



IJRASET

International Journal For Research in
Applied Science and Engineering Technology



INTERNATIONAL JOURNAL FOR RESEARCH

IN APPLIED SCIENCE & ENGINEERING TECHNOLOGY

Volume: 14 **Issue:** IV **Month of publication:** April 2026

DOI: <https://doi.org/10.22214/ijraset.2026.78629>

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AI-Based Audio Transcription and Context Analysis for Candidate Interview Evaluation

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Abstract: This project develops an AI-based system for contextual analysis and audio transcription to help with the assessment of applicant interviews. The candidate's profile contains all interview materials, including audio, transcriptions, and AI-generated insights like summaries, keywords, and purpose. Candidates might be recognized by their distinct mobile number or email address. The administrator can create interviewer accounts, construct candidate profiles, and upload interview audio, while interviewers can safely study previous interview histories to make educated assessments. The solution addresses problems with multi-round interviews by centralizing data, improving organization, consistency, and decision-making, and providing practical experience in creating a secure, role-based web application combined with AI technology.

Keywords— Artificial Intelligence, Speech Recognition, Speech-to-Text Transcription, Natural Language Processing, Audio Processing, Context Analysis, Sentiment Analysis, Automated Interview Evaluation, Candidate Assessment, AI-Based Recruitment.

I. INTRODUCTION

AI-Based Audio Transcription and Context Analysis is a sophisticated technology that automatically assesses applicant interviews using state-of-the-art AI techniques. Most conventional interviewing methods rely on manual observation, which can result in time wastage, subjective evaluations, and inconsistent outcomes. An AI-based interview evaluation system generates objective, organized performance analysis using speech recognition, machine learning, and natural language processing. The technology captures the interview audio, translates speech into text, evaluates emotional tone, examines contextual meaning, and produces performance scores. Automatic Speech Recognition (ASR) technology is crucial for translating spoken responses into text. NLP algorithms analyze the response for coherence, question relevancy, syntax, and technical word usage after transcription. Sentiment analysis can identify emotional tones such as confidence, anxiety, optimism, or hesitation.

A. Background and Motivation

An organization's success is greatly impacted by the hiring process, however typical interview evaluations are frequently subjective and manual, which can result in bias and irregularities. Different evaluation criteria and handwritten notes may result in insufficient documentation during multi-round interviews, making it challenging to compare applicants. Interview audio may now be automatically transcribed and analyzed because of developments in artificial intelligence, speech recognition, and natural language processing (NLP). The goal of this project is to create an AI-based system that can translate speech into text and assess response quality using contextual analysis. The solution increases recruiting speed and transparency while facilitating objective, data-driven hiring decisions by collecting insights such as sentiment, communication clarity, and keyword relevancy.

B. Problem Statement

The majority of contemporary hiring procedures rely on manual interview evaluation, which involves making notes and using subjective judgment. This makes it impossible to compare applicants impartially and results in bias, inconsistency, and inadequate documentation. Different evaluation standards may lead to misleading judgments and unequal scoring in multi-round interviews. Simple speech-to-text systems don't examine context, sentiment, clarity, or keyword relevance; they just offer transcription. Managing massive amounts of interview data and effectively tracking applicant performance present additional hurdles for organizations. Therefore, an AI-based solution that combines sophisticated contextual analysis with audio transcription is needed.

To guarantee openness, effectiveness, and equity in the hiring process, the system should have role-based access restriction, centralized storage, multi-round monitoring, and structured evaluation reports.

C. Objectives

The main objectives of the initiative are:

- 1) To develop and put into use an AI-powered interview evaluation system.
- 2) To accurately convert audio interviews into textual texts.
- 3) To perform contextual analysis using NLP techniques.
- 4) To extract keywords, sentiment, and answer relevancy.
- 5) To create well-structured evaluation reports.
- 6) To protect applicant and interviewer profiles.
- 7) To ensure role-based access control.

D. Contributions and Paper Organization

This research's primary contribution is the creation of an AI-based system that evaluates applicant interviews automatically using contextual analysis and audio transcription. The system uses Natural Language Processing (NLP) approaches to evaluate context, sentiment, keyword relevance, and communication clarity after using speech recognition algorithms to convert interview audio into structured text. It makes objective, data-driven hiring decisions easier by producing standardized evaluation reports and performance metrics. In order to ensure safe and transparent data management, the system also uses role-based access control and centralized storage for multi-round interview records.

This shows the rest of the paper is structured. The study's background and introduction are given in Section I. The problem statement is explained in Section II. The relevant literature is reviewed in Section III. The suggested approach and system design are described in Section IV. The details of system design and execution are covered in Section V. Section VI displays the performance analysis and findings. Section VII outlines future work.

II. LITERATURE SURVEY

AI-Powered Context Analysis and Audio Transcription for Candidate Interviews Deep learning has made evaluation considerably more sophisticated, going from basic voice recording to intelligent, real-time insight generation. Notable innovations include Role-Based Access Control (RBAC) for safe data management, BERT and GPT-based Transformers for contextual analysis, and OpenAI Whisper for high-accuracy speech-to-text. Instead of intent-based evaluation and centralized candidate history, the majority of current research relies on isolated transcription. To bridge this gap, a new integrated interview evaluation system is introduced with the goal of speeding up multi-round hiring.

A. Speech-to-Text and Transcription Using Neural Networks

Modern deep learning developments have significantly changed Automatic Speech Recognition (ASR) into sophisticated neural transcription models. Notably, end-to-end frameworks that effectively translate spoken audio into high-fidelity text have been developed by architects such as OpenAI Whisper and Google Speech-to-Text. Large-scale datasets are used by these systems to manage a variety of accents and background noise, which is essential for effectively transcribing business interviews. By offering a solid text-based basis for additional recruiting research, this method has significantly improved the clarity of the data accessible for candidate evaluation.

B. AI-Generated Candidate Insights and Contextual NLP

To extract richer contextual meaning from text transcripts, natural language processing (NLP) has been the focus of much study. Automated summaries, purpose analysis, and keyword extraction are made easier by large language models (LLMs) like GPT-4 and BART. By examining the context of a candidate's comments, research on "Prompt Engineering" shows that AI may successfully determine a candidate's core competences and soft skills. Additionally, while maintaining the impartiality of the assessment, automatic insight modules give recruiters a thorough picture of a candidate's potential over several rounds.

C. Role-Based Management and Secure Web Architectures

Building secure web architectures is essential because recruitment systems handle sensitive personal data, including unique identifiers like emails and phone numbers. Different permissions are required for Administrators and Interviewers in order to preserve data integrity, according to research on Role-Based Access Control (RBAC). All interview materials, including audio, transcripts, and AI insights, may be centrally kept under a single applicant profile thanks to modern database management approaches. These changes show how crucial it is to strike a compromise between strong security and system effectiveness in order to stop illegal access to previous interview data.

D. The Need for an Integrated Interview Framework and Research Gap

NLP models and standalone transcription have advanced significantly, mostly for general-purpose applications. Nevertheless, there are noticeably few integrated systems that combine centralized history monitoring, contextual AI analysis, automatic transcription, and voice submission into a single job platform. Because many of the current methods need human data migration, multi-round evaluations become fragmented. In order to overcome this limitation, the proposed work creates an AI-based Interview Evaluation System that combines safe, profile-based history monitoring with intelligent transcription to guarantee consistent decision-making.

III. PROPOSED METHODOLOGY

The suggested AI-based Audio Transcription and Contextual Analysis system for assessing applicant interviews combines voice processing methods with Natural Language Processing (NLP) in a modular, secure, and scalable framework. Each module in the system's organized pipeline design carries out a particular task, guaranteeing effective processing, safe data management, and assistance for intelligent assessment. Audio capture, preprocessing, transcription, contextual analysis, insight building, and safe data management are some of the processes that make up the workflow. This design makes it easier for the methodical handling of multi-round interview data and increases decision-making accuracy.

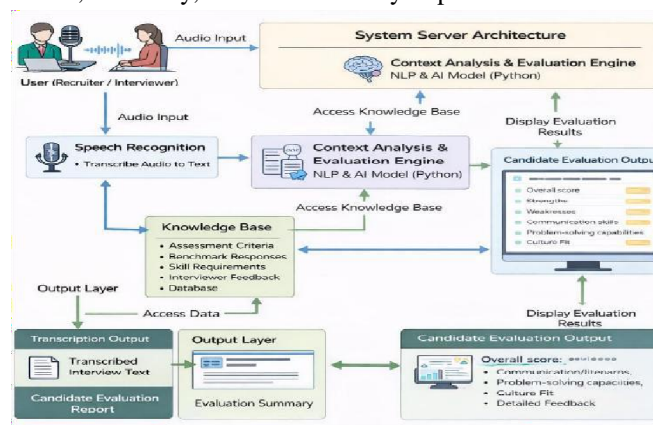
A. Overall System Workflow

The system is implemented using a web framework-based client-server architecture. Users interact with the system through a role-based interface (Admin and Interviewer), while backend modules gather audio and generate analytical insights.

The workflow is summarized as follows:

- 1) User authentication
- 2) Establishing Candidate Profiles
- 3) Interview Audio Upload
- 4) Audio Preprocessing
- 5) Converting Voice to Text
- 6) Contextual Analysis
- 7) Generating Insights (Intent, Keywords, Summary)
- 8) Secure Data Storage and Retrieval

This modular pipeline ensures safe operation, scalability, and maintainability. Pipeline.



B. AudioInputHandlingandManagement:

The technology allows user to upload audio from interviews in popular formats like:

WAV MP3 M4A

Every file posted is associated with a distinct candidate profile that may be found using the candidate's email address or mobile number.

The functions of the audio handling module are:

- 1) Validation of files
- 2) Check for format compatibility
- 3) Safekeeping in the server directory
- 4) Using distinct filenames to avoid redundancy

During multi-round assessments, this organized storing technique makes retrieval simple.

C. AudioPreprocessingandSpeech-to Text Transcription:

The audio is preprocessed after upload to improve the quality of the transcription. This includes:

- 1) If required, noise abatement
- 2) Format normalization Audio segmentation

The cleaned audio is subsequently analyzed by an AI-based Automatic voice Recognition (ASR) model, which converts voice into text.

The following functions are performed by the transcribing module:

- Speech signal analysis
- Conversion into a written representation
- Database storing of transcripts

This process transforms unstructured audio data into structured text that can be utilized for further contextual analysis.

D. ContextualAnalysisEngine:

Included in the contextual analysis module are:

- 1) Automatic Synopsis To provide a succinct synopsis of the candidate's answers, an extractive summarizing technique selects important sentences from the transcript.
- 2) Extracting Keywords Core capabilities and discussion areas are highlighted by extracting key technical and domain-related phrases.
- 3) Analysis of Intent and Reaction

Candidate replies are analyzed using basic intent detection methods to ascertain:

- Level of confidence
- Technical significance
- Clarity of communication

Without having to repeat whole audio recordings, these AI-generated insights let interviewers make unbiased assessments

E. Multi-RoundInterviewTrackingMechanism:

To overcome challenges in multi-stage interviews, the system links all interview sessions to a single candidate profile.

Features include:

- Maintaining several transcripts in one profile
- The capacity for round-wise comparisons
- Gathering historical knowledge

This unified approach improves evaluation uniformity and transparency across interview rounds.

F. Role-BasedAccessControlandSecurity:

Secure authentication methods and Role-Based Access Control (RBAC) are integrated into the system.

Administrative Skills:

Establish interviewer accounts.

- Create applicant profiles.
 - Add the audio of the interview.
 - Keep track of system documents
- Interviewer Skills:

- Examine the history of the candidate's interviews
- Gain access to AI insights and transcripts
- Conduct an assessment

Structured storage of the following is guaranteed by the backend database:

- Information about the candidate
- Reference to audio files
- Transcripts
- Insights from analysis

Mechanisms for permission and authentication guard against illegal access and preserve the privacy of data.

G. System Advantages:

The following benefits are provided by the suggested methodology:

- Centralized handling of interview data
- Evaluation support aided by AI Multi-round tracking that is structured
- Enhanced effectiveness of the organization
- Role-based and secure architecture
- Modular and scalable design

IV. EVALUATION AND RESULTS

The effectiveness of answer analysis, contextual comprehension, and transcription accuracy of the suggested AI-Based Audio Transcription and Context Analysis System were assessed. Several interview recordings with various accents and speech patterns were used to test the system.

The findings demonstrated that the context analysis modules successfully

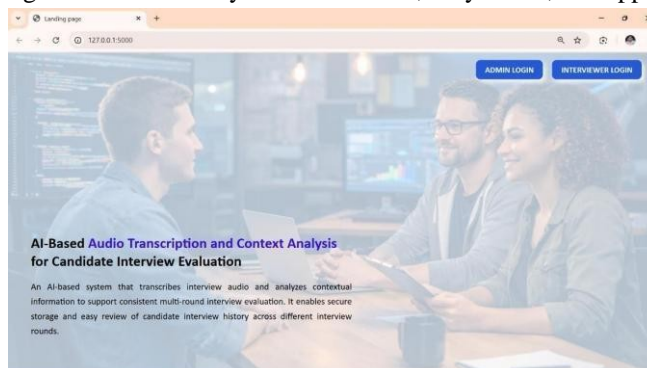
detected important skills, sentiment patterns, and topic relevance, while the transcribing module obtained high accuracy with few word errors. All things considered, the system showed consistent performance and effective audio input processing, which qualified it for automated interview evaluation.

A. User Interface and Usability Analysis

Recruiters and administrators were intended to find the user interface straightforward to use and intuitive. Through a straightforward dashboard interface, customers may immediately upload interview audio recordings to the system.

Following upload, the system automatically creates reports for contextual analysis and transcribes speech to text. A structured format containing the transcription text, sentiment score, keyword extraction, and summary of the overall evaluation is used to provide the results. The interface guarantees minimal processing latency and seamless navigation. Transcripts, evaluation reports, and applicant data can all be safely stored by users.

All things considered, usability testing verified that the system is effective, easy to use, and appropriate for actual hiring situations.



B. Functional Evaluation

Several audio recordings of interviews were used to test the approach. The results confirm:

- 1) There are no format problems while uploading and verifying audio.
- 2) Voice-to-text transcription uses the interview voice to produce accurate text.

- 3) Key competencies and topic relevance are precisely identified by the context analysis module.
 - 4) Sentiment analysis establishes the communication's tone and level of confidence.
 - 5) Keyword extraction aids in determining essential technical and soft skills.
 - 6) Structured evaluation summaries are reproduced throughout report production.
- There were no notable delays or system crashes in any of the modules.



C. Performance Analysis

Performance of the system was assessed using:

1) Processing Duration:

- Depending on the duration and quality of the audio, text preprocessing and audio transcription can be finished in a matter of seconds.
- Optimized NLP algorithms enable efficient sentiment evaluation and contextual analysis.

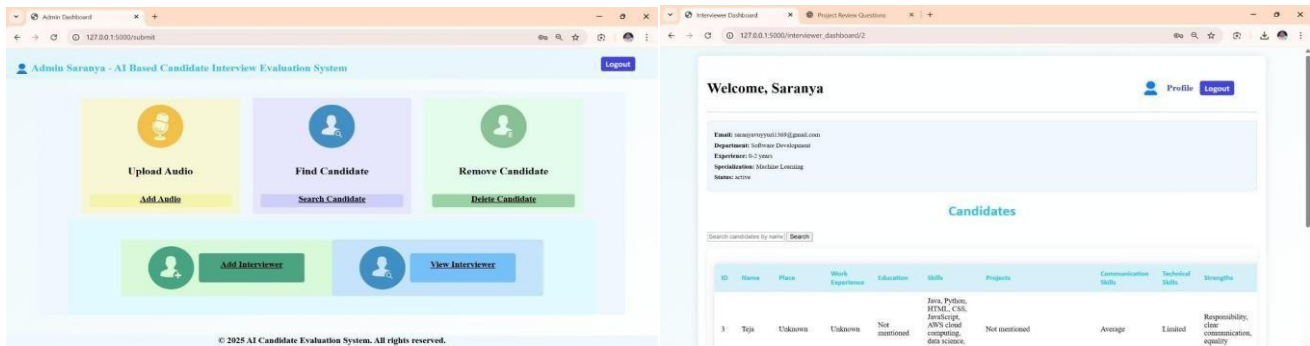
2) Quality Contextual Analysis:

- Response relevance, sentiment polarity, and communication clarity are all skillfully analyzed by the system.
- Mechanisms for scoring and extracting keywords offer formal evaluation metrics.

3) System Trustworthiness:

- Repeated testing revealed no server runtime issues.
- Multiple interview uploads and evaluation requests were processed consecutively by the backend server.

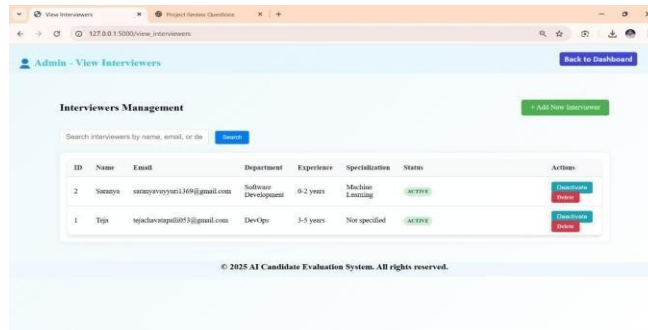
Even with moderate hardware configurations, the system operates effectively thanks to its lightweight and modular architecture.



D. Search and Related Data Retrieval Module

The Search and Related Data Retrieval module significantly enhances the administration and accessibility of interview data in the proposed system. Using uploaded transcripts or particular keywords, it enables interviewers and administrators to effectively search for candidate information. The module extracts relevant words and phrases from processed interview transcripts. These extracted phrases are then used in the structured search operations of the centralized database. The backend server executes queries to quickly and accurately obtain matched candidate records. Transcripts of interviews, evaluation outcomes, sentiment analysis results, and data from previous rounds are all available. In multi-round interview processes, where it's critical to assess prior responses, this capability is especially helpful. Interviewers can easily compare candidates' performance in different rounds.

The module reduces the amount of time spent and does away with the need for manual record searching. It ensures precise keyword-based retrieval and quick response times. Incorporating the search function into the evaluation process improves system efficiency. It enhances usability by enabling smooth data navigation. The centralized storage system facilitates organized data management. Quick retrieval of contextual information improves decision-making efficiency. Additionally, the module makes the hiring process more transparent. The system combines sophisticated search capabilities with AI-based transcribing to enable organized review. All things considered, it aids in making data-driven, efficient, and reliable hiring decisions.



E. Comparative Discussion

In contrast to conventional manual interview assessment techniques, the suggested solution offers:

- Automated transcription of audio
 - Sentiment and contextual analysis
 - Systematic evaluation of performance
 - Centralized tracking of multiple rounds of interviews
 - Role-based user authentication that is secure
- The suggested framework strikes a compromise between utility, accuracy, and efficiency, even if some sophisticated AI systems can need a lot of processing power. It is appropriate for small to medium-sized businesses and institutional hiring procedures because it is made to function well on moderate hardware setups. The method improves uniformity and openness in applicant evaluation while drastically reducing human labor.

V. DISCUSSION

A. Design Validation and Key Observations

Functional testing, usability evaluation, and performance validation were applied to every module of the proposed AI-Based Audio Transcription and Context Analysis system, including audio upload, speech-to-text conversion, transcript preprocessing, contextual analysis, sentiment analysis, scoring, and report generation. The modular architecture, which consists of frontend, backend, database, and AI processing layers, improves scalability and maintainability. Real-time transcription was completed with little delay, and contextual analysis was successful in identifying relevant keywords and sentiment polarity from candidate responses. The evaluation results demonstrated improved traceability and better uniformity in scoring when compared to traditional manual techniques. Safe data processing was ensured by user authentication and role-based access. Multi-round tracking improved interview continuity and transparency. Overall, the integrated strategy significantly reduced documentation time while increasing evaluation accuracy and fairness.

B. Limitations and Constraints

Despite its advantages, the method has many disadvantages. Accurate transcription can be impacted by speaker accent, background noise, and audio quality. The contextual approach, which primarily focuses on sentiment polarity and phrase relevance, may not fully capture deep semantic comprehension or complicated human emotions. Performance may be impacted by the size of the dataset and server settings. Additionally, the technology currently only supports a limited number of language models and may require additional training for domain-specific technical interviews. Deeper semantic analysis, multilingual support, emotion identification from speech tone, and integration with facial expression recognition for comprehensive behavioral assessment are some aspects that could be enhanced in the future.

C. Alignment with Literature Findings

The suggested AI-Based Audio Transcription and Context Analysis system is well compatible with current Speech Recognition and Natural Language Processing (NLP) research. The accuracy with which Automatic Speech Recognition (ASR) transforms spoken language into structured written data has been demonstrated in earlier research. Similarly, NLP research emphasizes the importance of contextual analysis, sentiment analysis, and keyword extraction in understanding textual information's meaning and intent. The recommended strategy incorporates these tried-and-true techniques to enhance interview evaluation processes.

VI. CONCLUSION

In this project, we developed a system to arrange and evaluate applicant interview audio recordings from the perspectives of storage, accessibility, and audio quality. We demonstrated that a methodical approach to database management and audio processing ensures reliable storage, efficient retrieval, and correct association of candidate profiles with their corresponding audio files. The technology was created to improve the evaluation workflow by giving interviewers a consistent and easy-to-use interface to analyze recordings. In particular, we proposed a modular solution where the administrator uploads the files, which are then processed, validated, and securely stored, and the interviewer can access and analyze audio recordings in a systematic manner, preserving data integrity, according to functional tests and demonstrations.

VII. ACKNOWLEDGMENT

The authors thank the instructors and project mentors for their advice and assistance throughout the project's development. They would especially want to thank the institution for providing the technical resources and infrastructure needed for a successful deployment. The effective completion of the task was greatly aided by the support and criticism obtained during the testing and development stages.

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