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Community Space using WebRTC

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Abstract: Human interaction is important in today's internet-based world. To effortlessly approach each other to communicate people can use the new technology i.e. web real-time communication (WebRTC). Our real-time communication system will provide virtual voice rooms to users to audio chat in groups or peer to peer using technologies like WebRTC, Node.js, Web Sockets and React.js. On the basis of a range of topics, users can create public/private rooms and join them as well. The implications of this study are better than the previous projects.
Keywords: WebRTC, React.js, Node.js, Web Sockets

I. INTRODUCTION

High-quality interpersonal communication is crucial given the quick advancement of current technologies in the transmission of communications in multimedia and computers. A real-time method of communication had been created to meet this expanding need.

Web browsing will be expanded by enabling online access to social media, chat, video conferencing, and television thanks to a brand-new standard and industry-wide initiative dubbed real-time communication (RTC). Users can view, record, comment on, or merely keep track of the motion of video content using these technologies. Web Real-Time Communication (WebRTC) combines To allow one-to-one audio, video, and data communication across browsers, There are JavaScript APIs, standards, and protocols out there.

Users of these systems can now see video content, record it, broadcast it, and leave comments on it thanks to the open source WebRTC initiative. Real-time communication between web browsers is also made possible via WebRTC. WebRTC, a real-time communication protocol, has increased API (Application Programming Interface) standards, enabling high-quality peer-to-peer communication for web developers using some JavaScript codes and real-time multimedia sharing for web browsers without the need for a plugin. The open-source WebRTC standard for multimedia web platforms was created by Google in 2011. Currently, browsers like Firefox, Chrome, Opera, etc. can access it. When this technology is eventually implemented into every browser, it won't require plug-in components and might perhaps lower the danger of infection while boosting interactive communication.

This paper presents a system that, on various hardware and OS based on WebRTC, transmits services like audio, recognizes the required user, and finds any further users connected to the system can feel confident using it without having to go through time-consuming installation or setup procedures inside a web browser.

II. LITERATURE REVIEW

The expansion of the web browsing paradigm and the Information can now be found in places like social media, chat, OTT and conference calling. Thanks to the development of real-time communication (RTC), a new standard and widespread industry effort[1]. Thanks to these systems' time-sensitive cloud infrastructures, users may watch, record, annotate, or alter the progression of video and audio material while still obtaining high-quality services. To create multipoint video conferencing systems, a variety of proprietary protocols and codecs can be employed, however they are neither scalable nor easily interoperable[8]. Modern open technology known as WebRTC allows for the transmission of data, audio, and video in real-time using web browsers. Plug-in-free JavaScript APIs are used by WebRTC. Leveraging Mozilla Firefox and the Scale Drone service, A web-based peer-to-peer real-time communication system is created by us that connects users with high-speed data transfer through the communication channel using HTML5 and a Node.js server address. Socket.io, an event-driven library, is made for real-time web applications. It offers real-time, communicating back and forth between web servers and clients. It includes a Node.js server-side library and a browser-based client-side library[3]. The APIs of the two components are rather similar to one another. Real-time analytics, binary streaming, instant messaging, and document collaboration are all made possible via socket.io. When it is possible, Socket.io will immediately upgrade to WebSocket and manage the connection secretly. This implies that for utilizing Socket.io, the developer doesn't require to be familiar with the WebSocket protocol[7][6].

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Socket.io does not offer a WebSocket library with backwards compatibility for other real-time protocols . Additional to earlier real-time protocols, It utilizes a special real-time transfer mechanism. A WebSocket client that does not employ A server cannot be contacted via Socket.io. Clients that support Socket.io cannot connect to a WebSocket or Long Polling Comet server without it[11]. To utilize Socket.io, both the client and server sides must make use of Socket.io libraries.

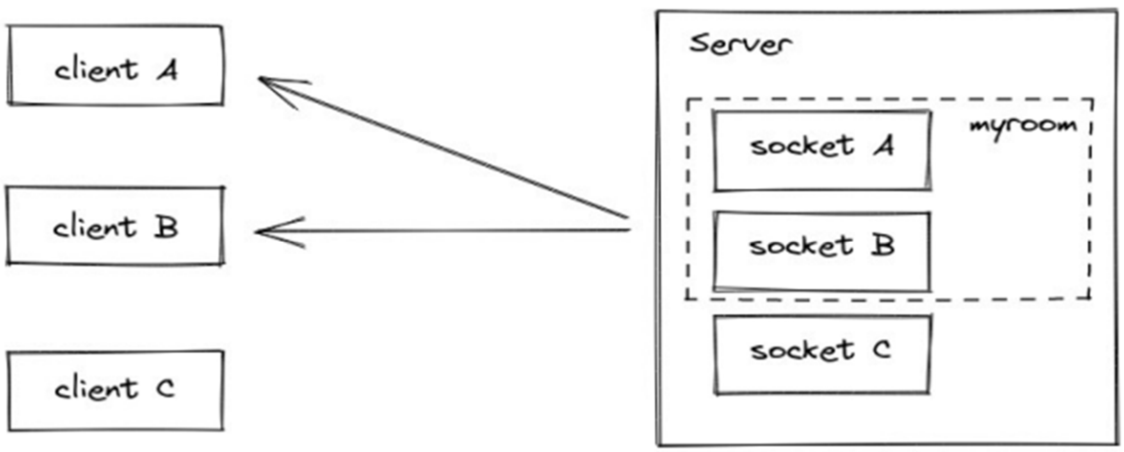

Fig.1 Client-Server Communication Model

III. METHODOLOGY

The example below demonstrates how the client-server semantics framework is maintained while incorporating WebRTC's notion of peer-to-peer communication across many machines.. Direct flow between browsers is made possible by the connection's media channel management. For Web Sockets or HTTP to be able to modify, interpret, or control signals as needed, web servers broadcast network signals. WebRTC signals between the browser and server were discovered to be intermittent when they constitute a component of the software. SIP (Session Initiation Protocol) and Jingle are examples of common signaling protocols that can be used by web servers to interact. To do this, a property signaling system could also be used. Network multimedia communication is the main goal of WebRTC. Network transmission, audio, and video technologies are all required to allow multimedia connectivity across browsers input and output devices for audio: Using this, multimedia data is recorded and played back.

Internet connections Peers must exchange a lot of data while participating in online video conferencing. Because that is the assurance of data delivery, it needs a dependable and continuous network connection.

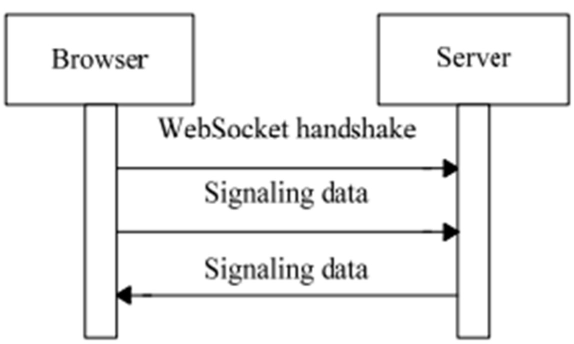

Fig.2 Brower-Server Signalling

Mechanism for WebRTC: Session negotiation is required prior to establishing a connection between browsers on the basis of a reliable data channel. Through the WebRTC signaling system, this part of the task has been finished. Prior to session negotiation, it is necessary to ascertain whether data can be successfully delivered to the next peer and whether the next peer is in a condition that allows for the establishment of a connection. The Offer signal must be sent by the session's first peer, and the Answer signal must be returned by the second peer.

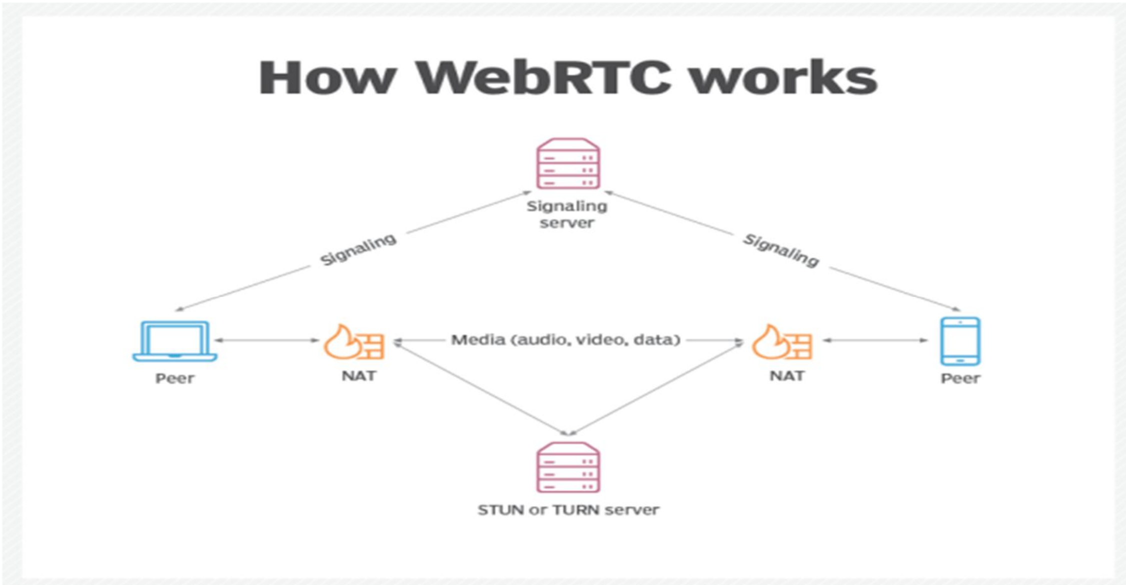

Fig.3 Architecture of WebRTC

IV. RESULT

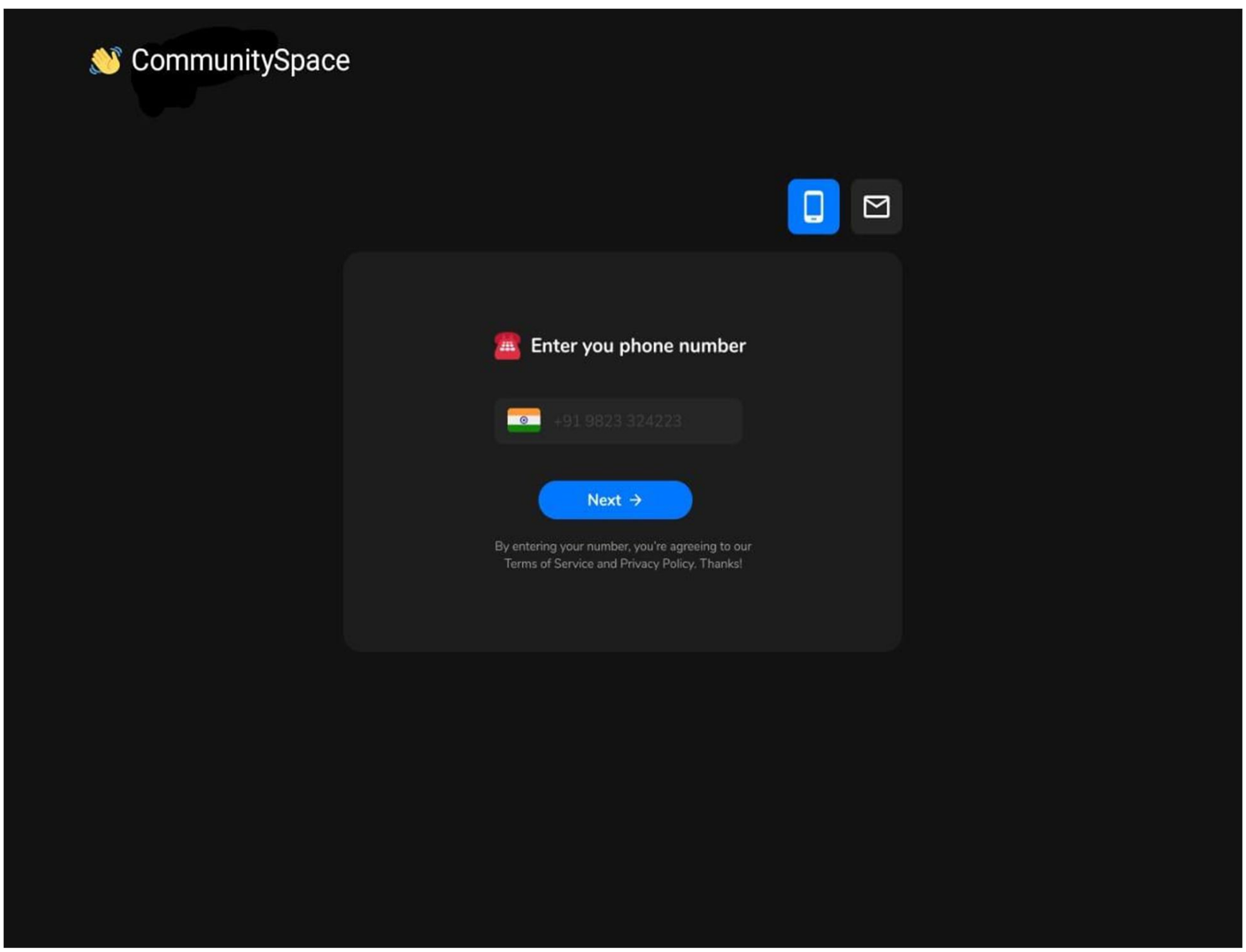

Fig.4.1 Login Screen

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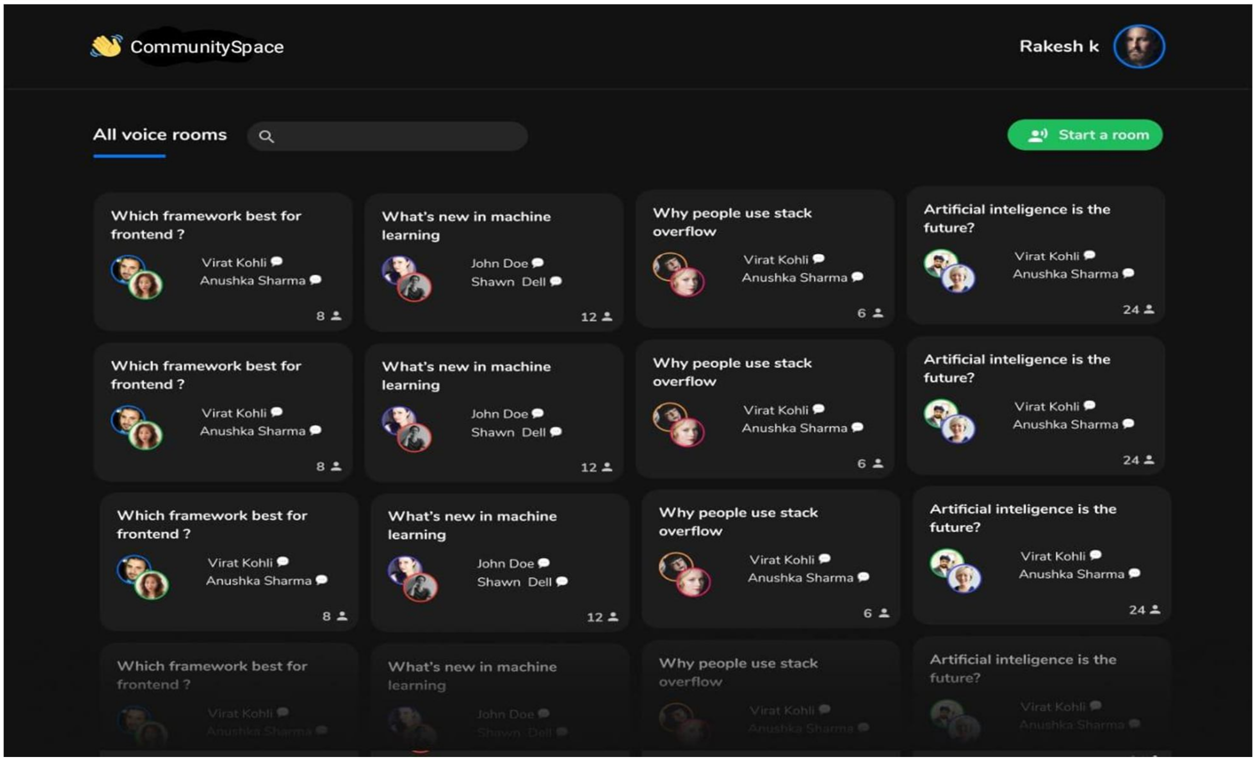

Fig.4.2 Home Screen

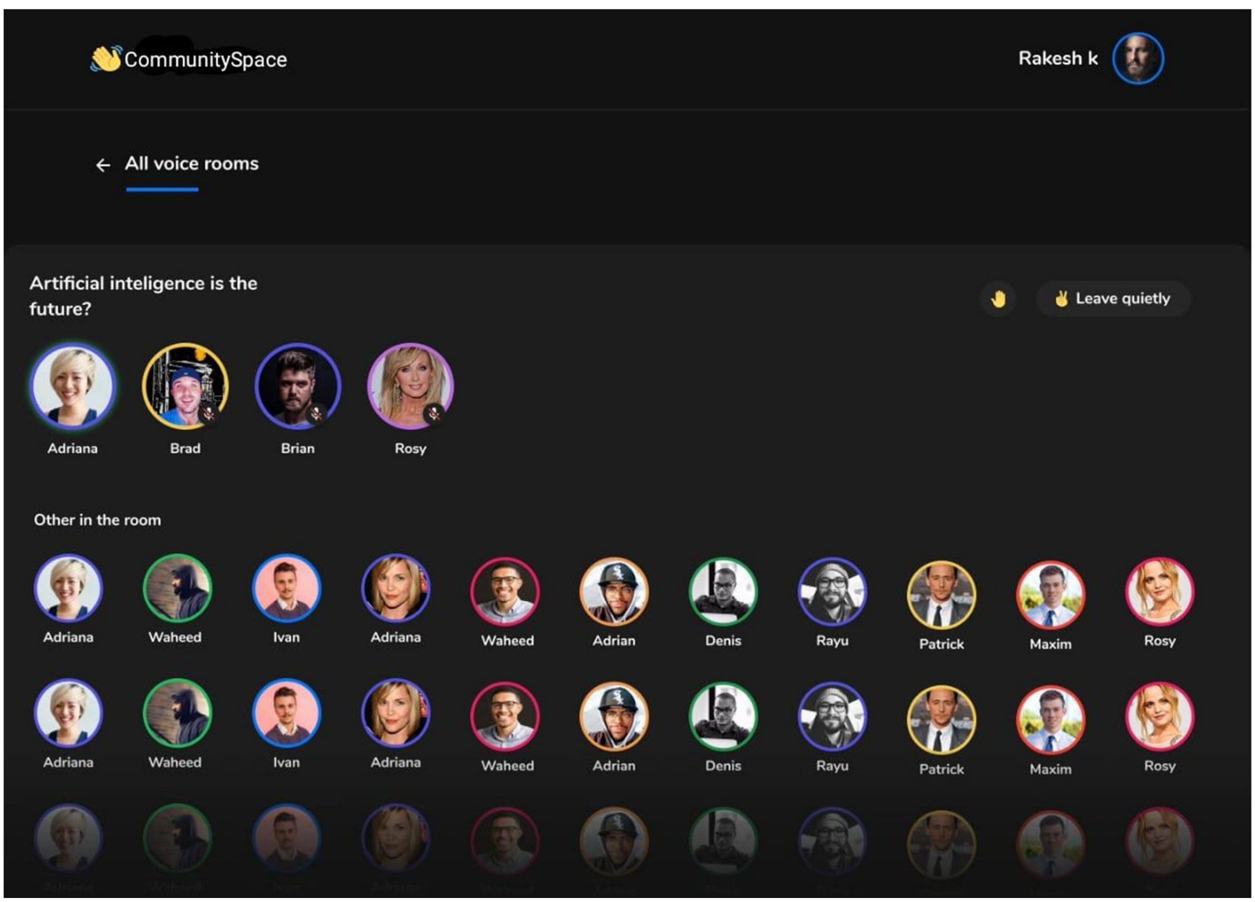

Fig4.3 Room Screen

V. CONCLUSION

Anyone can now construct their own Web page or application for real-time file sharing, real-time message chat, or video/audio conferencing thanks to the implementation of WebRTC technology that enables peer-to-peer connections for real-time communication.. As a result, programmers and developers can now compete for jobs in the real world alongside social media moguls. We can make a website with the greatest cutting-edge functionalities, allowing every user to get connected with each other through text messaging and video or audio communications, with the aid of simple JavaScript APIs and Node JS. the machine that houses the STUN and TURN servers for Google.

Finally, based on the method, we developed a real-time audiovisual connectivity. The system performs well. The article provides a theoretical for WebRTC signaling work, and a solid practical base. Both the system's ability to use mobile internet and the ability of mobile smart devices to connect with one another are essential.. The framework for many browsers' communication will be studied in the project's subsequent stage due to the low connectivity efficiency brought on by a large number of connections.

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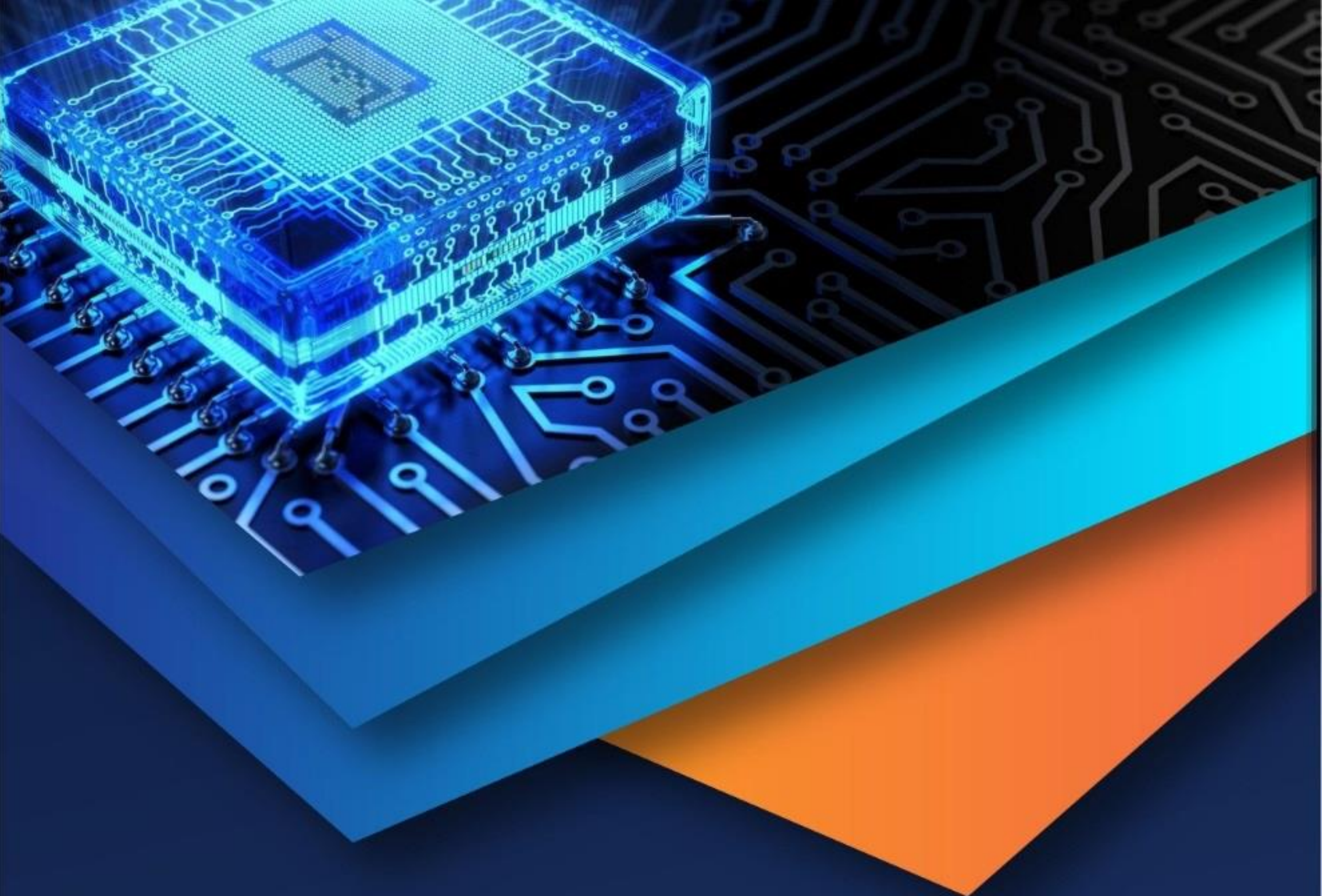

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