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Hybrid WebRTC Signalling System for Bidirectional and Unidirectional Video Conferences

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Abstract*: The present research captures a multipurpose signaling protocol that can be utilized to deliver a hybrid WebRTC environment that can support a single direction video conference as well as bi-directional video conference. Since the traditional WebRTC signaling is based on the use of centralized servers, its scalability may be limited and cause latency. Being the blend of the decentralized and the centralized methods of signaling, the introduced system is more flexible, more celeb, and more reliable. The system can fulfill the broadcast-style one-manner communication in the unidirectional mode that serves the webinars or operates the types of broadcasting, and the bidirectional mode is more universal where both the participants are expressed with two-way roughly voice and video channels. The hybrid model dynamically sets the signaling path according to user role, network condition as well as type of communication. According to experimental tests, there is less traffic on the server, less connection setting delay and enhanced user experience. This architecture suits with scalable real time video applications that are applied in the media broadcasting, business, and instructional sectors.*

Keywords*: Wide area network (WAN), Quality of experience (QoE), Local area network (LAN), Mesh topology, Socket. IO, The real-time web communication (WebRTC), Web new signalling mechanism (WebNSM)*

I. Introduction

Removing the requirement that such connectivity be established with plugins, WebRTC (Web Real-Time Communication) has revolutionised the ability to communicate in real-time via audio and video over the internet, gravity peer-to-peer. WebRTC also has an important signalling mechanism that facilitates easier setting-up of channels due to exchange of metadata such as description of sessions and network capabilities. The traditional signalling methods tend to rely on centralised servers that may cause the reduction of fault tolerance, the rise in latency, and bottlenecks. To eliminate these disadvantages, this study proposes a similar hybrid webRTC signalling system that could support both unidirectional and bidirectional video conference. To increase scalability and efficiency, the system provides a mix between decentralised, peer-assisted communication paths and centralised signals, to control and session initiation. It enables two-way interactive user communication and enables optimal transmission of webinar or real-time events on one-sided communication. This hybrid method is suitable in multiple real-time protocol applications because it has a faster connection establishment, it achieves maximum resource diffusion and provision of high overall quality of service.

II. Related Work

There have been several studies that improve the WebRTC signalling in a bid to optimize the performance, scalability, and flexibility of real time communications. Traditional WebRTC architecture is principally based upon centralised signalling servers and WebSocket, SIP, or XMPP applications which may be bottlenecks due to traffic levels. Loreto et al research state that large-scale applications need to have scalable signalling systems supporting them. To enhance fault-tolerance at the cost of latency several papers have proposed decentralised approaches using a distributed ledger, or a peer-to-peer signalling structure. Various signalling models are a combination of peer-assisted communication and centralised control, called hybrid signalling models; hybrids have been explored in Janus and Jitsi. However, most existing systems focus on one or the other while not focusing on both streaming and bidirectional communication. This paper provides an addition to the previous studies by proposing a single hybrid signalling scheme that supports both unidirectional and bidirectional video conferencing. By changing to various levels of communication and networking consistently, the proposed system aims at addressing the pitfalls of existing models.

III. Methodology and Implementation

Hybrid WebRTC Type Signalling System incorporates both centralised and decentralised signalling methods in order to support unidirectional (broadcast) and bi-directional (interactive) video conference connections. The strategy can be split into a number of relevant steps. It consists of a signalling controller, peer clients and a centralised signalling server [1-2]. The server performs user authentication, room administration and initiates session pre-configuration. The signalling can operate on a decentralised model after joining peers to communicate directly. Both users send and receive video and audio feeding. Sharing of ICE candidates, as well as offer and responses, is simplified using the signalling server. When the session is established, the media can be broadcasted peer-to-peer using WebRTC.

Fig 1: Web page based on forefox

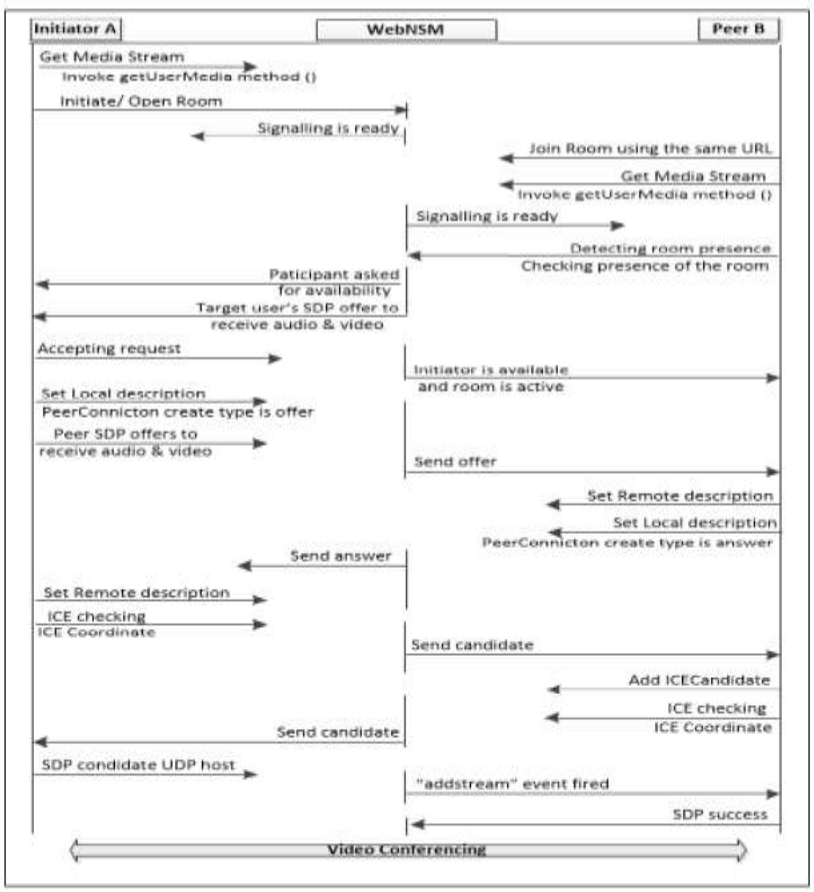

Fig 2: Broad casters signalling

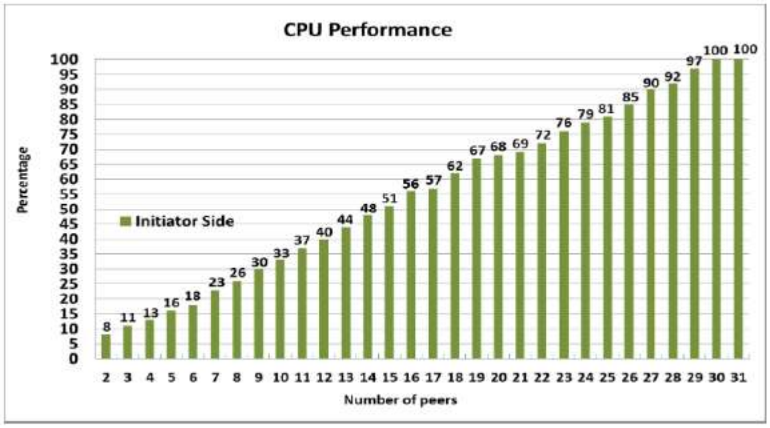

Fig 3: LAN and WAN using broadcasters

There are viewers and a single individual does the broadcasting. Distributing streams in an efficient way, the signalling is initially centralised and subsequently optimised through the use of peer relays referred to jointly as SFUs (Selective Forwarding Units) [1-4]. Depending on the kind of session and the roles of the users, the system will dynamically determine the signalling path. The strongest suggestion made in reducing latency with small-scale connections is through peer-to-peer signalling. In order to ensure scalability of large broadcasts, centralised or SFU-based signalling is used.

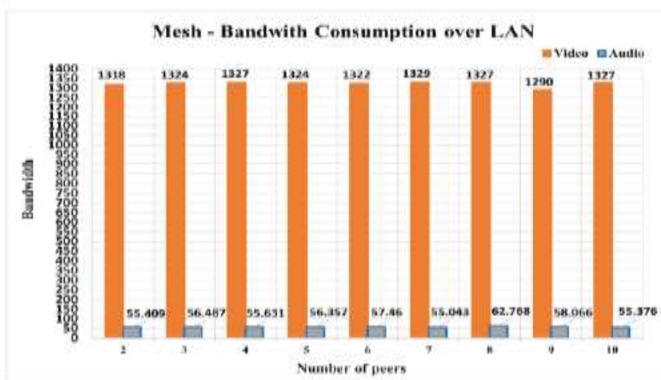

Fig 4: LAN as vroadcasters

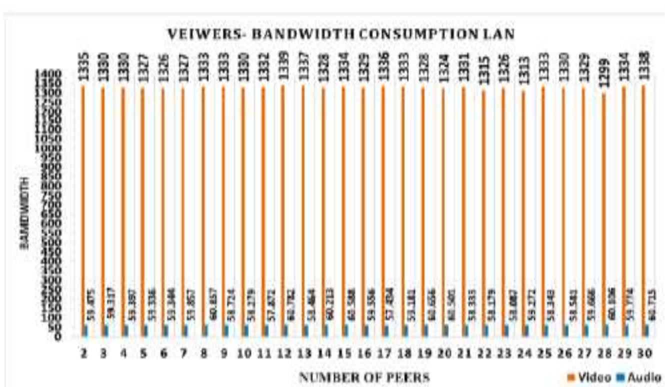

Fig 5: LAN as viewers

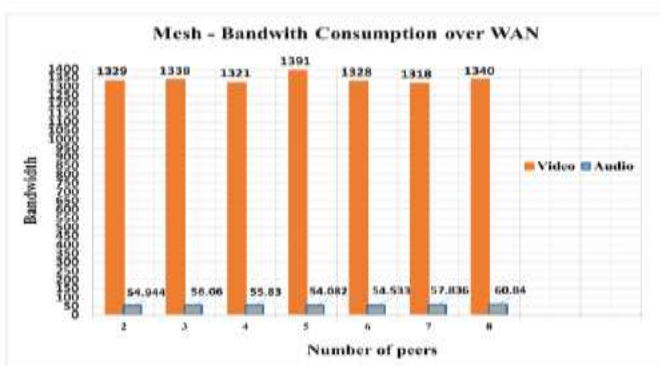

Fig 6: WAN as broadcasters

Technologies that are utilised are WebRTC APIs (RTC Peer Connection, get User Media), Node.js as a signalling server and WebSocket as a means of exchanging initial messages. There is a straightforward front-end user interface with which users can select roles (broadcaster, spectator, or participant) and enter sessions. The switching in between communication modes on the fly with regard to network conditions and circumstance is enabled in this hybrid signalling architecture, which enhances flexibility and maximises the usage of resources [3].

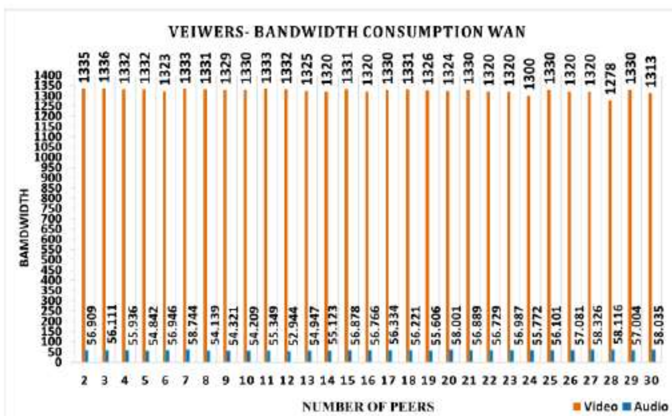

Fig 7: WAN as viewers

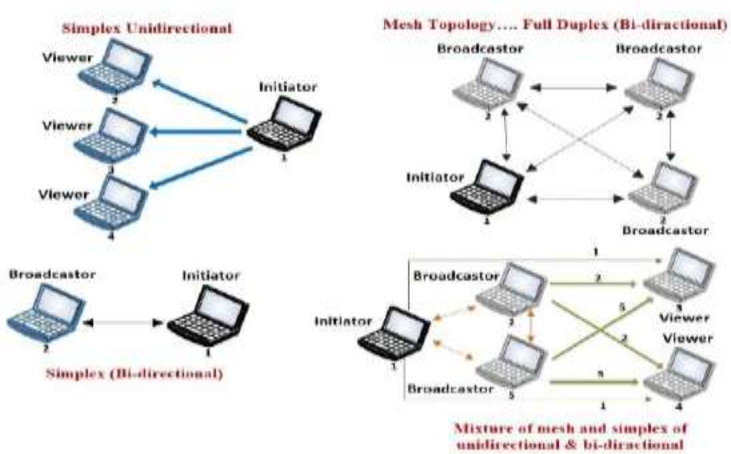

Fig 8: Hybrid system architecture

IV. Conclusion

To sum up, combining both centralised and decentralised methods, the proposed Hybrid WebRTC Signalling System manages to eliminate disadvantages associated with traditional signalling methods and make it possible to perform unidirectional and bidirectional video conferences. The benefits of this dynamic adaptation of this flexible architecture to user roles, types of sessions, and network situation are better scalability, reduced latencies, and more efficient resource use. The system is capable of handling such broad broadcasts whereas ensuring easy peer-to-peer communication in a small group conversation using centralised or SFU-assisted signalling to control the number of peers in a multicast transmission across many peers. In testing and implementation, it is found that this hybrid solution can enhance quality of experience and connection reliability across many real-time communications environments. It gives a viable and scalable solution to modern applications in media streaming, live event attendance, off-site work, and education.

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