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A Lightweight and Efficient Text-to-Speech Framework for Educational Applications

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Abstract: Text-to-Speech (TTS) systems play a crucial role in modern e-learning and assistive technologies by facilitating effective human-computer interaction. This paper discusses the creation of an AI-based multilingual TTS framework using Flask and MySQL, which transforms textual input into natural-sounding speech via Google Text-to-Speech (gTTS). The system features intelligent modules, including user authentication, file-based text extraction from formats such as PDF, DOCX, and TXT, automatic text summarisation, and multilingual translation. Its architecture comprises modules for text preprocessing, language mapping, translation handling, neural speech synthesis, and audio generation. Supporting English, Hindi, and Telugu, the system enhances accessibility for diverse learners. Experimental validation showed efficient real-time voice generation with reduced latency, improving usability in educational settings. By combining translation and summarisation with speech synthesis, this framework boosts digital content accessibility and offers a scalable solution for interactive learning. Future developments aim to integrate advanced neural TTS models for enhanced expressiveness and support offline deployment.

Keywords: Text-to-Speech, Multilingual Speech Synthesis, Natural Language Processing, Flask Framework, gTTS, Speech Generation, E-Learning Systems.

I. INTRODUCTION

The rapid growth of digital technologies has transformed information access, with Text-to-Speech (TTS) systems enhancing accessibility for visually impaired individuals and language learners. Traditional TTS systems often produced robotic outputs, limiting usability. Advances in Artificial Intelligence (AI) and Natural Language Processing (NLP) have led to modern TTS systems that generate more natural speech. However, many systems lack features like multilingual support and document extraction. This paper proposes an AI-based multilingual TTS framework for educational applications, utilizing the Flask framework and MySQL for user management. It supports text extraction from various formats and includes translation, summarization, and speech generation using Google Text-to-Speech (gTTS). The goal is to create an efficient, user-friendly system to enhance accessibility in digital learning environments.

A. Background and Motivation

The expanding landscape of digital education has heightened the need for sophisticated content delivery systems, particularly Text-to-Speech (TTS) technology, which converts text into spoken words. This technology is vital for visually impaired users, language learners, and those who favour auditory learning. Recent advancements in Artificial Intelligence (AI) and Natural Language Processing (NLP) have enhanced the quality of synthesised speech, making it more natural and intelligible. However, many TTS solutions merely provide basic voice conversion without integrating essential functionalities such as multilingual translation, document text extraction, and intelligent summarisation. Users in educational contexts require robust systems capable of processing various document formats, translating different languages, and delivering real-time speech. This necessity has led to the creation of a versatile, lightweight multilingual TTS framework designed to improve accessibility and usability in e-learning environments.

B. Problem Statement

Numerous speech synthesis systems currently face several limitations that hinder their effectiveness. Key issues include limited multilingual support in basic text-to-speech (TTS) applications, the lack of integrated document processing capabilities for formats such as PDF, DOCX, and TXT, and the absence of built-in text summarisation features prior to speech generation. Additionally, advanced neural TTS models exhibit high computational requirements, and simple TTS tools often lack user authentication and structured backend management. These challenges underscore the necessity for a robust, scalable, and user-friendly solution that can seamlessly integrate text extraction, translation, summarisation, and speech synthesis in a single platform, all while ensuring real-time performance and ease of deployment.

C. Objectives

The primary objectives of the proposed system are:

1) To design and implement an AI-based multilingual Text-to-Speech system.
2) To support text extraction from multiple document formats including PDF, DOCX, and TXT files.
3) To integrate automatic text summarization for improved content consumption.
4) To provide multilingual translation support for diverse users.
5) To generate natural and clear speech output using efficient speech synthesis techniques.
6) To develop a secure backend system with user authentication and database integration.
7) To ensure lightweight processing and real-time response suitable for educational applications.

D. Contributions and Paper Organisation

The proposed work presents significant contributions, notably the creation of a comprehensive multilingual Text-to-Speech framework that integrates various functions such as text extraction, translation, summarisation, and speech synthesis into a single platform. The backend system is developed using the Flask framework, with MySQL for secure user authentication and activity management, ensuring a lightweight and real-time performance. It supports multiple document formats, including PDF, DOCX, and TXT for versatile content processing. The framework includes multi-language translation capabilities to produce speech output in English, Hindi, and Telugu. Furthermore, it employs Google Text-to-Speech (gTTS) for efficient and natural speech generation, which is particularly suitable for educational use. The architecture is designed to be scalable and user-friendly, enhancing accessibility for a wide range of users, including those who are visually impaired. Collectively, these contributions deliver a practical solution for intelligent, speech-enabled learning systems.

II. LITERATURE SURVEY

Text-to-Speech (TTS) technology has advanced due to deep learning, moving from robotic outputs in early systems to more natural speech. Notable developments include Tacotron 2 for generating mel-spectrograms, FastSpeech 2 for faster speech generation, PromptTTS for expressive speech via textual prompts, and LEF-TTS for real-time applications. Recent approaches like speech vectorisation aim to enhance model efficiency. Current research tends to optimise models without integrating multilingual translation and summarisation. A new lightweight multilingual TTS framework is proposed to address this gap, aiming for real-time educational use.

A. Neural Network-Based Text-to-Speech Model

Recent advancements in deep learning have fundamentally transformed traditional Text-to-Speech (TTS) systems into advanced neural speech synthesis models. Notably, Tacotron 2 has introduced an innovative end-to-end architecture that efficiently converts text into mel-spectrograms, subsequently generating waveforms using neural vocoders. This approach has markedly enhanced the naturalness and clarity of speech. Additionally, FastSpeech 2 has contributed further improvements by employing a non-autoregressive technique, which reduces latency while preserving high-quality voice output. Collectively, these models have laid the groundwork for contemporary neural TTS systems.

B. Expressive and Prosody-Controlled Speech Synthesis

To enhance the expressiveness of synthesised speech, significant research has concentrated on prosody modelling and the generation of controllable speech. PromptTTS allows for the control of speech styles through textual prompts, which facilitates the generation of varied tones and expressions. Investigations into the granularity of prosodic representation reveal that fine-grained prosody modelling significantly improves the emotional and expressive quality of speech outputs. Additionally, approaches such as Stable-TTS and VITS focus on refining speaker adaptation and duration modelling, ensuring that voice synthesis remains stable and consistent across various contexts.

C. Lightweight and Efficient TTS Architectures

Neural text-to-speech (TTS) models generate high-quality speech but face significant computational efficiency challenges. Lightweight architectures like LEF-TTS strive to minimise model complexity without sacrificing speech clarity, thus enhancing suitability for real-time applications.

Additionally, advancements in beam search optimisation and speech vectorisation techniques have further accelerated generation speed and scalability. These developments underscore the critical need to balance speech quality with system efficiency in TTS research.

D. Research Gap and Need for Integrated Educational Framework

Significant advancements have been achieved in neural speech synthesis, predominantly in enhancing model architecture and improving speech quality. However, there remains a noticeable lack of attention towards creating a unified system that encompasses multilingual translation, document text extraction, summarisation, and secure user management. This deficiency is particularly relevant in educational contexts, which necessitate a robust framework capable of processing diverse content and delivering real-time multilingual speech output. To address this shortcoming, the proposed work aims to develop an AI-based multilingual Text-to-Speech framework. This framework will incorporate intelligent content processing modules with effective speech synthesis, specifically designed for educational applications.

III. PROPOSED METHODOLOGY

The proposed AI-based multilingual Text-to-Speech system is a modular, scalable, and lightweight framework that combines intelligent text processing with real-time speech synthesis. It utilises a structured pipeline architecture, with each module executing a specific function to enhance efficiency and produce natural voice output. The workflow encompasses several stages: input acquisition, preprocessing, translation, speech synthesis, audio management, and secure backend handling, ensuring a seamless operation throughout the system.

A. Overall System Workflow

The system utilises a client–server architecture built on the Flask web framework. It allows users to engage with the system via a front-end interface, enabling them to input text manually or upload documents in various supported formats. The backend server then handles requests through a series of modules, ultimately producing the final speech output.

The workflow can be summarised as follows:

1) User Authentication
2) Text Input or Document Upload
3) Text Extraction
4) Text Preprocessing and Cleaning
5) Optional Text Summarisation
6) Multilingual Translation
7) Speech Synthesis
8) Audio File Generation and Delivery

This structured approach ensures modularity, maintainability, and scalability.

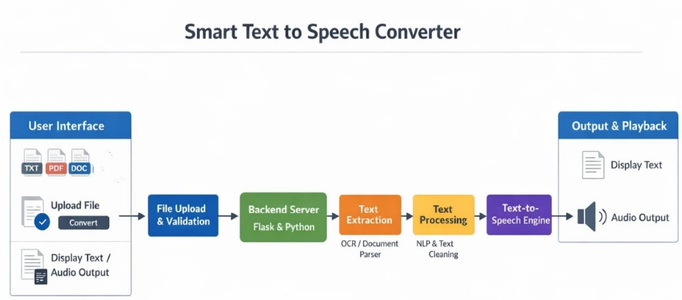

Fig 01: Architecture Of System

B. Input Handling and Text Extraction

The system accommodates both direct text input and document uploads, supporting various file formats such as PDF, DOCX, and TXT. It employs text extraction techniques tailored to each format, merging the extracted content into a single text format for streamlined processing. This functionality increases flexibility, enabling users to convert educational materials into speech directly, bypassing the need for manual retyping.

1) Portable Document Format (PDF)
2) Microsoft Word Documents (DOCX)
3) Plain Text Files (TXT)

C. Text Preprocessing and Summarisation

After extraction, the text undergoes preprocessing steps to improve clarity and structure. These steps include:

1) Removal of unnecessary whitespace
2) Sentence segmentation
3) Basic normalisation

To enhance usability in educational settings, a straightforward extractive summarisation mechanism has been developed. This system identifies and selects crucial initial sentences from the source text to create a concise summary. Such an approach minimises cognitive load for users and facilitates quicker content consumption prior to audio generation.

D. Multilingual Translation Mechanism

To enhance accessibility, the system incorporates a multilingual translation module that enables users to choose their preferred output language through a predefined language mapping system. If the chosen language is different from the original input, the system automatically translates the text before proceeding to speech synthesis.

Currently supported languages include:

1) English
2) Hindi
3) Telugu

This feature enables cross-language learning and ensures broader usability in diverse educational settings.

E. Speech Synthesis Module

The speech synthesis module transforms processed text into natural audio via a neural-based text-to-speech engine, specifically leveraging Google Text-to-Speech (gTTS) to create MP3 audio files.

The module performs the following steps:

1) Receives processed or translated text
2) Converts text into phonetic speech representation
3) Generates waveform audio output
4) Saves the audio file with a unique identifier

The generated audio is securely stored in a designated directory, accessible for download or playback through a backend endpoint. The speech generation process is optimised for minimal latency and real-time response, making it ideal for interactive applications.

F. Database Integration and User Authentication

To ensure secure access and user management, the system integrates a MySQL database. The backend includes:

1) User registration
2) Login authentication
3) Password validation
4) Login history tracking

Database connectivity ensures structured data management and enhances system reliability. Authentication mechanisms prevent unauthorised access and maintain secure operation.

G. System Advantages

The proposed methodology offers several advantages:

1) Integrated text extraction, translation, summarisation, and speech synthesis
2) Lightweight and efficient architecture
3) Real-time audio generation
4) Multilingual support
5) Secure backend implementation
6) Scalable and modular design

IV. EVALUATION AND RESULTS

The evaluation of the proposed AI-based Multilingual Text-to-Speech system focused on several key aspects: functionality, performance efficiency, usability, and speech output quality. The assessment utilised various document formats and multilingual text inputs to test the system's reliability and its ability to process information in real-time.

A. User Interface and Usability Analysis

The front-end interface was designed for intuitiveness and user-friendliness, featuring a homepage that allows text entry or file uploads. The summary page displays processed text clearly prior to speech generation. Users can preview and download voice output with adjustable parameters like speed and pitch, facilitating better control over the speech. The system also includes login authentication and history tracking for enhanced personalization and data management. Overall, user interaction was smooth with minimal response time during regular usage.

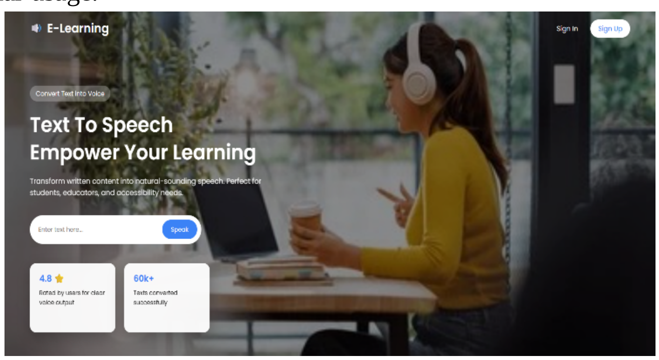

Fig 02: Home Page

B. Functional Evaluation

The system was tested with different types of inputs, including direct text entry and document uploads (PDF, DOCX, and TXT). The results confirm that:

1) File upload and validation process functions correctly without format errors.
2) Text extraction from documents is accurate and preserves meaningful content.
3) The summarization module generates concise outputs by extracting key sentences.
4) Multilingual translation successfully converts text into selected languages such as Hindi and Telugu.
5) Speech synthesis generates clear and intelligible audio output in MP3 format.

All modules operated sequentially without system crashes or major latency issues.

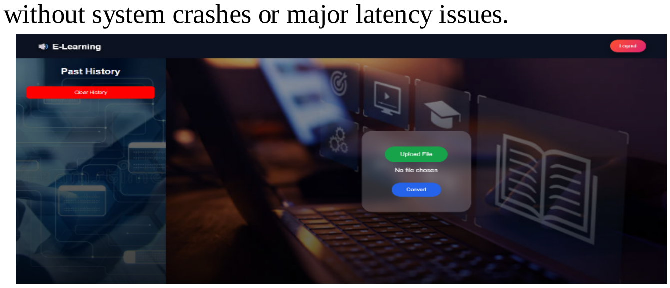

Fig 03: Upload Page

C. Performance Analysis

The system performance was evaluated based on:

1) *Processing Time:*

- Text extraction and preprocessing are completed within a few seconds, depending on document size.
- Speech generation latency remains low due to a lightweight processing architecture.

2) *Audio Quality:*
- Generated speech output is clear and natural.
- Pronunciation accuracy is maintained for supported languages.

3) *System Reliability:*
- No significant runtime errors were observed during repeated testing.
- Backend server successfully handled multiple user requests sequentially.

The lightweight architecture ensures that the system performs efficiently even on moderate hardware configurations.

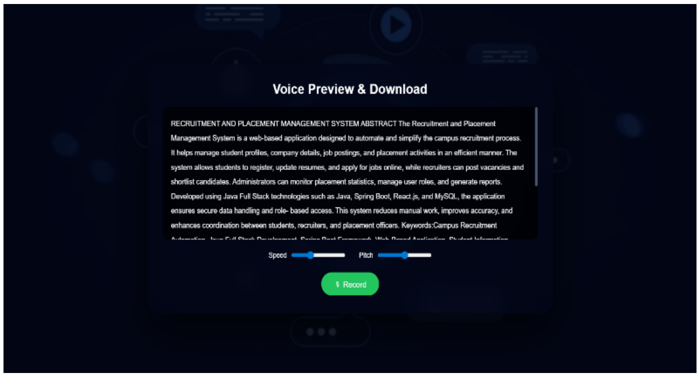

Fig 04: Voice Preview and Download

D. *Search and Related Data Retrieval Module*

The proposed system features a Search Related Data module that enhances content accessibility and user interaction by allowing users to search for relevant information based on uploaded files or entered text. This functionality facilitates quick retrieval of contextually related data, thereby improving usability. The search module operates by accepting keywords or phrases derived from the processed document, with the backend server executing the query to retrieve matching data from the system database or content repository. This capability supports educational scenarios where users seek additional information related to their uploaded material. Evaluation results indicate that the search functionality delivers accurate keyword-based retrieval, minimal response times, smooth integration with the system workflow, and increased user engagement through interactive data exploration. Furthermore, by integrating search capabilities within the Text-to-Speech framework, the system transcends basic speech synthesis, promoting an intelligent learning support environment and enhancing content discoverability for a comprehensive e-learning experience.

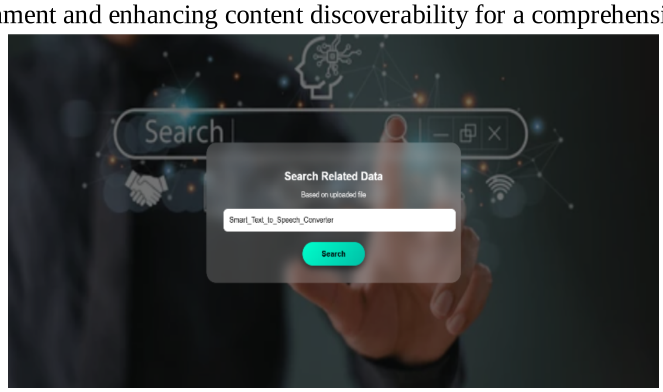

Fig 05: search related data

E. *Comparative Discussion*

Compared to traditional rule-based TTS systems, the proposed system provides:
1) Multilingual translation capability
2) Integrated document extraction
3) Built-in summarisation
4) Secure user authentication

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5) Real-time audio generation

While advanced neural TTS models may offer higher expressive control, they often require high computational resources. The proposed framework balances efficiency and functionality, making it suitable for educational applications and small-scale deployment.

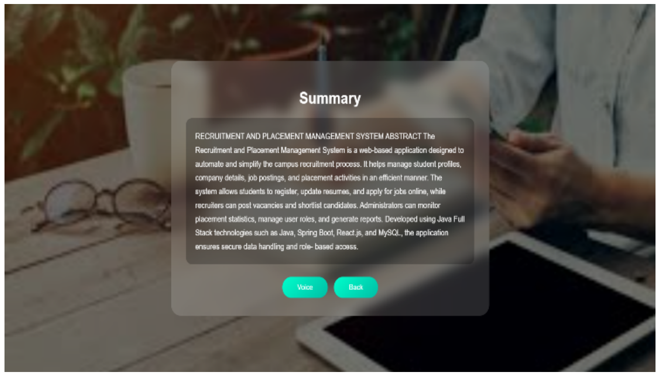

Fig 06: summary

V. DISCUSSION

A. Design Validation and Key Observations

The proposed AI-based Multilingual Text-to-Speech (TTS) system underwent functional testing, usability evaluation, and performance analysis, with all modules—including file upload, text extraction, preprocessing, translation, summarization, and speech synthesis—functioning sequentially without critical errors. The system's modular architecture, which differentiates between frontend, backend, and processing layers, enhances maintainability and scalability. Real-time speech generation was achieved with minimal latency, showcasing the efficacy of the lightweight Flask-based backend. Key evaluation findings indicate high document parsing accuracy for PDF, DOCX, and TXT formats, enhanced accessibility through multilingual translation, satisfactory speech output clarity for educational use, and improved system personalization via user authentication and history tracking. Overall, the integrated workflow significantly reduces manual effort in comparison to standalone TTS tools, validating that the design meets its objectives for educational and assistive applications.

B. Limitations and Constraints

Despite its successful implementation, the system exhibits limitations, such as dependence on external Text-to-Speech (TTS) services, necessitating internet connectivity. Its summarisation technique is extractive, lacking deep semantic understanding. Additionally, it does not incorporate emotional tone or expressive speech modulation. Performance varies with document size and server configuration, and language support is restricted to specific regional options. These limitations highlight opportunities for future enhancements and optimisations.

C. Alignment with Literature Findings

The proposed system is in line with contemporary research trends in neural Text-to-Speech (TTS) systems, focusing on real-time performance, multilingual capabilities, and lightweight architectures. It shares similarities with efficient TTS models like FastSpeech by emphasizing reduced latency and enhanced usability. Unlike many neural architectures that prioritize model optimization, this system uniquely integrates document processing, translation, summarization, and authentication into a cohesive framework. This integration responds to practical educational needs identified in recent studies concerning accessible learning technologies. Consequently, the system effectively bridges the gap between theoretical advancements in neural TTS and their practical deployment in real-world applications.

VI. CONCLUSION

This paper discusses the design and implementation of an AI-based Multilingual Text-to-Speech framework aimed at educational applications. The system incorporates various processes such as document text extraction, preprocessing, translation, summarization, and speech synthesis, which are organized within a lightweight client-server architecture. Evaluation results indicate that the framework produces natural and intelligible speech output while maintaining real-time performance and usability.

The incorporation of multilingual support and secure backend management improves accessibility and reliability. The framework is presented as a practical and scalable solution for intelligent speech-enabled learning environments. Future research will focus on integrating advanced neural TTS models, enhancing semantic summarization techniques, introducing emotional speech synthesis, and enabling offline deployment capabilities.

VII. ACKNOWLEDGMENT

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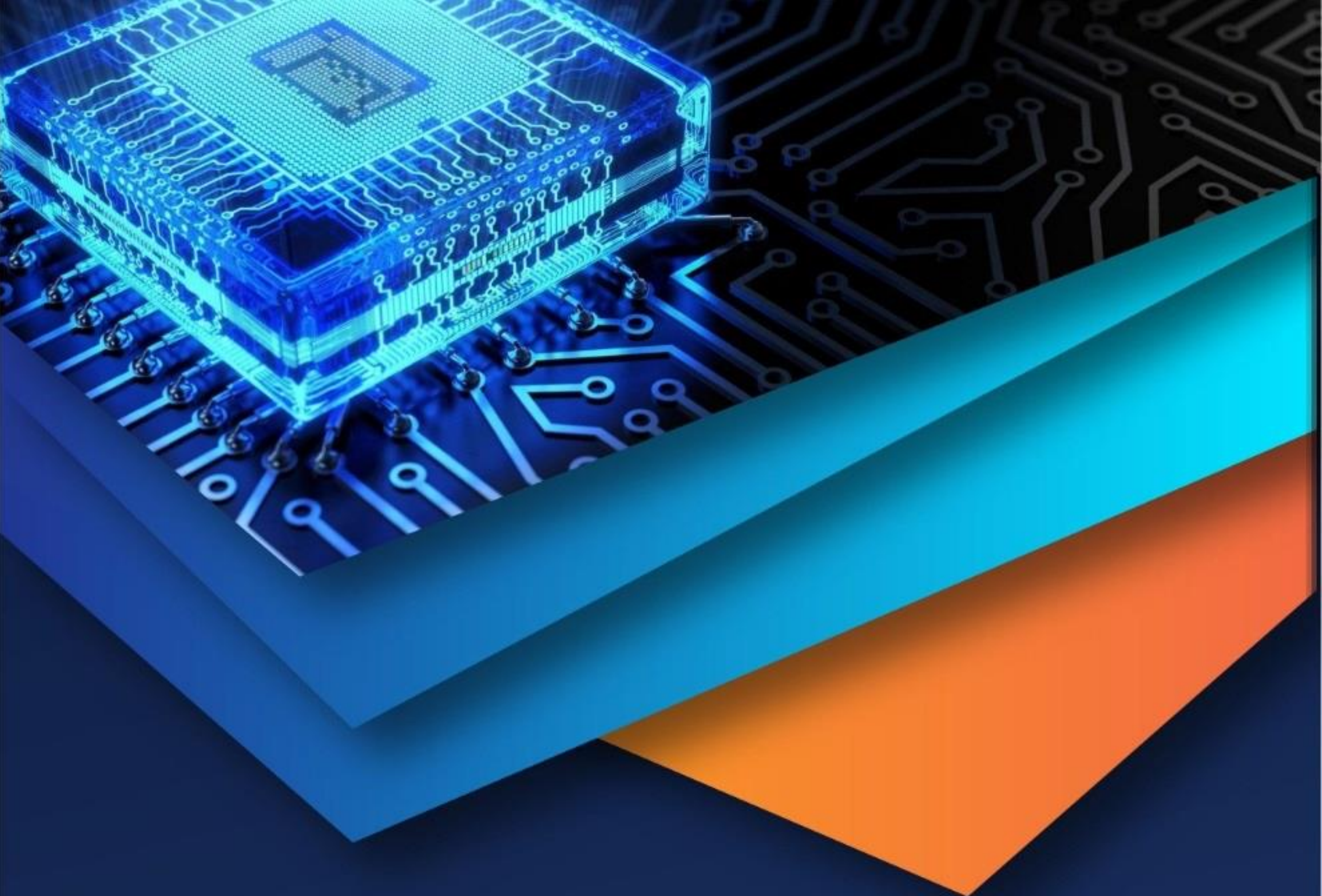

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