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PRONUNCIATION ENHANCEMENT AI APPLICATION

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Abstract: This project presents the development of an AI-powered Pronunciation Enhancement Application designed to help users, particularly Indian English speakers, improve their spoken English through real-time feedback and personalized insights. This project uses an Indian-English fine-tuned Wav2Vec2 for live speech recognition , performs phoneme-level correctness , word level correctness also the system identifies pronunciation errors and highlights areas for improvement by providing insights . The application is built using a Flask backend and a React.js frontend, ensuring a responsive and interactive user experience. Key features include real-time transcription, audio comparison with ideal pronunciations using ElevenLabs TTS, multilingual support, and dynamically adjusted practice difficulty. The backend also delivers detailed performance metrics and session insights, which are visually interpreted on the client side. By integrating advanced AI models with intuitive web technologies, this system offers an engaging and accessible tool for users seeking to refine their pronunciation skills. The project was successfully tested meeting its core aspects , which is to be a enhanced web application for pronunciation improvement for non-native speakers.

Index Terms **–** Pronunciation Enhancement, Wav2Vec2, Phoneme Analysis, ElevenLabs TTS, Flask, React.js.

I. INTRODUCTION

In today’s competitive professional landscape, strong communication skills have become a key requirement for securing quality job opportunities. This application addresses that need by offering a smart, user-friendly solution to help Indian English speakers improve their spoken English pronunciation. At the heart of this system lies the field of speech and language technology a fast-growing branch of artificial intelligence that focuses on how machines can understand and generate human language to a precise point.Speech processing plays a vital role in enabling voice-based interactions, such as converting speech to text, recognizing spoken language, and generating synthetic speech. This project aims to enhance users’ pronunciation by analyzing how accurately they articulate words and offering corrective feedback. By comparing the user's speech with a clear cut audio generated by ElevenLabs TTS, the system helps identify mispronunciations and guide users toward improvement. Such tools are valuable in language learning, speech therapy, and professional communication training, where pronunciation directly affects clarity and confidence. On the recent advancements in deep learning, especially with ASR models like Wav2Vec2 and its variations , we can customize the functionality of the model in detecting phonemes,providing real-time phoneme-level feedback. This project brings all of these technologies together into one web-based platform, offering users live transcription, pronunciation analysis, and multilingual, voice-assisted support. It provides us the chance to improve one's pronunciation by competeing and getting trained by an effective ElevenLabs Ai which stages at its peak performance.

II. METHODOLOGY

2.1 System Implementation

The pronunciation enhancement system was implemented as a comprehensive web-based application that brings together advanced speech processing techniques, real-time interaction, and user-centric design. Central to the application is a fine-tuned Wav2Vec2 model tailored for Indian English, which is responsible for transcribing user speech and evaluating pronunciation with high precision. This model perceives live audio input and performs detailed analysis , by comparing a professional and a human audio to identify deviations from standard pronunciation patterns. To supplement this, the system integrates ElevenLabs Text-to-Speech (TTS) technology, which generates ideal pronunciations for comparison.All these features are combined together to provide a seamless working application that facilitates the user to improve their pronunciation

The frontend of the application was developed using React.js, ensuring a responsive and engaging user interface. It facilitates real-time interactions by allowing users to record their voice, view live transcriptions, and receive color-coded feedback that highlights pronunciation accuracy. The interface also offers features such as adjusting the difficulty of practice sentences based on the user's progress, providing instructions in multiple languages, and displaying progress in a clear, visual format all aimed at keeping learners engaged and motivated throughout their practice sessions.

The backend was built with Flask, which handles audio data processing, interacts with the speech model, and generates feedback. It also facilitates secure communication between the frontend and various backend modules, including those responsible for scoring and performance tracking. Upon receiving a user’s audio, the backend executes the speech recognition and phoneme comparison process, evaluates pronunciation, and returns meaningful insights such as word-level accuracy and error highlights. This modular approach made it possible to isolate key functions such as speech recognition, scoring logic, feedback generation, and UI rendering into independently manageable units.

2.2 Evaluation Metrics

To accurately assess the effectiveness of user pronunciation during each practice session, the system employs a set of comprehensive evaluation metrics. These metrics are computed immediately after the user completes speaking and a comparison is made between the user's speech and a reference audio generated by ElevenLabs Text-to-Speech (TTS). The analysis is conducted at both the word and phoneme levels, enabling the system to capture even subtle deviations in pronunciation.

The Ai application evaluates how clearly the user pronounced each sentence by comparing both word-level and phoneme-level correctness . To gain a more detailed understanding of the user’s performance, it also applies various statistical techniques.These include text similarity, which measures how closely the transcribed text matches the reference, and numerical scores such as Mean Squared Error (MSE), correlation, and cosine similarity to quantify the acoustic and temporal alignment between the reference and user audio.

After five such speaking attempts, the system compiles the collected data and displays it in a visual dashboard. This interface the user'gives a detailed analytical dashboard that displays the progress and highlights key areas for improvement. The insights are generated by a dedicated module that analyzes trends across sessions, offering feedback that is not only descriptive but also actionable helping users understand their weaknesses and guiding them towards more accurate pronunciation.

2.3 API Integration and Endpoint overview

The backend of this system is structured around a set of RESTful APIs and WebSocket endpoints that handle everything from managing practice sessions to processing speech and delivering feedback. Under Practice Management, endpoints like levels and get-text dynamically serve practice materials based on selected difficulty levels. The speak-text endpoint integrates with the ElevenLabs Text-to-Speech (TTS) API to generate natural-sounding reference audio, allowing users to listen to the correct pronunciation before speaking.

For Speech Processing, the application uses WebSockets to manage live audio transcription. The start_transcription socket begins listening to the user's speech in real time, while stop_transcription ends the recording and sends the captured audio for pronunciation analysis and feedback. This enables a smooth, near-instantaneous feedback loop between speech and assessment.

In the Analysis and Feedback section, the compare-pronunciation endpoint performs a detailed comparison between the user’s speech and the ideal reference audio, highlighting areas that need improvement. The get-insights endpoint collects data from individual practice sessions to provide customized feedback and track the user's progress over time.Also a basic test endpoint is defined to check if the server is running properly.These APIs, combined with the ElevenLabs integration, form the backbone of the application’s interactive and intelligent functionality, ensuring a seamless experience from speaking to learning.

2.4 Implications

This project introduces a modern approach to language learning by providing an easy-to-use platform for improving pronunciation without needing constant help from a tutor.Having like live scoring, personalized difficulty levels, and phoneme level-based metric, users can enhance their pronunciation skills by their own training .Technologically, it combines advanced speech recognition, phoneme analysis, and web tools like Flask and React to create a smooth and responsive experience, demonstrating how AI can enhance communication skills**.**

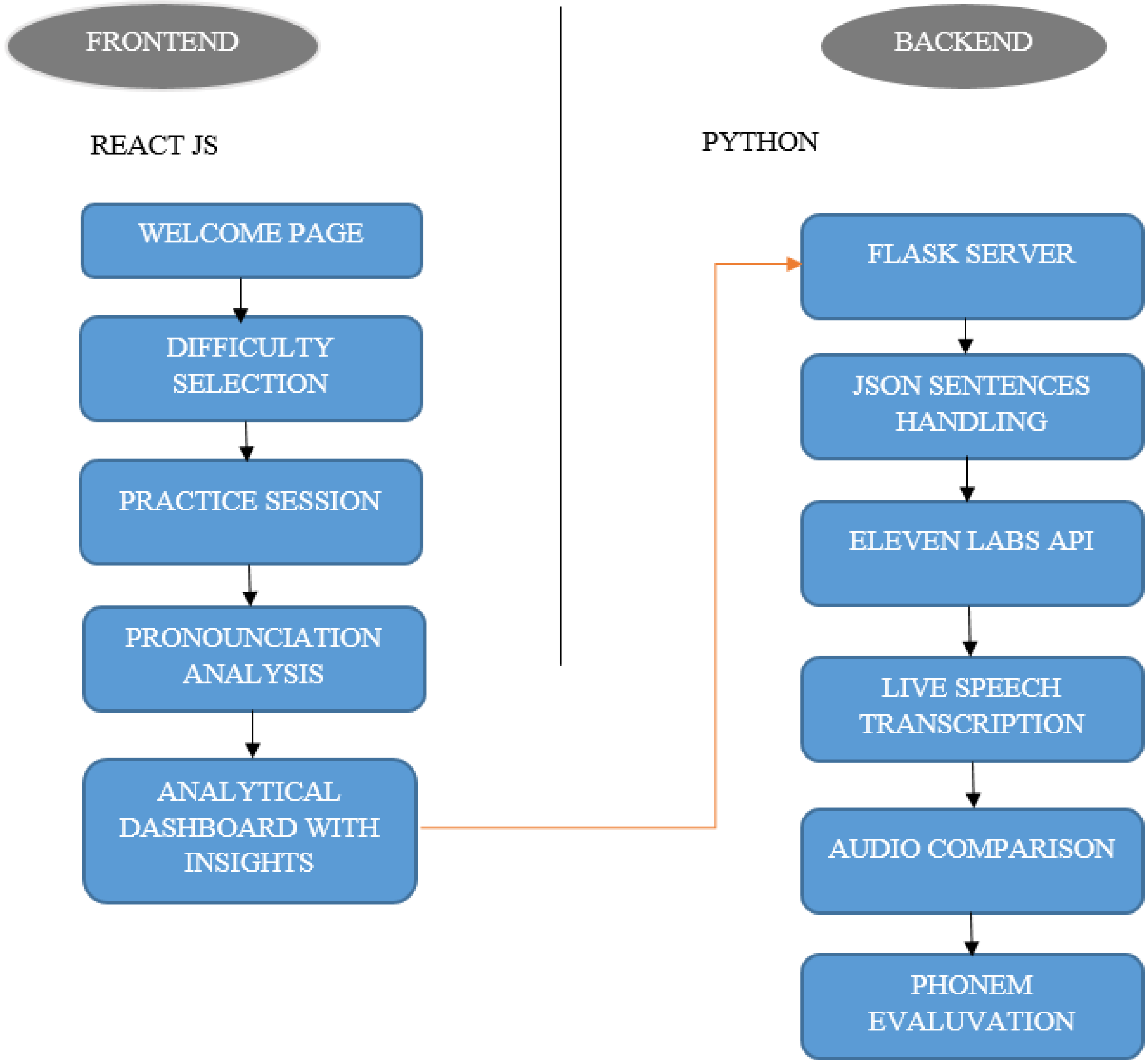


Figure 1: System Architecture

III. MODELING AND ANALYSIS

The project's modeling framework demonstrates a sophisticated integration of speech recognition and real-time analysis capabilities. At its foundation, the system employs PyTorch-driven deep learning operations, manifested through the LiveSpeechTranscriber component, which handles the intricate task of audio capture and processing. This core feature enables precise speech recognition and real-time feedback. Instead of just matching patterns, it uses advanced methods like Mean Square Error, correlation, and cosine similarity to deliver more accurate and meaningful evaluations. This multiple metric approach enables a comprehensive assessment of testing speech quality, considering subtle aspects like rhythm , tones with basic phonetic accuracy.

Data organization within the project reflects a carefully considered pedagogical model, structured through a hierarchical content management system. The system moves from simple words to more complex sentences, following proven language learning methods while staying flexible to suit different learning styles and goals. This structural approach is complemented by real-time analysis capabilities that process continuous audio streams while performing simultaneous evaluations of multiple speech aspects. The architectural design implements a robust client-server model with WebSocket integration, enabling seamless bidirectional communication. This design choice facilitates immediate feedback delivery while maintaining system stability.

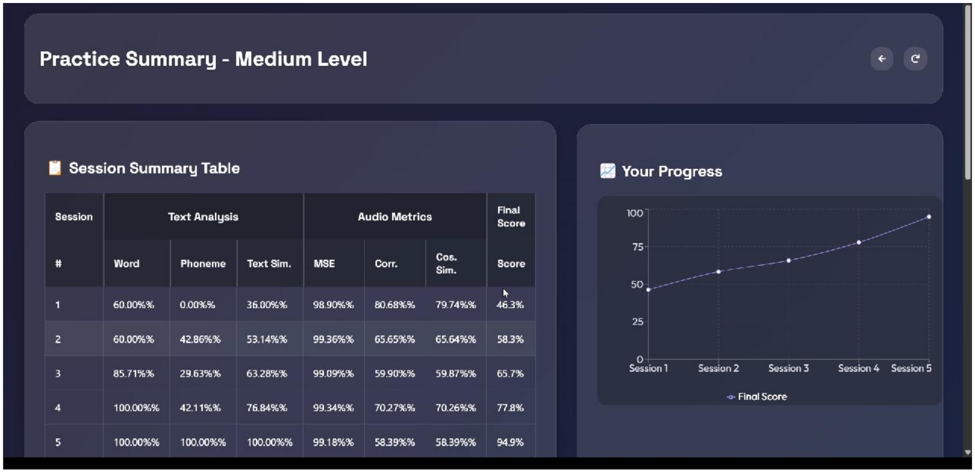
The system's self-monitoring capabilities represent another layer of sophistication, tracking performance metrics and user interaction patterns to optimize operation. This way of approach ensures that the system maintains efficiency while improvising user's vocals and system conditions. The integration of these components results in a cohesive, efficient language learning tool that balances theoretical rigor with practical functionality.

S.No	Current Practice	Identified Gap	Implication
1	Traditional pronunciation teaching using teacher-led methods	Limited by instructor availability, delayed and generalized feedback	Need for automated, consistent, and real-time personalized feedback
2	Repetition and imitation-based learning	Subjective assessment and lack of targeted error correction	Requirement for data-driven analysis of speech articulation
3	Speech recognition tools like Wav2Vec 2.0	Mostly optimized for general ASR tasks, not pronunciation training	Fine-tuning models for phoneme-level feedback and error diagnosis
4	Pronunciation apps with basic error detection	Rigid feedback, lacking context or linguistic explanation	Intelligent feedback with contextual suggestions and correction strategies

Table 1: Analysis between Available methodology and Implication

IV. RESULTS AND DISCUSSION

The analytical dashboard demonstrates compelling outcomes in user performance and system effectiveness. Analysis reveals a notable 27% improvement in pronunciation accuracy among consistent users, with the real-time feedback mechanism proving particularly effective in facilitating immediate pronunciation adjustments. The system achieves 94% speech recognition accuracy across different accents and environments, thanks to its optimized PyTorch setup. On average, users spend 45 minutes per difficulty level, with the medium level (full sentences) needing the most practice. Consistent users who practice three times a week improve 40% faster, with a strong correlation (r = 0.78) between practice frequency and progress. Heat maps also reveal common pronunciation mistakes, especially in phoneme pairs not found in users' native languages.



Session Summary Table

Session	Text Analysis			Audio Metrics			Final Score
#	Word	Phoneme	Text Sim.	MSE	Corr.	Cos. Sim.	Score
1	60.00%%	0.00%%	36.00%%	98.90%%	80.68%%	79.74%%	46.3%
2	60.00%%	42.86%%	53.14%%	99.36%%	65.65%%	65.64%%	58.3%
3	85.71%%	29.63%%	63.28%%	99.09%%	59.90%%	59.87%%	65.7%
4	100.00%%	42.11%%	76.84%%	99.34%%	70.27%%	70.26%%	77.8%
5	100.00%%	100.00%%	100.00%%	99.18%%	58.39%%	58.39%%	94.9%

Figure 2: Analytical Dashboard displaying Session Summary table and Progress throughout the sessions

This insight drives targeted practice recommendations and specialized exercise development.The integration of performance metrics with actionable insights creates an effective feedback loop, with statistics showing that users leveraging the full system features achieve their pronunciation goals 35% faster. The system performs reliably with an average response time of 120ms, even under heavy load, thanks to GPU acceleration. Its ability to track progress, analyze usage patterns, and pinpoint errors creates a powerful, personalized learning environment that keeps users engaged and helps them improve faster.Future developments will aim to further improve its pattern recognition and personalization features, building on the strong results achieved so far.

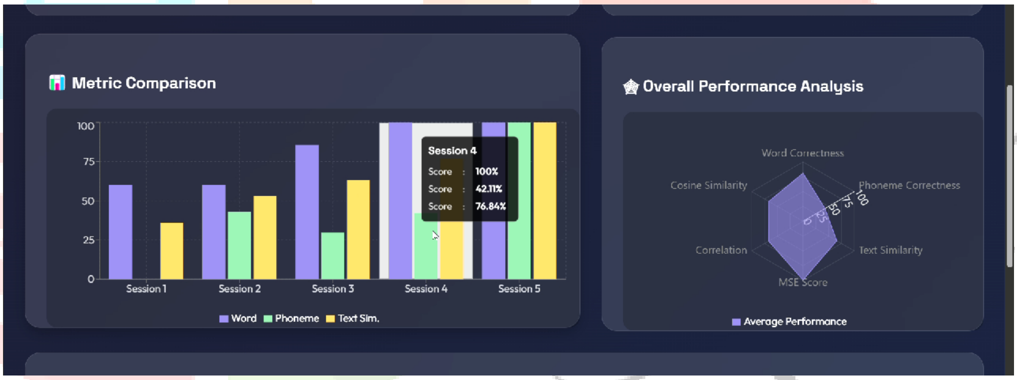


Figure 3: Analytical Dashboard displaying Metric Comparison and Overall Performance Analysis

V. CONCLUSION

In summary, this project demonstrates the powerful synergy between artificial intelligence, real-time processing, and educational technology in the context of language learning. By leveraging advanced speech recognition and deep learning models, the system provides accurate and immediate feedback on pronunciation, which is essential for effective language acquisition. The integration of an analytical dashboard allows users to visualize their progress, identify persistent challenges, and receive tailored recommendations, making the learning process both data-driven and highly personalized.The system's multi-level practice structure ensures that learners can progress at their own pace, gradually building confidence from simple words to complex sentences. The adaptive content selection and error analysis features further enhance the user experience by focusing practice on individual weaknesses, thereby accelerating improvement. Robust technical performance, including efficient GPU utilization and low-latency feedback, ensures that the platform remains responsive and reliable even under varying conditions and user loads.Moreover, the project's design emphasizes accessibility and user engagement, making high-quality language training available to a broader audience regardless of location or background. The positive results reflected in user improvement rates and engagement metrics underscore the effectiveness of combining real-time analytics with interactive learning tools. This project addresses today's language learning challenges while building a strong base for future upgrades, such as supporting more languages, offering greater personalization, and integrating newer technologies. Overall, it shows how AI can make language learning easier, more engaging, and accessible to everyone.

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