OPEN CONCURRENT NETWORK COMMUNICATION METHODS IN BUILDING DISTRIBUTED WEB APPLICATIONS

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(received: 10.08.2021; revised: 22.10.2021;
accepted: 04.11.2021; published online: 30.12.2021)

Abstract: The attractiveness of real-time multimedia communication as part of an e-learning platform largely depends on the quality of the telecommunications infrastructure and on the services that support the exchange of audiovisual data. The research subject of this work is communication between stationary and mobile devices using distributed services such as Web Real-Time Communication (WebRTC) running in HTML5 compliant web browsers. The test connections were carried out in a peer-to-peer architecture over a local wireless WiFi network and a mobile network supporting the Long Term Evolution (LTE) standard for data interchange. Several audiovisual sessions between two clients were analyzed for different connection scenarios. Parameters responsible for the transmission quality, such as delay, jitter, packet loss, or the speed of sending and receiving video frames were measured for each scenario. Open audiovisual communication system performance experiments were conducted under real operating conditions. The obtained results indicated potential applications in the development of e-learning websites.

Keywords: WebRTC, e-learning, LTE, LAN, WiFi, video transmission, jitter, delay, bandwidth

DOI: https://doi.org/10.34808/tq2021/25.4/c

1. Introduction

Recently, due to the epidemic, the demand for ICT systems for remote learning has increased. These systems require multimedia data transmission services. Such services include video chat, audio-video streaming, and peer-to-peer sharing of information. The video chat service is particularly demanding as it requires real-time data exchange. Many such ICT solutions are offering real-time communication. Such applications can be found in videoconferencing and remote learning systems [1, 2], controlling the UAVs [3, 4], and operating medical robots

[5, 6]. These solutions function mainly as native applications in a given operating system. The environment that provides universal access to the multimedia Internet content is the Internet browser environment. Currently, both stationary and mobile devices can use this environment to communicate with web services. However, what has remained outside the sphere of the web browser is the direct client-client communication. Nowadays, the WebRTC protocol developed by the World Wide Consortium (W3C) and the Internet Engineering Task Force (IETF) is the closest to the implementation of such a solution. This protocol allows real-time audiovisual communication of the browser-browser type. A system based on WebRTC still needs a telecommunications infrastructure to operate. The network technology that is widely available today are WiFi wireless networks. In big cities, there is a whole network of free access points, WiFi spots. This wireless network solution is used by both internet service providers and households. The second widely used technology for fast data transfer is the LTE service provided by mobile operators. The theoretical data transfer for the LTE system is 3 Gbps for downloading and 1.5 Gbps for uploading data [7]. The overall quality of the LTE video and sound transmission is satisfactory for the user. However, in the case of real-time calls, it is not only the data transfer speed that is significant, but also delays, which may result from changing the reception conditions of a wireless network. The genesis of work on the presented solution was the need to create a remote communication system for training purposes. The parameters characterizing this system should be low cost of implementation, the possibility of using it without installing additional software on the client's side, the quality of data acquisition allowing the ongoing communication to present the teaching content, and the possibility of archiving the collected audiovisual content.

2. Review of available solutions and literature

The communication software development has been using the WebRTC system since 2011. In [8], one of the multimedia communication solutions is presented, which allows the identification of users and takes into account the possibility of authorization. Additionally, installation of extra components is not required, but having access to a web browser is sufficient. Additional features presented in the system are the functionalities allowing single-cast and multicast transmission. The implementation of the solution uses Javascript and the Node JS library, and the server part is associated with Google STUN and TURN Server services. The authors of the study implemented a web application for group communication and presented the possibilities of using audio, video, and text messages. Although the RTC protocol does not provide session maintenance mechanisms, nor does it guarantee the quality of services, it can be the basis for the implementation of systems requiring high availability. The possibility of using additional methods of session control and monitoring the quality of data acquisition enables the use of the WebRTC protocol also in such areas. These issues are presented in [9], which describes the possibilities of using the WebRTC protocol for telemedicine implementation. In the presented work, the authors are using the STUN service to establish the connection between nodes in separate local networks. They set up NAT address translation on the routers to allow the nodes to communicate with each other. The empirical performance tests confirmed the usefulness of the application. Interestingly, it was possible to use the system with the use of modest hardware resources. The server responsible for maintaining the session was using the Linux Ubuntu Server 14.04 x64 operating system on a device with 512 MB of RAM and a 20 GB SSD drive. The paper [10] presents the requirements for teleconference systems in the WebRTC technology and problems that arise when implementing such solutions. In addition to such elements as the need to implement the required functionalities, such as recording resources, mixing audio and video channels, they mentioned the need to adapt to the changing conditions of the Internet network. The proposed solution is the introduction of multi-point transmission quality analysis. The study suggests a solution to improve session maintenance based on a centralized architecture, which can be considered a limitation concerning the p2p architecture. Multimedia data acquisition systems are a proper choice to extend the offer of educational units. This area of WebRTC applications is presented in the study [11]. The authors analyze the possibilities of integrating the created application with the Moodle e-learning platform. The proposed solution ensures the functioning of the so-called Virtual rooms, audiovisual broadcasts, and chat functionality. The use of the WebRTC protocol has allowed us to obtain a solution that works with the most popular web browsers. The created system is also capable of running on smartphones. An intriguing and innovative approach to build the application based on the WebRTC protocol in the areas of electronic learning is presented in the study [12]. The authors describe a system that uses the concept of 3D virtual teachers and methods of interaction between students and real teachers using the WebRTC protocol. The presented e-learning system has well-known functionalities from traditional audiovisual data acquisition systems, but also virtual presentations with the use of avatars. For this purpose, it uses text to speech modules. The study shows that the use of such modern technologies improves the perception of the educational content by students. One of the basic functionalities of group work systems and multimedia systems for sharing educational content is the functionality of communication via text messages. Currently, in applications such as text messengers, the dedicated Signal protocol [13] is often used. It is designed to transmit text messages and uses data encryption methods between nodes. The messengers using this technology are WhatsApp, Wire, Facebook Messenger, and many others. This protocol has a set of innovative features increasing its security (”future secrecy” or ”post-compromise security”). The improvement of safety is influenced by an innovative solution that allows updating the session key with each sent message.

3. WebRTC protocol

WebRTC is a protocol that enables the communication between web browsers and applications without the need to install additional applications. Currently, it becomes part of the standards for creating websites and web portals. The WebRTC communication project is developed under an Open Source license. It ensures data streaming between applications and web browsers through the use of peer-to-peer technology. This standard supports the most popular Internet browsers. The Google corporation initiated work on this solution, but at the moment, also the creators of Firefox, Opera, and Edge browsers have joined the project. There are also extensions for other web browsers that allow this method of communication through web pages. SIP solutions using RTP are also available on the market now. Soon, this may constitute the development of VoIP services, which are currently formidable competition for standard mobile telephony. The great advantage of WebRTC is the wide availability of free tools, ensuring the entire technological stack to use this standard. Among them are free decoding tools for G.711 and VP8 standards - they are made available without charging any license fees, which allows free of charge development of software. Collaboration tools through WebRTC have open access to the code whereby an increased number of their applications can be anticipated soon. This solution also has its limitations, such as the lack of support for Quality of Service (QoS) methods. The lack of guarantee of the QoS may constitute a significant barrier to the commercial implementation of solutions based on this standard. A serious advantage of WebRTC is the ability to support transmission encryption and authorization protocols. This functionality increases the security of the system based on the presented standard. All these features of the WebRTC protocol predestine this standard to the quick preparation of a system enabling audiovisual communication. This open set of tools allows configuring intermediary nodes in handling Video transmission and client nodes based on the browser itself without the need to install plugins.

4. Experimental setup

The purpose of this research is to evaluate the time efficiency of using WebRTC in video call tasks. In the actual conditions of using this type of application, individual nodes (client devices) are in the corporate infrastructure of the enterprise, various networks connected via the Internet, or local networks. Figure 1 shows the architecture for testing such configurations. As we can see in this Figure, the connection between Workstation 1 and Workstation 2 depends on the connection scenario. It can be a connection within the organization or via the Internet. If the workstation nodes cannot establish a direct call, WebRTC data transmission will be via an external ICE server. If the workstations are in the network of one organization, the provision of network ports for communication is related only to the configuration of access control lists and NAT services on individual routers.

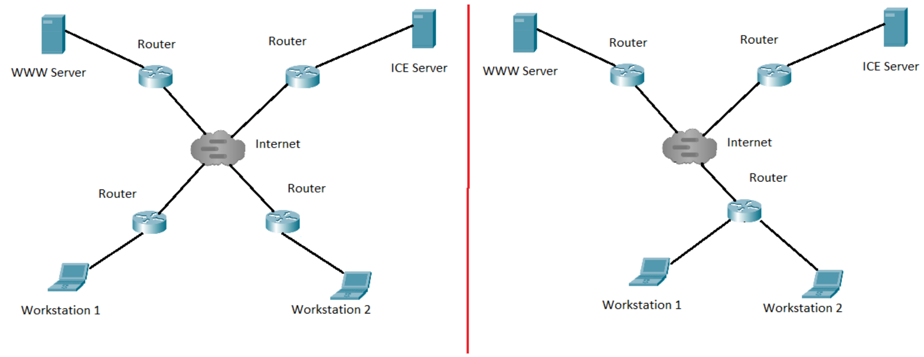

Figure 1. The research architecture for the solution

When using the WebRTC protocol in the field of e-learning, there is also a situation where both nodes do not have public IP addresses, and the required ports cannot be made available on the public network. In this situation, the role of an intermediary in the transmission is taken by the ICE server. It affects the delays and thus the time efficiency of data transmission. The presented network infrastructure architecture made it possible to evaluate the transmission time efficiency in various configurations, taking into account the acquisition and processing times of data between web browsers on individual workstations. Our experimental setup can be used as a platform of direct conversation between student and teacher, for example in the case of oral exams. We used the web application developed by us to conduct this research. The front-end of this application applies JavaScript and the WebRTC library. In the back-end, the PHP scripting language is applied to the signaling server. The application runs on the Apache HTTP server. We uses an everyday life computer network to test the time efficiency of the peer-to-peer transmission used in remote education applications. Due to the necessity to use a home computer network during teleworking and electronic education, in the research we used the LTE CAT 4 network and a wireless network in the 802.11n standard with an omnidirectional antenna. For this purpose we used Workstation1 and Workstation2 portable computers with the following configuration.

Workstation1:

HP OMEN i5-6300HQ/8GB/1TB, GTX960M graphics, HP Wide Vision HD Camera with a two digital microphone array.

Network interface: Intel Dual Band Wireless-AC 7265, TP-Link TL-WR1043ND router.

Workstation2:

HP Pavilion Gaming 15-dk1009nw i5-10300H 15,6"MattFHD slim IPS 250nit 8GB DDR4 SSD512 GTX 1650Ti_4G.

Network interface: Realtec RTL8822x, Netiaspot Netia Spot ADSL2+ router.

We connected these two workstations using the LTE network, sharing from a mobile phone, and the home wireless network. The default gateway used during the tests is a standard wireless network router provided by Netia (Netiaspot Netia Spot ADSL2 +), which works in the 802.11abg and 802.11n standards. LG Zero (LTE CAT 4) and Huawei Honor 6A (LTE CAT 4) mobile phones equipped with a VoLTE-enabled modem and a WiFi b/g/n module were used as access devices to the LTE network. The LG phone was connected to Workstation1 and the Huawei phone to Workstation2 using the USB 3.0 interface. The tests were performed with a single workstation in a home wireless network (low load), and with four workstations operating simultaneously (high load). We measured such parameters of the WebRTC transmission as: delay, jitter, packet loss, outgoing and incoming frame-rate speed to evaluate the effectiveness of communication. The duration of the entire measurement was 30 minutes, and the acquisition took place every second, which gave a total of 1,800 measurement points. We compared the obtained measurement results with the measurements carried out on the network infrastructure consisting of one router and workstations operating in the local network.

5. Results

The obtained measurement results are in the form of diagrams showing the dependence of the Cumulative Distribution Function (CDF) on parameters determining the quality of transmission. Figure 2 shows the CDF of the instantaneous packet delay in various network configurations. The Figure shows that the highest delays are in the case of transmission via the LTE network with the use of the infrastructure of two different network access service providers. The maximum instantaneous delay in this case is 2149 ms. For 50% of the connection time, the average latency is 330 ms. In the case of using the LTE network and the network provided by the service provider using the wired infrastructure, the number of temporary increases in delays was lower. When the connection was in the local wireless network, the delay in 86% of the cases did not exceed 300 ms. The packet loss during transmission using the WebRTC protocol occurred only when transmitting between two networks of different service providers. The highest packet loss occurred when using the LTE network and the wireless home network (Fig. 3). For 50% of cases, it was 1.13%. The packet loss for the LTE-LTE connection was 0.07% in 50% of cases. This result indicates strong signal fading and is closer to the LTE connection scenario number 4, proposed by Carullo et al. [14]. In other cases, during the experiment, the packet loss in the measured time was equal to zero.

The parameter that is showing the instantaneous link quality due to delays is the network jitter. In Figure 4, we can see the CDFs of this parameter in different configurations of the laboratory environment. The highest jitter occurred in the case of two LTE networks from two separate service providers. In this case, the instantaneous maximum value of the jitter was 113 ms. However, on average, for

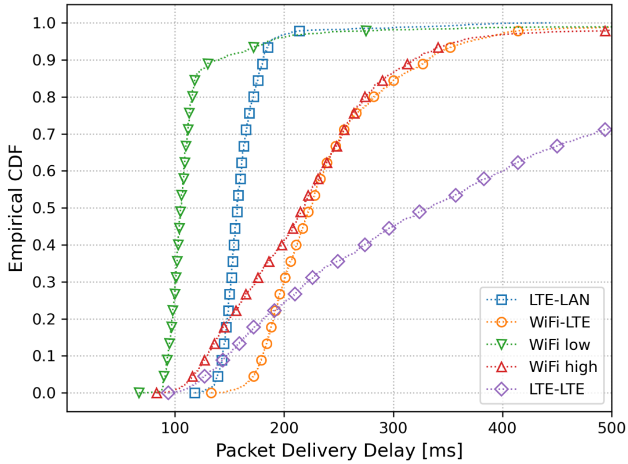

Figure 2. The CDFs of the packet delay

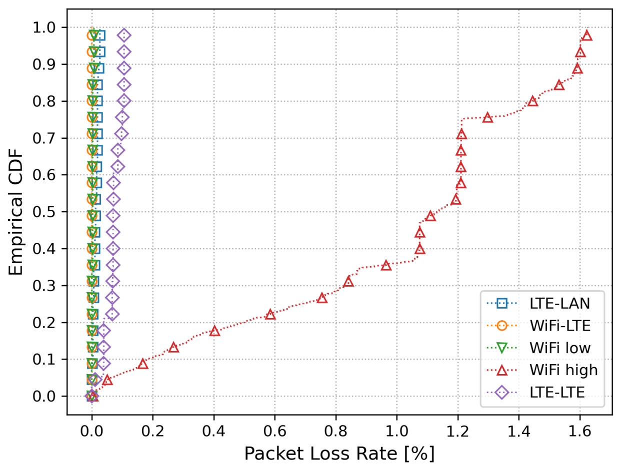

Figure 3. The CDFs of the packet loss rate

50% of the cases, this parameter remained at the level of 6 ms. The highest delay fluctuations occurred for the WiFi and LTE connections, where the jitter was equal to 12.5 ms for 50% of the measurements. A significantly high jitter was also observed in the local network in the case of a significant network load with additional traffic. Its maximum value was 57 ms.

The impression of smooth transmission and the possibility of effective use of WebRTC as a method of video transmission is related to the ratio showing the speed of received and transmitted video frames. In Figure 5, the measured values of these parameters in the form of a CDF plot can be seen. The worst situation

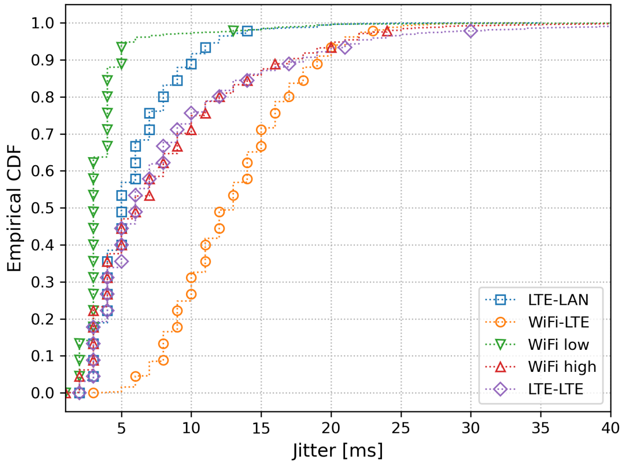

Figure 4. The CDFs of the jitter

was for the overloaded WiFi network and the WiFi-LTE connection, where the speed of receiving video frames for 10% of the connection time was at the level of 10 frames per second (FPS). The maximum FPS was 15 for a busy WiFi network. It is still acceptable for a video call from the client's point of view. We found that the best FPS rate was for the LTE-LAN connection and the low-load WiFi connections. The incoming and outgoing FPS strongly depends on the hardware, especially on the graphical subsystem and the camera device.

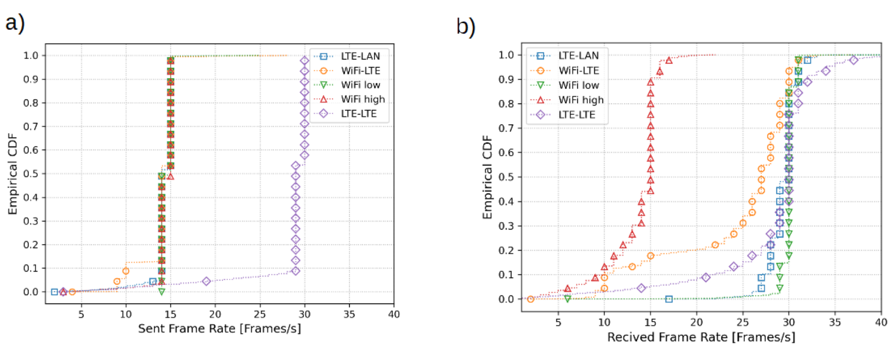

Figure 5. The CDFs of a) outgoing and b) incoming frame rates

6. Conclusions

In all the tested laboratory environment configurations, the speed of sending and receiving video frames enabled uninterrupted video transmission. The quality of the connection was acceptable for the real-time conversation, the measured average delays were under 400 ms. In the case of using the LTE network, the parameters of the speed of reading and transmitting video frames also made it possible to obtain a smooth transmission. The performed measurements showed that in each of the tested scenarios, the transmission parameters met the quality requirements for systems intended for remote communication in the area of electronic education.

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