Current state of the web browser-based data streaming

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Abstract. The article deals with the history of development and the modern state of the technology of receipt and transmission of streaming data with a web browser. It describes the current capabilities of data streaming through a web browser using various methods, starting with web browser add-ons and ending with integrated functionality; determines further prospects of these methods; and formulates the criteria of comparison and the methodology of selection of the technology for creation of a communication Internet service with the functionality of conventional VoIP.

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Introduction

One of the engrained trends of the modern times is the permanent growth of the SaaS (Software as a Service) market. At that, the user's web browser becomes the only point of entry for the whole set of applications. For example, the Gartner Agency predicts stable growth at the rate of 19.5% a year for SaaS [1]. At that, the sphere of CRM (Customer Relationship Management), ERP (Enterprise Resource Planning) use the SaaS model rather well, such area as audio and video communications are still the prerogative of conventional applications, as exemplified by Skype or VoIP clients. This situation is explained by the complexity of implementation of the necessary functionality in web browsers related to the fact that web browsers initially are oriented to the query-response model with displaying various textual content and images [2], which does not allow using them for data streaming. They also use the TCP protocol, which is way behind the UDP protocol used in conventional data streaming applications [3]. As the Internet has been developing, a number of technologies was implemented, which allowed circumventing these restrictions [4].

The history of development of web browser-based data streaming

The need in data streaming has resulted in implementation of various variants of arrangement of such streaming. Add-ons to web browsers that are common programs, which can operate inside the browser, were the first ones used for data streaming through web browsers. At that, the user needed to install them directly from the service website or download them and then install as separate installation packages. Java Applet was suggested for implementing voice communication in web browsers [5]. Add-ons to the Firefox web browser were developed that implemented the SIP protocol [6].

Currently, the usage of add-ons has decreased due to several reasons. Firstly, the widespread use of the Flash technology by the Adobe company. Secondly, one of the leading web browsers – Chrome [7] – was declared not to support add-ons anymore. According to the Adobe Company, Flash Player is installed with more than 1.3 billion Internet users [8]. By its essence, Flash Player is an add-on, but almost every user has it, so it does not require installation. At that, Flash Player also evolved: initially it used the RTMP protocol, which later was replaced with the more advanced RTMFP protocol [9]. Since recently, there appeared the trend of integration of the data streaming functionality directly into web browsers. The first technology that implemented this technology was the WebRTC project [10], which is the set of various protocols and models. Despite the rather long development (starting from 2011), standardization of this set of protocols has not been completed yet. An alternative to WebRTC is the ORTC protocol [11], which is similar to WebRTC, but is more low-level one and thus provides more freedom to web applications developers. Thus, the following protocols have prospects in terms of history:

1) Adobe Flash with the RTMFP protocol;
2) WebRTC;
3) ORTC.

Application and the comparison methodology

Before proceeding to the comparison of technology of streaming data exchange with the web browser of a user, it is necessary to identify the criteria, by which the comparison must be made.

The article suggests consideration of the technologies from the point of their application at development of a communicative Internet service with the following specifications:

1) Large number of end users using Internet communications.
2) Users use various hardware and software

platforms including mobile devices.

3) Users are territorially separated and locate in networks of different service providers with different topology.
4) Users need to communicate with both the service users and the users of VoIP, PSTN or cellular networks.
5) Users need protected data transmission.

The general scheme of a communication Internet service with user's access through the web browser is represented in Figure 1.

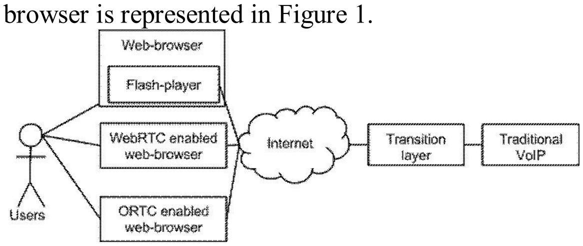

Figure 1. The general scheme of a communication Internet service with user's access through the web browser

In terms of creation of a communication Internet service with the opportunity to interact with the systems of conventional VoIP, the following properties of the technology are quite interesting:

The simplicity of the initial deployment and installation of the client components. This property is important for a mass Internet service and for cloud-based contact centers, as it allows cutting the deployment costs.

Support of various software and hardware platforms. This includes operation systems, web browsers, support of mobile devices, such as tablets and smartphones. At that, it is necessary to consider both the current state and the future prospects.

Infrastructure requirements, namely the resources of the hardware platform, on which the web browser will operate, the requirements for the communication channels to the web browser, as well as opportunities to communicate with networks that use network address translation (NAT). Qualification requirements must be carried out by personalized assessment of the quality of the received audio (assuming the symmetry results) on a 5-point scale with a description of the problems encountered by criteria similar to the Mean Opinion Score [12]. Personalized assessment is suitable in this case, because they allow comparing the tested technologies. The characteristics must have the following grades:

1) Impossibility of communication.
2) Problems resulting in distraction close to the impossibility to communicate.
3) Problems (regular interruptions or jamming) resulting in distraction.
4) Sufficient, the sound is pure, though some deficiencies are noticed.
5) Perfect quality like when talking face to face.

In order to determine the requirements, it is necessary to perform the following experiments:

1) In order to assess the requirements for the hardware platform, it is needed to compare the quality of the reproduced sound in your web browser and evaluate it. To level off the influence of the network, the source of the audio stream must be located in the same sub-network as the PC, on which the audio is listened to. To model the load of the CPU, one can use the CpuKiller freeware [13] with the following gradation of the CPU load: 5%, 20%, 40%, 60%.

2) In order to test the bandwidth, one can use the Wide Area Network Emulator freeware [14]. At that, the focus is not on the channel capacity, as in the majority of cases, it is sufficient, but such parameters as jitter and losses. It is necessary to provide assessment of the following parameters of the network by the 5-point grading scale: Latency 150 msec and jitter 0, 10 msec, 20 msec, 40 msec. Losses 0, 5%, 10%, 20%.

The characteristics will be suitable if their score is not less than three points.

The possibility to connect to traditional VoIP systems. Support of signal protocols as well as protocols of media data streaming. Support of this requirement will allow ensuring interaction with the PSTN and cellular networks subscribers. The conventional VoIP uses the SIP [15] for establishing connection and RTP [16] for the streaming data.

The used formats of data encoding. The formats have different areas of application, and usage of coordinated formats of data transmission allows avoiding transcoding when data are transmitted to other information systems.

Methods of the streaming data protection. This means protection of both the protocol of the session establishment and the protocol of data streaming itself.

Support of the UDP protocol. This criterion is necessary to provide for high quality data streaming, as the alternative TCP protocol does not allow to provide for sufficient latency at unstable channels.

Based on the formulated criteria, we can compare the current technology of streaming data receipt and transmission with a web browser.

Conclusions

Based on the historical analysis, the prospective technology of streaming data exchange with web browsers were determined:

1) WebRTC
2) ORTC
3) Adobe Flash

Then, we determined the area of application of this technology in order to create a communication Internet service with the functionality allowing interacting with conventional VoIP, and the methodology for further comparison and selection of the most suitable technology was formulated.

Credits

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