A feasibility study on the web conferencing system using WebRTC

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1. Introduction

The Recent web technologies provide various remote conferencing functions for individuals or groups who need to conduct a conference with remote individuals or groups. The technology regarding web conferencing allows groups or individuals to communicate real-time via voice and images over the internet browser. It can provide live video-based interaction with remote participants. WebRTC is the web technology for the real-time communications over the internet browsers, and it is one of the interesting topics in the field of web conferencing. Web conferencing solution using WebRTC enables users to connect over video without having to download a plug-in or client program. In order to run the solution, all users need is an internet connection, a webcam device, and the compatible internet browsers. Then users can communicate with others at the simple procedure. But WebRTC technology is still under development and it has a serious problem regarding codec which will be supported by all internet browsers. Thus we study the feasibility of adopting WebRTC in the proposed web conferencing system. In this paper, we describe the result of the feasibility test of WebRTC multicasting and the proposed web conferencing system.

2. Proposed web conferencing system

The proposed web conferencing system is a chrome browser-based system that allows users to instantly share their live video/audio on the web through screen devices. The system can provide peer to peer real-time communication on the internet browser using WebRTC technology. It doesn't require to download any plug-in or client program. It consists of three functional modules, which are Video-ui, Peer, and WebRTCManager module. The work flow of the proposed system is as follow.

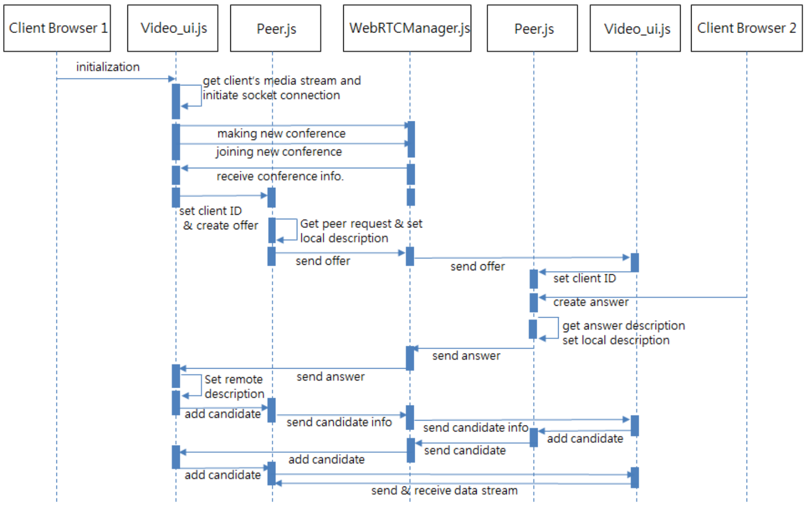

Figure 1. Work flow of the proposed system

To make two-way communication channel, Websocket server is used and control messages between peers is processed. In order to initiate a session, WebRTC uses PeerConnection to communicate data stream. Peer needs local & remote configuration information, and then those configuration information are sent/controlled by WebRTCManager. To make and join the conference room in the html page, local description and answer description is processed. In this workflow, the session description sent by the caller is referred to an offer, and the response from the callee is referred to an answer.

3. WebRTC multicasting test

We implement an application to perform multicast connectivity test using WebRTC. It is intended for checking whether the multicast operation is possible under the private network without using STUN/TURN server. The application to be developed in the chrome browser via Javascript APIs and HTML5 show us the successful multicast connection for the p2p communication. We use five screen devices to do 1:N multicasting test in private network. We measure the bandwidth and transmission rate of those devices respectively. First, we check the server and client side. The following figure demonstrates the feasible send/receive bandwidth for server and client side respectively.

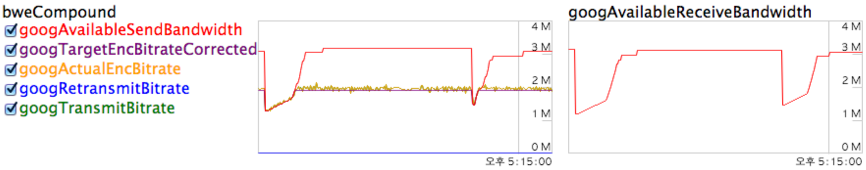

Figure 2. Feasible send/receive bandwidth

Next, we get the description information when performing the individual multicast connection. The following figure describes the output information which comes from the initialization and generation of peer connection object respectively.

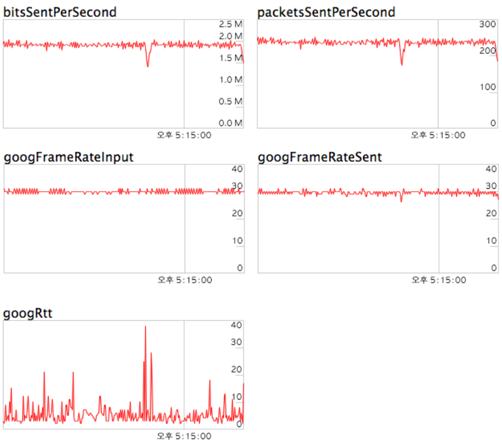

Figure 3. Description information when multicasting

4. Conclusion

We implement a web conferencing system using WebRTC technology. The system to be developed in the chrome browser show us the successful peer to peer communication through WebRTC multicast connection. Whenever peer connects to server, peer connection object is generated and it takes the feasible bandwidths. Then CPU usage of the server PC increase by up to 25%, memory usage and network usage increase by up to 200 megabyte, 25%, respectively.

5. Acknowledgment

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6. References

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