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Random Video Call and Chat Application Using Web RTC and Firebase Based on Mobile

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ABSTRACT

Humans as social beings, require communication and interaction. Communication technology found in smartphones plays a crucial role in accelerating access to social media. However, traditional social media often only provides written and asynchronous communication, which can reduce the motivation to participate. This research aims to develop a mobile application that integrates video call and random chat features using WebRTC and Firebase technology. The research method includes developing the application using React Native as the framework, Firebase for real-time database management and authentication, and WebRTC for handling real-time video communication. The research results show that the developed application can provide optimal performance in terms of latency and video call quality, as well as functional login, registration, chat, and video call features. This application opens the door to more direct, interactive, and natural human engagement by offering a more spontaneous and unpredictable meeting experience. The conclusion of this research is that integrating WebRTC and Firebase in the development of video call and random chat applications can provide an efficient and enjoyable communication experience for users.

Keyword : Application, Video Call, Chat, WebRTC, Firebase.

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1. INTRODUCTION

The development of information and communication technology has brought significant changes in the way we interact and communicate. In this digital era, the need for applications that enable real-time communication is increasing, especially with the widespread use of mobile devices (Ulucak et al., 2020). The digital revolution that has occurred in the last few decades has brought about various technological innovations that support faster, more efficient and accessible communication. Mobile devices, such as smartphones and tablets, have become an integral part of everyday life (Wang et al., 2020). Advances in network technology, such as 4G LTE and now 5G, have accelerated data transfer and enabled users to access multimedia content and communicate with high quality instantly. Video and audio compression technologies have also undergone significant improvements, enabling high-resolution video calls and clear sound even on unstable networks (Al-Rahmi et al., 2020).

In addition, the development of cloud computing technology has had a great impact on communication applications. With data storage and processing in the cloud, applications can offer faster and more reliable services (Alashhab et al., 2021). Platforms such as Firebase provide end-to-end solutions for application development, including real-time database management, user authentication, and hosting, all of which contribute to improved efficiency and user experience. Video calling and chat applications have become a popular solution to meet real-time communication needs (Koehler et al., 2020). From services that initially only allowed voice calls, it has now evolved into multifunctional applications that support video calls, text chats, file sharing, and more. Communication platforms such as Zoom, Microsoft Teams, and WhatsApp have proven how important this technology is in supporting daily interactions, both for personal and professional purposes (Jiang, 2020).

The use of WebRTC (Web Real-Time Communication) technology is a breakthrough in the development of communication applications. WebRTC allows direct audio and video communication through a web browser without the need for additional plugins, so it can be utilized into a fast and responsive video call application (Shreya et al., 2021). The technology supports a wide range of audio and video codecs, ensuring optimal communication quality across different network conditions. WebRTC was developed by Google and first announced in 2011, and has since been widely adopted by various communication platforms (Perna et al., 2022). One of the main advantages of WebRTC is its ability to overcome latency, which is often an obstacle in real-time communication. By using a peer-to-peer (P2P) protocol, WebRTC enables direct data transmission between users, reducing delay and improving interaction quality (Andi & Kara, 2023).

In addition, WebRTC provides mechanisms for sophisticated network handling such as dynamic bitrate adjustment techniques, which allow video and audio quality to be maintained despite fluctuating network conditions (Smirnov & Tomforde, 2024). The technology also supports end-to-end encryption, ensuring that data sent and received are secure from unauthorized access, which is crucial in maintaining user privacy and security. With rich and flexible APIs, developers can easily implement real-time communication features into their applications without having to worry about compatibility with specific hardware or operating systems (Lee et al., 2022).

Various previous studies have explored the development of random video call and chat applications. For example, research by Sakinah et al. (2023) in "Application of WebRTC Technology in E-Learning Applications" showed that the integration of WebRTC into an e-learning platform is able to increase the interactivity and effectiveness of distance learning. This study found that the use of WebRTC allows students and teachers to communicate directly through video calls, hold group discussions, and share layers to contribute to a collaborative learning experience. Shiddiqramzy & Sediyono (2023) in "Real-Time Chat Application Using Firebase" elaborated that Firebase is effective as a real-time database management solution, but this application does not include video call functions that can enrich the user experience. Thus, in this research, further development is carried out with the aim of combining WebRTC and Firebase technology to create a mobile application that provides video call and random chat services. This application is expected to overcome the limitations of written and asynchronous communication that are often found in conventional chat applications.

2. RESEARCH METHOD

This research uses a software development methodology consisting of several stages, namely requirements analysis, system design, implementation, testing, and evaluation. Requirements analysis includes literature studies related to WebRTC, Firebase, and React Native technologies to determine application specifications (Kaligis & Fatri, 2020). System design involved designing an architecture that integrates WebRTC for video calls, Firebase for real-time data management, and React Native as a mobile application development framework, as well as a user interface (UI/UX) design that was tested with several users. Implementation included application development using React Native, implementation of the video call module with `react-native-webrtc`, and Firebase integration for chat and authentication features, with unit testing performed on each component. Testing involved functional testing to ensure all features worked to specification, performance testing to measure latency, video call quality, and connection stability, and user acceptance testing to obtain feedback. Evaluation involved analyzing the test data to identify the strengths and weaknesses of the application and preparing a final report documenting the research process, test results, analysis, and recommendations for further development.

3. RESULTS AND DISCUSSION

A. Discussion

Random Video Call and Chat Application Using WebRTC and Firebase Based Mobile can be defined as a communication solution that combines WebRTC technology for video calls and Firebase for real-time data management. The main purpose of developing this application is to provide a platform that allows users to interact directly and randomly with others through video calls and chats. The implementation of this application also aims to overcome the limitations of existing video call and chat applications by offering richer and more responsive functionality.

B. Definition of technology

Technology refers to the development and use of various devices or systems to solve problems that humans face in everyday life. Technology plays an important role in facilitating and improving efficiency in various areas of life, such as communication, transportation, health, and education. Through technological innovation, humans can find more effective and efficient solutions to various challenges, thereby improving the overall quality of life (Sudiantini et al., 2023).

C. Definition of React Native

React Native is an open-source framework developed by Facebook to create mobile applications that can run on iOS and Android platforms. Using React Native, developers can write code in JavaScript that is then converted into native applications, offering optimal performance across multiple devices. The framework also provides a hot reloading feature, which speeds up the development and debugging process by allowing developers to see code changes directly without having to restart the application (Rohman & Bhakti, 2024).

D. Definition of Firebase

Firebase, developed by Google, serves as the real-time database in this app. As an application development platform, Firebase offers a fast and reliable database management solution with instant data synchronization among users. In addition, Firebase provides various additional services such as user authentication and push notifications, which support efficient data management and user interaction (Panjaitan & Pakpahan, 2021).

E. Definition of WebRTC

WebRTC (Web Real-Time Communication) is used as the medium for video calls in this application. This technology enables audio and video communication directly through a web browser without the need for additional plugins. Developed by Google, WebRTC supports various audio and video codecs to ensure optimal communication quality in various network conditions. Using a peer-to-peer (P2P) protocol, WebRTC reduces latency and improves interaction quality, making it an ideal solution for real-time video calls. The combination of React Native, Firebase, and WebRTC aims to provide responsive, secure, and high-quality video call and random chat services, as well as provide an optimal user experience on various mobile platforms (Suciu et al., 2020).

F. Design system

F.1. Register and login

This sequence diagram illustrates how the register and login process uses firebase.

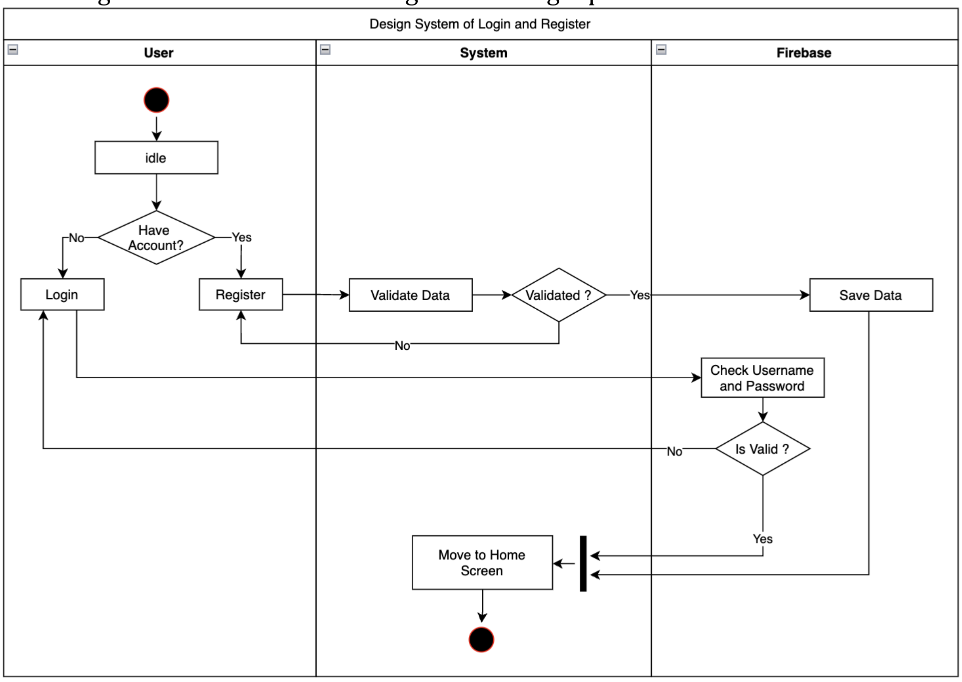

Fig 1. Sequence diagram system of login and register

F.2. Chat

This sequence diagram illustrates the process of connecting the two users and communicating via chat using firebase.

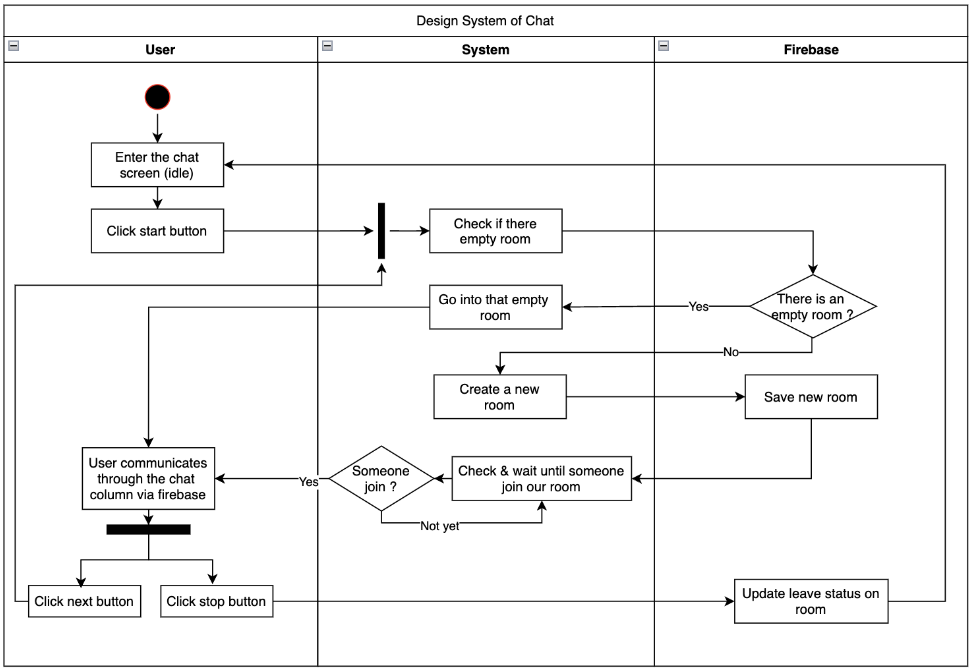

Fig 2. Sequence diagram system of chat

F.3. Video call

For this video call sequence diagram, the logic is the same as the chat sequence diagram, but one more level is added, namely WebRTC for the process of exchanging media such as video and sound to produce a video call feature.

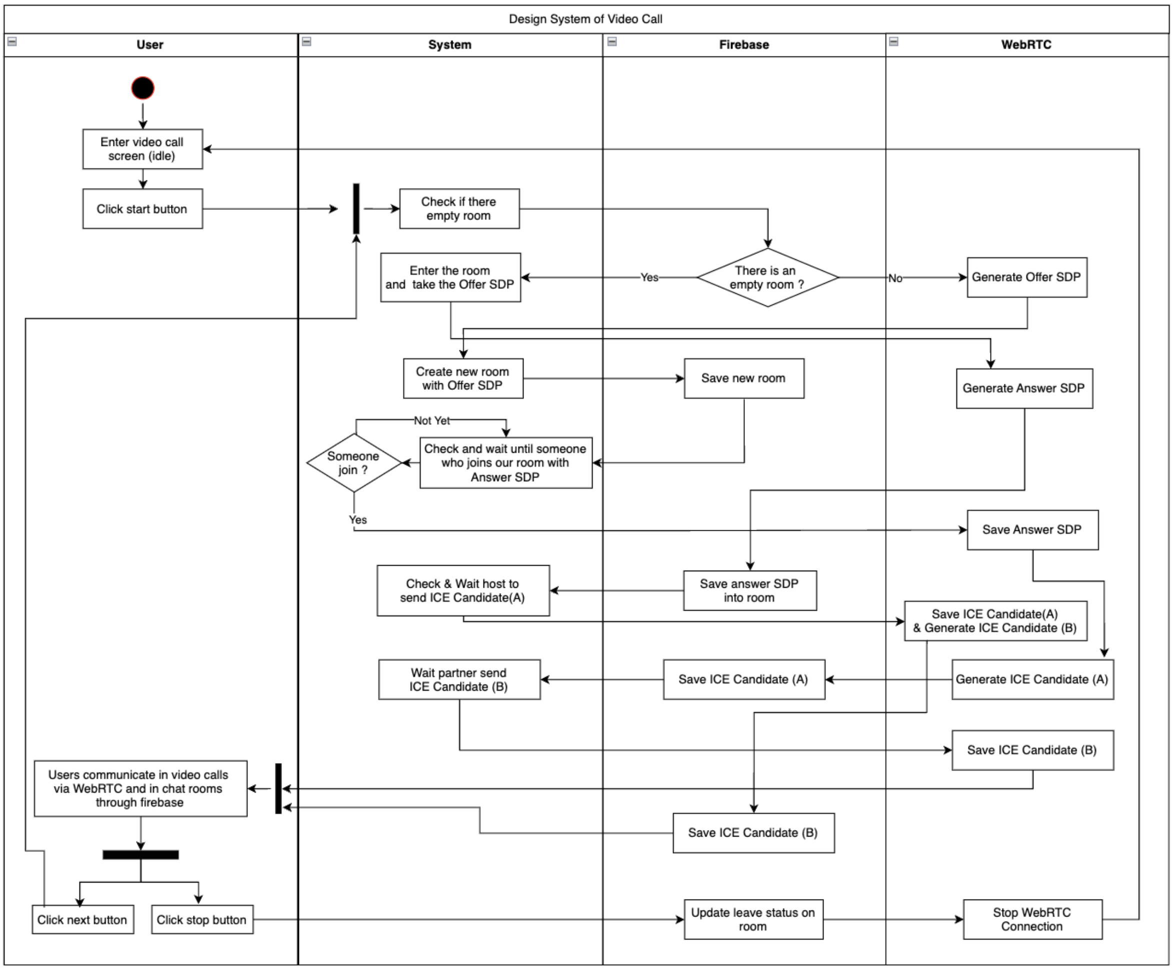

Fig 3. Sequence diagram system of video call

G. Features

G.1. Login screen

The Login Screen is specifically designed for users who already have an account, allowing them to enter credentials such as username and password to access the application. For users who do not have an account, there is a “Sign In” button that will direct them to the Register Screen to register.

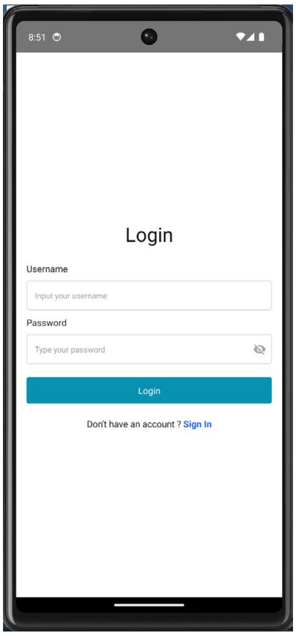

Fig 4. UI/UX Login screen

G.2. Register screen

Register Screen is a page for users who do not yet have an account. On this page, users can register themselves by filling in the required information, such as user name, username, and password. After the registration process is complete, users can use the newly created account to log into the application.

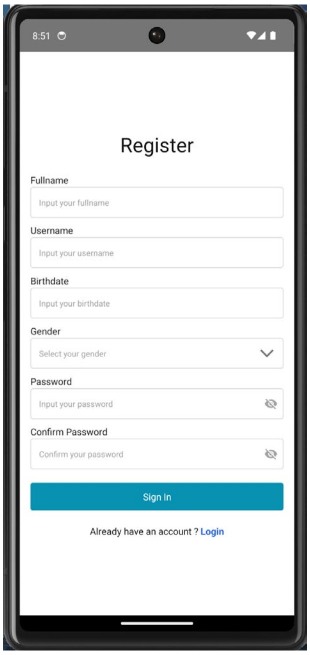

Fig 5. UI/UX Register screen

G.3. Chat screen

The Chat Screen allows users to interact with random people through the chat feature. There are several main actions that can be performed: the Start Button is used to start a chat by searching for a random partner, the Stop Button serves to stop the chat and disconnect with the current partner, the Next Button to disconnect with the current partner and search for a new partner, and the Message Input which allows users to send text messages to chat partners.

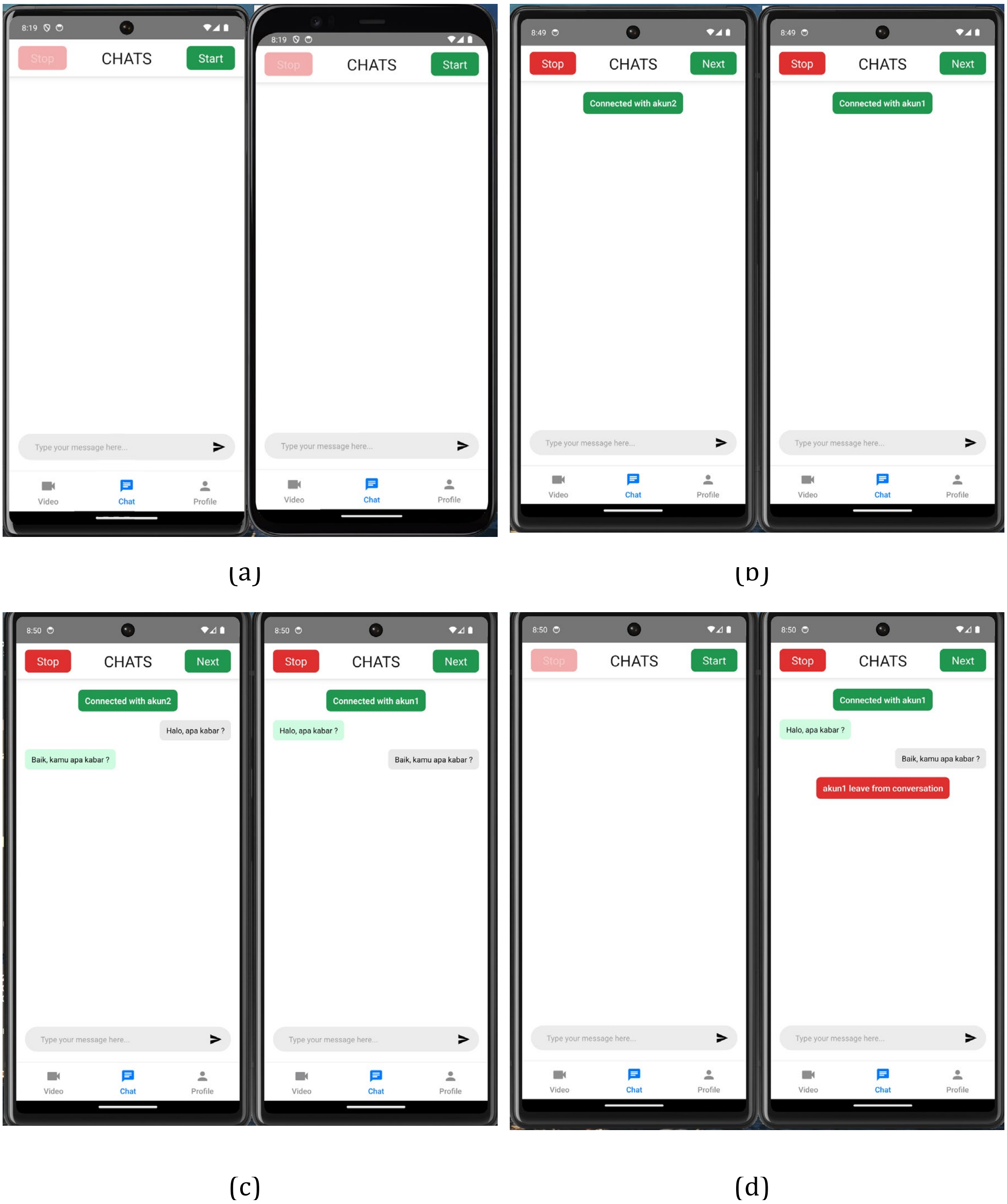

Fig 6. UI/UX Chat screen,
(a) Click Start Button, (b) Connected with partner, (c) Chatting with partner, (d) Stop Button

G.4. Video Call screen

The Video Call Screen is similar to the chat page, but adds a video call feature. On this screen, users can use the Start Button to start a video call by searching for a random partner, the Next Button to break the video call with the current partner and search for a new partner, and the Stop Button to end the video call and disconnect. In addition, users can also use the Message Input to send text messages during the video call. With these features, the app provides interactive and flexible communication options through both chat and video calls.

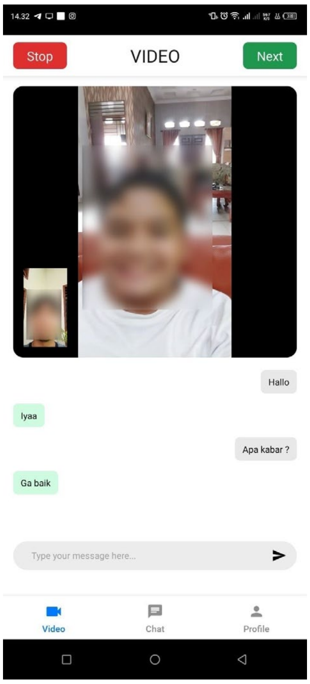

Fig 7. UI/UX video call screen

4. CONCLUSION

The development of a random video call and chat application based on WebRTC and Firebase with React Native as a mobile framework presents an innovative and effective communication solution. With key features such as login and register for user authentication, chat with random partners, and video calls that support visual and audio interaction, the app provides a more dynamic and interactive communication experience. WebRTC integration ensures optimal video call quality with low latency, while Firebase supports real-time data management and reliable authentication. React Native facilitates consistent mobile app development across multiple platforms. Overall, the app offers a useful solution for users who are looking for new ways to interact with others in a random and direct way, as well as enhancing the communication experience in an increasingly connected digital age. As such, the app is expected to fulfill modern communication needs and provide significant benefits in the way people interact virtually.

5. SUGGESTION

To improve the usability and attractiveness of the Video Call and Chat Random application, there are several things that can be developed. The development of a web version of the Video Call and Chat Random application will increase the flexibility of use and allow access from various devices with a browser without the need for additional installation. In addition, the addition of image or media sending features in chat will enrich the communication experience, allowing users to share visual information and other media.

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