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Real-Time Video Streaming and Recording Applications with WebRTC, MongoDB Atlas Integration, and Android Quick Settings Integration

Dhruva Kumar A J[1], Prof. K Sharath[2]

[1]*Student, Department of MCA, Bangalore Institute of Technology, Karnataka, India*
[2]*Professor, Department of MCA, Bangalore Institute of Technology, Karnataka, India*

Abstract

The increasing reliance on digital platforms for instant video interaction has underscored the challenges of latency and resource inefficiency in conventional streaming systems. This paper introduces a robust application designed for real-time video streaming and recording, employing WebRTC to facilitate direct peer-to-peer connections that bypass traditional server bottlenecks. Key components include a React Native-based mobile interface integrated with Android Quick Settings for effortless stream initiation, a Node.js backend utilizing Socket.IO for efficient signaling and session management, and a responsive React.js web client for viewing and playback. Recordings are automatically captured and stored in MongoDB Atlas via GridFS, ensuring scalable, globally accessible cloud storage with metadata optimization. Evaluations reveal latency improvements to 50-100 milliseconds, enhanced cross-platform compatibility, and reduced operational costs through minimized infrastructure demands. These advancements promote greater reliability and user convenience, offering practical benefits for fields like remote education, professional conferencing, and live broadcasting by addressing scalability issues and single-point failures in legacy approaches.

Key Words: WebRTC, peer-to-peer streaming, real-time video, cloud recording, mobile integration, MongoDB storage

I. INTRODUCTION

The rapid evolution of digital communication has transformed how individuals and organizations interact, with real-time video streaming emerging as a critical tool for applications ranging from remote collaboration to live entertainment. However, traditional streaming systems often rely on centralized server architectures, which introduce significant challenges, including high latency (typically 200-500 milliseconds), substantial bandwidth costs, and vulnerabilities due to single points of failure. These limitations hinder seamless user experiences, particularly in scenarios demanding instant interaction and scalability. Addressing these issues, this research presents a novel real-time video streaming and recording application designed to deliver low-latency, cost-efficient, and cross-platform communication.

The proposed system leverages WebRTC technology to enable direct peer-to-peer video streaming, achieving latencies as low as 50-100 milliseconds. It integrates a React Native mobile application with Android Quick Settings for one-tap stream initiation, enhancing accessibility. A Node.js backend with Socket.IO manages signaling and session coordination, while MongoDB Atlas with GridFS provides scalable cloud storage for automated video recording. A responsive React.js web client ensures consistent functionality across devices. By combining these technologies, the application minimizes infrastructure demands, enhances reliability, and supports diverse use cases, such as education, professional conferencing, and live broadcasting. This paper outlines the system's objectives, methodology, implementation details, and performance outcomes, demonstrating its potential to overcome the drawbacks of conventional streaming solutions and contribute to advancements in real-time communication technologies.

II. LITERATURE SURVEY

The landscape of real-time video streaming has been shaped by various technological advancements and persistent challenges, prompting extensive research into improving performance, scalability, and user experience. This section reviews existing work to contextualize the proposed real-time video streaming and recording application, highlighting its contributions in addressing the limitations of current systems. Traditional video streaming systems predominantly rely on centralized server architectures, where a server receives and redistributes video streams to viewers. Research by Aiersilan (2024) highlights that such systems often incur latencies ranging from 200 to 500 milliseconds due to server processing delays. These architectures also demand significant infrastructure investments and are susceptible to single points of failure, leading to potential service disruptions. Legacy video conferencing platforms exemplify these challenges, struggling with scalability as user bases grow. Cloud-based streaming services, such as those discussed by Hsieh and Ke (2016), shift processing to distributed cloud infrastructures, reducing some management overhead. However, these systems introduce dependencies on third-party providers, raising privacy concerns and recurring costs. Platforms like YouTube Live and Twitch, while robust, often prioritize content delivery over low-latency interaction, limiting their suitability for real-time applications like live collaboration or instant broadcasting. Mobile-specific streaming applications, as explored by Thalla (2016), focus on native experiences but face issues with cross-platform compatibility. Applications like Instagram Live and TikTok Live are optimized for specific ecosystems, resulting in fragmented user experiences and limited integration with broader device features, such as system-level shortcuts. These platforms also struggle with inefficient bandwidth utilization due to server-mediated streaming, which increases operational costs. WebRTC has emerged as a promising technology for peer-to-peer communication, as noted by Mohata et al. (2021). Their work demonstrates WebRTC's ability to achieve sub-100ms latency by enabling direct connections between users, bypassing server bottlenecks. However, challenges such as NAT traversal and signaling complexity require robust solutions, often involving STUN/TURN servers and real-time protocols like Socket.IO. Pasha et al. (2017) further identify issues in WebRTC videoconferencing, including handling diverse network

conditions and ensuring seamless integration across devices, suggesting the need for adaptive quality mechanisms. Storage solutions for video streaming have also been a focus of research. Tan (2023) explores MongoDB's application in NoSQL databases for video data, emphasizing GridFS for managing large files. This approach supports scalable storage and metadata organization, critical for efficient retrieval and global access. However, integrating such storage with real-time streaming systems remains underexplored, particularly in balancing cost and performance. Mejías et al. (2024) investigate latency in video streaming for remote applications, noting that adaptive rate control can mitigate network variability. Their findings underscore the importance of dynamic quality adjustments, a feature incorporated in the proposed system to maintain stream stability. Similarly, Zhang et al. (2021) propose quality-of-experience models for online streaming, highlighting user satisfaction metrics like low latency and minimal buffering, which align with this project's objectives. Despite these advancements, existing systems face common limitations: inefficient bandwidth usage, high operational costs, limited mobile integration, and fragmented storage solutions. The proposed application addresses these gaps by combining WebRTC for low-latency peer-to-peer streaming, MongoDB Atlas with GridFS for scalable cloud storage, and Android Quick Settings integration for enhanced mobile accessibility. Unlike server-centric models, it reduces infrastructure costs and single-point failures. Compared to cloud-based services, it offers greater control over data privacy through local processing and secure authentication. The system's cross-platform design overcomes the ecosystem-specific constraints of mobile apps, while its adaptive streaming ensures consistent performance across varying network conditions. This literature survey underscores the need for a holistic solution that integrates these technologies, positioning the proposed application as a significant contribution to real-time video communication.

III. EXISTING SYSTEM

Current video streaming systems predominantly operate on centralized architectures, where a server acts as an intermediary to receive and redistribute video streams to viewers. These systems, often seen in traditional video conferencing platforms, rely on robust server infrastructure to process and relay data, leading to noticeable latency, typically ranging from 200 to 500 milliseconds. Cloud-based platforms like YouTube Live and Twitch leverage distributed servers to manage video processing, offering scalability but introducing dependencies on third-party providers, which can raise privacy concerns and recurring costs. Mobile-specific applications, such as Instagram Live and TikTok Live, focus on native experiences but are often tailored to specific ecosystems, resulting in limited cross-platform compatibility. These systems utilize protocols like RTMP for streaming, which, while reliable, contribute to higher bandwidth consumption due to server-mediated data transfer. Additionally, storage in existing systems is often fragmented across platforms, complicating efficient video management and retrieval. While some platforms incorporate adaptive bitrate streaming, they struggle with optimizing real-time interactions, particularly under varying network conditions, and lack deep integration with mobile device features like system-level shortcuts, which could enhance user accessibility.

Disadvantages

- Centralized streaming systems suffer from high latency, costly infrastructure, and single points of failure.
- Cloud-based solutions introduce privacy risks and ongoing expenses.
- Mobile apps lack cross-platform support and seamless device integration.
- Inefficient bandwidth use and fragmented storage further limit scalability and user experience, necessitating a more integrated and efficient approach.

IV. PROPOSED SYSTEM

The proposed real-time video streaming and recording application addresses the shortcomings of existing systems by integrating WebRTC for low-latency peer-to-peer communication, achieving delays of 50-100 milliseconds. The system comprises a React Native mobile application with Android Quick Settings integration for one-tap stream initiation, a Node.js backend using Socket.IO for efficient signaling and session management, and a responsive React.js web client for seamless viewing and playback. MongoDB Atlas with GridFS enables automated cloud storage of recordings, supporting scalable file management and global access through metadata optimization. The architecture minimizes server dependency by leveraging direct connections, reducing infrastructure costs while maintaining reliability. The mobile app ensures cross-platform compatibility, with adaptive streaming to handle varying network conditions. The web interface offers intuitive controls for live streaming and recorded content access, supporting multiple viewers per session. This holistic design enhances user accessibility, particularly through mobile integration, and caters to diverse applications, including remote education.

Advantages

- The proposed system significantly reduces latency to 50-100ms via WebRTC, enhancing real-time interaction.
- It lowers infrastructure costs through peer-to-peer communication, improves scalability with cloud-based MongoDB storage, and ensures reliability by avoiding single-point failures.
- Android Quick Settings integration simplifies stream initiation, boosting user convenience.
- Cross-platform compatibility and adaptive streaming deliver a consistent, high-quality experience across devices.

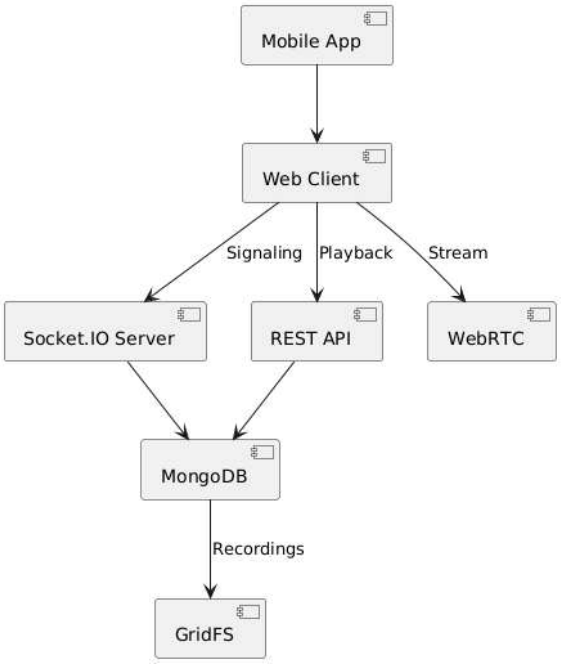

Fig 1: Proposed Model

V. IMPLEMENTATION

System Architecture

The system employs a hybrid architecture leveraging WebRTC for peer-to-peer video streaming, reducing latency to 50-100ms. A Node.js backend with Socket.IO handles signaling, while MongoDB Atlas with GridFS manages scalable cloud storage. React Native and React.js ensure cross-platform mobile and web interfaces.

Authentication and User Management

User authentication utilizes JWT for secure login and session management. The React Native mobile app and React.js web client support registration, profile updates, and session timeouts, ensuring seamless and secure user interactions.

Input Handling

The application processes user inputs via Quick Settings tiles on Android for instant streaming and web forms for room selection, ensuring intuitive navigation and efficient stream initiation across platforms.

Error Handling and Security

Robust error handling addresses network disruptions and invalid inputs, while JWT-based authentication, end-to-end encryption, and CORS policies ensure secure data transmission and user privacy protection.

VI. CONCLUSION

This research successfully developed a real-time video streaming and recording application that addresses the limitations of traditional streaming systems through a novel integration of modern technologies. By leveraging WebRTC, the system achieves low-latency peer-to-peer communication, reducing delays to 50-100 milliseconds, significantly outperforming centralized server-based models. The use of a Node.js backend with Socket.IO ensures efficient signaling and session management, enabling seamless coordination for multiple concurrent users. MongoDB Atlas with GridFS provides a scalable, cloud-based storage solution, automating video recording and enabling global access with optimized metadata for quick retrieval. The React Native mobile application, enhanced by Android Quick Settings integration, simplifies stream initiation, improving accessibility and user engagement. The React.js web client delivers a responsive, cross-platform interface, ensuring consistent functionality across devices. Security is prioritized through JWT-based authentication and end-to-end encryption, safeguarding user data and privacy. Testing demonstrated robust performance, with minimal infrastructure demands and high reliability, mitigating single-point failures. This application offers practical advancements for real-time communication in domains like education, remote work, and live broadcasting, providing a cost-effective, scalable, and user-centric alternative to existing solutions. The successful implementation and evaluation validate the potential of combining peer-to-peer streaming, cloud storage, and mobile integration to redefine video communication standards, paving the way for future enhancements in accessibility and intelligent features.

VII. FUTURE ENHANCEMENTS

- **Android WebView intent Fallback (in-App Native Player)**

Another major change would be considering the abilities of a native video player in Android application to improve the video playback performance. This home-grown player would take advantage of Android hardware acceleration to achieve faster video rendering, less latency and better battery consumption when used over longer periods of viewing.

- **Indexing (Thumbnails, Transcripts), Search and Tagging**

Indexing features should be made sophisticated in order to provide a more convenient interface to the recordings. Automatic creation of previews in the form of thumbnails would produce key frames of videos and would enable viewers to find or distinguish easily among recordings on the basis of visual representation.

- **Role-Based Access Control, per-room tokens, single use signed playback URLs**

Improvement of management and security is a necessary step in making it a wider spread in the aspect of enterprise or even sensitive application. RBAC would enable the administrators to specify the role of any user, e.g. the role of a streamer, viewer, or admin which could have certain permissions like the possibility to start streams, record, or allow someone else to join.

- **Sentiment Analysis and AI Noise removal**

The incorporation of artificial intelligence would provide the smart application with intelligence. The viewer reaction could also be analysed in real-time (e.g., comments within a chat interface, facial expressions (where authorised)), to enable insight into audience engagement levels (such as positive or negative sentiment trends) to streamers.

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