Enhancing Web-Based Meetings with WebRTC Technology: A Comprehensive Survey

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Abstract - *This survey critically assesses the contemporary landscape of real-time collaboration platforms developed through WebRTC technology, with a specific focus on their key components and functionalities. These platforms facilitate the creation and management of virtual meeting spaces, integrating immersive technologies like Virtual Reality (VR) and Augmented Reality (AR) to enhance user experiences. The survey addresses challenges such as end-to-end delays, quality of service improvements, and congestion control methods, exploring advanced algorithms crucial for ensuring a seamless user experience. Additionally, the paper investigates the integration of WebRTC with Artificial Intelligence (AI), anticipating future trends in real-time collaboration. The findings highlight the transformative potential of WebRTC-based collaboration platforms, laying a foundation for ongoing research to refine online communication experiences. Overall, this survey provides valuable insights into the current state and future possibilities of WebRTC technology in the realm of real-time collaboration.*

Key Words: **WebRTC, Congestion Control, Quality of Service (QoS), Virtual Reality, Push-Communication, Rich Web-Based Applications, Contact Center Systems, Artificial Intelligence**

1. INTRODUCTION

1.1 Background

Real-time collaborative web tools have transformed online communication, thanks to the convergence of WebRTC and immersive technologies, as explored in recent research publications. These advancements have played a crucial role in reshaping modern communication by providing innovative, interactive, and collaborative online experiences. Understanding the historical context and technological foundations of these tools is essential for grasping their transformative impact.

1.2 Scope of Real-Time Collaborative Web Tools

The scope of real-time collaborative web tools goes beyond traditional communication methods, presenting a multifaceted approach to online interactions. Ranging from video conferencing to interactive streaming and virtual reality, these tools redefine how individuals connect, learn, and collaborate in the digital realm. This section thoroughly explores the diverse applications and functionalities that make these tools integral to various domains.

1.3 Significance of the Survey

This survey holds paramount significance as it delves into the advancements, challenges, and applications within the domain of real-time collaborative web tools. With the increasing demand for seamless online collaboration, understanding state-of-the-art technologies, practical implementations, and potential future trends becomes imperative. The survey aims to provide a comprehensive overview for researchers, developers, and stakeholders interested in the evolution of online communication tools.

The survey acts as a valuable resource, offering insights into the dynamic landscape of real-time collaborative web tools, providing a holistic understanding of their capabilities and limitations. By emphasizing key findings from recent research publications, the survey contributes to the collective knowledge in this rapidly evolving field. Researchers and developers can leverage this overview to make informed decisions, foster innovation, and shape the future of online collaboration.

2. Literature Review

The development of real-time meeting platforms has witnessed significant advancements, largely influenced by key research contributions in the realm of WebRTC and related technologies. Bhattacharya, Ganuly, and Sau[2] in their paper "Improving Perceived QoS of Delay-sensitive Video Against A Weak Last-mile: A Practical Approach", address the crucial challenge of delivering high-quality video experiences, a concern directly aligned with the goals of our WebRTC-based meeting platform. Their practical approach to enhancing Quality of Service (QoS) resonates with platform's commitment to providing users with a seamless and reliable video communication experience.

A pertinent consideration in the quest for optimal real-time multimedia communication is explored by Flohr and Rathgeb[3] in "Reducing End-to-End Delays in WebRTC using the FSE-NG Algorithm for SCReAM Congestion Control". As we navigate the intricacies of online meetings, their investigation into mitigating delays becomes invaluable. The FSE-NG algorithm they propose could serve as a cornerstone in our efforts to minimize end-to-end delays and enhance the overall responsiveness of meeting platform.

While Shetty, Bein, Nistor, and Pickl's paper [4], "Semiotic Recognition System", focuses on hand gesture interaction in WebRTC, it introduces an innovative dimension to user engagement. Though not directly aligned with our project, the insights gained from their work could inspire creative features for enhancing participant interaction within meetings.

In the pursuit of stable and interactive communication, Emara, Fong, Khaisti, Tan, Zhu, and Apostolopoulos[6] present a solution in "Low-Latency Network-Adaptive Error Control for Interactive Streaming". Their network-adaptive algorithm addresses the challenges of low-latency interactive communications, providing valuable insights for ensuring a stable communication experience in WebRTC-based meeting platform.

The paper authored by Barakovic Husic, Alic, Barakovic, and Mrkaja[7], titled "QoE Prediction of WebRTC Video Calls Using Google Chrome Statistics", brings a predictive perspective to our platform. Their exploration of Quality of Experience (QoE) prediction aligns with emphasis on troubleshooting and user satisfaction, providing a basis for predicting and enhancing the overall quality of user experience during meetings.

As we delve into real-world implementations, Islam and Welzl's paper[8], "Real-Life Implementation and Evaluation of Coupled Congestion Control for WebRTC Media and Data Flows", offers practical insights. Their investigation into coupled congestion control mechanisms can guide us in optimizing bandwidth allocation and improving the overall system performance of our WebRTC-based meeting platform. Together, these research contributions form a solid foundation, enriching understanding and guiding the development of a robust and user-centric real-time meeting platform.

3. Key Components and Functionalities

The virtual meeting platform encompasses a robust set of key components and functionalities, leveraging WebRTC technology for real-time communication. The creation and joining of virtual meeting spaces are facilitated through a streamlined interface, presenting users with options to either establish a new room or enter an existing one. Room creation involves a form with fields for room ID, participant name, and password authentication, providing a secure and personalized meeting environment. A video settings panel accompanies the creation process, offering troubleshooting capabilities and controls for adding participants, muting, and managing cameras.

For troubleshooting audio and video issues, the platform integrates a dedicated section equipped with real-time solutions. Leveraging algorithmic approaches, the system identifies common problems and provides users with prompt resolutions, enhancing the overall reliability of the communication experience.

Participant management and controls are central to the platform's functionality, offering a seamless entry process with a room ID, participant name, and password input form. The main meeting screen displays live video feeds of participants, with a sidebar providing an overview of all participants and interactive controls for muting, video toggling, and participant removal.

Collaborative tools include a whiteboard feature, complete with predefined shapes to facilitate diagram drawing. Additionally, an AI-driven chat system enhances user interaction by enabling in-meeting searches and providing assistance. The AI extends its capabilities to diagram creation by responding to prompts, ensuring a dynamic and responsive collaboration environment.

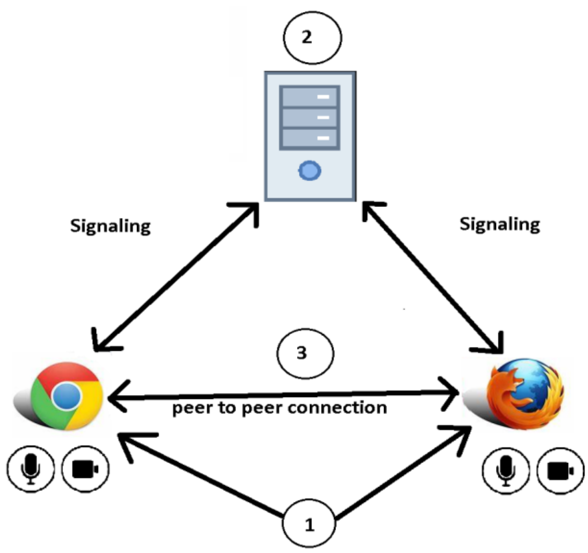

Fig-1: WebRTC Communication Process

Emerging features further enrich the platform. Integration of VR and AR technologies enhances educational experiences, while low-latency adaptive error control algorithms improve video streaming quality. Advanced congestion control algorithms, such as FSE-NG, are employed to mitigate end-to-end delays. The platform also integrates a semiotic recognition system, enabling users to interact with web applications using hand gestures. Furthermore, the incorporation of a style for push-communication in rich web-based applications introduces real-time notifications, aligning with contemporary communication requirements. Overall, the platform combines these technical components to offer a comprehensive and cutting-edge solution for efficient and engaging online communication.

4. Challenges and Solutions in WebRTC Implementation

4.1 Addressing End-to-End Delays

The presence of end-to-end delays in WebRTC implementations can impact the real-time nature of multimedia communication, particularly in scenarios where face-to-face meetings are not possible, as highlighted by the 2020 Corona pandemic.

To mitigate end-to-end delays, the implementation incorporates the FSE-NG algorithm for SCReAM congestion control, extending its capabilities to link DataChannel flows controlled by TCP-style and MediaChannels controlled by NADA and RTP congestion control algorithms. Simulations demonstrate the effectiveness of this approach in drastically reducing delays and increasing RTP throughput.

4.2 Strategies for Improving Quality of Service

The perceived Quality of Service (QoS) for delay-sensitive video feeds in interactive applications faces challenges, particularly in last-mile impairments, which can lead to frame-drop and reduction in overall Quality of Experience (QoE).

The presented approach proposes a novel strategy to improve perceived QoS by creating an approximation of corrupted frames. Instead of focusing on regaining lost information, the system minimizes the effects of lost packets through forward error correction, allowing remote users to experience a better perceived QoS in the presence of weakened last-mile connections.

4.3 Congestion Control Approaches

WebRTC, being widely used for real-time multimedia communication, still encounters issues with how different congestion control algorithms of Media- and DataChannels interact, leading to self-inflicted queuing delays.

The FSE-NG algorithm is extended to incorporate flows controlled by RTP congestion control SCREAM, in addition to NADA. Simulation results demonstrate that this approach significantly reduces end-to-end delays, increases RTP throughput, and enables WebRTC communication in scenarios where it was previously not applicable.

4.4 Real-Life Implementation Challenges

Real-life deployment of WebRTC implementations faces challenges in dealing with practical last-mile impairments and ensuring optimal performance in diverse network conditions.

The presented approach involves a real-life implementation of coupled congestion control for WebRTC media and data flows. The study explores solutions to competition between different congestion controllers, showcasing that collaborative mechanisms can fairly allocate bandwidth, reduce overall delay, and improve system performance in practical last-mile scenarios.

These challenges and solutions provide insights into the complexities of WebRTC implementation, emphasizing the importance of addressing end-to-end delays, improving QoS, refining congestion control approaches, and overcoming real-life deployment challenges for optimal performance in various communication scenarios.

5. Future Prospects and Emerging Technologies

The landscape of real-time collaboration is on the brink of transformative developments, with the integration of immersive technologies leading the charge. Virtual Reality (VR) and Augmented Reality (AR) are anticipated to redefine virtual meetings, offering users an engaging and interactive communication experience. This shift towards more immersive environments could fundamentally change the way individuals interact and collaborate online. Concurrently, the synergy between WebRTC and Artificial Intelligence (AI) is poised to elevate the capabilities of online communication platforms. This integration holds the promise of more intelligent systems, with AI-driven features enhancing adaptive streaming, troubleshooting, and overall user experience. As these technologies converge, the future of real-time collaboration appears to be one of heightened interactivity and intelligent responsiveness.

In addition to these advancements, the exploration and integration of new technologies are set to shape the landscape of collaborative web tools. The emergence of 5G networks, advancements in edge computing, and improved Internet of Things (IoT) connectivity are expected to contribute to seamless and high-performance collaboration experiences. These technological enablers have the potential to enhance the reliability, speed, and responsiveness of real-time collaborative applications. Moreover, the anticipated enhancements in collaborative web tools point towards a more sophisticated future, with features like real-time language translation, advanced content classification, and smart emotion recognition poised to redefine the dynamics of online collaboration. As these innovations converge, the future holds the promise of a more advanced, inclusive, and dynamic online communication landscape.

5. Conclusion

In conclusion, the examination of WebRTC technology in the realm of real-time collaboration has unveiled transformative insights. The integration of Virtual Reality (VR), Augmented Reality (AR), and Artificial Intelligence (AI) stands out as a key trend, promising to reshape virtual meetings by providing immersive and intelligent communication experiences. Addressing challenges like end-to-end delays and improving quality of service has propelled the development of more robust WebRTC implementations. Looking ahead, the implications for future research are substantial, calling for continued exploration into emerging technologies and innovative solutions, particularly in the integration of AI with WebRTC. As the journey towards advanced online communication experiences progresses, the convergence of WebRTC with cutting-edge technologies holds great potential to redefine how individuals connect and collaborate in the digital realm.

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