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WebRTC Signaling Mechanism Using npRTC Topology for Online Virtual Classroom

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ABSTRACT
Equitable quality of education is still a national strategic issue, especially since the Covid-19 pandemic has not yet ended. Learning through Online Virtual Classroom (OVC) is needed to fulfill equal access to quality education for the community. One of the technologies that can be used to build virtual classrooms online is WebRTC (Web Real-Time Communication). WebRTC is a real-time communication technology or web platform that can be run between browsers without the use of various plug-ins. The purpose of this study is to model the WebRTC topology through the signaling mechanism that works in the construction of the OVC. The identification of the OVC feature is integrated with WebRTC as a real-time communication medium, namely Electronic whiteboard, Screen sharing, File transfer, Recording, Chat room, Calendar Integration, and Moderation. Application is developed using Node.js as backend programming and React.js as frontend programming, which is Socket.io for signaling communication. This study proposes a topology using the signaling mechanism of the npRTC model. The research method is quasi-experimental with a forward engineering approach. System testing is carried out to measure network performance with testing parameters including bandwidth consumption, CPU performance, memory usage, throughput, delay, jitter, and packet loss. The results showed that by using the npRTC signaling mechanism, the CPU load, bandwidth requirements, and large memory usage by the client could be reduced, because throughput was increased, and delay, jitter, and packet loss were reduced. This research on the WebRTC signaling mechanism is that the intermediate server for interactive connectivity establishment using a STUN server does not yet involve a TURN server.

Keywords: *WebRTC, Online Virtual Classroom, npRTC Signaling Mechanism, Network Performance.*

1. INTRODUCTION

The first important thing for us to review the characteristics of the Online Virtual Classroom (OVC) is how well it affects student learning outcomes. Identify the OVC features that are integrated with WebRTC technology as a real-time communication medium, namely: Electronic whiteboard, Screen sharing, File transfer, Recording, Chatroom, Calendar Integration, and Moderation, is intended to facilitate the achievement of these learning outcomes.

Web Real Time Communication (WebRTC) [1] is an evolving open-source framework developed by the World Wide Web Consortium (W3C) and Internet Engineering Task Force (IETF) that can be used to build real-time communication applications without installing any plugins. WebRTC technology makes it easy to use because it does not require plugins or other applications to be installed, only a browser that supports it is needed [2].

Unlike in native peer-to-peer applications where nodes can establish connections by contacting their peers directly, WebRTC requires the use of separate channel signals to negotiate connections. Signaling mechanisms must be negotiated to establish connections between peers [3]. Signal processing provides a level of security by eliminating the need for nodes to keep ports open and enabling creative routing strategies that allow peers behind NAT devices or firewalls to connect. The signal between the browser and the server is not standardized in WebRTC, because it is considered part of the application [4]. Regarding data paths or media paths, peer connections allow media to transmit directly between browsers without applicable servers [5]. Server signaling is used for communication between clients before peer-to-peer communication is established [6].

The webRTC application itself has limitations on the number of clients that can connect at one time, especially in the form of video conferencing applications. This is because each connected client will consume quite high bandwidth, RAM, and processor along with the increase in the number of clients [7]. Signaling methods will affect the network performance such as throughput, delay, jitter, and packet loss [8]. Due to resource limitations and the use of different topologies for WebRTC video conferencing, a WebRTC hybrid signaling mechanism was created and implemented over a LAN and WAN network named WebNSM for video conferencing based on Socket.io (API) and Firefox mechanisms. WebNSM is designed with a combination of different topologies, such as simplex, star, and mesh. WebNSM takes an average of 89 (milliseconds) to be ready and 111 (milliseconds) to send requests and receive responses, even when the network is congested [9]. P2P-MCU approach to support multi-party WebRTC conferencing with common Android phones may cause some delay (< 500 ms), stable delay and perceptions were almost negligible for the participants. P2P-MCU performance is quite stable and success rate build connection on 3G network almost 90%, which is much higher than in normal WebRTC. Experimental results of eight-party video conferencing experiments show that our solution can reduce 64% CPU 35% bandwidth usage and consumption for each participants compared to a pure WebRTC mesh network [10]. To compare the performance of each SFU has been measured by bit rate and latency in video scalability testing conferencing use cases using a single SFU WebRTC media server. The results show that that such approach is viable, and provide unexpected and refreshingly new insights on the scalability of those SFUs, that the decrease of bit rate is sharper for Medooze [11].

The studies carried out as described [9], [10], [11] still have shortcomings with regard to the effectiveness of resources on WebRTC regarding audio and video media streams. This study aims to model the WebRTC topology that works in the development of the Online Virtual Classroom (OVC) with the npRTC mechanism model. Therefore, this article aims to answer the research question: How much influence does the npRTC signaling mechanism have on network performance and effectiveness of resource devices in WebRTC on end users?

This paper is structured and described as follows, in part 2 the material and research methods and their implementation and analysis are explained. In section 3, the results and discussion are described. Finally, section 4 presents conclusions and future work.

This research is used for the STUN server intermediary server in the establishment of interactive connectivity, not involving the TURN server.

2. THE MATERIALS AND METHOD

The computer and network devices used in this study are as shown in Table 1.

Table 1. The testing material of the WebRTC signaling mechanism

Component	Mode
Desktop PC	Gigabyte Technology Co., Ltd. H81M-DS2
Webcam; Audio device	Full HD webcam
Monitor	S24R35x
Audio device	Microphone (High Definition Audio Device)
Network infrastructure device	802.11n NIC
Sistem Operasi	Windows 10 Home
Processor	Intel® Core™ i5-4570 CPU @ 3.20GHz 3.20 GHz
RAM	8,00 GB
Operating System Type	64-bit Operating System, x64-based processor

The research method used quasi-experimental with a forward engineering approach. The forward engineering is the final step of the re-engineering process, while the forward engineering stage follows the stages of the general model of re-engineering software, namely conceptual, requirements, design, and implementation [12]. Forward engineering is the engineering which moves from high level to low level abstraction. In this the high level model or concepts are building to low level details [13]. The forward engineering approach becomes the research stage on the quasi-experimental method. The planning stage is the process of defining what is to be achieved from the research conducted. The design phase is the process of creating and designing a new system. The development stage is the process of achieving new knowledge and findings. The last stage is implementation, namely the act of practicing the design that has been formulated previously.

2.1. Consceptual

WebRTC applications require signaling services for peers to exchange network and media metadata. Once signaling occurs, video/audio/data is streamed directly between clients. Signaling is not defined by the WebRTC standard, not implemented by its API to allow flexibility in the technologies and protocols used. The signaling and the servers that handle it are left to the WebRTC application developer to handle. Each peer in a WebRTC connection tries to acquire a set of Interactive Connectivity Establishment (ICE) protocol candidates.

The candidate represents the combination of IP address, port, and transport protocol to be used.

Initiating a peer-to-peer connection requires some work such as a handshake mechanism to negotiate connection parameters, and it implicitly assumes that the destination server is reachable by the client—that is, the server has a publicly routable IP address or the client and server are on the internal network. the same one. Before a connectivity check or session negotiation can take place, we must find out if the other peer is reachable and if the peer is willing to establish a connection. We have to make offers, and partners have to return answers. To exchange offers and answers between clients, we need a shared signal channel.

In planning for WebRTC-based communication, it starts using the Session Description Protocol (SDP) to describe the parameters of the peer-to-peer connection. The first steps required to initiate a WebRTC connection: (1) initialize the shared signaling channel, (2) initialize the RTCPeerConnection object, (3) request video and audio streams from the browser, (4) register local video and audio streams with the RTCPeerConnection object, (5) generate SDP description (offer) of peer connection, (6) apply generated SDP as local description of peer connection, (7) send generated SDP offer to remote peer via signal line

2.2. Requirements

The design of the signaling mechanism is carried out to obtain the required network service quality parameters. Standardization of network performance based on the Throughput value is shown in table 2.

Table 2. Standardize network performance based on Throughput

Throughput Category	***Throughput (%)***	***Indeks***
Very good	100	4
Good	75	3
Moderately	50	2
Bad	< 25	1

Sumber: TIPHON

Standardization of network performance based on Packet Loss values is shown in table 3.

Table 3. Standardization of network performance based on Packet Loss values

Packet Loss Category	***Packet Loss (%)***	***Indeks***
Very good	0	4
Good	3	3
Moderately	15	2
Bad	25	1

Sumber: TIPHON

Standardization of network performance based on the delay value is shown in table 4.

Table 4. Standardization of network performance based on the Delay

Delay Category	***Delay (ms)***	***Indeks***
Very good	< 150	4
Good	150 s/d 300	3
Moderately	300 s/d 450	2
Bad	> 450	1

Sumber: TIPHON

Standardization of network performance based on the Jitter value is shown in table 5.

Table 5. Standardization of network performance based on the Jitter

Jitter Category	***Jitter (ms)***	**Indeks**
Very good	0	4
Good	0 s/d 75	3
Moderately	75 s/d 125	2
Bad	125 s/d 225	1

Sumber: TIPHON

2.3. Desaign

Several signaling mechanisms to establish connections between users in WebRTC have been developed by developers, including peer-to-peer, Star, Mesh, Selected Fowarding Unit (SFU), Multipoint Central Unit (MCU), Hybrid. The npRTC signaling mechanism is a combination of the Star and Mesh models shown in Figure 1.

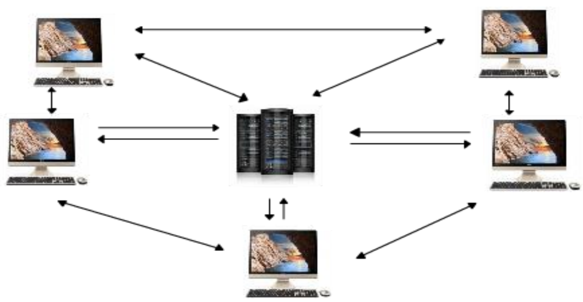


Figure 1 npRTC signaling mechanism

The Figure 2 is peer-to-peer signalling mechanism. WebRTC basically allows web apps go create Peer-to-

Peer (p2p) communication. When web application needs some data or a resource, it fetches it from some server and that's it, video chat, by directly connecting to someone else's browser. In order for two devices on different networks to find each other, this process, called signaling, involves both devices connecting to a mutually agreed upon third server. Through this third server, the two devices can find each other, and exchange negotiation messages.

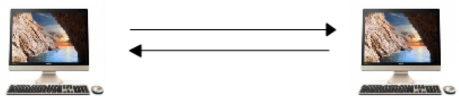


Figure 2 Peer to peer (p2p) signaling mechanism

In Mesh network all peers send their stream directly to other connected peers in network individually, such as figure 4.

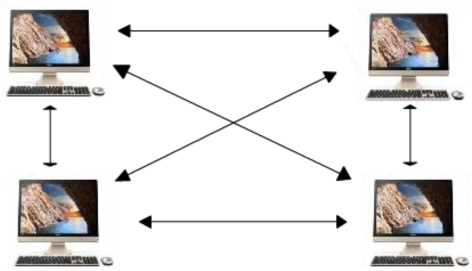


Figure 3 Mesh signaling mechanism

2.4. Implementation

The last stage is an implementating, which is an act of practicing a theory, method, and other things to achieve certain goals. The implementation of the test is carried out in a basic computer network laboratory as shown in Figure 5.

Figure 4 The basic computer network laboratory

The implementation of WebRTC with p2p signaling mechanism is carried out via the https url shown in figure 6. Tested the performance of computer resources used by the user, also tested the quality of network performance using wireshak.

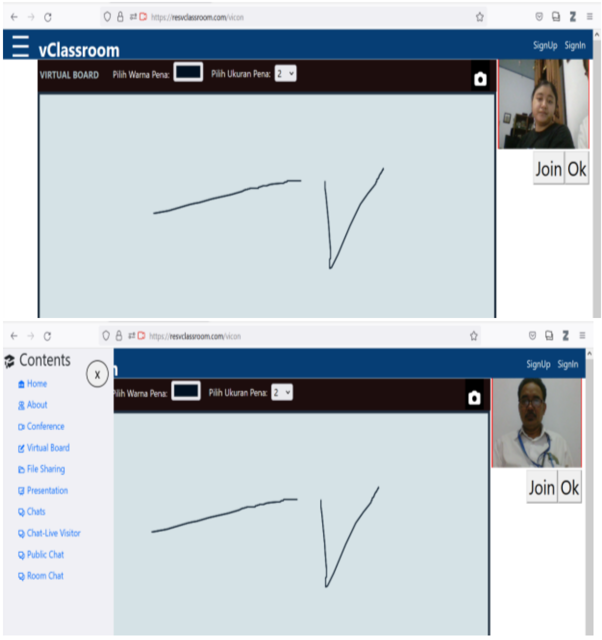


Figure 5. Peer to peer (p2p) in online virtual classroom

3. RESULTS AND DISCUSSION

To communicate with other peers via a web browser, each client web browser must go through the following steps: (1) agree to initiate communication, (2) know how to find each other, (3) bypass firewall security and protection, and (4) transmits all multimedia communications in real-time. The four steps are carried out through a signaling mechanism. In order to find out how much influence the npRTC signaling mechanism model has on network performance and device resource efficiency on end users, we have tested several parameters. Then we compared it with the mesh signaling mechanism model.

Testing of videoconferencing and whiteboard features is carried out in groups with a certain number of users, namely 3 users, 6 users, and 15 users. However, we previously tested a peer-to-peer (p2p) signaling mechanism, consisting of 2 users.

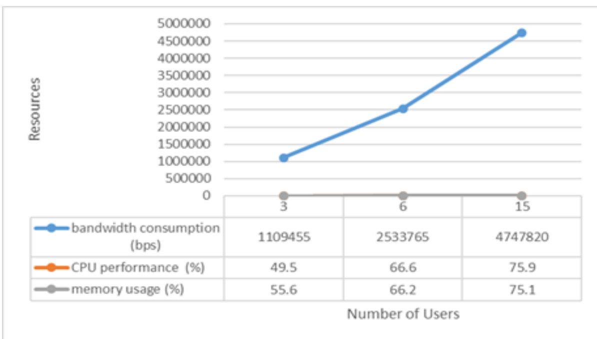


Figure 6. The Computer resource usage level for audio communication in mesh signaling mechanism

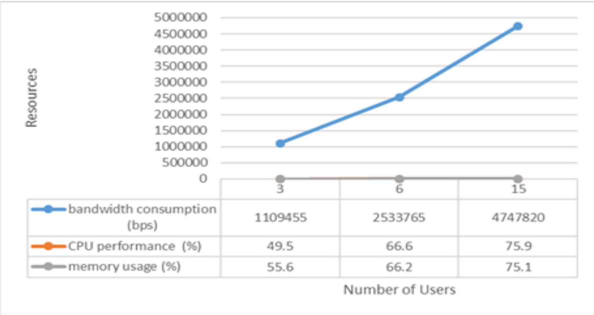


Figure 7. The Computer resource usage level for video communication in mesh signaling mechanism

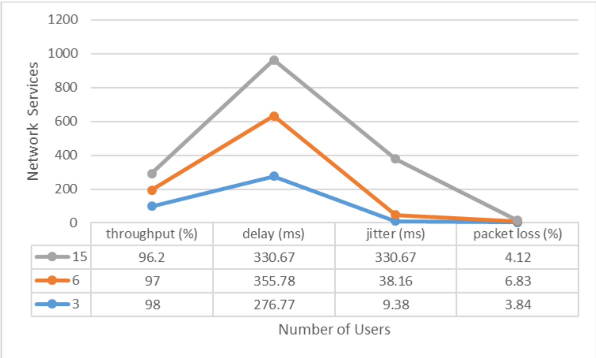


Figure 8. The network service of the quality for audio communication in mesh signaling mechanism

Figures 6 and 7 show that bandwidth consumption using a mesh signaling mechanism has an increasing similarity between audio streams and video streams, in line with the increasing number of clients, in this test 3 users, 6 users, and 15 users. To stream video requires more bandwidth than audio stream. The same goes for CPU performance and memory usage. The following in Figures 8 and 9 show the network service quality characteristics resulting from the mesh signaling mechanism.

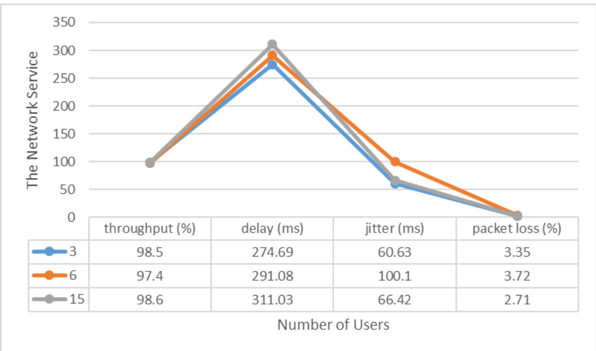


Figure 9. The network service of the quality for video communication in mesh signaling mechanism

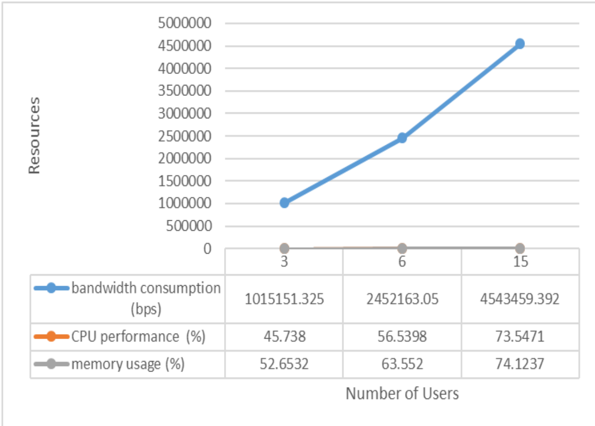


Figure 10. The Computer resource usage level for audio communication in npRTC signaling mechanism

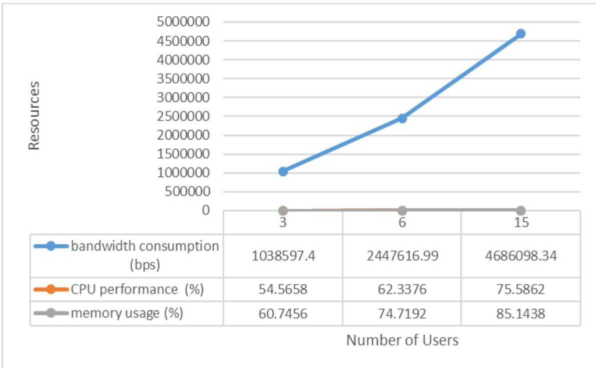
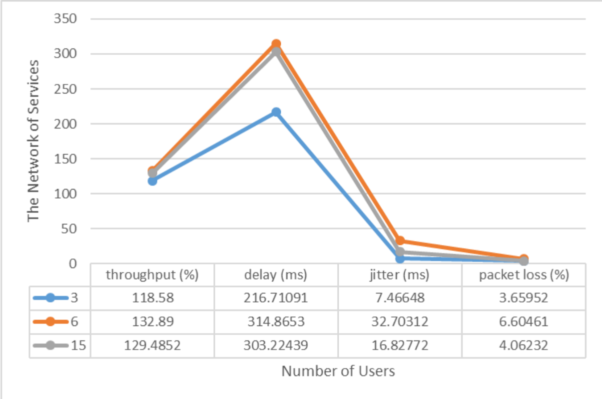


Figure 11. The Computer resource usage level for video communication in npRTC signaling mechanism

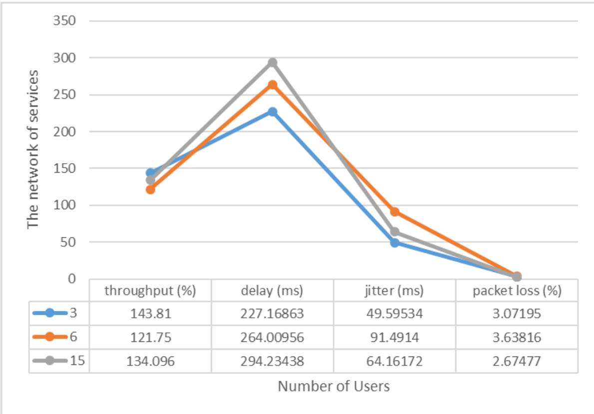
The Jitter characteristic is the average delay, but the graphs shown in Figures 8 and 9 do not describe the average delay. The author's assumption is that TURN Server has not been used in establishing connections in the WebRTC system. The throughput characteristics are shown to decrease in line with the increasing number of clients. The characteristics of delay, jitter, and packet loss increase in line with the number of clients.

Furthermore, testing has been carried out on the use of the npRTC model signaling mechanism with the same number of users, namely 3 users, 6 users, and 15 users. The level of use of computer resources for audio and video communication is shown in Figures 10 and 11. The results of the network service quality test are shown in Figures 12 and 13.



	throughput (%)	delay (ms)	jitter (ms)	packet loss (%)
3	118.58	216.71091	7.46648	3.65952
6	132.89	314.8653	32.70312	6.60461
15	129.4852	303.22439	16.82772	4.06232

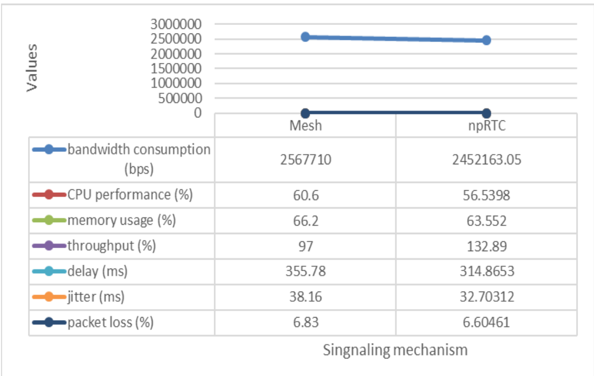
Figure 12. The network service of the quality for audio communication in npRTC signaling mechanism



	throughput (%)	delay (ms)	jitter (ms)	packet loss (%)
3	143.81	227.16863	49.59534	3.07195
6	121.75	264.00956	91.4914	3.63816
15	134.096	294.23438	64.16172	2.67477

Figure 13. The network service of the quality for video communication in npRTC signaling mechanism

Based on the test results as shown in the line chart above, the use of the npRTC signaling mechanism when compared to the mesh model shows an increase in



	Mesh	npRTC
bandwidth consumption (bps)	2567710	2452163.05
CPU performance (%)	60.6	56.5398
memory usage (%)	66.2	63.552
throughput (%)	97	132.89
delay (ms)	355.78	314.8653
jitter (ms)	38.16	32.70312
packet loss (%)	6.83	6.60461

Figure 14. The signaling mechanism from Mesh to npRTC model.

WebRTC communication performance, in terms of the performance of computer resource usage, including bandwidth consumption, RAM performance, and performance. memory usage, as well as in terms of network service quality. If it is described in a line chart, the comparison can be shown in Figure 14.

Testing of the npRTC model signaling mechanism and then comparing it with the mesh model signaling mechanism has been shown to improve the performance of the resources used by the user, as well as to improve the quality of network services. Thus our findings are novelty of the development of signaling mechanisms to establish real-time communication. However, further research needs to involve TURN Server as an intermediate server for establishing interactive connectivity.

4. CONCLUSION

This paper describes the methodology and data sources used for network performance testing with test parameters including: bandwidth consumption, CPU performance, memory usage, throughput, delay, jitter and packet loss.

The results presented in this report clearly that in this paper, the npRTC signaling mechanism in WebRTC is used for video conferencing and whiteboarding in an online virtual classroom and tested in a real implementation among 15 PCs. Other than that, npRTC signaling can be considered as a new signaling mechanism because it provides flexible communication among users. In addition, it can be applied in different applications, such as gathering a group of people on one call at a time, conferences between users, between teachers and students. The npRTC signaling requires an average of 76 (milliseconds) for ready and 103 (milliseconds) for sending requests and receiving responses, even when the network becomes dense. An in-depth explanation of CPU performance, memory usage, signaling performance, mesh topology, and p2p topology in physical implementation has been done.

The weak aspect of this research on the WebRTC signaling mechanism is that the intermediate server for interactive connectivity establishment using a STUN server does not yet involve a TURN server, so the media flow is still unstable.

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