A Study of Influencing Factors on the Quality of Experience Perception in Video Streaming

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***Abstract*—Video Services using Over-The-Top (OTT) becomes extremely popular in all countries but video streaming through unguaranteed path in Internet cause a lot of problems of keeping the Quality of Experience (QoE) on a suitable level. In the previous works of authors the influence of Quality of Service (QoS) parameters on QoE were studied using subjective assessment. As a result of this study objective QoE metrics: the frequency and duration of re-buffering were chosen as main parameters of customer perception. The dependency of these metrics from QoS parameters were obtained for separate influence factors (packet loss, delay, bitrate). In this paper the results of studies of complex influencing factors on QoE are presented. Combination of the influencing QoS factors cause the degradation of the threshold after which re-buffering starts. The comparison of resistance to complex influencing factors for different video streaming technologies (Pseudo-HTTP streaming, RTMP and HDS) were done. The results of comparison show that different streaming technologies react on combination of packet loss, throughput and delays in a various ways. The recommendations for using video streaming technologies with OTT are done.**

I. INTRODUCTION

The OTT (Over-The-Top) technology for providing video on demand services becomes more and more popular in all countries over the world. A number of users of streaming video services increase significantly from year to year. The biggest providers implement this technology in their IP networks for gaining more profits from the existing network infrastructure. This all leads to increase of competition between the providers seeking to capture the market. We have a lot of examples of such strategy. Thus American company Netflix announced the plan of coming to Russian market [1]. Due to information from the site of Izvestia newspaper [2] Netflix service is available in Russia from the beginning of 2016. This popular service was studied in [3].

The competition is always good to the users but with a lot of possibilities to choose the provider the quality of experience QoE becomes the most significant factor. For this reason a lot of research work was done by authors for evaluation of the influence factors of network performance to QoE. During the first phase of the research the experiments for subjective evaluation of QoE with different sets of QoS parameters (packet loss, delay, throughput) were completed on experimental platform. The results of experiments [4-6] show that the influence of IPER and IPTD is most significant to QoE. Unfortunately the subjective measurements have a great dispersion and are time consuming. For objective measurements was suggested to use such factors as re-buffering – the situation when receiving buffer is empty and video playback is stopped. The estimation of video quality can be done based on the frequency and mean duration of re-buffering [6].

Most of the providers [7-10] have instruments for monitoring the network performance parameters but they need to know what are the thresholds of QoS parameters. If the current value exceeds the threshold than video quality will decrease significantly. This research is devoted to the estimation of thresholds of packet loss and packet delay for Pseudo-HTTP, HDS and RTMP streaming technologies.

The paper is structured as following. In section II the experimental platform is described and characteristics of used video clips are presented. In section III the instantaneous parameters of video streaming are chosen. In section IV the influence of packet loss and packet delays on the start of re-buffering are presented and discussed. In section V conclusions are made. Section VI is devoted to future work.

II. EXPERIMENTAL PLATFORM

The previous studies were done on the experimental platform described in papers [5, 6]. For the new studies the platform was optimized in the following way. As a first step the Virtual Machines (VM) simulating the streaming servers were excluded and their functionality were transferred to the physical server with OS Windows Server 2008 R2 Standard (64-bit). We use Wowza Streaming Engine v 4.1.0 [11] to provide: Adobe Flash RTMP [12] and Adobe Flash HTTP Dynamic Streaming (HDS) [13], and Integrated IIS server to provide Pseudo-HTTP streaming [14]. The functional diagram of the optimized experimental platform is shown on a Fig.1. Detailed information about platform is presented in the Table I.

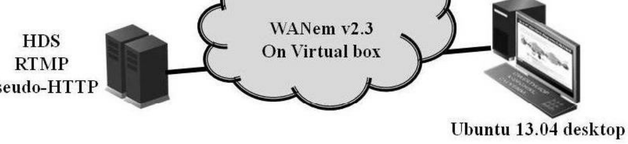

Fig. 1. Optimized experimental platform

The previous experience shows that shorter video files fragments can be used in the experiment. We have used video files with the duration of 140 s that give us a possibility to reduce total duration of the experiment twice. The experiment proved that reducing of the duration of video files did not influence on the estimation of the thresholds of re-buffering.

TABLE I. DETAILED INFORMATION ABOUT PLATFORM

Name	Description
PC Hardware Platform	CPU - Intel Core i7 – 4770 RAM – 16 Gbyte Video card - GeForce 210 Gainward PCI-E
OS Server	Windows Server 2008 R2 Standard (64 bit)
Virtual Box	Oracle cross-platform virtualization application
WANem	Wide Area Network Emulator version 2.3
Streaming Server	Wowza Streaming Engine version 4.1.0
OS Client	Ubuntu version 13.04 (64 bit)
Browser	Opera
Web Interface	Custom Interface

The following limitations were used in experiment based on the results of [6]:

- The throughput of the path was limited to 10 Mbit/s;
- Video clip was chosen with resolution of 1280x720 pixels, Variable Bit Rate (mean value of bitrate is equal 5.68 Mbit/s) and quickly changing scenes with dynamic objects, audio 192 Kbit/s.

The limitation of throughput and selection of the video clip are based on the following conditions:

- New results must be comparable with previously obtained in [5, 6] with 10 Mbit/s throughput;
- For video clips with resolution of 1080 pixels the throughput of 10 Mbit/s is a critical limitation which reduce of range of the influencing factors (any small change of the factor causes the re-buffering);
- Video clips with lower resolution can influence on the perception of viewers.

III. THE SELECTION OF INSTANTANEOUS PARAMETERS OF STREAMING VIDEO USABLE FOR FURTHER ANALYSIS

The first task of the research was to find the parameter applicable for forecasting of the re-buffering. On application layer we can use only 2 such parameters: current buffer size in bytes and buffer size estimation in seconds of viewing.

During the experiment was revealed that the re-bufferings start on certain timestamps of video playback. A timing chart of changing the bitrate during video playback was analyzed. As shown on Fig.2 the bitrate is changing greatly during the video playback with high peaks and downs.

Mean value of bitrate is equal 5.68 Mbit/s and during peaks the value is doubled. Meanwhile the re-buffering does not occur if peak's duration is not too long as we can see on Fig.2 near the timestamp of 40 seconds. But if bitrate is at high level relatively long than re-buffering can takes place as it can be seen at a timeframe 80-90 seconds – the buffer is nearly empty (see Fig.4).

On Fig.3 the fulfillment of the buffer is shown in Mbytes during video clip playback with packet loss of 0.5% and delay of 5 ms. As it can be seen on Fig.4 the current buffer fulfillment for Pseudo-HTTP, RTMP and HDS are rather similar and they follow the changing of bitrate on Fig.2.

This is a good illustration that during 140 s of this video clip playback we have enough range of the changing of bit rate to evaluate the threshold after which the re-buffering starts.

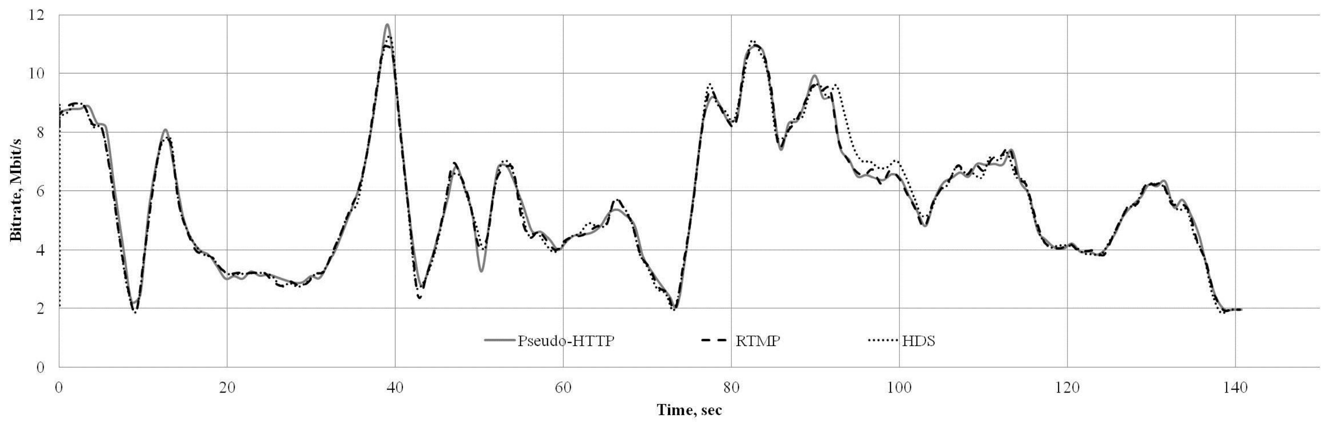

Fig. 2. A time chart of bitrate during video clip playback

On Fig.4 the fulfillment of the buffer in seconds is shown (how many seconds of video clip playback can be realized with current buffer fulfillment). The buffer size remains nearly the same when no problems occur for all types of streaming protocols. We consider that buffer size in seconds gives more representative view of the process than buffer size in bytes. Taking into account this consideration we will use for further analysis the buffer size in seconds.

As it can be seen from Fig.4 the most stable streaming technology is Pseudo-HTTP. The streaming technologies HDS and RTMP have drop down the buffer size at a timestamp of 80 ms with packet loss 0.5 % and 5 ms of delay. At this timeframe the buffer is nearly empty but re-buffering has not starts. If the delay increase up to 9 ms then re-buffering starts as it can be seen from Fig.5. At a timeframes of 90 ms and 110 ms the buffer becomes empty for RTMP and HDS streaming technologies. For Pseudo-HTTP technology the buffer becomes empty only once at a timestamp of 110 s.

According to all algorithms of video streaming at a time when buffer is empty the playback stops until the buffer size exceeds the limit. You can see these periods of filling the buffer as vertical lines on Fig.5. Immediately after reaching minimum buffer size the playback starts again. If the bitrate of flow is rather high than buffer size becomes smaller and smaller and at last the buffer becomes empty that cause re-buffering as it is shown on Fig. 5 for HDS. If the bitrate is lower than the buffer size increase to higher level. The behavior of Pseudo-HTTP streaming is rather similar to HDS but for RTMP it is really different.

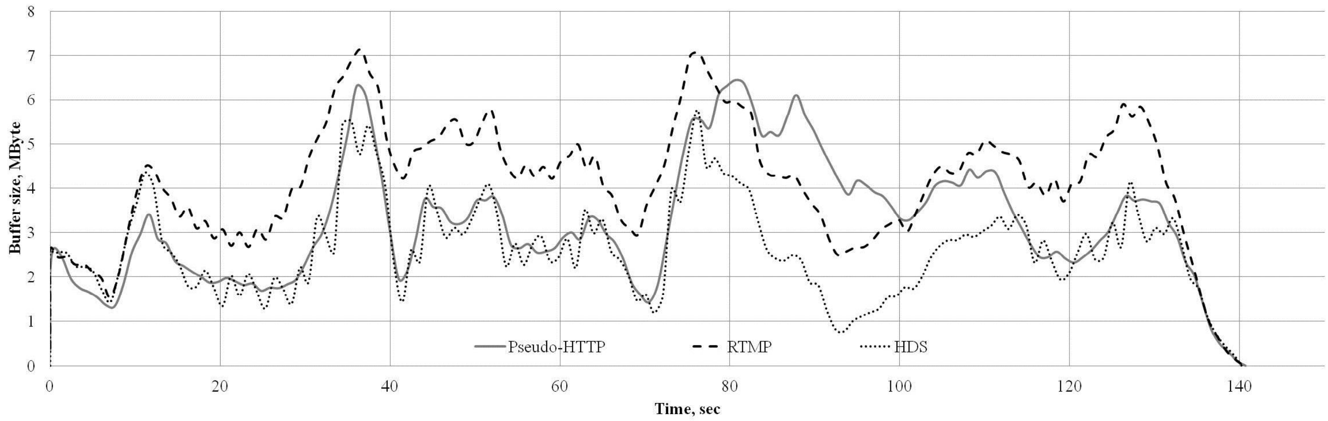

Fig. 3. The buffer size in bytes during video clip playback (packet loss – 0.5 %, delay – 5 ms

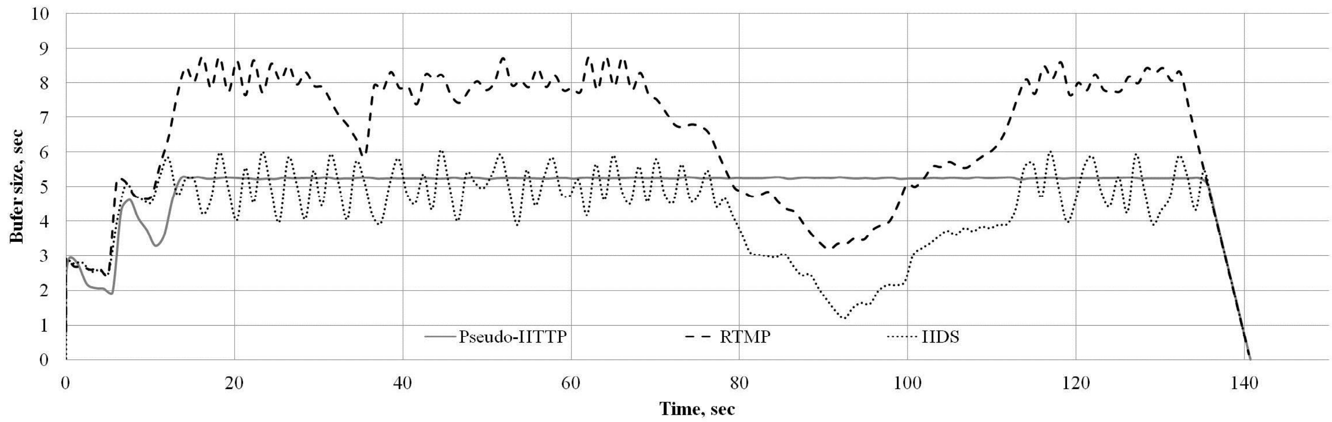

Fig. 4. The buffer size in bytes during video clip playback (packet loss – 0.5 %, delay – 5 ms)

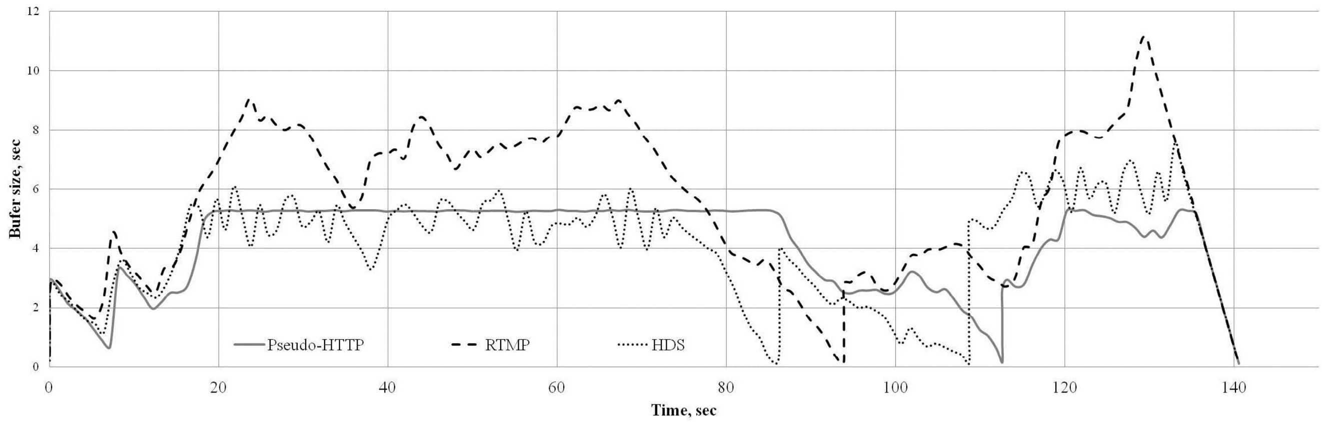

Fig. 5. The buffer size in bytes during video clip playback (packet loss – 0.5 %, delay – 9 ms)

The buffer size for RTMP is much higher than for Pseudo-HTTP and HDS (both in bytes and in seconds) during all period of playback. Never the less also for RTMP the buffer becomes empty at a time stamp of 90 s. According to description of RTMP [12] server can decrease the quality of video during playback that causes a great difference in user perception of video files. For Pseudo-HTTP and HDS streaming technologies the re-buffering reflects in user perception as a freeze picture. The duration of freeze picture is equal to the time needed to fill the buffer for minimum size at which playback restarts. No other effects as degradation of the picture occur because retransmission is used to recover packet loss. In RTMP it is different because before the buffer becomes empty the decoder decreases the quality of video by suspending the playback but audio track is still running. For the user it looks like that instead of the movie the slide show starts. According [5, 6] the users react on this slide show as rather bothering factor disturbing their more than freeze pictures.

IV. RESEARCH OF THE THRESHOLDS OF QOS PARAMETERS

The main goal of the current research is obtaining of the thresholds of QoS parameters (packet loss, delay, throughput) which excess would lead to the re-buffering. At the first stage was studied the influence of each factor separately. At the next stage the influence of the combination of both factors were studied. As it was shown in [6] the influence of separate factors on quality of experience QoE measured by MOS starts at rather high levels of packet loss ratio 2-3% and IP packet delay more than 50 ms. Meanwhile the combination of these factors decrease the thresholds significantly.

In ITU-T Recommendation Y.1541 [15] defined 6 classes of IP networks with specified objectives for each class. The video streaming with OTT technology should be classified as IP network of Class 4. According this Recommendation the network performance parameters should meet the following objectives for the IP network Class 4: IPLR = 0,1 %, IPER =0.1 %, IPTD = 1 s and IPDV is not limited.

We also can take into consideration the QoS class identifiers QCI 7 [15] for Video streaming with objectives for delay of 100 ms and packet loss of 0.1%.

Taking into account both considerations we have limited the range of IPLR from 0.5% to 3% and range of IPTD from 5 ms to 70 ms.

A. Influence of IPLR on the start of re-buffering

The first stage of experiment was based on using the platform with the throughput limited to 10 Mbit/s. During the experiment the minimum size of the buffer was fixed at which re-buffering starts for each of 3 streaming technologies.

The results of experiment are shown on Fig.6. As it can be seen from the Fig.6 for RTMP re-buffering starts from the packet loss of 2.8%, for HDS – 2.4% and for Pseudo-HTTP it starts from 3.1%. It means that the packet loss can cause the re-buffering only at the periods of network congestion when the value of IPLR exceeds objectives 10 times.

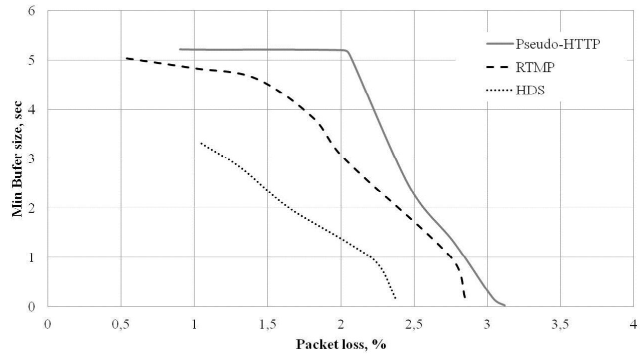

Fig. 6. Influence of the packet loss on the start of re-buffering

B. Influence of the packet delay on the start of re-buffering

The throughput of the experimental platform was limited to 10 Mbit/s. During experiment only the packet delay was changed without losses of the packets.

The results of the experiment are shown on Fig.7. As it can be seen from Fig.7 the packet delay starts to cause the re-buffering from 34 ms for RTMP, 40 ms for HDS and from 62 ms for Pseudo-HTTP. If we compare these values with the objective of 1 s valid for Class 4 networks (also for QCI7) than we can see that real values of the packet delays are much lower (in real networks). It means that the influence of the delay is greater than packet loss. Never the less the influence of only 1 factor is not too much and we should proceed with research of the influence of the combination of both factors.

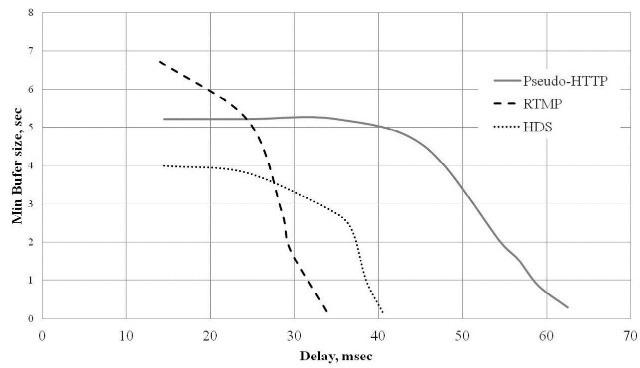

Fig. 7. Influence of packet delay on the start of re-buffering

C. Influence of combination of 2 factors on the start of re-buffering

In the real IP networks a lot of factors can influence on the quality of video streaming: changing of throughput, packet loss and packet delay. From the previous results [6] we have found that 10 Mbit/s is real threshold for bitrate for the picture quality of 720 pixels. It was the main reason to fix this value for our experiments. We also limit the values of packet loss to 2 values 0.5% and 1%. The range of delays were extended from 2 ms to 20 ms.

As it can be seen from Fig.8 if the packet loss is equal to 0.5% than the re-buffering starts from 12 ms for HDS, 14 ms for RTMP and from 15 ms for Pseudo-HTTP.

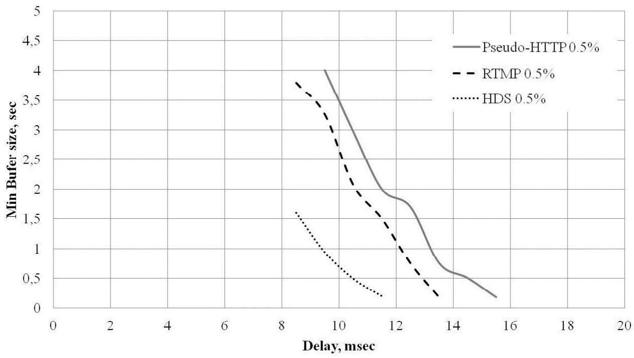

Fig. 8 Influence of the delay on the start of re-buffering (packet loss – 0.5%)

The influence of the delay on the start of re-buffering with packet loss of 1% is shown on Fig.9. As it is clearly seen from the Fig.9 the re-buffering starts at much lower values compare with the results on Fig.8. So for HDS this value is about 9 ms, for RTMP – 10 ms, for Pseudo-HTTP – 11 ms. The conclusion may be done that the combination of packet loss and packet delay significantly influence on the start of the re-buffering and cause the lowering of QoE.

We also can compare 3 streaming technologies using the results of our experiment Table II. The worst results show HDS streaming. In the same conditions of the network performance the re-buffering starts at lower values of packet loss and delays. The best results show Pseudo-HTTP technology – it is really resistant to influence of packet loss and delays.

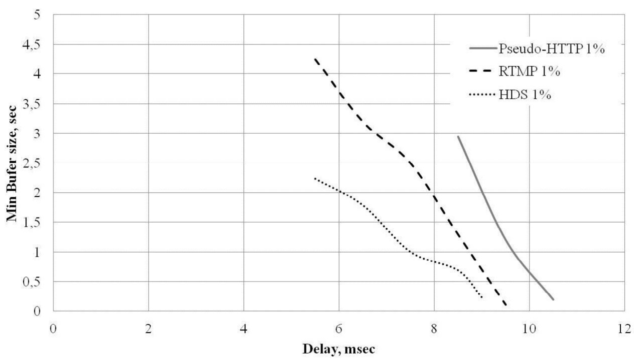

Fig. 9 Influence of the delay on the start of re-buffering (packet loss – 1 %)

TABLE II. COMPARE STREAMING TECHNOLOGIES (INFLUENCE OF A PARAMETER ON THE START OF RE-BUFFERING)

	Streaming Technologies		
	HDS	RTMP	HTTP
Packet Loss, %	2,4	2,8	3,1
Delay, ms	40	34	62
Delay, ms (packet loss – 0.5%)	12	14	15
Delay, ms (packet loss – 1%)	9	10	11

The interpretation of the results for RTMP is rather difficult because the degradation of the video quality carried out in suspension of images (we called it slide show) and then the re-buffering starts. If we take into account only re-bufferings then RTMP shows mean results just between HDS and Pseudo-HTTPP but we also had take in consideration the slide show as it influence on customer perception. For objective definition of this influence we could use the comparison of changing the bitrate of video file playback with different QoS parameters. For the demonstration of this effect we choose 2 samples: first - called RTMP-9 with 9 ms delay and 0.5% packet loss and second – called RTMP-5 with 5 ms delay and 0.5% packet loss. The results of playback of the same video clip are shown on Fig.10.

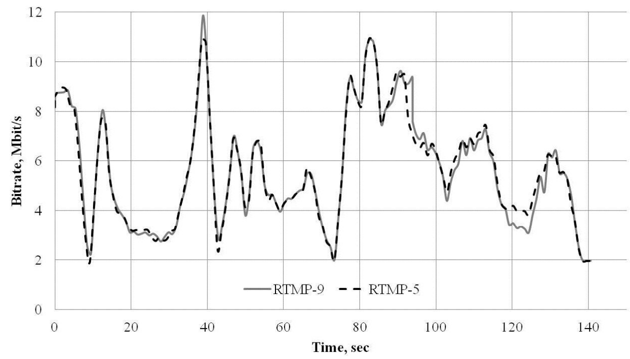

Fig. 10 The bitrate changing during the playback of video clip

If we compare two plots then it can be clearly seen that at a timestamp between 115 s and 125 s the bitrate decrease significantly for RTMP-9 and at this period the slide show was fixed by the experts. It means that during slide show the bitrate is go down but it prevents buffer from becoming empty. For further research we should take into account both factors of the degradation of video quality for RTMP.

V. CONCLUSION

As OTT becomes very popular in majority of national and international providers the quality of experience for streaming technologies becomes more and more important. For IPTV technology ITU-T Recommendation G.1080 [16] covers most important aspects of supporting QoE. For OTT there are no such regulatory documents. Meanwhile the mechanisms of supporting QoE for streaming over OTT become vital for video service providers. A lot of papers were dedicated to this subject [3, 7-10]. For practical needs providers need recommendation for monitoring and control specific QoS parameters to exclude the influence of network performance parameters on the quality of user perception. In this paper the research was performed for finding the thresholds for QoS parameters radically influencing on QoE. From the previous results the re-buffering was used as a main factor influencing on subscriber's perception. The evaluation of the influence of 2 parameters: packet loss and delay were performed for 3 streaming technologies Pseudo-HTTP, HDS and RTMP.

The research of the influence of separate packet loss and packet delay shows that the thresholds from which the re-buffering starts are rather high compare with objectives for IP networks Class 4 [15]. In real networks both factors are exist simultaneously. In case of combination of mentioned parameters the thresholds of delays was lower in 4-5 times compare to the thresholds for separate factors and are in the range of 12 – 15 ms for the packet loss of 0.5% and in the range of 9 – 11 ms for packet loss of 1%. These values are very close to the measured QoS parameters of the real IP networks.

The comparison of streaming technologies was done based on the influence of combination of both factors. Best results in resistance to packet loss and delays show Pseudo-HTTP, HDS was the worst. RTMP shows intermediate results but RTMP has specific functionality of the decoder at low bitrates which leads to the slide show and this mode also influence on user's perception [12].

VI. Future Work

Further study should be devoted for clarification of the thresholds in larger range of basic parameters. A lot of combinations of packet loss and delays should be investigated for researching of the range from which re-buffering starts. The influence of the content of the video clip should be analysed. Also the number of investigated streaming technologies should be extended according to new implementation.

Current research was performed for the resolution of 720 pixels. Nowadays providers deliver video content of the resolution of 1080 and 1440 pixels and start 4K with 2160 pixels. This is also a good motivation to continue our research.

The main goal of our future work is to create a new method for evaluating the thresholds for different combinations of QoS parameters for different streaming technologies.

Acknowledgment

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