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# Investigation of Stability of the Technologies of Streaming data Transmission from a Web Browser with Respect to the Factors Affecting the Transmission

**Anton Pavlovich Teykhrib***

***Abstract :*** The article deals with the issues of stability of the technologies of streaming data transmission from the user's Web browser with respect to various external factors affecting the quality of the streaming data transmission in the user's workplace and in the network infrastructure. The following factors in the user's workplace are considered: CPU load and the amount of free RAM. For the network infrastructure we consider: the delay jitter, losses in the transmission, transport capacity of the network. For various values of these parameters, we measure the quality of the streaming data transmission for the Adobe Flash RTMFP and WebRTC technologies, using a developed prototype of the software for contact centers. The obtained results have showed the superiority of the WebRTC technology over RTMFP with respect to all considered parameters.

***Keywords :*** Contact center, stability of the data transmission technology, WebRTC, Adobe Flash RTMFP.

## 1. INTRODUCTION

The quality of the streaming data transmission is one of the most important factors from the standpoint of the end user of the contact center services. The technology of the streaming data transmission from the user's Web browser, beginning from the advent of Adobe Flash RTMFP in 2006 and WebRTC in 2011, is gradually expanding the scope of its application. Therefore, when choosing a technology of the streaming data transmission used in the work of a contact center, it is necessary to determine the limiting conditions for the applicability of this technology.

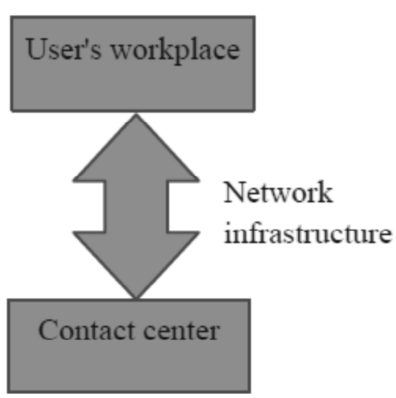


**Fig. 1. A general scheme of the streaming data transmission.**

To determine the factors that influence the transmission of streaming data from a Web browser during the contact center operation, it is necessary to consider the whole system in which streaming data is transmitted,

* The NAUMEN company (Nau-service) 49a Tatishcheva Street, 4th Floor, Ekaterinburg, 620028, Russian Federation

because the transmitted streaming data are influenced by both the environment and infrastructure. Taking into account the fact that, from the viewpoint of the data transmission, the audio data (the initial signal is a sound wave) and the video data (the initial signal is a light wave) are similar, so in the future we will consider, without any loss of generality, the transfer of the audio data. The transmission scheme is shown in Figure 1.

The factors, related to the user's workplace and the network infrastructure, directly affect the quality of the sound information transmission [1]. However, these factors cannot be influenced preventively; and therefore, to determine the boundary conditions, the following factors should be considered:

**(*a*) User's workplace:**

1. CPU load of the user's workplace;
2. the amount of free RAM in the user's workplace.

**(b) The parameters of the network in the segment from the user's workplace to the cloud contact center [2]**

1. The delay jitter;
2. Losses in the transmission;
3. Transport capacity of the network.

It is also necessary to determine to what extent the streaming data transmission technology is demanding of the server resources for the subsequent determination of the required computing power; thus the following characteristics should be considered:

1. CPU load at the side of the cloud contact center [3];
2. the amount of free RAM at the side of the cloud contact center, in addition to the considered in [3] CPU load;

This article is devoted to the analysis of stability of the WebRTC and Adobe Flash RTMFP technologies with respect to the considered factors.

**Method of studying stability of the technologies of streaming data transmission from a Web browser with respect to the factors affecting the transmission**

On the basis of the factors identified above, it is necessary to study the stability of the technologies of streaming data transmission with respect to the factors that are associated with the streaming technology. The study will be carried out over the independent groups identified in the previous section. To determine the stability, it is necessary to develop a method by which we will compare the technologies. Our general procedure will be as follows: we should measure the voice quality during the streaming data transmission for various values of the considered factors. At the same time, prior to the measurement, it is necessary to determine the range of variation of these factors. As a result, we will obtain the quality assessments for the considered data streaming technologies. Based on the comparison, we can talk about the superiority or shortcomings of the considered technologies of the streaming data transmission from a Web browser.

**User's workplace**

**As it was shown above, in the user's workplace the following factors have influence:**

1. CPU load of the user's workplace;
2. The amount of free RAM in the user's workplace.

Taking into account the differences in the workplace configuration of potential users of a cloud contact center, we will consider a typical configuration. As a typical configuration, we have chosen the one based on the Yandex company's recommendations for the office computer [4]:

1. One Intel or AMD processor with 2 cores;

2. The clock frequency is 2 GHz
3. RAM is 2 GB;
4. The disc larger than than 100 GB;
5. The operating system is Microsoft Windows 7.

The CPU load will be varied on a scale from 0% to 100% with the increments of 20%. It should be noted that the load values are rough estimates because it is impossible to provide uniform loading due to the multi-tasking character of operating systems and the presence of a large number of background processes that run at arbitrary times.

To ensure the processor load, we will use free software simulating the computing load, CpuKiller [5], which allows setting an arbitrary required load in the range from 0% to 100% with the step of 1%.

We will vary free RAM from 1GB to 0GB with the 500MB increments, and to set these values, we will use the TestLimit free software [6], recommended by Microsoft.

Thus, the result is a two-dimensional space of options, for each of which we will obtain a quality assessment according to the MOS method, presented in [7].

**Network infrastructure in the segment from the user to the cloud contact center**

This segment is very important from the point of view of the quality of the transmitted data, because the factors generated by this segment are directly reflected in the transmitted audio signal. Within this segment, the following factors are significant:

1. The delay jitter;
2. Losses in the transmission;
3. Transport capacity of the network.

To simulate changes in the factors of this segment, we will use the Wide Area Network Emulator free software [8], which allows changing all these factors. Consider the range and the increment of these factors.

The factor of delay jitter is defined in [9] as the deviation of the delay of a packet from a reference or baseline value, for example, from the average delay. The occurrence of jitter is directly related to the fact that the voice packets share the same channel with the data packets, and in the case of high network load, the time, during which the packets are awaiting processing in the network device, may vary randomly. Irregular intervals of the packets arrival to the recipient significantly reduce the quality and intelligibility of voice, as well as introduce additional delay in the presence of the jitter leveling algorithms. The admissible jitter value [10] in VoIP should not exceed 40 ms. Thus, we have to test the streaming data transmission technology for the jitters: 0 ms, 20 ms, 40 ms and 60 ms (as the excess over the recommended value by the step value).

Losses in the transmission. This is the percentage of packets that have not been delivered to the recipient by the network. In the case of loss of a part of the data packets, the recipient can simply request a retransmission of the lost data; for the voice data, this approach is not fully suitable, because it leads to a strong increase in the overall delay. If the voice packets are coming with a long delay, the receiving side ignores them because they may already be behind the current playback.

The losses of about 3% are considered to be admissible [11]. Moreover, it is an ambiguous factor, because depending on the nature of losses, the same value of this factor may have different effects on the final value of the quality assessment; as the delay character, we understand in this case the distribution of the lost packets in time: 1% losses concentrated at the same time leads to disappearance of a whole word, or they may be inobservable for a uniform distribution. Therefore, in the research concerning this factor, we will consider values of 0%, 3%, 6%, with a uniform distribution, since it will provide an objective description.

Transport capacity of the network. This factor has less impact on the transmission of streaming data, because the voice traffic does not need a broad band: for example, the traditional audio coding format G.711 requires 64 kbit/s in one direction, or 128 kbit/s in two directions, and, according to [12], this enables 98.7% of the users of stationary workstations to work.

Thus, within the study of the network section from the user's workplace to the cloud contact center, there should be studied the significance of two factors: losses and jitter. This generates a two-dimensional space of options, for each of which we will obtain a quality assessment in accordance with the MOS procedure, presented in [7].

**Functioning of the cloud contact center**

Given the fact that this section is completely under the control of the contact center, its factors determine the resource-intensiveness of the streaming data transmission technologies, as well as the specific instances of this technology implementations. As a result, we do not need measurements of the quality of the transmitted voice in its dependence on the values of these factors over the ranges, but we need an assessment of the streaming technology requirements to the server computing power.

**Evaluation of the consumed power will be conducted by the following algorithm on the server with a following configuration:**

1. The processor type is Intel Xeon with 4 cores;
2. The clock rate is 2.6 GHz;
3. RAM is no less than 8.0 GB;
4. The disc subsystem – 100 GB SATA-disc, 2 pieces;
5. RAID-controller with the support of the levels 0, 1;
6. The presence of two network interfaces with the carrying capacity 1 Gbit/s.

We perform the measurement of CPU load and free memory for the server without the streaming data transmission sessions. To improve the accuracy, we carry out 5 measurements, with the subsequent calculation of the average value and the error for the probability of 0.95.

To increase the accuracy of measurements and averaging the values, we organize 50 streaming data transmission sessions (in accordance with the technical specification), and also calculate the average value and the error, corresponding to one transmission session.

Thus, as a result we obtain the requirements to the server computing power, which are made by the streaming data transmission technology, which will allow evaluating the characteristics of the hardware environment of the cloud contact center to serve a given number of streaming sessions.

**The developed technique of studying the streaming data transmission technology allows looking at the streaming transmission technology from two sides :**

1. To what extent it is demanding of the computational power of the user's workplace and the communication channels;
2. To what extent it is resource-intensive in terms of the server computational power.

Carrying out research according to the given methodology will provide qualitative and quantitative characteristics of streaming data transmission technologies, which can then be used to select the streaming technology for each newly created communication session.

**The research results on the stability of the technologies of the streaming data transmission from a Web browser with respect to the factors influencing the transmission.**

**User's workplace**

By measuring the quality estimates for the streaming data transmission for the Adobe Flash RTMFP technology, the following average values of the estimates have been obtained, and they are shown in Table 1.

**Table 1. The average values of the estimates for the streaming data transmission quality for the Adobe Flash RTMFP technology**

| *CPU load, %\ free RAM, GB* | ***1*** | ***0.5*** | ***0*** |
|---|---|---|---|
| 0 | 4.1 | 4.1 | 0 |
| 20 | 3.5 | 3.5 | 0 |
| 40 | 2.5 | 2.5 | 0 |
| 60 | 1.5 | 1.5 | 0 |
| 80 | 0 | 0 | 0 |
| 100 | 0 | 0 | 0 |

By measuring the estimates of the streaming data transmission quality for the WebRTC technology, the following average values of the estimates have been obtained, and they are presented in Table 2.

**Table 2. Average values of the estimates of the streaming data transmission quality for the WebRTC technology**

| *CPU load, %\ free RAM, GB* | ***1*** | ***0.5*** | ***0*** |
|---|---|---|---|
| 0 | 4.1 | 4.1 | 0 |
| 20 | 4.1 | 4.1 | 0 |
| 40 | 3.5 | 3.5 | 0 |
| 60 | 2.5 | 2.5 | 0 |
| 80 | 1 | 1 | 0 |
| 100 | 0 | 0 | 0 |

To determine the superiority of one technology over another, we calculate the difference between these estimates in accordance with the formula (1).

$$S = M_{WebRTC} - M_{RTMFP}, \qquad (1)$$

In the case of the WebRTC technology being superior, the value is positive, and, conversely, it is negative when the RTMFP technology is superior. A comparative analysis is presented in Table 3.

**Table 3. Comparative analysis of the estimates for the streaming data transmission quality for the WebRTC and Adobe Flash technologies.**

| *CPU load, %\ free RAM, GB* | ***1*** | ***0.5*** | ***0*** |
|---|---|---|---|
| 0 | 0 | 0 | 0 |
| 20 | 0.6 | 0.6 | 0 |
| 40 | 1 | 1 | 0 |
| 60 | 1 | 1 | 0 |
| 80 | 1 | 1 | 0 |
| 100 | 0 | 0 | 0 |

Thus, it is seen that the streaming data transmission quality in the case of WebRTC is greater or equal to the data transmission quality while using Adobe Flash. This suggests that the use of the Adobe Flash technology from the standpoint of user's workplace only makes sense when the WebRTC technology is not supported in the user's Web browser.

**Network in the segment from the user to the cloud contact center**

According to the results of measurement of the estimates of streaming data transmission quality for the Adobe Flash RTMFP technology, the following average values of the estimates have been obtained, and they are presented in Table 4.

**Table 4. Average values of the estimates of the streaming data transmission quality for the Adobe Flash RTMFP technology**

| *The delay jitter, ms\Losses in the transmission, %* | *0* | *3* | *6* |
|---|---|---|---|
| 0 | 4.1 | 3.5 | 3 |
| 20 | 2.5 | 2 | 2 |
| 40 | 2 | 1 | 0 |
| 60 | 1 | 0 | 0 |

According to the results of measurement of the estimates of streaming data transmission quality for the WebRTC technology, the following average values of the estimates have been obtained, and they are presented in Table 5.

**Table 5. Average values of the estimates of the streaming data transmission quality for the WebRTC technology**

| *The delay jitter, ms\Losses in the transmission, %* | *0* | *3* | *6* |
|---|---|---|---|
| 0 | 4.1 | 4.1 | 3.5 |
| 20 | 4.1 | 3.5 | 3 |
| 40 | 3.5 | 3 | 2 |
| 60 | 2.5 | 2 | 1 |

To determine the superiority of one technology over another, we calculate the difference between these estimates in accordance with the formula (1).

In the case of the WebRTC technology's superiority, the value is positive, and, conversely, it is negative in the case when the RTMFP technology is superior. A comparative analysis is presented in Table 6.

**Table 6. Comparative analysis of the estimates for the streaming data transmission quality for the WebRTC and Adobe Flash technologies**

| *The delay jitter, ms\Losses in the transmission, %* | *0* | *3* | *6* |
|---|---|---|---|
| 0 | 0 | 0.6 | 0.5 |
| 20 | 1.6 | 1.5 | 1 |
| 40 | 1.5 | 2 | 2 |
| 60 | 1.5 | 2 | 1 |

Thus, it is seen that the streaming data transmission quality in the case of WebRTC is greater or equal to the data transmission quality while using Adobe Flash. Thus, it can be concluded that the use of Adobe Flash technology from the standpoint of the network segment from the user to the cloud contact center makes sense only when the WebRTC technology is not supported in the user's Web browser.

**Study of the technologies of the streaming data transmission in terms of the server resources consumption**

The consumption of server resources plays an important role in deciding whether to use this or that streaming technology. This is due to the fact that, on one hand, it affects the cost of service, and, on the other, it affects scalability of application of a streaming data transmission technology.

Based on the measurement of the server resource consumption, the following results have been obtained, see Table 7.

**Table 7. Comparative analysis of the server resources consumption.**

| | ***Adobe Flash RTMFP*** | ***WebRTC*** |
|---|---|---|
| CPU load per one connection, % | 2 | 1.5 |
| RAM load per one connection, MB | 1.5 | 1.5 |

Thus, it is seen that, in terms of loading the server computing power, both technologies demonstrate comparable results for one connection. However, in the case of a large number of connections, a scale effect will be significant, resulting in the fact that the WebRTC technology is ultimately preferable in terms of the server resources consumption. The obtained results are fairly predictable and can be explained by the fact that, for the operation of the Adobe Flash RTMFP streaming data transmission technology, one needs to have an additional conversion of the RTMFP protocol into the RTP, which is not used when using the WebRTC technology.

## 2. CONCLUSION

**The analysis of the applicability conditions of the technologies of streaming data transmission from a Web browser has yielded the following results and conclusions :**

1. The main factors affecting the quality of the streaming data transmission from a Web browser are identified and systematized.
2. It is established that the factors directly associated with the technology of streaming data transmission from a Web browser are localized in the following sections: user's workplace, the segment of the network between the cloud contact center and the user's workplace.
3. Based on the analysis of the distinguished factors, a methodology is developed of assessing the stability of the streaming data transmission technology with respect to the changes in these factors, as well as assessing the demands of technology to the server computing power. In accordance with the developed method, a study is conducted of the Adobe Flash RTMFP and WebRTC streaming data transmission technologies.
4. A comparative analysis of the study results has revealed the superiority of the WebRTC technology over the Adobe Flash RTMFP both in terms of stability and in terms of demands for the server capacity. Moreover, in the case of unavailability of WebRTC technology in the Web browser of the user, it make sense to use the Adobe Flash RTMFP technology, which has acceptable stability and reasonable requirements to the server computing power.

   The findings should be taken into account in the design and development of software for contact centers, using the streaming data transmission technologies.

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