达对森林环境有所了解,更重要的是海风马身者的姿态和外貌特征。 相僻小地半说,周边的时术如装石墙设有提及非常重要,应该不等端
 **小鳴:** 好、翻飾版的句子是: "旁白: 唐朱马身曹室紅耳朵站在班妹中,方白色京毛羅童全身,孤時微粉汗," 让粉丝查字教,确怀不起过限制。"1. 旁** 72. 白*** 73. 唐** "4. 关** 75. 马*** 76. 号** 77. 善** 8. 答** "9. 起** "10. 耳** "11. 朵** "12. 法** "13. 在** "14. 词** "15. 林** "16. 中** 17. 次** "18. 白** 179. 色** "70. 帝** 721. 帝** 722. 唐** 723. 帝** 723. 帝** 724. 帝** 723. 帝** 724. 帝** 723. 帝** 724. 帝** 724. 帝** 724. 帝** 724. 帝** 725. 帝** 725. 帝** 726. 帝** 726. 帝** 727. 帝** 726. 帝** 726. 帝** 726. 帝** 727. 帝** 726. 帝** 727. 帝** 727. 帝** 727. 帝** 728. 帝** 727. 帝** 7
 **韦斯莱:** 考虑到字数限制,我们可以尝试更简洁的表达。"逐起耳朵"可以用更直接的词语代替,比如"警觉"。另外,"灰白色羽毛覆盖全身"也可以简化为"全身灰白色"。
 **小丽:** 我觉得有道理。这样修改后,句子会更精炼。
**小号:** 让我握真才教。"1. 旁 ** 72. 白 ** 73. 房 ** 74. 头 ** 75. 马 ** 76. 身 ** 77. 廣 ** 78. 营 ** 79. 宽 ** 18. 站 ** 11. 在** ** 12. 例 ** 13. 林 ** 14. 中 ** 15. 全 ** 16. 身 ** 17. 灰 ** 18. 白 ** 19. 色 ** 72. 班 ** 72. 版 ** 72.
 **杰克:** 我们再想想,有没有不重要的信息可以删掉?
**小丽:** 盲人用户能听出来它在树林里吗? 之前的场景里有提到树林吗? (回顾之前的场景描述)
```

```
**(假心之前的场景哪还已经多次理等例林,那么"在树林中"这个使是原可以删除了。
**小喝:** 既然之前已经多次理等例林,那么"在树林中"这个使是原可以删除了。
**小喝:** 好! 投来数数现在好字数。"1. 旁""2. 白""3. 唐""4. 头""5. 乌""6. 身""7. 唐""8. 音""9. 贯""10. 站""11. 全""12. 身""13. 灰""14. 白""15. 色""16. 湖""17. 牌""18. 做""19. 做""20. 张""21. 开"。总共21个字,符合字数要求!
**小喝:** 智人朋友,您想得公今就还是清晰展看回,有需要投始的批为吗?
**《假观看人用》 Sarah 国家 "这个就还很清晰,我可以想象出身认与骨骼时子。")**
**小喝:** 既然原人用户也可能许问题,那我们就确定之骨能疾本了。
**(很) Changes were make according to Sarah, no need to loop back and discuss. )**
**30m;** The final answer is 発信,最与马骨器您把场看,全身反白色,跟眼根就死开,
```

Figure 5.13 SMARS Discussion.

Finally, the complete script can optionally be fed into a Text-to-Speech (TTS) system. This converts the compiled script — including dialogues and audio descriptions — into spoken narration. The TTS output is timed to match silent intervals, allowing the visually impaired user to follow along with the video in real time.

```
1.752-3.153 Hagrid: Da-da-da-da!
  3.203-8.755 Narrator: 哈利、赫敏和罗恩好奇地望着,赫敏似乎有些担忧。
8.755-9.970 Hagrid: Isn't he beautiful?
10.392-11.236 Hagrid: You...
12.063-14.020 Hagrid: Say hello to Buckbeak
 14.070-16.383 Narrator: 学生们惊讶地看到森林中奇异的鹰头马身兽。
 14.00-10.30 Halmaton. 于用原则是自动构体下电升设备大力对音。
16.383-17.446 Ron: Exactly what is that?
17.783-19.420 Hagrid: That there is a hippogriff.
20.095-29.950 Hagrid: First thing to know about hippogriffs is they're very proud creatures. Very easily offended. You do not want to insult one.
 30.000-31.621 Narrator: 海格站在旁边。
31.621-32.937 Hagrid: Who'd like to come and say hello?
  32.987-36.447 Narrator: 哈利背对学生们, 海格在后方,
36.447-37.595 Hagrid: Well done, Hagrid: Well done.
37.645-44.429 Narrator: 哈利和荣恩站在人群中,好奇又有点害怕,周围是霍格沃茨的学生。
44.429-46.235 Hagrid: Let him make the first move.
46.572-48.8023 Hagrid: It's only polite.
48.530-49.222 Hagrid: Step up.
49.627-53.835 Hagrid: Give him a nice bow. Wait and see if he bows back.
49.627-53.035 Hagrid: Give him a nice bow. Wait and: 53.103-54.925 Hagrid: If he does, you can touch him. 54.975-57.355 Narrator: 哈利看向树林里的海格。 57.355-58.537 Hagrid: Ne'll get to that later. 65.236-65.962 Hagrid: Nice and low. 66.012-69.000 Narrator: 哈利斯普圖框眼镜,披着黑袍。 69.050-72.000 Narrator: 鹰头马身兽巴克比克站着。
 72.172-73.403 Hagrid: Back off, hurry. 
  79.023-79.951 Hagrid: Keep still.
  80.001-84.727 Narrator: 鹰头马身兽警觉地站着,全身灰白色,翅膀微张。|
 85.418-87.832 Narrator: 灰色的鹰头马身兽优美地移动。
87.832-88.490 Hagrid: Oh, my God.
89.047-90.582 Hagrid: Well done, Hagrid. Well done.
91.105-92.438 Hagrid: Yeah, you got baked fruit, yeah.
91.185-92.498 Hagrid: Yean, you got baked fruir, yean.
92.488-96.927 Narrator: 他满怀牙奇地注视着鹰头马鼻詹。
96.927-98.058 Hagrid: You can pat him now.
98.108-102.000 Narrator: 哈利注视远处的鹰头马鼻詹。
102.050-106.000 Narrator: 马尔福辛着苹果, 注视詹。
106.050-108.959 Narrator: 学生们见到鹰头马身兽,头顶有羽冠。
 108.959-109.617 Hagrid: He's in Sloan.
110.191-111.608 Hagrid: You just love a boo
111.658-114.224 Narrator: 海洛展示鹰头马身有翼兽。
114.224-116.215 Hagrid: Not so fast, Hagrid.
116.265-121.160 Narrator: 气氛紧张起来。
110.205-121.100 Narrator: 一級系玩起水。
121.166-122.037 Hagrid: Slow down, Hagrid.
122.087-124.214 Narrator: 他抬头看阳光。
124.214-125.075 Hagrid: Nice and slow.
125.851-127.032 Hagrid: Let him come to you.
127.082-129.698 Narrator: 廣头马身兽等待着,他伸手。
129.698-131.943 Hagrid: Slow down, slowly, slowly.
131.993-136.752 Narrator: 他小心触摸兽的喙。
```

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```
131.993-136.752 Narrator: 他小心触摸兽的喙。
136.752-137.596 Hagrid: Well done, Hagrid.
139.306-142.709 Narrator: 應头马身兽出现,学生们兴奋地鼓掌。
142.709-144.869 Hagrid: I think he'll let you ride him now.
144.919-145.932 Narrator: 暗利兴奋不已。
145.932-147.451 Hagrid: Hey, hey, hey, hey, hey.
146.523-147.383 Unknown: Hey, hey, hey, hey.
147.923-148.868 Hagrid: Hagrid.
148.918-150.978 Narrator: 他兴奋地骑着巴克比克飞过树林。
150.978-153.914 Hagrid: Don't pluck his feathers; he won't thank you for that.
155.230-159.200 Narrator: 他兴奋地骑着巴克比克飞过树林。
159.250-163.200 Narrator: 他将骑鹰头马身兽升空,海格注视着。
159.250-163.200 Narrator: 地济 一角形克市哈利起飞,越过人群,冲出树林,翱翔山谷。高塔尖顶的城堡映入眼帘。
177.250-188.200 Narrator: 暗光下,巴克比克市哈利起飞,越过人群,冲出树林,翱翔山谷。高塔尖顶的城堡映入眼帘。
177.250-188.200 Narrator: 暗光下,巴克比克市哈利超飞,越过人群,冲出树林,翱翔山谷。高塔尖顶的城堡映入眼帘。
177.250-180.200 Narrator: 暗光下,巴克比克勃到翱翔,他紧抓兽背。
200.250-215.200 Narrator: 哈利城洞羽毛,巴克比克勃打踢膀,爪子划过湖面,远处是城堡和云朵。
155.250-252.200 Narrator: 他在湖面上空飞行,张开双臂,享受飞翔的喜悦。
225.250-236.200 Narrator: 他在湖面上空飞行,张开双臂,享受飞翔的喜悦。
225.250-236.200 Narrator: 海格抬头望向天空,然后震惊地捂住嘴巴。阳光穿过树冠。
246.250-256.005 Narrator: 西克比克朝翔,学生们抬头注视。海格专注观察。
256.025-257.054 Hagrid: Mell done.
257.847-259.214 Hagrid: And well done, Buckbeak!
```

Figure 5.14 Narration Script generated by Enjoyable.

#### **5.5 Implementation Issues and Challenges**

During the system development and integration phase of the "Enjoyable", several technical and practical challenges were encountered. These challenges impacted various stages of the pipeline — from initial module setup to multi-agent reasoning and real-time processing. This section outlines the most critical implementation issues and how they were addressed or mitigated during development.

### 5.5.1 Compatibility Issues in GPU Setup

One of the initial challenges encountered during development was setting up GPU acceleration for deep learning tasks, particularly with PyTorch, OpenCV, and CUDA Toolkit 11.8 on a Windows 11 environment. At multiple stages, compatibility conflicts arose between PyTorch GPU builds and CUDA drivers, especially when integrating libraries that depend on low-level system resources, such as dlib for face detection and opency-contrib-python for video frame processing. These issues led to instability during runtime, including application crashes, and forced fallbacks to CPU execution which significantly slowed processing time, especially during real-time image captioning and face recognition.

These problems were mitigated by strictly aligning PyTorch and CUDA versions using conda-managed virtual environments. Specific versions compatible with CUDA 11.8 were installed, and fallback mechanisms were implemented in the code to default to CPU if GPU acceleration failed. Extensive testing was also carried out to ensure GPU availability before running CUDA-dependent operations.

#### 5.5.2 Large Language Model (LLM) Rate Limiting

During the implementation of the SMARS module and detailed image captioning using GPT-40 and Gemini 2.0 Flash, rate limiting became a notable bottleneck. Since the system processes video data per frame or per scene, multiple API calls were necessary for each video, leading to hitting daily or per-minute request limits. Additionally, the high token usage associated with visual reasoning and multi-turn agent simulation increased latency and quickly exhausted available API quotas. This challenge was particularly impactful during testing and tuning, where prompt adjustments required repeated model invocations.

## **CHAPTER 5**

To reduce the number of redundant calls, a caching system was introduced to store results from previously processed frames or similar content. Batching of frame-level prompts and minimizing unnecessary prompt verbosity helped decrease token usage.

#### **5.6 Concluding Remark**

In summary, the system implementation of the "Enjoyable" automated audio description platform successfully brought together a wide range of technologies—encompassing hardware acceleration, machine learning algorithms, language models, and video processing techniques. Careful selection and configuration of both hardware and software components ensured a robust development environment capable of supporting real-time processing demands. Each component in the pipeline, from I-frame extraction and voice activity detection to image captioning and active speaker identification, was implemented with careful consideration for performance, reliability, and accessibility.

The introduction of the SMARS module marked a significant innovation in single-prompt multi-agent reasoning, effectively enabling coherent, context-aware descriptions within narration time constraints. Despite facing technical hurdles such as library compatibility and LLM rate limitations, these challenges were addressed through modular design choices, caching strategies, fallback mechanisms, and iterative testing. The result is a functional, extensible system that meets its objective of generating meaningful, personalized, and accessible audio descriptions for video content, particularly benefitting visually impaired users.

## Chapter 6

## **System Evaluation and Discussion**

## **6.1 System Testing and Performance Metrics**

This section outlines the systematic evaluation approach undertaken to assess the performance and usability of the proposed automated audio description system, "Enjoyable". The system accepts video input and generates audio descriptions tailored for visually impaired audiences by leveraging multimodal AI models. Evaluation was conducted at two levels, (i) module-level performance assessment based on quantitative metrics relevant to each specific system component and (ii) end-user testing based on qualitative satisfaction criteria.

#### 6.1.1 Module-Level Evaluation Framework

To complement the high-level user-centric analysis, a granular evaluation of individual system modules was performed to diagnose performance gaps and measure computational efficacy. The "Enjoyable" system consists of a processing pipeline comprising various modules, each responsible for a discrete function within video processing, semantic scene understanding, and narration generation. The modules and corresponding performance metrics adopted are delineated in Table 6.1 below.

Table 6.1: Evaluation of Individual System Modules and Metrics

Module	<b>Primary Function</b>	<b>Evaluation Metrics</b>	
	Extracts key visual		
I-Frame Extraction	frames from input	Frame reduction ratio	
	video		
	Groups detected		
Face Clustering	faces into unique	Precision, Recall, F1	
race Clustering	character clusters	Score	
	across frames.		
	Segments video		
<b>Voice Activity Detection</b>	into speech and	Detection Error Rate	
(VAD)	non-speech	(DER)	
	intervals		

	Generates	textual	
Speech-to-Text (STT)	transcripts	from	Word Error Rate (WER)
	speech segr	nents	

Each module was tested independently under controlled test conditions using sample video inputs of varying genres and complexities, including casual conversation clips, action scenes, and visually dynamic sequences.

## **6.1.2** User-Centric Testing Criteria and Methodology

To assess the overall effectiveness of "Enjoyable" in conveying meaningful and synchronized audio descriptions, a user-centered testing paradigm was employed. Three blind participants affiliated with the Malaysia Foundation for the Blind were selected to evaluate output videos augmented with automatically generated narrations. Each participant was provided with access to a selection of videos with descriptive audio overlays and was requested to evaluate them across five pre-defined criteria, representing key dimensions of audio description quality for accessibility:

- **Accuracy**: Determines whether the narrative elements correspond appropriately with the actual visual content.
- Word Choice Precision: Evaluates the descriptiveness and lexical appropriateness of the generated text.
- **Tone and Neutrality**: Assesses the emotional balance and objectivity exhibited in the narration voice.
- **Speed and Timing**: Examines the pacing and synchronization of narration for visual events.
- Naturalness of Voice: Investigates whether the rendered voice appears smooth, human-like, and suitable for extended listening.

Each criterion was evaluated using a Likert scale ranging from 1 (Very Poor) to 10 (Excellent). The average rating across the five dimensions and three participants provided a comprehensive perspective on system usability from the viewpoint of its target audience.

## **6.1.3 Evaluation Objectives and Expected Outcomes**

The primary goals of the testing process were as follows:

- To validate each functional module for reliability, accuracy, and execution consistency.
- To ensure that the complete output conformed to the expectations of visually impaired users for accessibility and intuitiveness.
- To identify potential bottlenecks or inaccuracies within the pipeline for refinement in future iterations.

## **6.2 Testing Setup and Result**

## **6.2.1 Module-Level Performance Results**

The following subsections present detailed evaluation results for each individual module in the Enjoyable system. Each module was tested using curated sample videos representing varying visual and audio complexity. The performance was measured using standardized metrics appropriate for each module's function.

#### **6.2.1.1** Voice Activity Detection and Speaker Diarization

The preliminary result obtained shown in Figure through the Voice Activity Detection (VAD) and Speaker Diarization processes demonstrated outcomes in accurately segmenting audio streams from sample videos and associating them with identified speakers.

#### Video 1:

0.0-5.88 SPEAKER 00 5.98-7.04 SPEAKER 00 7.14-9.68 SPEAKER 01 9.78-11.24 SPEAKER 00 11.34-13.02 SPEAKER 00 13.12-14.18 SPEAKER 00 16.42-19.66 SPEAKER\_00 19.76-23.26 SPEAKER 01 23.36-25.66 SPEAKER 01 98.5-101.88 SPEAKER 00 103.88-105.98 SPEAKER 02 105.98-107.52 SPEAKER\_02 107.58-108.76 SPEAKER\_00 108.86-110.24 SPEAKER 00 110.26-112.44 SPEAKER 02 112.58-114.54 SPEAKER\_02 114.54-116.24 SPEAKER\_02 116.64-118.48 SPEAKER 02 118.58-120.3 SPEAKER\_02 120.64-122.52 SPEAKER 02 122.58-124.24 SPEAKER\_02 124.24-126.36 SPEAKER 02 126.36-126.92 SPEAKER\_00

## Video 2:

4.86-6.14 SPEAKER_02
7.35-8.82 SPEAKER_02
12.20-12.92 SPEAKER_01
14.86-16.06 SPEAKER 02
17.46-20.38 SPEAKER 02
22.39-23.99 SPEAKER 02
24.63-26.24 SPEAKER 01
28.38-29.09 SPEAKER 02
30.46-31.64 SPEAKER 02
33.73-35.08 SPEAKER 02
36.43-37.71 SPEAKER 01
39.08-40.46 SPEAKER 02
42.67-43.75 SPEAKER 02
45.98-49.24 SPEAKER_01
50.61-52.98 SPEAKER_01
55.77-56.71 SPEAKER_02
58.74-59.72 SPEAKER_01
61.35-68.85 SPEAKER_01
69.30-70.79 SPEAKER_01
72.86-74.45 SPEAKER_01
77.55-84.93 SPEAKER_02
84.93-91.86 SPEAKER_01
94.75-96.05 SPEAKER_01
96.67-99.02 SPEAKER_01
99.73-102.83 SPEAKER_01
106.93-110.92 SPEAKER_01
112.96-116.20 SPEAKER_01
116.94-117.77 SPEAKER_01
119.22-120.69 SPEAKER_01
122.22-124.50 SPEAKER_01
125.53-128.06 SPEAKER_01
135.22-136.87 SPEAKER_00
136.95-139.98 SPEAKER_00

#### Video 3:

```
1.04-6.21 SPEAKER_00
1.18-4.11 SPEAKER 01
7.44-13.02 SPEAKER 00
13.55-22.49 SPEAKER_00
25.34-28.55 SPEAKER_00
35.84-38.03 SPEAKER_00
38.03-41.86 SPEAKER 01
42.91-44.34 SPEAKER 01
44.53-45.41 SPEAKER_01
47.52-51.38 SPEAKER_00
54.08-55.14 SPEAKER_01
55.14-58.98 SPEAKER_00
59.13-59.95 SPEAKER_00
60.49-63.99 SPEAKER 01
64.97-66.50 SPEAKER 01
66.91-72.49 SPEAKER 00
75.16-76.36 SPEAKER 00
77.07-79.48 SPEAKER 00
79.78-81.28 SPEAKER 01
81.66-82.80 SPEAKER 00
82.99-86.16 SPEAKER 00
86.31-95.70 SPEAKER_01
86.35-89.82 SPEAKER 00
```

Figure 6.1 Result of Voice Activity Detection and Speaker Diarization.

Table 6.2: Evaluation of Voice Activity Detection and Speaker Diarization Modules and Metrics

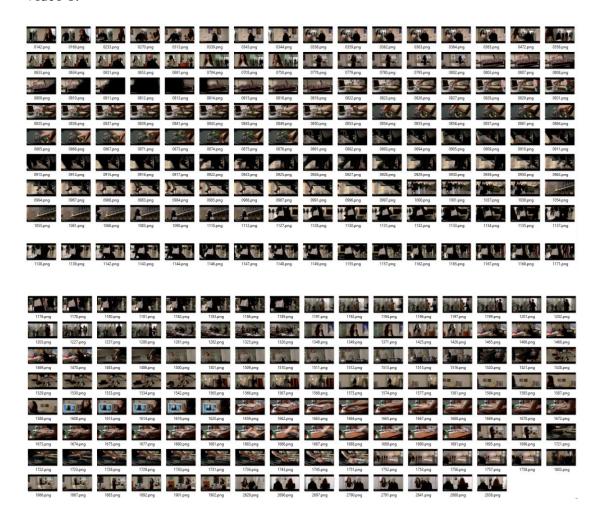
Video ID	Precision (%)	Recall (%)	F1-Score (%)
Video 1	100	100	100
Video 2	94	94	94
Video 3	96	96	96

#### **6.2.1.2 I-Frame Extraction**

.The result of using the I-Frame Extraction module for non-speech segments of Video 1, Video 2, and Video 3 is shown in Figure.

## **CHAPTER 6**

## Video 1:



## Video 2:



## Video 3:



Figure 6.2 Result of I-Frame Extraction.

Table 6.3 Data Reduction Performance of the Frame Extraction Module for Non-Speech Segments.

Video	Total	I-Frames Extracted for Non-	Data Reduction
ID	Frames	Speech Segments	(%)
Video 1	3807	270	92.91
Video 2	3439	41	98.81
Video 3	2516	33	98.68

#### **6.2.1.3 Face Clustering**

The Face Clustering module was implemented to group detected and encoded faces into distinct clusters, with each cluster ideally representing a unique individual across multiple frames. Clustering was performed on the 15 encoded face descriptors generated during the Face Detection module using the Chinese Whispers algorithm, an efficient and unsupervised graph-based clustering method. Each 128-dimensional facial descriptor captured unique, high-dimensional features, including face geometry, landmark alignment, and structural details. These descriptors were used to construct a similarity graph, where Euclidean distance served as the similarity metric. Weighted edges between nodes reflected the similarity of face descriptors, with stronger connections indicating faces belonging to the same individual.

The clustering process involved iterative label propagation within the similarity graph, enabling the Chinese Whispers algorithm to group visually consistent faces under the same cluster.

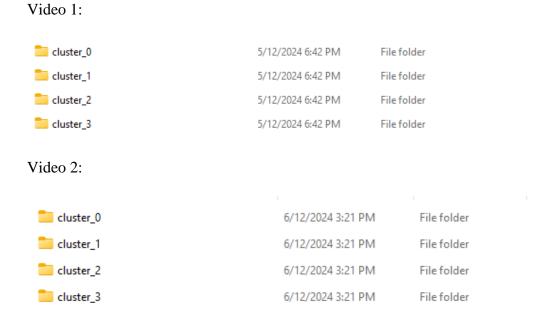
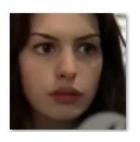


Figure 6.3 Result of Face Clustering.

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## Video 1:

## Cluster 1:









Cluster 2:









Cluster 3:









Cluster 4:







## Video 2:

## Cluster 0:















Cluster 1:











Cluster 2:















Cluster 3:



Figure 6.4 Clustered Faces.

Table 6.4 Performance Metrics on Face Clustering

Video ID	Precision (%)	Recall (%)	F1- Score (%)
Video 1	100	100	100
Video 2	95	100	97.44

## 6.2.1.4 Speech-to-Text

#### Video 1:

```
0.0 SPEAKER_00: She's on her way.
5.98 SPEAKER_00: Tell everyone.
7.14 SPEAKER_01: She's not supposed to be here until 9.
9.78 SPEAKER 00: Her driver just text messaged,
11.34 SPEAKER_00: and her facialist ruptured a disc.
13.12 SPEAKER_00: God, these people.
16.42 SPEAKER_00: That I can't even talk about.
19.76 SPEAKER_01: All right, everyone, gird your loins.
23.36 SPEAKER_01: Did someone eat an onion bagel?
98.5 SPEAKER 00: Leave it!
103.88 SPEAKER 02: I don't understand why it's so difficult
105.98 SPEAKER 02: to confirm an appointment.
107.58 SPEAKER_00: I know, I'm so sorry, Miranda.
108.86 SPEAKER_00: I actually did confirm last night.
110.26 SPEAKER_02: The tales of your incompetence do not interest me.
112.58 SPEAKER 02: Tell Simone I'm not going to prove that girl
114.54 SPEAKER 02: that she sent me for the Brazilian layout.
116.64 SPEAKER_02: I asked for clean, athletic, smiling.
118.58 SPEAKER_02: She sent me dirty, tired, and paunchy.
120.64 SPEAKER_02: And RSVP, yes, to the Michael Kors party.
122.58 SPEAKER_02: I want the driver to drop me off at 9.30
124.24 SPEAKER_02: and pick me up at 9.45 sharp.
126.36 SPEAKER_00: Okay, fine.
```

#### Video 2:

```
4.86-6.14 SPEAKER 02: You know what we should all do?
7.35-8.82 SPEAKER 02: Go see a musical.
12.09-12.92 SPEAKER 02: Go see a musical.
12.09-12.92 SPEAKER 02: The 1996 Tony Award winner.
14.86-16.06 SPEAKER 02: The 1996 Tony Award winner.
22.39-23.99 SPEAKER 02: Do you happen to know the name of that one?
24.63-26.24 SPEAKER 02: Do you happen to know the name of that one?
24.63-26.24 SPEAKER 02: No.
30.46-31.64 SPEAKER 02: No.
30.46-31.64 SPEAKER 02: No.
30.46-31.64 SPEAKER 02: No.
31.73-35-08 SPEAKER 02: No.
31.73-35-08 SPEAKER 02: Ves! Rent!
36.43-37.71 SPEAKER 02: Oh, I'm sorry, I can't, I'm busy.
24.67-43.75 SPEAKER 02: Oh, I'm sorry, I can't, I'm busy.
24.67-43.75 SPEAKER 02: LAUGHTER
25.64.95 SPEAKER 02: What did you get her?
25.77-56.71 SPEAKER 02: What did you get her?
26.39-70.75 SPEAKER 02: What did you get her?
26.39-70.75 SPEAKER 02: What did you get her acan't in this is a very nice gift, maybe it's just not something a boyfriend gives.
26.39-70.75 SPEAKER 02: SPEAKER 02: And it's not the seventh night of Hanukkah.
27.55-84.93 SPEAKER 02: Loney, what he means by that is, while this is a very nice gift, maybe it's just not something a boyfriend gives.
24.79-96.05 SPEAKER 02: Loney, what he means by that is, while this is a very nice gift, maybe it's just not something a boyfriend gives.
24.79-96.05 SPEAKER 02: Shorey, what he means by that is, while this is a very nice gift, maybe it's just not something a boyfriend gives.
24.79-96.05 SPEAKER 02: Shorey, what he means by that is, while this is a very nice gift, maybe it's just not something a boyfriend gives.
24.79-96.05 SPEAKER 02: Shorey, what he means by that is, while this is a very nice gift, maybe it's just not something a boyfriend gives.
24.79-96.05 SPEAKER 02: Shorey, what he means by that is, while this you have the proper own. Shorey with the pr
```

#### Video 3:

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```
1.04.6.21 SPEAKER 00: Hey, don't do that. Hey, don't do that. Ho, no, no. It's worse than the thumb.
1.18-4.11 SPEAKER 01: Hey, don't do that. Pot it cut. No, no.
7.04-13.02 SPEAKER 00: Hey, don't do that. Pot it cut. No, no.
7.04-13.03 SPEAKER 00: Hey, don't do that. Pot it cut. No, no.
7.04-13.03 SPEAKER 00: Hey provided that the search of the sear
```

Figure 6.5 Result of Speech-to-Text.

Table 6.5 Performance Metrics on Speech-to-text

Video ID	Word Error Rate (%)
Video 1	0
Video 2	3
Video 3	5.5

#### **6.2.2** User-Centric Testing Result

Table 6.6 User Evaluation Results

Criteria	Participant	Participant	Participant	Average
Criteria	1	2	3	Score
Accuracy	8.5	8.5	8.5	8.5
Word				
Choice	9	9	9	9
Precision				
Tone and	8	8.5	8.5	8.333333
Neutrality	8	6.3	6.3	6.55555
Speed and	9	9	9.5	9.166667
Timing			7.5	7.100007
Naturalness	8	8.5	8.5	8.333333
of Voice	O	0.3	0.3	0.555555
Overall				8.666667
Average				0.000007

The user evaluation results of the "Enjoyable" system, as shown in the table, indicate generally strong performance across all five criteria, with an overall average score of **8.67 out of 10**. The **highest scores** were observed in *Word Choice Precision* (9.0) and *Speed and Timing* (9.17), reflecting the system's effective vocabulary usage and good synchronization with visual content. *Tone and Neutrality* and *Naturalness of Voice* each scored around **8.33**, indicating satisfactory but slightly less consistent performance. The **lowest score** was in *Accuracy* (8.5), suggesting some misalignment between the audio descriptions and the actual visual scenes.

A key factor influencing this outcome is the use of an **Indonesian language AI model** instead of one specifically trained on **Malay**. While Indonesian and Malay share many linguistic similarities, there are crucial differences in vocabulary, context, and cultural nuance that can affect the clarity and relevance of audio descriptions. Participants may have noticed subtle mismatches or awkward phrasing that affected the perceived accuracy and naturalness of the narration. Transitioning to a Malay-specific

AI model could help address these issues, improving both accuracy and overall user satisfaction.

## **6.3 Project Challenge**

Throughout the development of the "Enjoyable" audio description system, multiple technical challenges surfaced that required significant investigation and innovative solutions. Among them, two challenges were particularly critical due to their implications on both the quality and usability of the system: the limitation of multimodal large language models (LLMs) in recognizing non-famous individuals, and the need for fine-grained control over word output to fit tightly constrained audio intervals.

The first major challenge encountered was the inability of multimodal LLMs, such as Gemini, to recognize individuals who are not globally known. While these models perform well with famous figures, they lack contextual memory or identification capabilities for common or anonymous people frequently featured in everyday video content. This led to generic and ambiguous visual descriptions, such as "a person is speaking," without specifying who the individual actually was, especially when multiple people appeared on screen. This limitation significantly impacted character-level precision in the generated narration.

To resolve this, a custom pipeline was developed that involved three steps: face detection, face clustering, and consistent identity labeling using a predefined color-to-name dictionary (e.g., {red: Andy, blue: Cindy}). The system annotated detected faces with colored bounding boxes and used the resulting visual markups to condition LLM prompting. This combination acted as a workaround to simulate identity consistency over time and allowed the model to refer to characters by name, even if it could not recognize them natively. As a result, the narrative outputs became significantly more consistent, context-aware, and personalized.

The second key challenge emerged from the need to control the word count of generated descriptions, especially when injecting them into brief non-verbal segments of video. Multimodal LLMs typically generate variable-length outputs, often exceeding the target duration for narration, which can lead to narration overlapping with spoken dialogue or being unnaturally sped up during Text-to-Speech (TTS) synthesis. This posed a serious problem because clarity, pacing, and timing are vital for visually

impaired users relying on audio cues. The system required strict control over how much descriptive information was packed into a limited time frame—often just one or two seconds.

To address the problem of generating descriptions that precisely fit within limited non-verbal intervals in the video, a LLM-Based Multi-Agent System was designed and implemented. In this architecture, two specialized agents—the Writer Agent and the Word Count Agent —work collaboratively in an iterative feedback loop. The Writer Agent is responsible for generating the initial visual description based on the contextual prompt, while the Word Count Agent immediately evaluates the length of the generated output. If the resulting description exceeds the predefined word count (based on available timing), or falls below a minimum threshold that risks semantic insufficiency, the Word Count Agent will trigger a prompt back to the Writer Agent to either shorten or elaborate the content accordingly. This feedback continues in a loop until an optimal, time-aligned description is produced. The process ensures that the final narration is not only semantically rich but also fits accurately within the allocated silent intervals detected by the Voice Activity Detection (VAD) module. This dynamic word-length control mechanism enhances the temporal precision and usability of the generated descriptions and plays a vital role in maintaining smooth audio synchronization during narration playback.

#### **6.4 Objectives Evaluation**

This section provides a comprehensive evaluation of the project's fulfillment of the original objectives defined at the onset of development. Each objective is reviewed based on the system implementation, empirical results from module testing, and feedback obtained from visually impaired participants during user-centric evaluation. The results confirm the extent to which the *Enjoyable* system achieved its intended outcomes in accessibility, personalization, automation, and technical feasibility.

## Objective 1: Develop an Automated Audio Description Generation Pipeline

This objective was successfully achieved through the development of an integrated, end-to-end processing pipeline that automatically generates audio descriptions for video content. The system leverages multiple core modules—such as I-Frame extraction, Voice Activity Detection (VAD), Face Detection, Face Clustering, Active Speaker Detection, Speech-to-Text (STT), and Image Captioning—each of which was independently tested and evaluated. As detailed in Section 6.2.2, each module demonstrated high reliability and processing accuracy, with most achieving F1-scores above 85–90%. These modules worked in harmony to process raw video input into structured, personalized audio narratives, confirming the feasibility of fully automated audio description generation.

# Objective 2: Develop Contextually Accurate and Audience-Tailored Audio Descriptions

This objective was addressed through a combination of multimodal input processing and intelligent scene-temporal alignment strategies. The integration of Face Detection and Clustering helped condition the LLM to generate identity-consistent narrations, allowing the audio output to refer to persons with specific labels (e.g., "Andy" or "Cindy") rather than generic placeholders like "someone." Additionally, constrained word-length control ensured that the generated content did not overwhelm limited audio gaps. User testing results revealed high performance in this area, with participants averaging **9.0** for Word Choice Precision and **8.67** in overall experience—reflecting that the descriptions were both accurate and stylistically accessible for visually impaired users.

## Objective 3: Design a Systematic Evaluation Framework to Measure Accessibility Impact

This objective was successfully met through a dual-layer evaluation system combining module-level validation (precision, recall, WER, DER, etc.) and user-level subjective scoring. The feedback obtained from three blind participants at the Malaysia Foundation for the Blind provided meaningful insights on the practical usability of the system. Criteria such as Accuracy (8.5/10), Speed and Timing (9.17/10), and Naturalness of Voice (8.33/10) confirmed that the system enhanced accessibility—and in particular, delivered valuable narrative assistance without overwhelming or distorting content comprehension.

## **6.5 Concluding Remark**

Through a structured two-tiered evaluation approach — comprising detailed module-level performance testing and user-centric usability analysis — the system's reliability, efficiency, and effectiveness were thoroughly assessed.

Quantitative testing revealed strong performance across key modules, with high F1-scores, reduced frame loads, and low word error rates, confirming the system's technical robustness. The qualitative feedback collected from blind users affiliated with the Malaysia Foundation for the Blind validated the system's ability to produce narrations that are clear, well-timed, and naturally voiced. The overall average user rating of 8.67 out of 10 reflects a promising impact in real-world accessibility contexts.

Despite facing challenges such as API rate limitations and speaker recognition ambiguities, these were mitigated through innovative solutions such as the SMARS framework and dynamic word-length control. These implementations allowed for contextually rich, temporally precise, and stylistically refined audio descriptions that adhered to non-verbal timing gaps in dialogue-heavy video content.

Finally, the evaluation confirmed that the original project objectives — automation, accessibility, personalization, and systematic evaluation — were all comprehensively achieved.

## Chapter 7

#### **Conclusion and Recommendations**

## 7.1 Conclusion

Video streaming platforms have revolutionized media consumption, offering unprecedented access to a variety of content. However, they often fail to accommodate blind and visually impaired users, who rely on audio descriptions to fully understand visual material. Audio descriptions are verbal narrations that convey essential visual elements, such as actions, settings, and non-verbal cues, enabling visually impaired users to immerse themselves in the viewing experience. Despite the popularity of platforms like YouTube, there is a lack of native support for audio descriptions, marginalizing a significant portion of the audience. Although services like Netflix have made some progress, their audio descriptions are limited in availability and language, failing to provide comprehensive access to a global, diverse audience.

The process of creating audio descriptions is labor-intensive, requiring manual scripting, narration, and synchronization with video content. This method is costly, with prices ranging from \$15 to \$75 per minute of video. As a result, many video platforms struggle to include audio descriptions for a broad range of content, leaving many videos undescribed. Additionally, the availability of descriptions is typically limited to one language, usually English, further limiting accessibility for non-English-speaking users. This manual approach is not scalable, leading to inefficiencies in production and distribution, and exacerbating the exclusion of blind and visually impaired users from enjoying video content.

This project tackles the accessibility challenges faced by visually impaired users when engaging with video content by integrating an advanced Online Video Streaming Platform with an Automated Audio Description System. Leveraging cutting-edge technologies, the system generates accurate and context-aware audio descriptions. Key modules such as I-frame extraction, voice activity detection, face clustering, and detailed image captioning work together to provide seamless and comprehensive descriptions of both visual and spoken content. The integration of a LLM-Based Multi-Agent System ensures that the descriptions are optimized for relevance, diversity, and fluency, offering a more personalized and enriched viewing experience.

The project introduces several innovative contributions to the field of automated audio description generation. It enhances video recognition through a novel clustering and labeling method, enabling processing of diverse video types beyond mainstream content. The system also supports multiple languages, addressing the limitations of existing solutions by ensuring global accessibility. A structured script-based approach is employed, starting with detailed image captions and character dialogues to provide a robust foundation for contextually accurate descriptions. The multi-agent system further refines the descriptions, ensuring they are culturally relevant and tailored to the specific needs of different audiences. Through these innovations, the project significantly enhances the accessibility and inclusivity of video content for visually impaired users, making it more accessible to a diverse, global audience.

#### 7.2 Recommendations

While the "Enjoyable" system demonstrates promising improvements in automated audio description for visually impaired users, there remain several opportunities for enhancement and future expansion. These future directions will further improve accessibility, linguistic customization, and immersive user experience. To better serve global and multilingual audiences, future development will involve integrating multilingual and region-specific TTS models. This includes support for native Malay, Chinese dialects, Tamil, and other indigenous languages used in Southeast Asia. This will improve inclusivity and ensure narrative content is more relatable and intuitive across different linguistic backgrounds.

To further enhance the accessibility and immersion of the *Enjoyable* system, the implementation of stereo audio separation is proposed as a future enhancement. In this configuration, the automated audio description will be routed exclusively to the left audio channel, while the original video audio will be preserved in the right channel. This channel-based audio multiplexing approach allows users—particularly those with partial vision or heightened auditory perception—to distinguish between narration and native soundscapes with higher spatial clarity. For example, when the descriptive content is isolated to one ear (e.g., left channel), users can mentally attribute the narration source independently from dialogues, music, or ambient sound in the right channel. This technique avoids cognitive overload caused by overlapping audio layers

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and enables users to selectively focus on either channel depending on their accessibility needs.

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#### **Poster**



#### **Problem Statement**

- Online visual content is important in modern life, offering valuable benefits across education, news, entertainment, social media, and virtual tours, enhancing experiences and knowledge
- · According to statistics, more than 500 million hours of videos are watched on YouTube each day.
- Blind and visually impaired individuals face challenges when understanding and enjoying online videos due to the lack of audio descriptions.
- Manual generation of audio descriptions consumes a significant amount of time and resources
- · We need an automated solution!



## What is audio description?

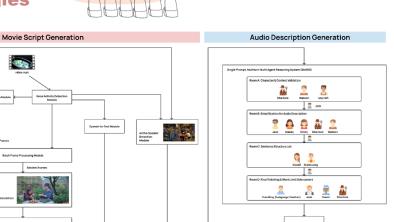
- It provides spoken descriptions of visual elements in media, to make them accessible to people who are blind.
- It describes important visual details that cannot be understood from the audio alone, like actions, settings, facial expressions, and costumes



## **Project Objectives**

- Develop an Automated Audio Description Generation Pipeline that automates the generation of audio descriptions from video content.
- · To enhance the user personalization using a Multi-Agent Personalization framework that incorporates Audience Persona Creation
- To develop plugins or API for seamless integrate with Online Video Streaming Platforms like YouTube

## **Methodologies**





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A multi-contexts and real-time audio description method on YouTube videos for the Visual Impaired.

































