FoS(Focus on Speaking)

Group 12

https://github.com/april2901/ai-assistant-for-presentation

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Abstract— In modern business and educational environments, meetings and presentations are key means of communication, and their importance continues to grow. However, presenters and participants often experience cognitive overload as they manage speech delivery, script reference, slide transitions, and time management while also trying to follow complex discussion flows and decisions. This leads to topic drift, loss of focus, and unclear outcomes. To address these issues, this project proposes an **LG display–linked real-time meeting AI prompter** that extends a traditional teleprompter into an active, context-aware assistant. The system listens to participants' speech, interprets the meeting context in real time, and presents the next required information—such as agenda structure, current topic, decisions, and action items—on LG displays, while providing a private coaching dashboard for the presenter or host. Core features include real-time STT, flexible speech-to-script matching, keyword omission detection, a real-time feedback dashboard, agenda visualization, decision and action-item extraction, and fact-check widgets. A Meeting Summary Report summarizes key topics, ideas, decisions, and action items after the session. Through these functions, the project aims to improve both individual presentation quality and overall meeting efficiency, and to explore integration within LG's smart office ecosystem.

Keywords—Speech Recognition, Real-Time STT, Script Synchronization, Real-Time Teleprompter, Slide Automation, Agenda Tracking, Presentation Feedback, Human-Computer Interaction, Meeting Intelligence

Role Assignment -

Roles	Name	Task description and etc.
User	Daeun Lee, Minhyuk Jang	Tests the prototype from the user's perspective, focusing on interface usability, speech synchronization accuracy, and overall user experience. Provides qualitative feedback for refinement.
Customer	LG Electronics (Assumed Client)	Defines requirements for smart office presentation support software and evaluates its feasibility for integration with LG's webOS-based business ecosystem.
Software Developer	Sangyoon Kwon, Dohoon Kim (Backend), Hyeyun Kwon, Seohyun Kim (Frontend)	Responsible for system implementation including backend server logic, database management, API communication, and frontend interface development. Ensures real-time synchronization and stable slide automation.
Development Manager& UI Designer	Daeun Lee, Minhyuk Jang	Oversees project planning, documentation, and communication between development teams. Manages task allocation, schedule tracking, design of interface and quality assurance.

I. INTRODUCTION

A. Challenges in Modern Presentations and Meetings

In professional and academic settings, presentations and meetings have become essential tools for sharing ideas and making decisions. Yet presenters and facilitators must simultaneously handle speech delivery, script reference, slide transitions, and time tracking, while participants struggle to follow complex discussion flows and remember key points. This often causes interruptions in the presentation flow, omission of important content, topic drift during discussions, and unclear conclusions, ultimately reducing communication efficiency.

B. Limitations of Existing Solutions

Existing tools such as teleprompters, timers, and subtitle features mainly provide static information or simple transcription. They are useful for displaying text but do not actively intervene in real time to prevent topic drift or support decision alignment. Many AI-based meeting services focus on post-meeting summaries or minutes, which help review what happened but do not improve the efficiency of the meeting while it is in progress. As a result, core issues such as cognitive overload and live meeting inefficiency remain unresolved.

C. Project Goals and Proposed Solution

This project proposes an integrated support system that covers preparation, live delivery, and post-meeting feedback. The core concept is an **"LG Display–Linked Real-Time Meeting AI Prompter."** The system continuously listens to meeting audio, analyzes the semantic context of each utterance, and surfaces what the meeting needs next: agenda structure, current topic, decisions, action items, and fact-check results. At the same time, it provides a private teleprompter and coaching dashboard for the presenter or host, helping with script tracking, omission alerts, pacing, and gesture suggestions. The ultimate goal is to reduce cognitive load and improve meeting focus and decision-making speed.

D. Dual-Screen Architecture for LG Displays

To support both individual coaching and shared awareness, the system adopts a dual-screen architecture. **Screen 1 (Presenter Dashboard)** is shown on the presenter's personal device (e.g., LG Gram) and provides a private teleprompter, omission alerts, pace metrics, and AI suggestions. **Screen 2 (Shared Meeting Board)** is shown on LG signage, LG One:Quick, or conference room TVs and visualizes slides, the real-time agenda map, decisions and action items, and fact-check widgets for all participants. This separation allows the presenter to receive rich guidance without overwhelming the audience, while participants share a clear view of where the meeting is and what has been decided.

II. Requirements

A. Before Presentation

This phase focuses on the preparation process before a presenter begins their presentation. Users interact with the system to upload materials, create a script, and adjust content to fit the presentation environment.

*1) **Slide-Script Alignment Recognition and Consulting***

When a user uploads a PPTX or PDF file, the system extracts textual and visual elements using python-pptx and the Google Vision API (OCR). A multimodal LLM processes these elements to interpret textual and graphical contexts, generating a coherent draft script for each slide.

- Acceptance Criteria

• OCR text recognition accuracy ≥ 95%
• Draft script grammatical accuracy ≥ 95%
• Slide–text coherence ≥ 95%

- Input & Output

• Input**:** Presentation file (.pptx, .pdf)
• Output**:** Structured text/image metadata, draft script (3–6 sentences per slide)

- Constraints

• Max 100 slides
• Max file size 200 MB
• Max 10 images per slide
• Supported formats: PPTX, PDF

*2) **Presentation Environment-Specific Script Adjustment***

The system adjusts the script's vocabulary level, tone, and length based on the audience type (non-expert / practitioner / expert), target presentation time, and speaker's pace. Using TensorFlow.js-based vision models, audience facial expressions are analyzed every 3 seconds to compute a "focus score" (0–100). When time is running short or audience engagement decreases, an LLM provides real-time summaries or interactive remarks.

- Acceptance Criteria

• Script adjustment time ≤ 5s
• Focus detection accuracy ≥ 85%
• Timing deviation ≤ ±5%
• Script fluency ≥ 90%

- Input & Output

• Input**:** Audience type, target duration, speech rate(WPM), tone, audience video data
• Output**:** Adjusted script (.txt), recommended timing table, real-time teleprompter feedback

- Constraints

• Camera ≥ 720p
• Max 10 audience members detectable
• LLM request frequency ≤ 1 per 10 s
• Presentation ≤ 60 min

B. Live Delivery Phase

This phase involves real-time interaction between the user and the system during the actual presentation. The system detects the presenter's speech and performs instant support tasks like synchronization, feedback, and suggestions.

*1) **Real-time Teleprompter(Meeeting Mode Support)***

The system transcribes the presenter's and participants' speech in real time using Google Cloud or Naver Clova STT and aligns it with the prepared script via KoSentence-BERT semantic similarity. The current sentence is visually highlighted on the teleprompter. In meeting mode, the STT pipeline can capture speech from multiple people in the

room as a single mixed audio stream; downstream modules operate on this combined transcript.

- Acceptance Criteria

• Speech–script synchronization delay ≤ 1 s
• Highlight accuracy ≥ 95%
• Alignment deviation ≤ 1 sentence

- Input & Output

• Input**:** Microphone audio (.wav, .mp3, ≥16 kHz), script file (.txt)
• Output**:** Real-time highlighted script text and STT logs

- Constraints

• Session length ≤ 60 min
• STT throughput ≥ 50 words/s
• API cost ≈ $0.006/min

*2)**Automatic Slide Transition***

Through the Microsoft PowerPoint COM API, slides are automatically advanced when the script reaches predefined transition points.

- Acceptance Criteria

• Transition latency ≤ 0.5 s
• Transition accuracy ≥ 95%
• Failure rate ≤ 5%

- Input & Output

• Input**:** Slide file (.pptx), predefined transition IDs
• Output**:** Automatically advanced slide display

- Constraints

• Max 100 slides
• Requires PowerPoint 2016 or later

*3)**Flexible Speech-to-Script Matching***

The system maintains synchronization even when speech deviates lexically from the script. Primary matching uses KoSentence-BERT vector similarity (threshold ≥ 0.8), followed by secondary LLM-based contextual verification if necessary.

- Acceptance Criteria

• Matching success rate ≥ 90%
• False match rate ≤ 5%
• Matching latency ≤ 0.3 s per sentence

- Input & Output

• Input**:** STT transcript, script text
• Output**:** Matched sentence ID and highlight position

- Constraints

• LLM call limit ≤ 1 per second
• STT buffering ≤ 5 s

*4)**Key Content Omission Detection***

Detects missing predefined key phrases using cosine similarity (threshold ≥ 0.75) and alerts the presenter within 3 seconds.

- Acceptance Criteria

• Detection precision ≥ 95%
• False alarm ≤ 5%
• Alert delay ≤ 2 s

- Input & Output

• Input**:** STT transcript, key phrase list (≤50)
• Output**:** Omission alert and log file

- Constraints

• Max 500 sentences compared
• STT buffering interval: 5 s

*5)**Real-time Script Reconstruction***

When omissions are detected, missing segments are asynchronously sent to an LLM that generates supplementary sentences within 5 seconds. Approved sentences are integrated into the script in real time.

- Acceptance Criteria

• Supplement generation ≤ 5 s
• Contextual coherence ≥ 90%
• Integration success ≥ 90%

- Input & Output

• Input**:** Missing sentence ID, context text, LLM API key
• Output**:** Supplementary sentence, updated script

- Constraints

• Max 10 API calls per minute
• Sentence length ≤ 100 characters

*6)**Real-time Presenter Dashboard***

The system visualizes metrics such as words per minute (WPM), voice volume, and progress rate using the Web Audio API, and applies TensorFlow Lite models for basic emotion recognition (e.g., tension/calmness).

- Acceptance Criteria

• Data refresh ≤ 2 s
• Emotion inference error ≤ ±5%
• Visualization accuracy ≥ 95%

- Input & Output

• Input**:** Audio stream, STT logs
• Output**:** Live dashboard showing progress, WPM, emotional state

- Constraints

• Sampling rate ≥ 16 kHz
• Dashboard latency ≤ 1 s

*7)**Speech Gesture Suggestions***

Before the presentation, the system analyzes slide images with a multimodal LLM to detect key visual elements (graphs, photos, diagrams) and map them to related keywords. During speech, when such keywords appear in STT, gesture icons (e.g., pointing, emphasis) are displayed on the teleprompter.

- Acceptance Criteria

• Suggestion latency ≤ 2 s
• Gesture relevance ≥ 85%
• Recognition accuracy ≥ 90%

- Input & Output

• Input**:** Slide images, script keywords
• Output**:** Gesture icons displayed on teleprompter

- Constraints

• Max 3 visual mappings per keyword
• Display duration: 2–3 s

*8)*Real-Time Context and Intent Tagging

The system analyzes STT text in real time and classifies each utterance according to its intent and type. This information is used as the foundation for the agenda map, decision board, and fact-check triggers.

- Supported Tags

• General comment
• Idea proposal
• Negative/Positive feedback
• Decision
• Request / Action Item
• Question
• Fact-check request

- Acceptance Criteria

• all utterances receive at least one tag from the predefined set
• Intent classification accuracy ≥ 85% on test samples

- Input & Output

• Input: STT transcript segmented into utterances
• output: Tagged utterance stream (text + tag(s) + timestamp)

- Constraints

• Classification latency ≤ 0.5s per utterance
• the tagging model must operate within the WebSocket round-trip time budget

9)Real-Time agenda Map

Using the tagged utterance stream, the system builds a real-time agenda map to prevent topic deviation and improve shared awareness. Each emerging topic is registered as a node in a network graph displayed on **Screen 2**

- Acceptance Criteria

• Screen 2 displays a live STT log and its mapping to agenda nodes
• Utterances tagged as “Idea,” “Decision,” or “Action Item” are grouped under appropriate agenda nodes
• Nodes are color-coded by agenda type
• The active topic is clearly highlighted
• Node detail view shows STT snippet and timestamp

- Input & Output

• Input**:** Tagged utterance stream (from Requirement 8), semantic embeddings
• Output**:** Real-time agenda network graph on Screen2

- Constraints

• Graph update interval ≤ 2s
• Max 30 agenda nodes per session

10) Dual-Screen synchronizatino between Presenter Shared Display

The system keeps Screen 1(presenter Dashboard) and Screen 2 synchronized while respecting privacy boundaries

- Acceptance Criteria

• Slide transition latency between Screen 1 and Screen 2 ≤ 0.5 s
• No private elements (teleprompter text, omission alerts, AI suggestions) appear on Screen 2

- Input & Output

• Input**:** Slide control events, layout state, synchronization messages
• Output**:** Consistent view of the current slide and agenda state on Screen 2

- Constraints

• WebSocket synchronization interval ≤ 1s

11) Real-time Decisions and Action Item Widget

The system captures utterances tagged as decision or action item and surfaces them in a dedicated widget on screen 2, often placed below or beside the agenda map.

- Acceptance Criteria

• Detection coverage for decision-like and action-like utterances ≥ 90% on test scenarios
• New items appear in the list within 2 seconds of the utterance

- Input & Output

• Input**:** Tagged utterance stream (Decision / Action Item tags)
• Output**:** Real-time decision & action-item list on Screen 2

- Constraints

• Max 100 items per session
• Each item stored with its text content and timestamp, with a link to the original transcript segment

12) Real-Time Fact-Check and Research Widget

When the system detects a **Fact-check request** tag, it triggers a lightweight research pipeline (RAG or web search) and surfaces the result on Screen 2

- Acceptance Criteria

• Successful keyword extraction for at least 90% of fact-check requests
• Research results displayed within 5 seconds

- Input & Output

• Input**:** Tagged utterance stream (Fact-check request tag), knowledge base or web search API
• Output**:** Fact-check result widget with short answer and source link(s)

- Constraints

• Max 30 fact-check queries per session
• Each query result limited to brief, conference-friendly summaries

C. Post-Presentation Phase

This phase involves the user receiving feedback on their presentation and conducting a Q&A session after the presentation has concluded.

1) Q&A Auto-Response

In Q&A mode, the system uses Retrieval-Augmented Generation (RAG) to search a pre-built database and generate 2–3 candidate answers, each referencing supporting slides or pages.

- Acceptance Criteria

• Answer generation ≤ 5 s
• Relevance score ≥ 0.85
• Slide reference accuracy ≥ 98%

- Input & Output

• Input**:** Question (speech/text), presentation DB (JSON)
• Output**:** 2–3 candidate answers with referenced slides

- Constraints

• Max 20 questions per session
• Max 300 tokens per answer
• RAG cosine similarity ≥ 0.8

2) Presentation and Meeting Analysis Report

After the meeting or presentation, the system automatically analyzes collected data (speech logs, agenda map, decisions, action items, and referenced research results) and generates a Meeting Summary Report. This report covers time management, speech habits, content delivery, and meeting outcomes such as major topics, ideas, decisions, action items, and external facts referenced during the discussion.

- Acceptance Criteria

• Report generation ≤ 10 s
• Analysis accuracy ≥ 95%
• User satisfaction ≥ 4.2/5.0

- Input & Output

• Input**:** Speech logs, slide transitions, emotion data, agenda map, decision/action-item list, fact-chcek logs
• Output**:** Analysis report (.html,.pdf)

- Constraints

• Max presentation time: 60 min
• Max 100,000 words processed

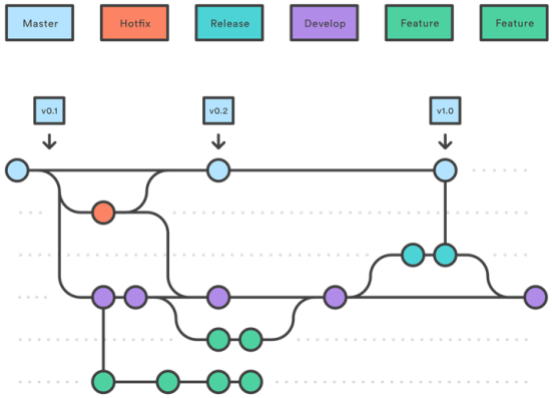
III. VERSION CONTROL SYSTEM

To manage source code and documentation, a version control system based on **Git** was established.
A public repository named **"ai-assistant-for-presentation"** was created on **GitHub**.

The following source code and documents have been uploaded to the repository:

- All backend and frontend source code
- Shared documents including this file (*project_documentation.md*) and other design files
- Configuration files and project-related assets

All team members (development, project management, and UI/UX design) have been granted access to the repository. For efficient GitHub management, the team will adopt a **branch management strategy** as illustrated in the diagram below.



IV. Development Environment

A. Choice of Software Development Platform

1) Platform Selection and Rationale

The project adopts a web-based client–server architecture as its primary development and deployment platform. The web environment ensures platform-independent accessibility without requiring users to install additional software, while providing seamless integration with cloud-based APIs such as Google Cloud Speech-to-Text and large-language-model (LLM) services. Modern web technologies, including WebSockets, Web Audio API, and TensorFlow.js, enable the implementation of essential real-time features such as live teleprompting and synchronized feedback dashboards.

2) Programming Languages and Rationale

- Backend: Python (version 3.11 or higher) using FastAPI for asynchronous API and WebSocket communication. Python is the de facto standard for AI and machine learning workflows, offering a mature ecosystem that includes python-pptx, sentence-transformers, and google-cloud-speech libraries.
- Frontend: TypeScript/JavaScript (Node.js 20 or higher) with React 18.2. JavaScript is the only natively supported browser language and is indispensable for client-side interaction. React combined with TypeScript supports modular,

maintainable UI components and ensures type safety.

3. *Cost Estimation*
   The estimated total development cost is approximately USD 30.00, as summarized below:

- Hardware: Personal laptops (MacBook Air/Pro) – USD 0.00
- Software and IDE: Visual Studio Code (free), Cursor Pro (USD 20 per month)
- Cloud Services and APIs: AWS Free Tier, GCP STT and Vision API, OpenAI GPT Realtime Mini (approx. USD 10)

4. *Development Environment Details*

- Operating Systems: Windows 11, macOS 14 (Sonoma)
- IDEs: Visual Studio Code (v1.90 or higher), Cursor (v1.7 or higher)
- Version Control: Git (v2.39 or higher) and GitHub public repository ("ai-assistant-for-presentation")
- Backend Stack: Python 3.11+, FastAPI 0.110+, PostgreSQL 16 (Render hosted)
- Frontend Stack: Node.js 20.10+, npm 10.2+, React 18.2+, TypeScript 5.2+
- Major Libraries: python-pptx, sentence-transformers, websockets, TensorFlow.js, Web Audio API
- Hardware Resources: Three personal laptops (two MacBook Airs, one MacBook Pro) used for development and testing.

5. *Use of Commercial Cloud Platforms*

- Google Cloud Platform (GCP): Utilized for Speech-to-Text and Vision OCR services to enable high-accuracy transcription and slide text extraction. Services operate within free-tier quotas.
- Amazon Web Services (AWS): EC2 for backend deployment, S3 for file storage, RDS for database hosting, and Route 53 for domain management. Free-tier services are used for prototype deployment and demonstration.

B. Software in Use

Several existing software solutions and research studies were referenced in designing the system:

- PromptSmart (VoiceTrack): A commercial teleprompter offering real-time voice tracking. The proposed system extends its capabilities by adding semantic matching, omission detection, and automated slide control.
- Microsoft PowerPoint (Live Subtitles): Provides speech transcription but lacks contextual synchronization with scripts and automated slide transitions.

These benchmarks highlight the project's improvements in real-time adaptivity and AI-driven presentation assistance.

C. Task Distribution

Role	Members	Responsibilities
Backend Development	Sangyoon Kwon, Dohoon Kim	System architecture design, FastAPI server and WebSocket implementation, database schema (PostgreSQL), AI logic integration (STT, LLM, BERT), cloud deployment (AWS, Render)
Frontend Development	Hyeyun Kwon, Seohyun Kim	React-based UI implementation, client-side state management, real-time dashboard (Web Audio API), teleprompter interface, client-side AI (TensorFlow.js)
Project Management & UI Design	Daeun Lee, Minhyuk Jang	Project planning and scheduling, UI/UX design (Figma), documentation and VCS management, user testing and feedback analysis

V. Specification

1) Requirement 1. Real-Time Teleprompter

This process involves a tightly coordinated real-time loop between the client (web browser) and the server (Python backend) through WebSocket communication.

1. **Client Initialization**
   When the user presses "Start Presentation," the React frontend requests microphone access using navigator.mediaDevices.getUserMedia() and establishes a secure WebSocket (wss://) connection to the backend API server.
   The Web Audio API initializes an AudioContext and ScriptProcessorNode to capture raw audio chunks.
2. **Client-Side Real-Time Audio Streaming**
   The ScriptProcessorNode continuously triggers onaudioprocess events (e.g., every 500 ms).
   Each raw audio buffer (16-bit PCM) is sent to the backend server through WebSocket.
3. **Server-Side STT and Synchronization**
   The backend receives the audio chunks and streams them to the Google Cloud Speech-to-Text API.
   The API returns *interim* (fast but less accurate) and *final* (slower but more accurate) transcripts.
   When a final sentence is received, it is appended to the full transcript of the session.
   The backend then calls the FlexibleSpeechMatcher service to locate the new currentSentenceIndex within the user's script.
4. **Server Broadcast**
   The server immediately sends a WebSocket message to

the client:
{ "action": "UPDATE_TELEPROMPTER", "index": currentSentenceIndex }

5. **Client Update**
   The frontend WebSocket listener receives the message and updates the highlighted text accordingly, scrolling the teleprompter to the current index in real time.

2) Requirement 2. Flexible Speech-to-Script Matching

This algorithm implements a hybrid matching mechanism combining fast vector similarity and fallback large-language-model (LLM) validation.

Pseudocode Overview

```
SIMILARITY_THRESHOLD = 0.75
SEARCH_WINDOW = 5
LLM_VALIDATION_THRESHOLD = 0.60

Function FindCurrentPosition(fullTranscript, scriptSentences, lastIndex):
   latestTranscript = GetLastNWords(fullTranscript, 10)
   transcriptVector = KoSentenceBERT.encode(latestTranscript)

   searchStart = lastIndex
   searchEnd = min(lastIndex + SEARCH_WINDOW, len(scriptSentences))
   searchWindow = scriptSentences[searchStart : searchEnd]

   bestMatchIndex = -1
   highestSimilarity = 0

   # Step 1: Fast vector similarity
   for index, sentence in enumerate(searchWindow):
      similarity = CosineSimilarity(transcriptVector, sentence.vector)
      if similarity > highestSimilarity:
         highestSimilarity = similarity
         bestMatchIndex = searchStart + index

   # Step 2: Omission check
   if bestMatchIndex > lastIndex:
      CheckForOmissions(lastIndex, bestMatchIndex, scriptSentences)

   if highestSimilarity >= SIMILARITY_THRESHOLD:
      return bestMatchIndex

   # Step 3: LLM fallback validation
   if highestSimilarity >= LLM_VALIDATION_THRESHOLD:
      prompt = CreateLLMPrompt(latestTranscript, searchWindow)
      AsyncCallLLM(prompt, HandleLLMResult)
      return bestMatchIndex


   return lastIndex
```

This process ensures low latency while maintaining semantic accuracy through adaptive matching.

3) Requirement 3. Key Content Omission Detection

This function integrates directly with the flexible matching process.
When a skipped section is detected, the system checks whether any omitted sentence contains a predefined *key phrase* and alerts the presenter.

1. **Database Preparation**
   When users upload a script, each sentence is marked with a boolean attribute isKeyPhrase (true / false) and stored in the database.
2. **Server-Side Omission Detection**
   When FindCurrentPosition() identifies a new bestMatchIndex greater than lastIndex + 1, the server calls CheckForOmissions().
3. **Omission Logic**
   If skipped sentences are found between the two indices, each is inspected for isKeyPhrase == true.
   When detected, the omitted sentence index is flagged, and the following message is broadcast:
   { "action": "OMISSION_DETECTED", "index": omittedSentenceIndex }
4. **Client Notification**
   The frontend highlights the corresponding part of the script (e.g., flashing red or adding a border) to visually warn the presenter in real time.
   Simultaneously, an asynchronous script-reconstruction task (Requirement 4) is triggered.

4) Requirement 4. Real-Time Script Reconstruction

This asynchronous process generates short “bridging sentences” whenever key content omissions are detected, ensuring smooth narrative flow without latency in the main loop.

1. **Asynchronous Trigger**
   CheckForOmissions() launches HandleOmissionAsynchronously() in a separate asynchronous task.
2. **Prompt Generation for LLM**
   The function combines three inputs:
   (a) the omitted sentence, (b) the current context sentences, and (c) an instruction prompt such as:
   “You are a presentation coach. The presenter accidentally omitted '[omittedSentence]' and is now moving to '[contextSentences]'. Please generate one short, natural bridging sentence in Korean that connects these topics smoothly.”
3. **LLM API Invocation**
   The backend asynchronously requests an LLM (e.g., GPT or Gemini) to generate the bridging sentence.
4. **Response Delivery**
   Upon success, the server sends the message:

{ "action": "SCRIPT_SUGGESTION", "text": llm_generated_sentence }

5. **Client Interface Behavior**
   The React frontend displays the generated sentence in the AI Suggestion section as an alert message: “Do you approve this suggestion?” with **Accept (#0064FF)** and **No (#E0E6EA)** buttons.
   If the user selects *Accept*, the updated script is applied, and a small alert “💡 Update complete” appears at the top-right corner.