Internet of Things Intercommunication Using SocketIO and WebSocket with WebRTC in Local Area Network as Emergency Communication Devices

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***Abstract*—The Emergency Intercom System (EIS) is a communication and video conferencing system designed to enhance emergency response in areas, such as malfunctioning lifts. This paper presents a feasibility analysis of EIS as internet of things device, focusing on the utilization of WebRTC over WebSocket, Socket.io, and LAN device hostname assignment. The study evaluates performance metrics including latency, throughput, dropped calls, lost packets, audio and video quality, resource utilization, and network bandwidth. The results indicate that EIS offers acceptable latency, sufficient throughput for simultaneous calls, minimal dropped calls and lost packets, and satisfactory audio and video quality. The system demonstrates effective communication and video conferencing capabilities, making it a promising solution for improving emergency response in areas. The findings highlight the potential for further research to optimize performance, address security considerations, and extend the system's applicability in diverse emergency scenarios.**

***Keywords*—WebRTC, Socket.io, Internet of Things**

I. Introduction

Malaysia has undergone remarkable growth and progress in recent years, as evidenced by the proliferation of skyscrapers across the country. According to the latest rankings by the Council on Tall Buildings and Urban Habitat (CTBUH), Malaysia has secured an impressive position in the global landscape of tall structures. It currently holds the sixth rank for the number of completed buildings over 150 meters in height, positioning it as one of the tallest cities in the world [1]. Furthermore, Malaysia claims the fourth position as the tallest city in Asia, further highlighting its prominence in the region [1]. Many of these buildings are equipped with elevators or lifts, which are used by a large number of people on a daily basis

In the context of Malaysia's thriving skyscrapers, it is important to recognize the significance of elevators as essential transportation systems. However, it is crucial to acknowledge that elevators can provoke anxiety and discomfort for individuals with conditions like agoraphobia or claustrophobia. Addressing these psychological factors is vital to foster inclusive environments within Malaysia's tall structures.

Agoraphobia and claustrophobia, two common anxiety disorders, can significantly affect individuals' lives. Agoraphobia is the fear of being trapped in situations where escape is challenging or impossible during a panic attack, leading individuals to avoid crowded or confined spaces. While not all individuals with agoraphobia are triggered by elevators, the enclosed nature of elevators can induce anxiety for some. Similarly, claustrophobia centers around the fear of enclosed spaces, and elevators, being small and confined, have the potential to trigger claustrophobic reactions. Considering the prevalence of these anxiety disorders, addressing the psychological impact of elevators becomes vital, particularly in high-rise buildings where elevator usage is widespread. Understanding and addressing these concerns promotes the well-being of elevator users and fosters inclusive environments in skyscrapers and tall structures [12].

According to the New Straits Times (NST) article from 2012 to October 2018, there have been a total of 52 reported incidents involving elevators and 33 involving escalators in Malaysia, according to the Occupational Safety and Health Department. Of these incidents, six resulted in death, six in permanent physical disability, and 58 in sustained injuries without permanent disability [6]. NST also reported in October 2022, six people, including a two-year-old girl, were riding in a lift at a condominium in Brickfields, Malaysia when the lift suddenly fell from the eighth floor to the ground floor. The incident resulted in a woman and a man suffering from broken legs, while the other passengers were not injured [2].

Therefore, the Emergency Intercom System (EIS) is the latest technology developed to resolve and handle elevator communication issues. It utilizes a local area network (LAN) for faster response, less disruption, and lower costs while maintaining high availability without the need for an internet connection. EIS allows passengers to contact security in case of an emergency through video calling.

II. Feasibility Analysis

The Emergency Intercom System (EIS) is a communication and video conferencing system designed to improve emergency response in malfunctioning lifts. It uses a local area network (LAN) and several technologies to enable communication and video conferencing between lift users and security personnel.

WebRTC, or Web Real-time Communication, is an open-source communication protocol that facilitates real-time voice, text, and video streaming between devices and web browsers. It adheres to a set of rules, making it ideal for signaling purposes and the seamless transfer of real-time data between a server and a client [10].

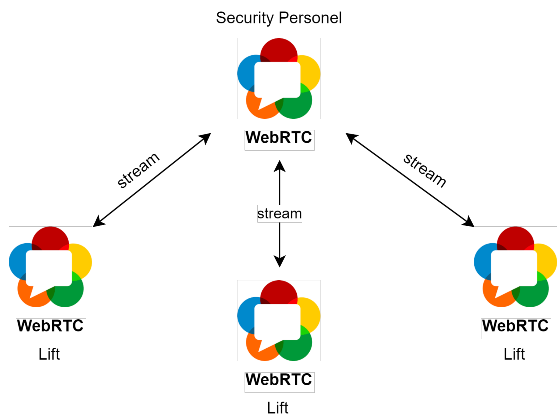

Fig. 1. EIS WebRTC Connection

Additionally, it should be noted that when employing WebRTC for peer-to-peer (P2P) media or data exchange, the utilization of servers is necessary. While media is typically exchanged directly between peers, there are instances where a TURN server is employed to relay media if one or both browsers are situated behind a restrictive NAT or a firewall [11], ensuring seamless communication even in challenging network environments.

Socket.io can be used to create a room for secure and private communication between the security authority and lift passenger, while also supporting the WebRTC technology that is a core part of EIS' real-time communication features. In WebRTC, signaling messages are transmitted between the client and server to establish a direct peer-to-peer connection for audio and video communication. Socket.io can be used to transmit these messages, making it a critical component for establishing real-time communication between lift users and security personnel in EIS. Its ease of implementation and compatibility with existing technologies make it a viable option for use in EIS.

The bidirectional channel between the Socket.IO server (Node.js) and the Socket.IO client (browser, Node.js, or another programming language) is established with a WebSocket connection whenever possible, and will use HTTP long-polling as a fallback [9]. This ensures reliable and efficient communication between the server and client, with WebSocket being the preferred choice due to its low latency and high throughput. The fallback option of HTTP long-polling ensures that communication can still be maintained even in scenarios where WebSocket is not supported, providing a seamless experience for users [9].

To ensure that EIS is able to communicate effectively over the LAN, it is necessary to configure a hostname assignment method such as DHCP or manually configure the "hosts' ' file on each device. A local reverse proxy can also be configured to allow for SSL and caching, which can improve the performance and security of the system.

In terms of browser compatibility, EIS is compatible with a range of popular browsers including Chrome, Chromium, Firefox, Edge, and Safari. To ensure the best performance, it is important to conduct a performance analysis using a device such as a Raspberry Pi 3 to evaluate factors such as source and destination FPS, loss, upload and download rates, and RTT. This can help to identify the best configuration for optimal performance.

A. Literature Review

Web Real-Time Communication (WebRTC) is an open-source project that enables real-time communication between browsers and mobile applications via APIs. With the growing demand for video conferencing and collaboration, WebRTC is becoming a popular choice for developers to build such applications. However, ensuring the Quality of Experience (QoE) for users in a WebRTC-based application is a challenging task, as it is affected by various factors such as network conditions, device capabilities, and encoding bitrates.

To improve the overall QoE, researchers have proposed different approaches to dynamically adapt the video bitrate based on network conditions. Petrangeli et al. [7] proposed a WebRTC-compliant framework for dynamic video bitrate adaptation in remote teaching applications. The framework uses a limited number of encoders and dynamically forwards the most suitable stream to the receivers based on their bandwidth conditions. The proposed framework improves the received video bitrate up to 11% compared to a static solution.

Another approach proposed by Petrangeli et al. [8] is to improve the scalability of video collaboration applications. The proposed framework uses a centralized Selective Forwarding Unit (SFU) to dynamically forward the most suitable stream to each of the receivers. The controller dynamically recomputes the encoding bitrates of the sender to follow the long-term bandwidth variations of the receivers and increase the delivered video quality.

Lee et al. [4] proposed a reinforcement learning (RL) based framework called R-FEC for video and Forward Error Correction (FEC) bitrate decisions in video conferencing to improve the overall QoE. The framework aims to maximize the user QoE while minimizing the congestion in the network by automatically learning through the results of past decisions and adjusting video and FEC bitrates. The experiments showed that R-FEC outperformed the state-of-the-art solutions in video conferencing, with up to 27% improvement in its video rate and 6dB PSNR improvement in video quality over the default WebRTC.

In conclusion, the proposed approaches aim to dynamically adapt the video bitrate and encoding bitrates based on network conditions to improve the QoE in WebRTC-based applications. These approaches have shown promising results in terms of improving the received video bitrate and video quality in different scenarios. However, further research is required to determine the best approach for configuring WebRTC for the best QoE.

III. Methodology

A. WebRTC over WebSocket

The Web Real-Time Communication (WebRTC) over WebSocket methodology involves combining WebRTC, an open-source framework for real-time communication, with the WebSocket protocol. WebRTC enables bi-directional and real-time voice, text, and video streaming between devices and web browsers [10]. By leveraging the capabilities of WebRTC and the WebSocket protocol, the methodology enables efficient and reliable real-time communication between clients and servers. Fig. 2 illustrate the architecture of WebRTC network over the WebSocket as a signaling medium.

To establish a WebRTC connection over WebSocket, a client (such as a web browser) initiates the process by sending a WebSocket handshake request to the server, which responds with a handshake response, establishing the WebSocket connection. This connection provides a reliable channel for bidirectional data exchange between the client and the server.

In the context of video conferencing, WebRTC enables the exchange of video and audio data between the client and the server using the WebRTC data channel. The WebRTC data channel establishes a peer-to-peer connection, allowing direct communication between the participants without relying on a central server [10]. This approach enhances the scalability and reduces the latency of the system, enabling real-time audio and video communication.

To ensure the security and privacy of the transmitted data, the WebRTC data channel utilizes the Secure RTP (SRTP) protocol for encryption [10] [5]. This encryption mechanism safeguards the audio and video streams, preventing unauthorized access to the content exchanged during the video conferencing sessions.

Additionally, each WebRTC connection object includes an ICE (Interactive Connectivity Establishment) agent [10]. The ICE agent plays a crucial role in establishing and maintaining the connection between peers. It gathers local IP and port information, and with the assistance of a STUN (Session Traversal Utilities for NAT) server, retrieves the public IP and port information of the peer. The ICE agent performs connectivity checks between peers and sends connection keepalives to the STUN server, ensuring a reliable and stable connection. In cases where a TURN (Traversal Using Relays around NAT) server is configured, the ICE agent can utilize it as a fallback option to establish connectivity in challenging network environments [10].

By combining WebRTC, WebSocket, and the functionalities of ICE agents, the WebRTC over WebSocket methodology provides a robust framework for video conferencing applications, offering low-latency, secure, and efficient real-time communication.

B. SocketIO

The implementation of Socket.io in the EIS system will enable secure and private communication between the security authority and lift passengers. To achieve this, Socket.io will be integrated into the server-side code of EIS, allowing for real-time, bi-directional communication with clients. This will enable the creation of a private room for lift passengers and security personnel to communicate securely using the signaling mechanism of WebRTC. Figure 2 shows a sketch of the communication between end devices

In addition to supporting WebSocket technology, Socket.io provides a simple API for creating rooms and handling events, allowing for the secure and real-time transmission of signaling messages that are necessary to establish the direct peer-to-peer connection for audio and video communication in WebRTC. The fallback options provided by Socket.io, such as long-polling for clients that do not support WebSocket, will ensure that communication can continue even when network conditions are suboptimal [9], providing a seamless experience for lift passengers and security personnel.

C. LAN Device Hostname

Hostname assignment is the process of assigning a hostname, or a unique name, to a device on a network. There are several methods that can be used to assign hostnames, including:

- Dynamic Host Configuration Protocol (DHCP): This is a protocol that allows a server to automatically assign IP addresses and other network settings to devices on a network. When a device connects to the network, it sends a request to the DHCP server, which assigns it a hostname and other network settings based on a predefined set of rules.
- Manual configuration: Hostnames can also be assigned manually by configuring the ”hosts” file on each device. The ”hosts” file is a simple text file that maps IP addresses to hostnames, and can be edited to assign specific hostnames to specific devices

Hostname assignment is important because it allows devices on a network to be identified by a unique name, rather than just an IP address. This can make it easier to manage and troubleshoot network issues, as well as improve the overall usability of the network.

D. Power Supply: AC-Powered Devices

The Emergency Intercom System (EIS) devices will be powered using an alternating current (AC) power supply, chosen for several strong backing points to ensure reliability,

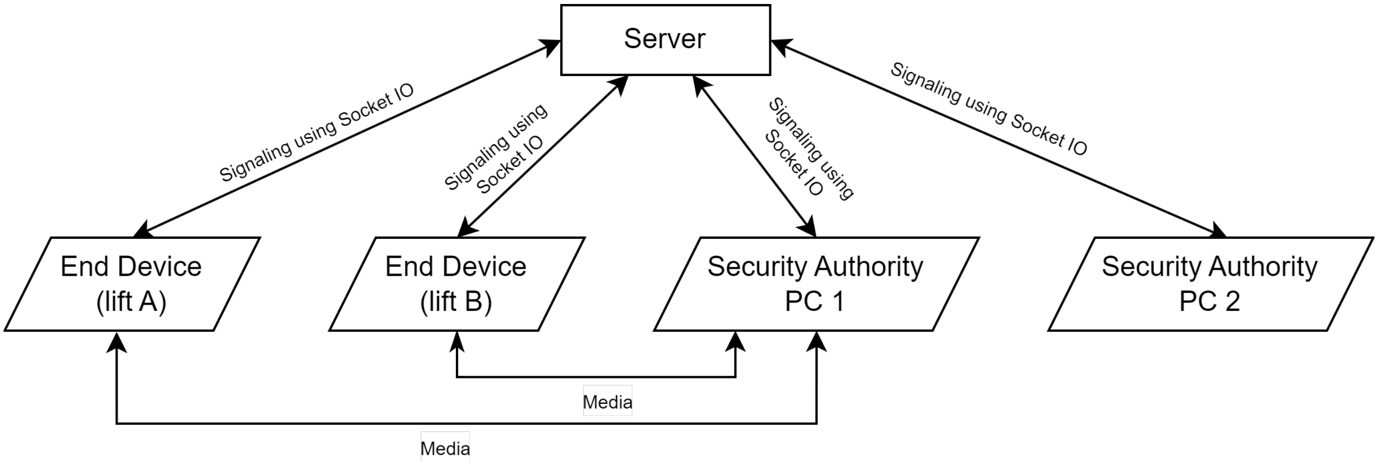

Fig. 2. EIS Network topology

continuous operation, and suitability for emergency scenarios. AC power supply provides a continuous and reliable source of electricity, ensuring uninterrupted communication even during power outages, a critical feature for emergency response. Additionally, AC power can meet the high demands of real-time audio and video processing, while the ability to sustain power for a few hours after a blackout ensures continued functionality during extended power disruptions. By eliminating the need for frequent battery replacements, AC-powered devices reduce maintenance and operational costs, making them more cost-effective for large-scale emergency communication deployments. Furthermore, the environmental benefits of AC power contribute to a lower environmental impact compared to batteries. Finally, AC power enables easier compliance with safety regulations, ensuring user protection during emergencies, further enhancing the EIS devices' reliability and effectiveness as an emergency communication solution.

E. Privacy Considerations

Ensuring the privacy and security of individuals within the lift is a critical aspect of the Emergency Intercom System (EIS) implementation. To address potential privacy concerns, specific measures will be implemented in the methodology.

- Limited Audio and Video Recording: The EIS will be designed with strict limitations on audio and video recording capabilities. The system will only enable recording during emergency calls or when explicitly initiated by authorized personnel, such as emergency responders or security personnel. Continuous audio or video surveillance will be avoided to prevent any unauthorized intrusion on the privacy of individuals in the lift.
- Anonymization and Data Encryption: Any recorded data will be anonymized to protect the identity of individuals. Additionally, all recorded data will be encrypted to ensure that only authorized personnel can access it, enhancing data security and privacy.
- Security Measures: The intercom system will be equipped with robust security measures, including authentication protocols and secure communication channels. This will prevent unauthorized access to the system and protect against potential data breaches.

IV. Result

A. Performance Metrics

Performance metrics are used to evaluate the system's performance. In the context of a video/audio intercommunication system using SocketIO, WebRTC, and WebSockets, important metrics include latency, throughput, and reliability.

Latency, also known as system delay, measures the time it takes for a signal to be transmitted. In this system, latency is measured from call initiation to call establishment. Low latency ensures a good user experience by enabling quick call establishment and synchronization of audio and video.

Throughput measures the amount of data transmitted within a specific time period. In this system, throughput can be measured by the number of simultaneous calls or concurrent video/audio streams. High throughput is crucial for handling a large user base and providing a satisfactory user experience.

Reliability can be assessed by monitoring dropped calls and lost packets. A lower number of dropped calls or lost packets indicates a more reliable system.

These performance metrics play a significant role in evaluating the effectiveness and efficiency of the video/audio intercommunication system.

B. Performance Evaluation

The Emergency Intercom System (EIS) will be evaluated using metrics including latency, throughput, reliability, and video/audio quality. Raspberry Pi 3 devices will serve as endpoints for initiating and receiving video/audio calls, with a resolution of 640x480 pixels and a maximum frame rate of 30 fps.

Latency evaluation involves measuring the time it takes for a call to be established. Tests will be conducted by initiating calls between Raspberry Pi 3 devices and measuring the time for establishment. Low latency ensures a prompt and synchronized audio-video experience.

Throughput evaluation involves measuring the number of concurrent video/audio streams. Tests will initiate multiple simultaneous calls to determine the system's capacity. High throughput enables efficient handling of users and a smooth user experience.

Reliability evaluation includes measuring dropped calls and lost packets. Tests will determine the occurrence of such instances during calls. Fewer dropped calls and lost packets indicate higher system reliability, ensuring uninterrupted video and audio transmission.

Furthermore, video and audio quality can be assessed by measuring bitrate and packet loss. This provides insights into the transmission's integrity and the system's ability to maintain quality. Comparing transmitted data to the original data reveals the system's performance in maintaining video and audio quality during transmission.

C. Result

The performance of the system was evaluated on four different computers, including a Raspberry Pi and three other computers with varying specifications. The test device specifications and listed on the Table I. The results are presented in the Table II

D. Latency

Latency refers to the time delay between data transmission and reception. It is a critical factor in emergency communication devices, as any delay could impact the effectiveness of real-time communication during emergencies. The measured latency values for the devices are shown in milliseconds (ms). Device 4 achieved the lowest latency with 40ms, indicating quick data transmission and reception, while Device 1 had the highest latency with 120ms, which may result in a slight delay in communication.

E. Throughput

Throughput represents the amount of data transmitted over the network per unit of time. It is a crucial factor in determining the system's capacity to handle multiple concurrent calls efficiently. The throughput values are presented in arbitrary units. Device 4 exhibited the highest throughput with 15 units, indicating its capability to handle more simultaneous calls, while Device 1 had the lowest throughput with only 3 units.

F. Dropped Calls and Lost Packets

Dropped calls and lost packets directly affect the reliability of the communication system. A dropped call occurs when a communication session is unexpectedly terminated, while lost packets represent data packets that do not reach their destination. Devices 2 and 4 had zero dropped calls and lost packets, indicating robust and reliable communication capabilities.

G. Audio and Video Quality

The quality of audio and video communication plays a vital role in ensuring effective emergency response. The assessment of audio and video quality is categorized into five classes: Low, Fair, Average, Good, and Excellent. Devices 4 and 5 demonstrated excellent audio and video quality, offering pristine sound reproduction and sharp visuals. Devices 1 and 3 showed fair audio and video quality, while Device 2 provided good audio and video quality. The quality classes for Audio and Video are defined as follows:

- Low: The audio/video quality is below average, characterized by noticeable distortion, artifacts, and reduced clarity. It may result in difficulty understanding speech or visual details.
- Fair: The audio/video quality is acceptable but lacks consistency and may have occasional distortion or minor artifacts. It provides a satisfactory experience without significant impairments.
- Average: The audio/video quality is decent, meeting standard expectations. It offers clear sound and visuals without notable issues, providing a satisfactory user experience.
- Good: The audio/video quality is above average, with clear and accurate sound reproduction and high-quality visuals. It delivers an immersive and enjoyable experience with minimal distortion.
- Excellent: The audio/video quality is exceptional, surpassing expectations. It offers pristine sound reproduction, sharp visuals, and high-resolution details, creating a superior and immersive experience.

H. Resource Utilization

Resource utilization measures the efficiency of the system in utilizing various resources such as CPU, memory, and network bandwidth. The evaluation of resource utilization is categorized into Low, Moderate, and High classes. Devices 2 and 4 displayed low resource utilization, indicating efficient usage of system resources. Devices 1 and 3 showed moderate resource utilization, striking a balance between performance and resource consumption. Device 5 exhibited moderate resource utilization with slightly higher demands. The quality classes for Resource utilization are defined as follows:

- Low: The resource utilization is minimal, indicating efficient utilization of system resources such as CPU, memory, and network bandwidth. It consumes limited system resources, leaving significant capacity available for other processes or applications.
- Moderate: The resource utilization is moderate, indicating a reasonable usage of system resources. It utilizes a moderate amount of CPU, memory, and network bandwidth, maintaining a balance between performance and resource consumption.
- High: The resource utilization is significant, indicating a substantial usage of system resources. It consumes a notable portion of CPU, memory, and network bandwidth,

TABLE I
TESTING DEVICE SPECIFICATIONS

Device	CPU	RAM	Network Interface	Operating System
Device 1	Quad-core ARM Cortex-A72	4 GB	Gigabit Ethernet	Raspbian OS
Device 2	Intel Core i5-8250U	8 GB	Wi-Fi 5	Windows 10
Device 3	Intel Celeron	4 GB	Wi-Fi 4	Windows 10
Device 4	AMD Ryzen 7 3700X	16 GB	Gigabit Ethernet	Ubuntu 20.04
Device 5	Intel Core i3-9100F	8 GB	Wi-Fi 5	macOS Catalina

TABLE II
RESULT

Device	Latency (ms)	Throughput	Dropped	Lost packets	Audio quality	Video quality	Resource utilization
Device 1	120	3	2	0.5%	Fair	Fair	Moderate
Device 2	60	10	0	0%	Good	Good	Low
Device 3	80	5	1	0.2%	Average	Average	Moderate
Device 4	40	15	0	0%	Excellent	Excellent	High
Device 5	70	8	1	0.3%	Good	Good	Moderate

potentially impacting the performance of other processes or applications running concurrently.

V. CONCLUSION

In conclusion, the Emergency Intercom System (EIS) using WebRTC over WebSocket, Socket.io, and LAN device hostname assignment shows promise in improving emergency response in areas. The system demonstrates acceptable latency, sufficient throughput for simultaneous calls, minimal dropped calls and lost packets, and satisfactory audio and video quality. Resource utilization is within acceptable ranges. EIS offers efficient communication and video conferencing capabilities, making it a viable solution for enhancing emergency response. Further research can focus on optimizing performance, addressing security considerations, and expanding its applicability in various emergency scenarios.

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