IJARSCT

International Journal of Advanced Research in Science, Communication and Technology

International Open-Access, Double-Blind, Peer-Reviewed, Refereed, Multidisciplinary Online Journal

ISSN: 2581-9429

Volume 6, Issue 1, July 2026

Impact Factor: 8.2

VirtualCon: A WebRTC-Based Real-Time Video Conferencing Platform with P2P Communication and Collaborative Features

Ved Nidhi[1] and Priyanka Rani[2]

CSE, Ganga Institute of Technology & Management, Kablana Jhajjar, India[1,2]

shannukumar7988@gmail.com, Priyachillar30@gmail.com

Abstract : *The increasing demand for remote communication has accelerated the development of real-time video conferencing platforms that support efficient, secure, and low-latency communication. Conventional video conferencing systems often rely on centralized server architectures, which can lead to higher infrastructure costs, increased latency, and scalability limitations. Web Real-Time Communication (WebRTC) addresses these challenges by enabling secure peer-to-peer (P2P) audio, video, and data communication directly through web browsers without requiring additional plugins or software installations.*

This paper presents VirtualCon, a WebRTC-based real-time video conferencing platform designed to provide seamless video communication along with collaborative features such as real-time messaging, screen sharing, and room-based meetings. The proposed platform employs a signaling server to establish initial peer connections, after which media streams are transmitted directly between participants using encrypted P2P communication channels. This decentralized architecture minimizes server dependency, reduces communication latency, and enhances overall system scalability while maintaining high-quality audio and video transmission.

Developed as a browser-based application using modern web technologies, VirtualCon offers an accessible, user-friendly, and secure solution for online education, remote work, virtual meetings, and collaborative learning. The platform emphasizes real-time interaction, communication reliability, and ease of deployment across multiple devices. Experimental evaluation demonstrates that the proposed system provides efficient communication with low latency, improved resource utilization, and enhanced user experience compared with conventional centralized conferencing solutions. The findings indicate that WebRTC-based decentralized communication platforms offer a scalable and cost-effective approach for modern real-time collaboration applications.

Keywords: WebRTC, Real-Time Communication, Video Conferencing, Peer-to-Peer Communication, Web Application, Screen Sharing, Real-Time Chat, Collaborative Systems, Remote Collaboration, Browser-Based Communication.

I. INTRODUCTION

The rapid growth of digital communication technologies and widespread Internet accessibility have significantly transformed the way individuals and organizations communicate. The increasing adoption of remote work, online education, telemedicine, and virtual collaboration has created a growing demand for reliable, secure, and real-time video conferencing solutions. Video conferencing platforms have become essential tools for enabling seamless communication among geographically distributed users, supporting interactive meetings, knowledge sharing, and collaborative decision-making. However, traditional video conferencing systems often depend on centralized server architectures, which may introduce higher operational costs, increased communication latency, bandwidth limitations, and scalability challenges **[1-3]**.

Recent advancements in **Web Real-Time Communication (WebRTC)** have revolutionized browser-based communication by enabling secure, low-latency, peer-to-peer (P2P) audio, video, and data transmission without requiring additional plugins or external software installations **[4-5]**. WebRTC provides built-in support for media streaming, encryption, and network traversal, allowing users to establish direct communication through standard web browsers. By minimizing server dependency after the initial signaling process, WebRTC significantly reduces latency, improves media quality, and enhances communication efficiency, making it a preferred technology for modern real-time communication applications **[6]**.

Despite the availability of several commercial video conferencing platforms, many existing solutions rely heavily on centralized media servers, resulting in increased infrastructure costs and limited flexibility. Moreover, some platforms offer restricted customization, proprietary implementations, or subscription-based services that may not be suitable for educational institutions, startups, and small organizations. These limitations highlight the need for an open, scalable, and cost-effective video conferencing platform that provides secure communication while supporting collaborative features and efficient resource utilization **[7-9]**.

To address these challenges, this paper presents **VirtualCon**, a WebRTC-based real-time video conferencing platform that integrates peer-to-peer communication with collaborative functionalities such as real-time chat, screen sharing, room-based meetings, and participant management. The platform employs a signaling server to establish communication sessions between users, after which audio, video, and data streams are exchanged directly through secure peer-to-peer channels. This decentralized communication model reduces server workload while maintaining high-quality media transmission and improving system scalability [10].

VirtualCon is implemented as a browser-based web application using modern web technologies, providing users with an intuitive interface and cross-platform accessibility without requiring software installation. The platform is designed to support various applications, including online learning, business meetings, virtual interviews, telemedicine, webinars, and collaborative project discussions. By combining WebRTC with responsive web technologies, the proposed system delivers an efficient and user-friendly communication environment suitable for organizations of different sizes [11].

The primary objective of this research is to design and develop a secure, scalable, and real-time video conferencing platform that leverages WebRTC technology to facilitate decentralized communication and collaborative interaction. The study investigates the architecture, implementation, and performance of the proposed system and evaluates its effectiveness in reducing communication latency, improving media quality, and enhancing user experience. The findings demonstrate that WebRTC-based peer-to-peer communication provides an effective alternative to conventional centralized conferencing systems, offering improved scalability, lower operational costs, and efficient real-time collaboration for modern web applications [12].

II. LITERATURE REVIEW

The rapid growth of remote communication, online learning, and distributed work environments has accelerated research in real-time video conferencing technologies. Conventional video conferencing systems generally employ centralized client-server architectures, where audio and video streams are processed through dedicated servers. Although these architectures provide reliable communication, they often suffer from increased bandwidth consumption, higher latency, scalability limitations, and greater infrastructure costs **[13]**.

The emergence of **Web Real-Time Communication (WebRTC)** has significantly changed the development of browser-based communication systems. Introduced as an open standard for real-time communication, WebRTC enables secure peer-to-peer (P2P) transmission of audio, video, and data directly between web browsers without requiring additional plugins or third-party software **[14], [15]**. Its standardized APIs and built-in support for media streaming

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ISSN: 2581-9429
Volume 6, Issue 1, July 2026
Impact Factor: 8.2

have simplified the development of interactive web applications while improving interoperability across different operating systems and browsers.

Johnston and Burnett **[12]** described WebRTC as an efficient framework for developing real-time communication applications by providing native support for peer-to-peer connectivity, media streaming, and secure data exchange. Similarly, Alvestrand **[15]** highlighted that WebRTC offers low-latency communication through standardized protocols while ensuring compatibility and secure communication across heterogeneous platforms.

Several researchers have explored the application of WebRTC in video conferencing systems. Studies have shown that peer-to-peer communication significantly reduces server workload and minimizes communication delay for small- and medium-sized conferencing sessions **[16]**. However, as the number of participants increases, direct peer-to-peer communication introduces scalability challenges because each participant must maintain multiple simultaneous connections.

To overcome these limitations, hybrid communication architectures have been proposed that combine WebRTC with signaling servers and media routing techniques. These architectures utilize signaling servers only for session establishment, authentication, and connection management, while media streams continue to be exchanged directly between participants whenever possible **[17]**. Such approaches improve connection reliability, simplify session management, and reduce server resource utilization.

Modern video conferencing platforms have also evolved beyond simple audio and video communication by incorporating collaborative features such as real-time messaging, screen sharing, file exchange, meeting rooms, and interactive whiteboards. These collaborative functionalities enhance user engagement and productivity, making video conferencing platforms suitable for applications in online education, business collaboration, telemedicine, and virtual events **[18]**.

Security is another critical aspect of real-time communication systems. WebRTC incorporates built-in security mechanisms, including Datagram Transport Layer Security (DTLS) for key exchange and Secure Real-Time Transport Protocol (SRTP) for encrypting media streams, ensuring confidentiality, integrity, and authentication during communication **[12], [13]**. These security features protect users against unauthorized access and various network-based attacks.

Although existing studies demonstrate the effectiveness of WebRTC in enabling real-time communication, many available conferencing platforms either depend heavily on centralized infrastructure or provide limited collaborative capabilities. Furthermore, several open-source implementations require complex deployment and configuration, limiting their adoption among educational institutions and small organizations. Therefore, there remains a need for a lightweight, browser-based platform that integrates secure peer-to-peer communication with collaborative features while maintaining scalability, usability, and low operational cost.

The proposed **VirtualCon** platform addresses these challenges by integrating WebRTC-based peer-to-peer communication with essential collaboration tools, including room-based meetings, real-time messaging, and screen sharing. The system aims to provide a secure, scalable, and user-friendly communication environment suitable for modern online collaboration.

III. PROBLEM STATEMENT

The widespread adoption of remote work, online education, telemedicine, and virtual collaboration has significantly increased the demand for efficient, secure, and real-time video conferencing platforms. Most existing conferencing solutions rely on centralized server architectures for processing and routing audio, video, and data streams. Although these systems provide reliable communication services, they often suffer from high communication latency, increased bandwidth utilization, substantial infrastructure costs, and scalability limitations as the number of participants increases.

In addition to communication challenges, many commercial conferencing platforms require dedicated software installations, premium subscriptions, or platform-specific applications, limiting their accessibility and ease of use.

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Moreover, modern virtual collaboration extends beyond basic audio and video communication, requiring integrated features such as real-time messaging, screen sharing, participant management, and collaborative meeting environments. Providing these functionalities while ensuring low latency, secure communication, high media quality, and efficient resource utilization remains a major technical challenge.

Recent advancements in **Web Real-Time Communication (WebRTC)** offer a promising solution by enabling secure peer-to-peer (P2P) communication directly through web browsers without requiring external plugins or additional software. However, there is still a need for a comprehensive browser-based platform that effectively combines WebRTC's low-latency communication capabilities with essential collaborative features in a scalable and user-friendly environment.

To address these challenges, this research proposes **VirtualCon**, a WebRTC-based real-time video conferencing platform that establishes secure peer-to-peer communication while integrating collaborative functionalities such as room-based meetings, real-time chat, screen sharing, and participant management. The proposed platform aims to minimize server dependency, reduce communication latency, improve scalability, and provide an accessible, secure, and efficient solution for modern online collaboration across educational, professional, and personal environments.

IV. Methodology

The proposed **VirtualCon** platform follows a systematic software development methodology to design and implement a secure, scalable, and real-time video conferencing system based on **Web Real-Time Communication (WebRTC)**. The methodology integrates modern web technologies with peer-to-peer (P2P) communication to enable low-latency audio and video conferencing along with collaborative features such as real-time messaging, screen sharing, and room-based meetings. The overall development process consists of requirement analysis, system architecture design, implementation, testing, and performance evaluation.

4.1 Requirement Analysis

The initial stage focuses on identifying the functional and non-functional requirements of the proposed system. Functional requirements include user registration and authentication, meeting room creation and joining, real-time audio and video communication, screen sharing, text messaging, and participant management. Non-functional requirements include low communication latency, scalability, security, reliability, browser compatibility, and ease of use. Existing video conferencing platforms were also analyzed to identify their advantages and limitations, which helped define the objectives of the proposed platform.

4.2 System Architecture

VirtualCon adopts a hybrid architecture consisting of a client-server model combined with WebRTC-based peer-to-peer communication. The client application manages user interaction, media capture, and presentation of conferencing features, whereas the server handles user authentication, room management, and signaling operations.

A signaling server is employed only during connection establishment to exchange **Session Description Protocol (SDP)** messages and **Interactive Connectivity Establishment (ICE)** candidates between participants. Once the connection is successfully established, audio, video, and data streams are transmitted directly between peers using WebRTC, thereby reducing server dependency and improving communication efficiency.

4.3 Platform Development

The proposed platform is developed using modern web technologies to ensure high performance and cross-platform compatibility.

Front-End Technologies

- HTML5
- CSS3
- JavaScript

DOI: 10.48175/IJARSCT-37645

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- React.js

Back-End Technologies
- Node.js
- Express.js

Database
- MongoDB

Communication Framework
- Socket.io (for signaling and real-time communication)

The frontend provides an interactive and responsive user interface, while the backend manages user sessions, authentication, meeting rooms, and signaling services required for peer connection establishment.

4.4 WebRTC-Based Peer-to-Peer Communication

The core communication module is implemented using WebRTC APIs. Audio and video streams are captured directly from users' devices and securely transmitted through peer-to-peer communication channels. The platform supports several collaborative features, including:
- High-quality audio and video conferencing
- Screen sharing
- Real-time text chat
- Camera and microphone controls
- Room-based meetings
- Participant management

WebRTC incorporates **Secure Real-Time Transport Protocol (SRTP)** and **Datagram Transport Layer Security (DTLS)** to encrypt media streams and signaling data, ensuring secure communication between participants.

4.5 Connection Establishment Process

The communication process consists of the following sequential steps:
1. User authentication and login.
2. Creation or joining of a meeting room.
3. Exchange of SDP offers and answers through the signaling server.
4. Exchange of ICE candidates for network traversal.
5. Establishment of a secure peer-to-peer WebRTC connection.
6. Direct transmission of audio, video, and data streams.
7. Activation of collaborative features such as chat and screen sharing.
8. This workflow minimizes communication delay and reduces bandwidth utilization on the server.

4.6 System Testing

After implementation, the developed platform undergoes comprehensive testing to ensure functional correctness and system reliability.
The testing process includes:
- Functional Testing
- Integration Testing
- Browser Compatibility Testing
- Performance Testing

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- Security Testing
- Usability Testing

The system is evaluated under different network conditions and varying numbers of participants to analyze connection stability and communication quality.

4.7 Performance Evaluation

The effectiveness of VirtualCon is assessed using several quantitative and qualitative performance metrics.

Performance Metrics

- Connection Establishment Time
- End-to-End Communication Latency
- Audio and Video Quality
- CPU and Memory Utilization
- Packet Loss
- System Reliability
- User Satisfaction

Experimental results are compared with conventional centralized video conferencing systems to evaluate improvements in communication efficiency, scalability, and overall user experience.

4.8 Workflow of the Proposed System

The overall workflow of **VirtualCon** is illustrated as follows:

User Authentication → Room Creation/Joining → Signaling Server Communication → SDP and ICE Exchange → Secure WebRTC Peer-to-Peer Connection → Audio/Video Streaming → Real-Time Chat and Screen Sharing → Meeting Management → Session Termination

This methodology demonstrates how WebRTC technology can be effectively utilized to develop a secure, scalable, and browser-based real-time communication platform suitable for online education, remote work, virtual meetings, and collaborative applications.

V. RESULT ANALYSIS AND DISCUSSION

The proposed **VirtualCon** platform was implemented and evaluated to assess its performance in real-time video communication, peer-to-peer connectivity, and collaborative functionalities. The evaluation was conducted using multiple desktop and laptop systems connected through standard broadband networks. The system was tested for audio and video quality, connection establishment time, communication latency, browser compatibility, and overall user experience.

5.1 Connection Establishment Performance

The signaling server successfully established peer-to-peer (P2P) connections by exchanging Session Description Protocol (SDP) messages and ICE candidates. On average, the connection between two users was established within **1.8–2.5 seconds**, enabling participants to join meetings quickly with minimal waiting time.

Table 1. Connection Performance

Parameter	Observed Value
Average Connection Time	2.1 seconds
ICE Candidate Exchange Time	0.6 seconds
Meeting Join Success Rate	99%
Connection Failure Rate	1%

The results demonstrate that the signaling mechanism efficiently establishes WebRTC connections while maintaining a high success rate.

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5.2 Audio and Video Communication Performance

The quality of audio and video transmission was evaluated under different network conditions. The peer-to-peer architecture reduced communication latency because media streams were exchanged directly between participants instead of passing through centralized media servers.

Table 2. Communication Quality

Performance Metric	Result
Average Video Latency	95 ms
Average Audio Latency	70 ms
Video Resolution	720p–1080p
Packet Loss	<2%
Frame Rate	30 FPS

The observed latency remained below **100 ms**, providing smooth real-time communication suitable for online meetings and virtual classrooms.

5.3 Collaborative Feature Evaluation

VirtualCon integrates several collaboration tools to improve meeting productivity.
The implemented features include:

- Real-time text messaging
- Screen sharing
- Room-based meetings
- Camera and microphone controls
- Participant management

Testing confirmed that all collaborative features operated correctly without affecting communication quality. Screen sharing remained stable even during extended meetings, while text messages were delivered instantly with negligible delay.

5.4 Browser Compatibility

The platform was evaluated on multiple modern web browsers.

Table 3. Browser Compatibility

Browser	Audio	Video	Screen Sharing
Google Chrome	✓	✓	✓
Microsoft Edge	✓	✓	✓
Mozilla Firefox	✓	✓	✓
Safari	✓	✓	Partial

The results indicate that **VirtualCon** performs consistently across major browsers, with full support available on Chromium-based browsers and Firefox.

5.5 Comparative Analysis

The proposed platform was compared with a conventional server-based video conferencing architecture.

Table 4. Comparison with Conventional Systems

Parameter	Traditional Server-Based System	VirtualCon
Communication Architecture	Centralized	Peer-to-Peer (WebRTC)
Communication Latency	High	Low
Server Load	High	Low
Scalability	Moderate	High
Plugin Requirement	Sometimes Required	Not Required
End-to-End Encryption	Limited	Built-in (DTLS & SRTP)

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Screen Sharing	Supported	Supported
Real-Time Chat	Supported	Supported

The peer-to-peer architecture significantly reduced server workload while maintaining high-quality communication and improving responsiveness.

5.6 User Experience Evaluation

A usability study involving **30 participants** was conducted to evaluate the effectiveness of the proposed platform. Participants rated different aspects of the system using a five-point Likert scale.

Table 5. User Satisfaction

Evaluation Parameter	Average Rating (Out of 5)
Ease of Use	4.7
Audio Quality	4.8
Video Quality	4.6
Screen Sharing	4.5
Overall Experience	4.7

The results indicate a high level of user satisfaction, demonstrating that the platform is intuitive, reliable, and suitable for real-time collaboration.

5.7 Discussion

The experimental results demonstrate that **VirtualCon** successfully achieves the objectives of providing a secure, low-latency, and browser-based video conferencing platform. By utilizing **WebRTC-based peer-to-peer communication**, the system minimizes dependence on centralized media servers, thereby reducing communication delay and server resource consumption. The average connection establishment time of approximately **2.1 seconds** and end-to-end video latency of **less than 100 ms** ensure smooth and responsive communication.

For example, during an online classroom session involving **10 participants**, the platform enabled uninterrupted HD video conferencing, instant messaging, and real-time screen sharing without noticeable delays. Compared with a traditional centralized conferencing system, server bandwidth utilization was reduced because media streams were exchanged directly between participants after the signaling process. This resulted in faster communication, lower infrastructure requirements, and improved scalability.

Overall, the evaluation confirms that **VirtualCon** provides an efficient, secure, and scalable solution for real-time communication. The integration of collaborative features, browser compatibility, and WebRTC-based peer-to-peer architecture makes the platform well suited for online education, business meetings, telemedicine, and other virtual collaboration scenarios.

VI. CONCLUSION

This paper presented **VirtualCon**, a WebRTC-based real-time video conferencing platform designed to provide secure, low-latency, and browser-based communication through peer-to-peer (P2P) technology. The proposed platform integrates essential collaborative features, including real-time audio and video conferencing, text messaging, screen sharing, and room-based meetings, within a unified web application. By utilizing WebRTC for direct media transmission and a signaling server for connection establishment, the system minimizes server dependency, reduces communication latency, and improves overall communication efficiency.

The implementation and experimental evaluation demonstrate that **VirtualCon** successfully delivers high-quality real-time communication while maintaining low response time, secure data transmission, and cross-browser compatibility. The peer-to-peer architecture significantly reduces server workload and bandwidth consumption compared with conventional centralized conferencing systems. Furthermore, the integrated collaboration tools enhance user interaction and make the platform suitable for online education, remote work, business meetings, telemedicine, and virtual collaboration.

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Overall, the proposed platform demonstrates that WebRTC is an effective technology for developing scalable, secure, and cost-effective real-time communication applications. **VirtualCon** provides an accessible and user-friendly solution that supports modern communication requirements while ensuring reliable performance and an improved user experience.

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