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Design of a Customer Service System Based on Video Conference Using WebRTC

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Abstrak

Perkembangan teknologi telah mengubah cara customer service beroperasi, dari interaksi tatap muka ke platform digital yang lebih efisien. WebRTC muncul sebagai teknologi utama dalam membangun sistem customer service berbasis video conference untuk komunikasi real-time dan peningkatan pengalaman pelanggan. Penelitian ini berfokus pada pengembangan sistem video conference dengan WebRTC, melalui tahapan perancangan, implementasi, dan pengujian. Hasil dari penelitian ini menunjukkan bahwa pengujian pada Microsoft Edge memberikan total waktu inspeksi 14.629 ms dengan penggunaan memori 584,1 MB dan CPU 4,4%. Sedangkan, pada Google Chrome, total waktu inspeksi tercatat 14.845 ms dengan penggunaan memori 656,1 MB dan CPU 0,8%. Hasilnya menunjukkan penggunaan sumber daya yang efisien.

Kata kunci: *Customer Service, Sistem, Web Real Time Communication (WebRTC), Video Conference*

Abstract

Technological developments have changed the way customer service operates, from face-to-face interactions to more efficient digital platforms. WebRTC is emerging as a key technology in building video conferencing-based customer service systems for real-time communication and enhanced customer experience. This research focuses on developing a video conferencing system with WebRTC, through the stages of design, implementation, and testing. The results of this research show that testing on Microsoft Edge provides a total inspection time of 14,629 ms with 584.1 MB memory usage and 4.4% CPU. Meanwhile, on Google Chrome, the total inspection time was recorded at 14,845 ms with memory usage of 656.1 MB and CPU of 0.8%. The results show an efficient use of resources.

Keywords : *Customer Service, System, Web Real Time Communication (WebRTC), Video Conference*

1. Introduction

Customer service is an activity that aims to provide services to customers, both before, during, and after they purchase products or services. These activities are designed to meet the needs or answer customer questions, so as to increase their satisfaction [1]. Along with the development of technology, customer service has undergone major changes, especially through the use of online platforms such as video conferencing. WebRTC technology enables peer-to-peer real-time communication in video conferencing [2], and simplifies the process of developing and managing video conferencing by providing solutions that are efficient and easy to integrate [3].

WebRTC is a solution in the development of real-time video conferencing due to its ability to simplify and enable implementation directly on the web without requiring complex installation or configuration. This technology facilitates real-time audio and video communication by providing features that support direct peer-to-peer communication. By overcoming technical challenges such as the need for intermediate servers and integration complexity, WebRTC offers an efficiently integrated framework. It also performs end-to-end encryption and optimizes bandwidth usage, which contributes to the development of a more responsive, stable, and manageable video conferencing system. As a result, users get a better and more reliable experience in video communication [4].

The development of WebRTC-based video conferencing applications shows advantages in various aspects. It is capable of connecting multiple users while effectively supporting screen sharing and recording features [5]. WebRTC is also used to create web-based video conferencing rooms that are easily accessible and can be embedded as extensions of larger web services, enabling seamless real-time communication between mobile devices and personal computers [6]. The utilization of JavaScript, WebRTC protocol, and other components further strengthens real-time communication services integrated directly in the browser [7]. WebRTC also offers bandwidth efficiency, low latency, improved user data security, and significant advantages over other video conferencing systems [8]. A video conferencing system refers to a communication platform that enables real-time interaction between two or more parties through electronic transmission of voice, video, and data. Systems built using WebRTC enable efficient one-to-one methods, and support features such as screen sharing to share content directly during the conversation.

Based on the previous explanation, the development of WebRTC-based video conferencing system will be focused on one-to-one communication model. The reason for choosing this one-to-one model is because the system is designed to mimic the way customer service works, which usually serves one customer in one communication session. The system is expected to replicate the customer service process in the form of video conferencing, so that interactions can take place in a personalized and effective manner, in line with the service methods applied in conventional customer service. WebRTC will be utilized to support smooth and high-quality real-time communication, enabling a more optimal interaction experience for users.

2. Research Method / Proposed Method

The research method begins with the process of collecting data from relevant literature studies, followed by the development of the main functions of the system which are the core of the research. Next, the development of additional features that support the main function is carried out, and ends with a thorough system testing stage to ensure all components function properly.

2.1 Literature Study Data Collection

Data collection from the literature study was the first step in this research process. This stage involves searching and collecting relevant references from various sources, including journals, books, and websites related to the research topic.

2.2 Design of the System

The second method in this research process is system design. At this stage, the system is designed with the help of a flowchart before starting the manufacturing process. The goal is to understand the workflow of the system that will be implemented later.

Figure 1 is a flowchart of how the system operates. Customer service starts by creating a conference room and entering the room first. After that, customer service shares the URL with the customer. Customers use the URL to access the form page and fill in their personal information before joining the conference without the need to create an account first. During the conference, customers can communicate with customer service and use the available features. Upon completion, the customer can close the conference. If the customer forgets or does not close it, customer service can close the conference if necessary.

2.3 System Main Function Development

The third method in this research process is the development of the main functions of the system. At this stage, the program code that supports the development of the core functions of the customer service system is created. The main functions developed include creating a real-time one-to-one communication method through video conferencing, developing a function to create a video conference room (create conference), and creating a function to end the video conference session (end conference).

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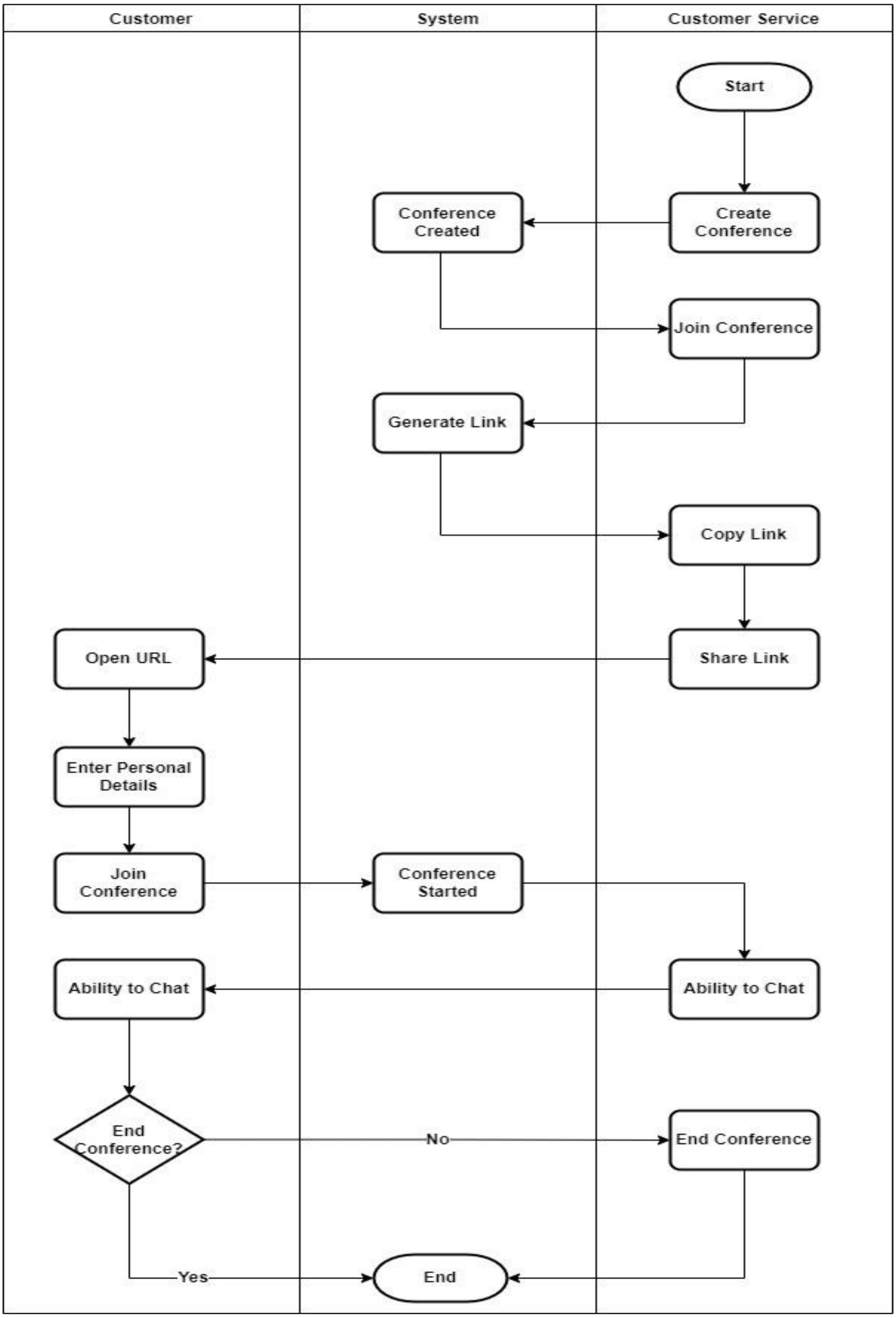

Figure 1. Flowchart System

2.4 Additional Features Development

The fourth method in this research process is the development of additional features. At this stage, program code is created that supports the addition of features to the customer

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service system. The features developed include functions to enable or disable the camera, set the microphone, and share the screen.

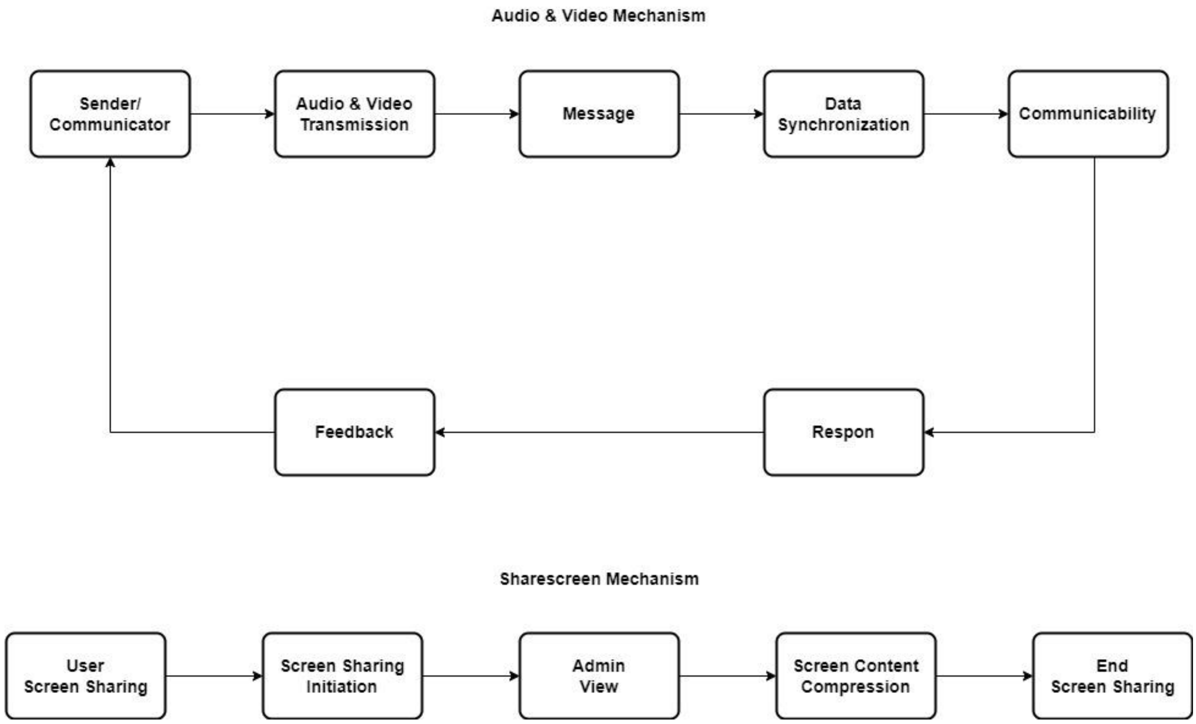

Figure 2. Audio, Video, Screensharing Mechanism

Figure 2 shows the mechanism of how audio, video, and sharecreen work. Messages are sent using input devices such as cameras and microphones to capture video and audio, which are then converted into a more efficient format for transmission. The data is processed to maintain synchronization between audio and video to keep them in sync and quality throughout the conference. In addition, the screen sharing feature allows the screen display to be shared and compressed to ensure optimal transmission. Screen sharing can be stopped at any time through the available features.

2.5 System Test

The last method in this research process is system testing. At this method, a test of the system that has been developed is carried out by checking each of the main functions and features that have been made. In addition, this stage also includes data collection about the advantages and disadvantages of the system based on the results of the trials conducted

3. Literature Study

This research uses various related references, such as journals, books, the internet, as well as company reports and research, as a reference in developing existing ideas and concepts.

3.1 Customer Service

Customer Service includes various activities that aim to provide satisfaction to customers through services performed by someone, so as to meet customer needs and expectations [1]. Customer service can be done through various communication channels, such as telephone, email, live chat, or social media. The purpose of this service is to maintain good relationships between organizations and customers, increase customer satisfaction levels, and build loyalty to the products or services offered. Efficient and responsive service can also improve the organization's image in the eyes of customers.

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3.2 Video Conference

Video conference is a communication method that connects two or more locations by transmitting sound, images, and data electronically, allowing real-time interaction. It is more personalized and effective than audio conference, as it allows participants to see facial expressions and body language. This technology involves both hardware and software. Video conferencing can take place between two connected locations or involve multiple locations. Communication can take place in a dedicated studio, a home computer with a webcam, or a third-generation mobile phone. In addition to audio and video transmission, the technology also supports document sharing and displaying information on a whiteboard [9].

3.3 Web Real-Time Communication

WebRTC is a technology that enables direct real-time communication between web browsers. The main uses of WebRTC include video calling, web conferencing, and direct data transmission. Different from other real-time systems, communication through WebRTC is managed directly through the web server using JavaScript APIs [10]. WebRTC is fundamentally designed for peer-to-peer communication that is optimal for 2 to 4 users in a direct connection. Performance will decrease because the device has to manage multiple connections simultaneously.

4. Result and Discussion

This research resulted in the implementation of the final system and the results of system testing on various browsers. The following is a review of the findings of this research.

4.1. Server Build

The first phase of this research involved the creation of a server using WebSocket, which serves for real-time exchange of data and information between devices. This server was built with the help of the Flask framework, which simplifies route management, HTTP request handling, and configuration settings. Flask also provides extensions such as Flask-SocketIO to enhance the functionality of WebSocket, enabling real-time communication between users and servers through connection management, sending and receiving events, and sending complex messages and data. In addition, Flask-Session is used for session management on the server side. The following is an overview of the server in the system.

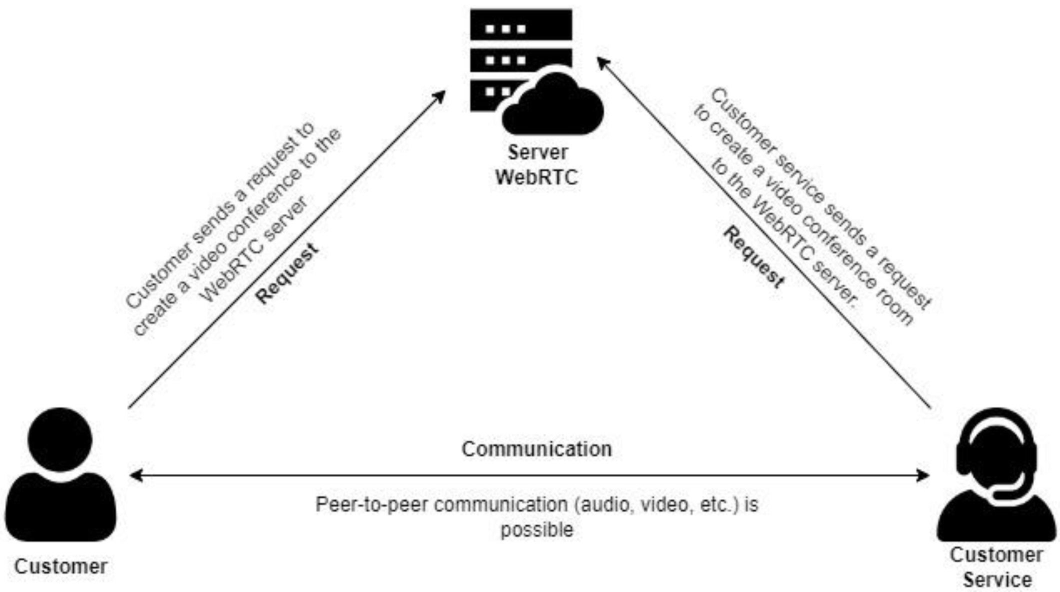

Figure 3. how does the system work

Figure 3 shows how the server works in the system. Requests from customer service are sent to the system and processed to form a meeting room. Meanwhile, the customer who submitted the request will be directed to the meeting room that has been formed according to the submitted request.

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4.2 Create Room Page Creation

The next stage is the creation of a create room page, which aims to facilitate the creation of video conference meetings. This page allows users to set various meeting details such as title, password, description, and customer search, to ensure the meeting can be organized properly and effectively.

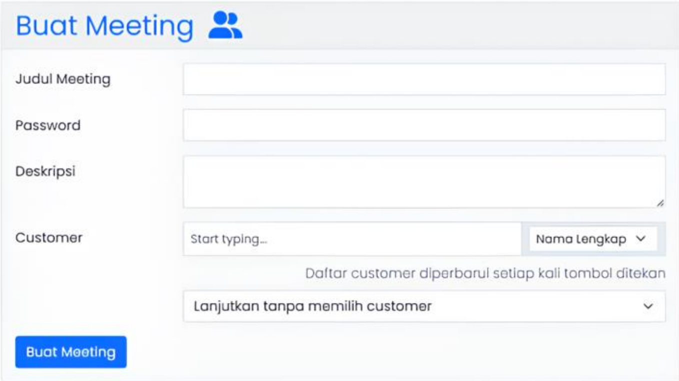

Figure 4. Create Room Page

Figure 4 is a view of the create room page. This page displays a form that must be filled in with the meeting title, password, description, and customer search options. There is also a "Create Meeting" button to confirm the completion of the form. In addition, this page also comes with an error display to handle any input issues that may occur.

4.3 Join Meeting Page Creation

The next stage was the creation of the join meeting page, which was designed to make it easy for customers to join a video conferencing meeting. This page ensures that customers can easily enter the required information, such as the registered phone number and meeting password, to access the desired conference session. The meeting can only be conducted by one admin and one subscriber, if there are other subscribers who want to join the meeting that has been created, a notification will appear stating that the meeting is full. If you cannot join, then the user can contact another admin so that a new room can be created because the room system is one-to-one.

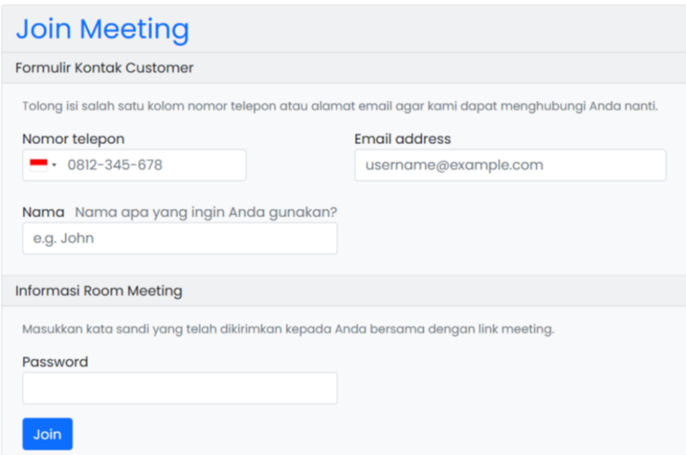

Figure 5. Join Meeting Page

Figure 5 is a view of the join room page. This page displays a form requesting the meeting phone number and password. The form comes with validation to ensure the phone number is filled in with the correct format, using the intl-tel-input plugin for international phone

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number validation. In addition, there is an error display that handles problems in filling out the form.

4.4 Meeting Page Creation

The last stage in the implementation of the video conference-based customer service system is the creation of the meeting page. This page involves important elements such as microphones and videos that support the main functions of the system, as well as providing connections with WebSocket for communication with the server and setting up WebRTC Peer Connection to manage communication between users. The status of connected users is sent to the server, and user information is updated as needed, with the server sending commands to start the video conference including access to the camera, microphone, and screen sharing. In addition, this stage handles status changes of features such as camera, microphone, and screen sharing, with the status sent to the server and the display updated. It also manages the receipt of ICE candidates to keep the WebRTC connection stable, adding candidates to the Peer Connection and disabling features if candidates are null. Session termination is performed by sending a notification to the server, closing the connection, and redirecting the user out of the session, ensuring an efficient and responsive video conferencing experience.

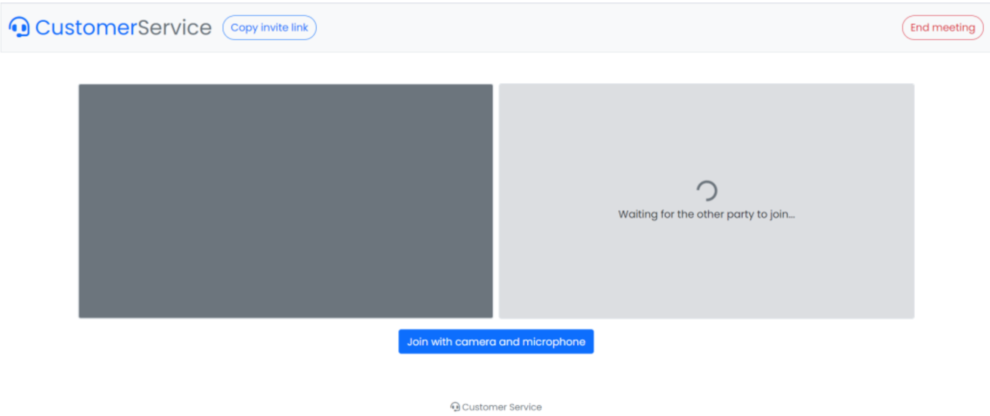

Figure 6. Waiting Meeting Page (Both Side)

Figure 6 is the initial view showing empty video areas on both sides of both the admin and customer views, indicating that the conference has not yet started. There is a button labeled “Join camera and microphone” to start participation in the conference, as well as an “End meeting” button to end the conference if necessary.

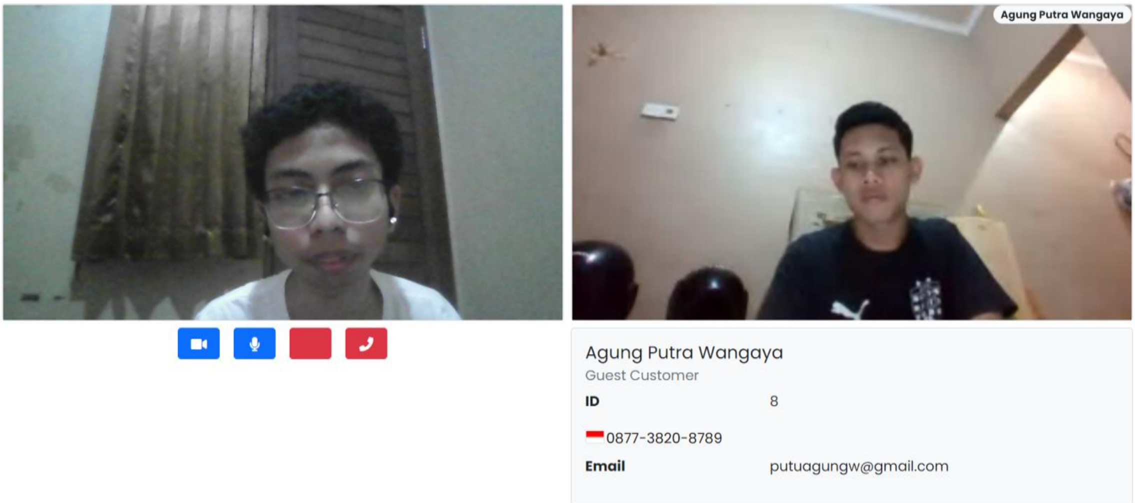

Figure 7. Meeting Page (Admin Side)

Figure 7 is a view of the meeting room page from the admin side. The display shows buttons with different colors to indicate their function status, blue buttons indicate active functions, while red buttons indicate inactive functions. In addition, the display also includes personal information from other users according to the form on the join meeting page.

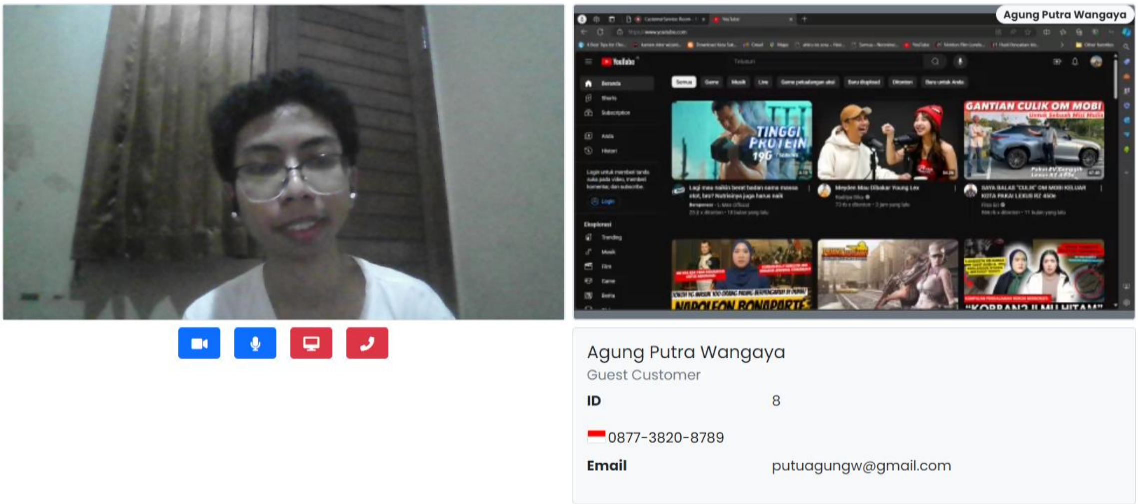

Figure 8. Screen Sharing Session (Admin Side)

Figure 8 shows the view from the admin when another user does a screen share, the video that previously showed the user's face, changes to the screen being shared. This change shows the content of the screen being shared by the other user, replacing the user's face.

4.5 Resource Requirement Testing

Resource requirement testing aims to evaluate the use of system resources during operation. The tests were conducted on a local server with port 5000 as the default in Flask, and used a tunneling system through ngrok to allow remote access by creating a secure tunnel from the server to the testing device. This allowed testing with a variety of devices. During testing, two different browsers, Microsoft Edge and Google Chrome, were used to evaluate the differences in resource usage and system performance on each platform. This test aimed to provide a comprehensive overview of how the system operates in various browser environments.

4.5.1 Microsoft Edge Testing

Testing on Microsoft Edge was done using two methods: the 'Inspect' feature in the developer tools for in-depth analysis, and Task Manager for detailed information on CPU, memory, and network usage.

Table 1. Microsoft Edge Task Manager Results

CPU	Memory	Disk	Nework
4.4%	584 MB	0,1 MB/s	3,1 Mbps

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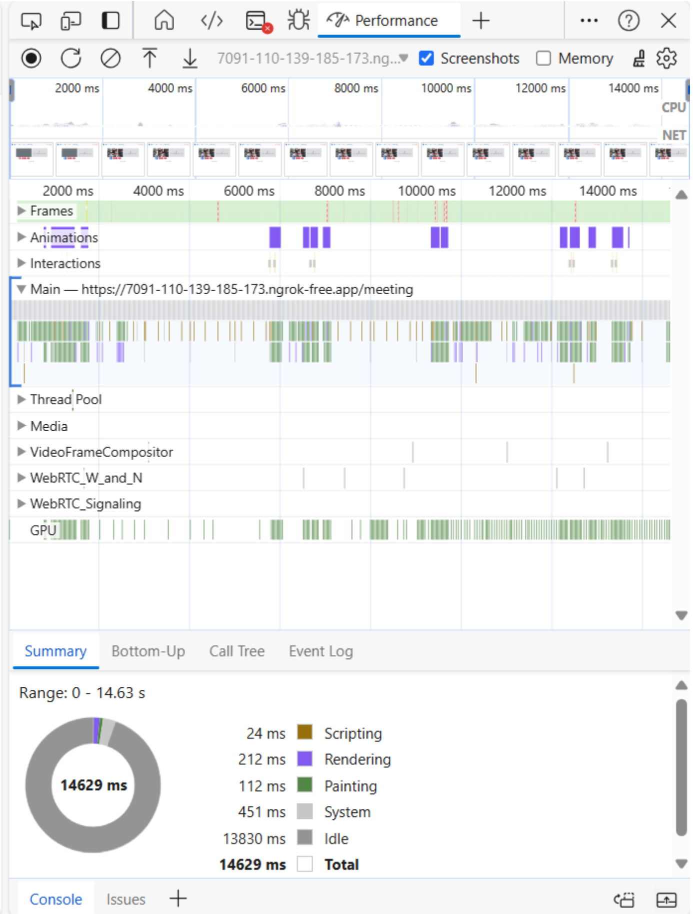

Figure 9. Microsoft Edge Performance Inspect Results

Figure 9 performance test results on Microsoft Edge browser. Based on the results of 15 seconds of recording, it shows the following times: scripting 24 ms, rendering 212 ms, painting 112 ms, system 451 ms, and idle 13,830 ms, totaling 14,629 ms. Table 1 shows the results of resource usage checked using the task manager. CPU usage was recorded at 4.4%, memory at 584.1 MB, disk activity at 0.1 MB/s, and network at 3.1 Mbps. This data shows that the system uses resources efficiently.

4.5.2 Google Chrome Testing

System testing on Google Chrome was conducted using two methods: the 'Inspect' feature in developer tools for in-depth analysis, and Task Manager for details on CPU, memory, and network usage.

Table 2. Google Chrome Task Manager Results

CPU	Memory	Disk	Nework
0.8%	656,1 MB	0,1 MB/s	0,1 Mbps

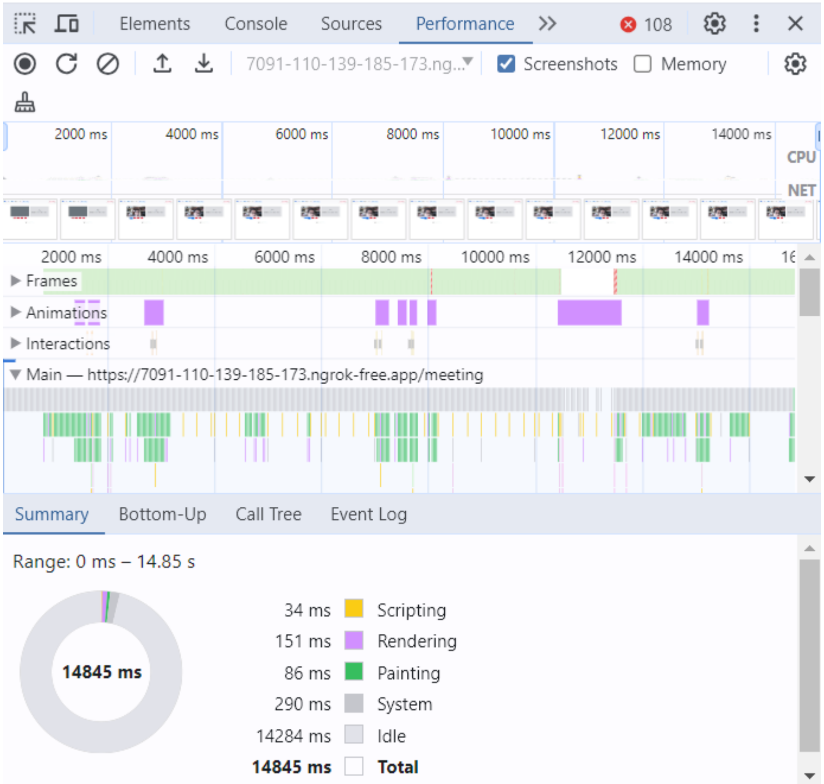

Figure 10. Google Chrome Performance Inspect Results

Figure 10 performance test results on Google Chrome browser. The test results with a 15-second recording showed the following times: scripting 34 ms, rendering 151 ms, painting 86 ms, system 290 ms, and idle 14,284 ms, totaling 14,845 ms. Table 2 shows the results of resource usage checked using the task manager. CPU usage was recorded at 0.8%, indicating a low processor load. Memory usage reached 656.1 MB, disk activity was minimal at 0.1 MB/s, and network usage was 0.1 Mbps, indicating efficient resource utilization.

5. Conclusion

This research shows that designing a videoconferencing-based customer service system requires a structured approach that includes user management, server management, and WebRTC-based real-time communication. WebRTC technology supports peer-to-peer communication, local and remote media streams, and real-time communication management through ICE candidates. Performance testing shows that Google Chrome is more efficient than Microsoft Edge. Although Edge has a faster execution time (24 ms vs. 34 ms), Chrome excels in rendering and painting (151 ms vs. 212 ms and 86 ms vs. 112 ms), CPU usage (0.8% vs. 4.4%), and bandwidth (0.1 Mbps vs. 3.1 Mbps). With these efficiencies, Google Chrome is more optimized to support video conferencing-based customer service systems.

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