# TCP is Bottleneck of Video Streaming via OTT

Alexey Nesterkin
IT HUB Group
Moscow, Russia
nnaaaa@yandex.ru, nesterkin@ithub.ru

Vladimir Deart
Moscow Technical University of Communications and Informatics
Moscow, Russia
vdeart@mail.ru

***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 to keep the Quality of Experience (QoE) on a suitable level. In the previous works of research, the influence of Quality of Service (QoS) parameters on re-buffering effect were done. Here we continue this research for video with higher resolution 1080p, 2K or also 4K. The primary purpose of this research is to investigate how delay and packet loss influence on re-buffering effect in HLS, RTMP and MPEG-DASH streaming protocols delivering high resolution video.**

## 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.

Based on the forecast analysis and trends of OTT market, it is possible to see increasing volume of OTT video traffic and using smartphones. From the analysis of perspective in Internet till 2022 made by Cisco Systems [1] we can see that people will less use Internet from PCs (personal computers). And PCs activity will decrease from 49% till 19% (Fig. 1). At the same time, smartphones become more and more aggressively used Internet devices, and they will generate 50% of total traffic in Internet by 2022 (Fig. 1).

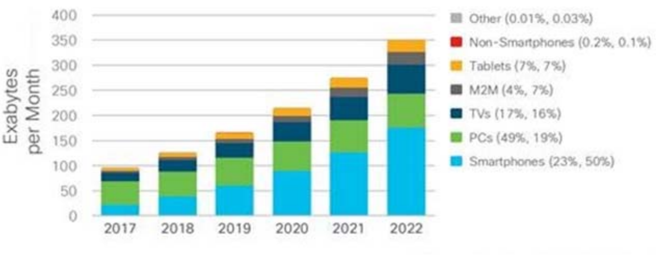


Fig. 1. Most active devices in Internet

Now the consumers primary use smartphone and tablets to view video [2]. As a result, the volume of IP video traffic will be increased from year to year. By 2022 IP video traffic will account for 82 percent of traffic [1] (Fig. 2).

The competition for the clients requires from OTT streaming video providers to offer new and new services. Most of them propose video with 1280×720 HD (High-Definition) or 1920×1080 FHD (Full High-Definition) resolution, but it is not enough now. Consumers want a better quality video with 3840×2160 UHD (Ultra-High-Definition) 4K or even 7680×4320 UHD 8K resolution. This trend showed in research of TV Technology [3].

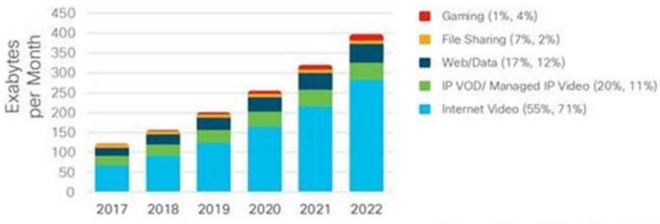


Fig. 2. Most active Internet application

This will increase Internet traffic, because videos with UHD resolution require more than twice bandwidth compare with FHD and more than ten times compare with SD (Standard-Definition).

From the analysis made by Cisco Systems [1], consumers will give preference HD and 4K UHD in IP VoD (video on demand) services (Fig. 3).

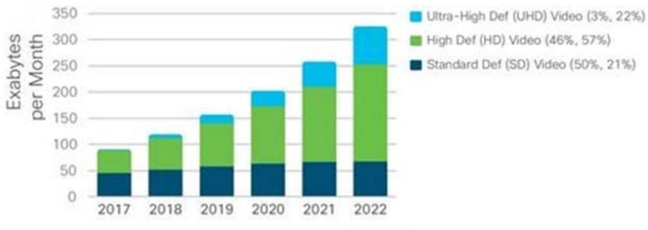


Fig. 3. Global IP video traffic

These trends in Internet traffic will change its pattern. The role of CDNs (Content Delivery Networks) will increase significantly. From the analysis of perspective in Internet till 2022 made by Cisco Systems [1], these services will carry 72 percent of total Internet traffic by 2022 (Fig. 4). Thus OTT service will be closer to the end user and will decrease delivery latency for OTT video streaming.

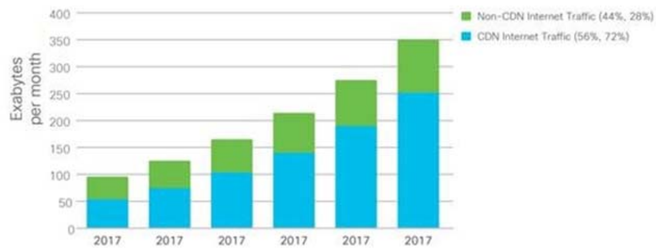


Fig. 4. The role of CDNs in content delivery

ISSN 2305-7254

Some modern users' devices such as smartphones, tablets, and TVs support 7680×4320 8K UHD resolution. 5G network is the forthcoming giving the ability to have high speed Internet connection to all consumers' devices. This new challenge creates the demand on new services from OTT streaming service providers [3] able to supply users with video with maximum resolution of their devices 3840×2160 4K or also 7680×4320 8K.

Our previous research of delivering streaming video with 1280x720 resolution and mean value of bitrate is equal 5.68 Mbit/s showed that combination of delay and loss give significant degradation of QoE [4]. Re-buffering effect appear with packet loss 0.5% and delay 12-15ms for different streaming protocols. Most clients' devices use wireless technologies to Internet access as last mile technology. These technologies have comparable loss rate and delay with received results.

This research is devoted to the estimation of packet loss and delay thresholds on re-buffering effect for HLS (HTTP Live Streaming), RTMP (Real-Time Messaging Protocol), and MPEG-DASH (MPEG Dynamic Adaptive Streaming over HTTP) streaming technologies for 1920*1080 HD video. The assessment is made for the cause so significant influence of packet loss and delay on re-buffering effect. The proposal is suggested for acceptable delay and packet loss for streaming video with 7680×4320 8K UHD resolution without re-buffering effect.

The paper is structured as following. In section II is reviewed current research of this problem. In section III the experimental platform is described and characteristics of used video clips are presented. In section IV the results of research are presented. In section V the influence of packet loss and packet delays on the start of re-buffering for different video resolution are analyzed and discussed. In section VI conclusions are made. Section VII is devoted to future work.

## II. REVIEW OF OTT TECHNOLOGIES RESEARCHES

### *A. Consumers satisfaction of OTT technologies*

At the beginning of implementation OTT technologies, consumers were very tolerant to re-buffering and delayed start. They have new capability – "I can watch, what I want and where I want". Extension of the OTT video service lead to rising consumers' request. Now they want to watch video content anywhere without any degradation.

There is some example of consumers' satisfaction research from different companies:

- "Viewers are increasingly less forgiving of re-buffering. The average number of times a viewer will let a video re-buffer before they stop watching has dropped from 2.7 times in 2016 to 2.2 times in 2018" [2];
- "OTT viewers now care more about QoE than choice of content" [5];
- "A total of 81% of consumers' experience buffering (a lag that disrupts playback while content is in-progress) or delayed video start (when a video takes significant time to initially load), according to this year's findings" [6];
- "Video buffering causes a decrease in positive emotions (happiness down 14%), a 9% increase in disgust, a 7% increase in sadness, and an 8% decrease in focus. Taken together, these represent an overall 16% increase in negative emotions" [7];
- "Today's promiscuous viewers will not tolerate any buffering. Intermittent outages and delays are out of the question. Just a few seconds is all it takes for them to switch off and move over to a competitor's service" [3].

Based on these research of consumers' satisfaction, it is possible to highlight three common problems, which emphasize clients:

- Re-buffering effect;
- Delayed start;
- Reducing video bitrate.

We also are continuing our study of consumers' QoE. From year to year customers become less tolerant to re-buffering effect and changing video quality. They don't want to have any distortion during watching OTT video stream. Based on this trend we change target of our research to finding threshold of QoS network parameters after the re-buffering effect could be seen for different video resolution and streaming technologies.

### *B. Results of our previous research*

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 works were 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 [8, 9, 10] show that the influence of IPLR (IP Packet Loss Ratio) and IPTD (IP Packet Transfer Delay) 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 [10].

Next 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 QoS parameters: packet loss and delay, have significant influence on re-buffering effect. In this work were compared three streaming technologies HDS (Adobe Flash HTTP Dynamic Streaming), RTMP and Pseudo-HTTP streaming. Video clip was chosen with resolution of 1280x720 pixels, VBR (Variable Bit Rate) (mean value of bitrate is equal 5.68 Mbit/s) and quickly changing scenes with dynamic objects, audio 192 Kbit/s. Results of it [4] presented in Table I. 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.

TABLE I. RESULTS OF PREVIOUS RESEARCH

| | Streaming Technologies | | |
|---|---|---|---|
| | HDS | RTMP | Pseudo-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 research of the influence of separate packet loss and packet delay showed that the thresholds from which the re-buffering starts are rather high compare with objectives for IP networks Class 4 [11]. In real networks both factors are existing simultaneously. In case of combination of mentioned parameters, the thresholds of delays were 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 both factors combination. 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].

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 (Internet Protocol Television) technology ITU-T Recommendation G.1080 [13] 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. 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's perception.

### C. Other authors works

Works of other authors are oriented to the design of bitrate adaptation algorithms for adaptive bitrate streaming (ABR) schema [14, 15, 16]. Many implementations of ABR streaming exist, including HLS [17], RTMP [12] and MPEG-DASH [18]. Each has its own proprietary implementation and slight modifications to the basic ABR technique. Most of research targeted on open-source standard for ABR streaming – MPEG-DASH. They usually assumed lower video bitrate than it usually present. For example, in [14] assumed bitrate 800 kbps for 720×576 SD and 2 Mbps for 1280×720 HD video resolution. This assumption don't allowable for our work. Netflix [19] gives these recommendations:

- 3.0 Mbps for SD video quality;
- 5.0 Mbps for HD video quality.

They focused only on ABR algorithm and did not consider current market demand and trends in consumers' request. Users want to watch video with FHD 1920×1080 resolution and without any video quality degradations. ABR technology mitigate re-buffering effect using video quality degradation, but this would be unacceptable for users.

## III. EXPERIMENTAL PLATFORM

In this section we describe our experimental platform and assumption for network parameters and video used in experiments. The experimental platform (Fig. 5) consists of three virtual machines:

- Ubuntu Server with installed ngnix web server [20, 21];
- WANem [22] – network emulator, used to set network QoS parameters packet loss and delay;
- Ubuntu Client with web browser.

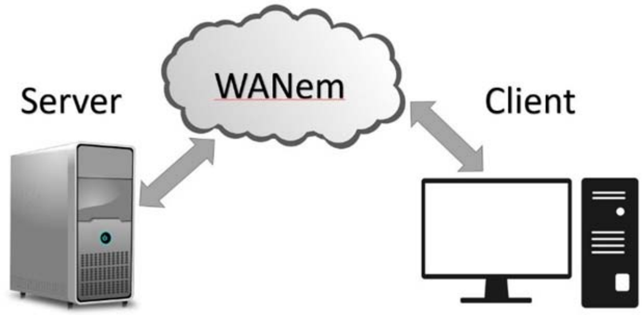


Fig. 5. Experimental Platform

Full description of experimental platform presents in Table II.

TABLE II. DETAILED INFORMATION ABOUT PLATFORM

| Name | Description |
|---|---|
| PC Hardware Platform | CPU - Intel Core i7 – 4770<br>RAM – 16 Gbyte<br>Video card - GeForce 210 Gainward PCI-E |
| OS Server | Ubuntu 16.04.4 LTS (64 bit) |
| Virtual Box | Oracle cross-platform virtualization application |
| WANem | Wide Area Network Emulator version 2.3 |
| Streaming Server | Ngnix web server with additional module for RTMP |
| OS Client | Ubuntu 16.04.4 LTS (64 bit) |
| Browser | Opera |
| Web Interface | Custom Interface |
| Video Player | JWPLAYER |
| Streaming technologies | HLS, RTMP, MPEG-DASH |

The following limitations were used in experiment platform based on the results of our previous researches [4] and new trends:

- The bandwidth of the path was not limited;
- We have changed streaming technologies to HLS, RTMP, MPEG-DASH, so they are more popular now than others [23];
- Video clip was chosen with resolution of 1920x1080 pixels, as the next step in our researchers [4], detail parameters are present in Table III.

## IV. Results of Research

### A. General research results

The study was conducted for the single factor effect of packet loss and delays on the start of re-buffering using the technique proven in previous research [4]. New streaming technologies HLS and MPEG-DASH were added to the experimental platform reflecting current trends in streaming video [23]. In the results of the study we identify the beginning of re-buffering in streaming protocols RTMP, HLS and MPEG-DASH, which are presented in the Table IV.

TABLE III. Detailed parameters of the video file

| Name | Description |
|---|---|
| Name | Big Buck Bunny |
| Resolution | 1920*1080 FHD |
| Video Codec | H.264 |
| Frame per second | 24 fps |
| Video bitrate | 9283 kbps |
| Audio Codec | MPEG4-AAC |
| Audio bitrate | 437 kbps |

TABLE IV. Streaming technologies comparison (influence of a single parameter on the start of re-buffering)

| | Streaming Technologies | | | ITU-T Y.1541 |
|---|---|---|---|---|
| | RTMP | HLS | MPEG-DASH | |
| Packet Loss, % | 4.2 | 4.7 | 5.3 | 0.1 |
| Delay, ms | 287 | 180 | 242 | 1000 |

We have analyzed the obtained results with the current requirements to ensure QoS for different traffic classes in the Recommendations of ITU-T Y.1541 (12/2011) [11]. Streaming video corresponds to class 4, for this class the maximum total allowable delay is 1000 ms and packet loss must be less than 0.1%. It means that delays are more valuable factor for re-buffering than packet loss. Moreover, exceeding the limits of delays indicated in Table IV is guaranteed to result in re-buffering also when the packet loss is zero. In the Recommendations rather big delays are allowed but in reality the delays of 180 – 287 ms cause the start of re-buffering. The main reason for this effect is that in Recommendations for streaming video transmission was supposed to use the UDP/IP stack and in the OTT technology the stack TCP/IP is used. While using TCP the retransmissions can occur which increases the delay of a single packet significantly causing empting of the buffer and starting re-buffering. It means that with mean values of the delay 200 – 300 ms the start of re-buffering can occur.

In modern computer networks the packet loss is less than 0.1% and delays are less than 100 ms meanwhile when both factors influence on video streaming transmission the thresholds for the start of re-buffering decrease significantly.

The research results of 2-factors (IPTD and IPLR) influence on the start of re-buffering are presented in Table V. During the research we fixed the delay and change packet loss until re-buffering starts.

TABLE V. Streaming technologies comparison (influence of the parameters combination on the start of re-buffering)

| Delay, ms | Packet loss, % | | |
|---|---|---|---|
| | HDS | RTMP | Pseudo-HTTP |
| 4 | 0.37 | 0.65 | 0.82 |
| 5 | 0.16 | 0.25 | 0.35 |
| 7 | 0.14 | 0.21 | 0.28 |
| 9 | 0.08 | 0.1 | 0.17 |

The values obtained are in the range of QoS parameters of real modern data networks that means QoS parameters should be monitored and controlled to enable streaming video transmission.

As it can be seen from the Fig.6 the best protocol for streaming video transmission is MPEG-DASH, second is HLS and the worst is RTMP. The slight difference between the streaming protocols can be explained by the peculiarities of the implementation of the algorithm of a specific protocol and the difference in the size of headers and service messages.

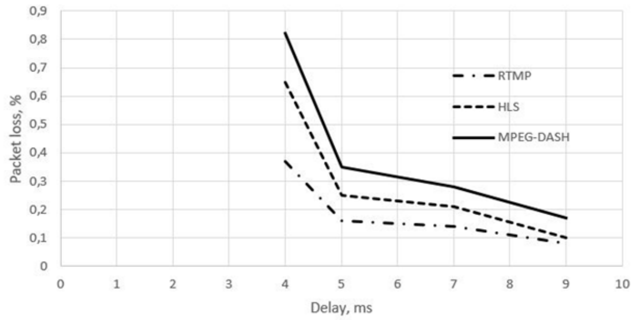


Fig. 6. The thresholds of re-buffering start for HLS, RTMP and MPEG-DASH

The other meaningful factor that influence on the start of re-buffering is the effective bandwidth of the TCP connection. In Annex 9 of ITU-T Y.1541 (12/2011) [11], the calculation of the effective TCP session rate is discussed in detail. We are using the formula for calculating the effective throughput of a TCP session presented in the recommendation in Section V of current work. Using it, we obtain theoretical maximum of TCP throughput from packet loss and delay. This will be upper limit for video AVG (average) bitrate.

### B. Re-buffering time frame

During our research we found out, that re-buffering effects occur only on specific time frame of video playback. We took three different with 1920x1080 FHD resolution to analyze their bitrate using Bitrate Viewer [24] software:

- Video1 – Big Buck Bunny (Fig. 7), from current research;
- Video2 – Sport (Fig. 8) from [4, 8-10];
- Video3 – Nature (Fig. 9) from [4, 8-10].

Result of video bitrate showed, that each tested video has its specific bitrate, but they all have variable bitrate (VBR). VBR is used by default for most video in Internet, because it gives much greater quality at the same AVG bitrates. It's an encoding method that enables the bitrate of a video to dynamically increase or decrease based on the complexity of the fragments. For more complex fragments bitrate will be hire and for less complex fragment bitrate will be lower. Constant bitrate (CBR) opposite saves a set bitrate over the entire video clip. Thus the image quality has degraded for complex video segments.

Analysis of these video present in Table VI. All of them have nearly the same AVG bitrate, but extremely different MAX and MIN value. Based on these analysis of video bitrate, we can predict at which time frames of video playback consumers could watch re-buffering effect more frequently than others. For Video1 re-buffering effect can occurred at the beginning, in the middle and at the end of the video. For Video2 – at the beginning and after the middle of the video. And for Video3 it could be at any time, because compensation buffer make chance of re-buffering the same.

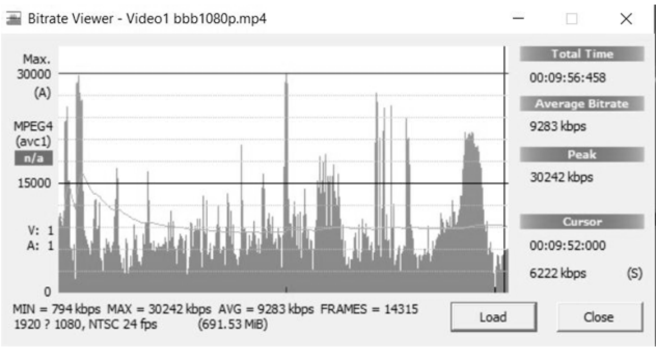


Fig. 7. Bitrate of Video1

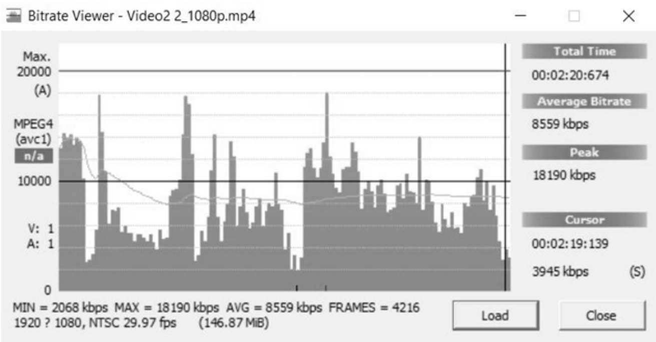


Fig. 8. Bitrate of Video2

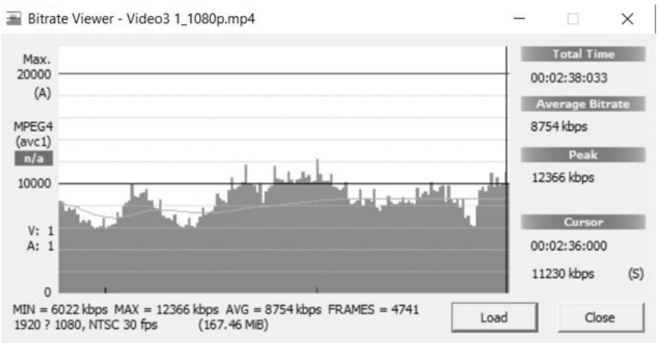


Fig. 9. Bitrate of Video3

## V. RESEARCH OF THE QOS PARAMETERS THRESHOLDS

The analysis of the results showed that TCP throughput have significant influence on the re-buffering effect. Using formula from [11] for calculating the effective speed of TCP session we can estimate opportunity of transfer streaming video over the network, with defined QoS parameters, without re-buffering effect.

TABLE VI. PARAMETERS OF VIDEO BITRATE FOR VIDEO 1/2/3

| Parameter | Video1 | Video2 | Video3 |
| --- | --- | --- | --- |
| AVG bitrate, kbps | 9283 | 8559 | 8754 |
| Min bitrate, kbps | 794 | 2068 | 6022 |
| Max bitrate, kbps | 30242 | 18190 | 12366 |

It is not possible to compare AVG of streaming video and throughput of TCP session straightforward. It is necessary to consider additional headers and service packets of streaming technologies. This will lead to even more significant reduction throughput for video transmission.

For creation of the graphic presentation TCP throughput as the function from packet loss and delay, we will use the formula of the TCP throughput simple model from [11].

$$B(p) \approx \min\left(\frac{W_{max}}{RTT}, \frac{1}{RTT\times\sqrt{\frac{2bp}{3}}+T_0 min\left(1,3\times\sqrt{\frac{3bp}{8}}\right)\times p\times(1+32\times p^2)}\right) \quad (1)$$

where Wmax is maximum window buffer size of receiver (packets), RTT – Round Trip Time (sec), b – number of packets that are acknowledged by a received ACK, p – probability that a packet is lost and T0 – time-out for re-transmitting an unacknowledged (lost) packet (sec).

We can use next simplification for formula 1, because packet loss is less than 0.1 % for modern network:

$$B(p) \approx \frac{1}{RTT\times\sqrt{\frac{2bp}{3}}+T_0 min\left(1,3\times\sqrt{\frac{3bp}{8}}\right)\times p} \quad (2)$$

From formula (2) we can receive formula for TCP throughput:

$$T = B(p)\times MSS = \frac{MSS}{RTT\times\sqrt{\frac{2bp}{3}}+T_0 min\left(1,3\times\sqrt{\frac{3bp}{8}}\right)\times p} \quad (3)$$

where MSS is max segment size.

In Table VII present recommendation from Google [25] to upload video on YouTube service. If we compare recommendation in Table VII and 3D chart of maximum TCP throughput on Fig. 10. we can to see that it is possible to watch video with resolution 4K for current standard network QoS parameters, if we will use stable cable connection with bandwidth more than 30 Mbps. But for 8K video we will have re-buffering effect with high probability. In modern telecommunications networks it is very difficult to ensure delay less 15 ms and packet loss less 0,03% continuously. This QoS parameters may be guaranteed only by CDN provider which have Servers in each city. This restrict possibility to expand 4k and more 8K service from OTT providers.

To watch 8K UHD video from OTT service providers, client must have stable Internet connection with minimum 50 Mbps speed without additional delay and packet loss on client side.

TABLE VII. VIDEO BITRATES FOR DIFFERENT RESOLUTION

| Name | Resolution | Bitrate, Mbps |
|---|---|---|
| SD | 720×576 | 2.5 |
| HD | 1280 × 720 | 6.5 |
| FHD | 1920 × 1080 | 10 |
| 4K UHD | 3840 × 2160 | 25 |
| 8K UHD | 7680 × 4320 | 40 |

## VI. 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.

In this work we confirm our previous results, that combination of delay and packet loss significant influence on re-buffering effect for used OTT technologies HLS, RTMP, MPEG-DASH.

The primary cause of re-buffering is that the effective throughput of TCP session depends on delay and packet loss. The calculated effective TCP throughput for given delay and packet loss allows to estimate possibility to deliver video stream with definite bitrate. Thus TCP session throughput is bottleneck of UHD video streaming via OTT.

Our research showed that using only one TCP session for transport have significantly restriction. Any protocol will be limited by TCP session throughput. This important limitation has got to take into consideration for the design of networking protocols and new NGN and FGN networks.

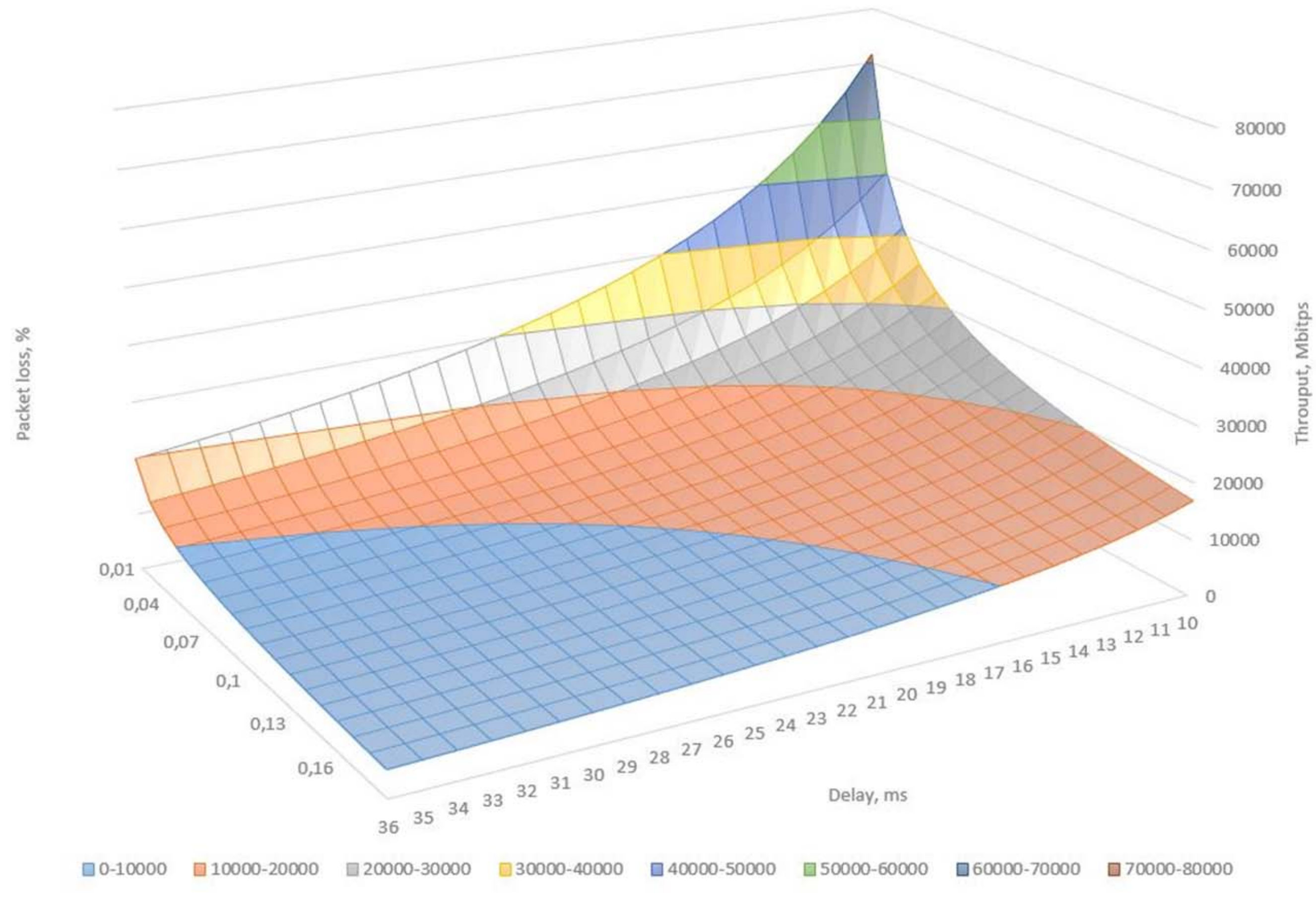


Fig. 10. Maximum TCP throughput from packet loss and delay

## VII. FUTURE WORK

Further study should be devoted for research how different TCP algorithms influence on re-buffering effect. A lot of combinations of algorithms should be investigated on client and server sides.

We are planning to extend experimental platform for research acceptable QoS parameters for 4K and 8K UHD video.

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, and to find correlation between AVG video bitrate and TCP throughput.

Also we need to research others alternatives for OTT video transport. QUIC (Quick UDP Internet Connections) is one of them.

## REFERENCES

[1] Cisco Systems Inc. official website, Cisco Visual Networking Index: Forecast and Trends, 2017–2022, Web: https://www.cisco.com/c/en/us/solutions/collateral/service-provider/visual-networking-index-vni/white-paper-c11-741490.html.

[2] Limelight Networks official website, The State of Online Video, Web: https://www.limelight.com/resources/white-paper/state-of-online-video-2017.
[3] TV Technology official website, The Future Of OTT Streaming, Web: https://www.tvtechnology.com/opinions/the-future-of-ott-streaming.
[4] V. U. Deart and A. A. Nesterkin, "A Study of Influencing Factors on the Quality of Experience Perception in Video Streaming", Proceedings of the 18th Conference of Open Innovations Association FRUCT, Apr. 2016, pp. 437-442.
[5] B. Flynn, Videonet official website, White Papers, OTT viewers now care more about QoE than choice of content, says Conviva. June 3, 2015, Web: http://www.v-net.tv/ott-viewers-now-care-more-about-qoe-than-choice-of-content-says-conviva.
[6] IBM official website, State of Streaming report, Web: http://info.video.ibm.com/State-of-Streaming-2017-Report.html.
[7] Akamai Technologies official website, Bit rate and business model https://www.akamai.com/us/en/multimedia/documents/white-paper/bit-rate-and-business-model.pdf.
[8] V. Deart, I. Kozhuhov "Research of the QoS parameters defining QoE for transmitting streaming video over Internet", *T-Comm*, no. 7, 2013, pp. 28-31.
[9] V. Deart, I. Kozhuhov, A.V. Pilugin "Experimental Platform for QoE research of streaming video services", *T-Comm*, no. 7, 2013, pp. 32-35.
[10] V. Deart, I. Kozhuhov, Y. Milicin, A. Pilugin "Analysis of experimental data on subjective assessment of QoE when transmitting video content using OTT", *T-Comm*, no.8, 2014, pp. 37-42.
[11] ITU-T official website, Recommendation ITU-T Y.1541 (12/2011) Network performance objectives for IP-based services, Web: http://www.itu.int/rec/T-REC-Y.1541.
[12] Adobe official site, Real-Time Messaging Protocol (RTMP) specification, Web: http://www.adobe.com/devnet/rtmp.html.
[13] ITU-T official website, Recommendation ITU-T G.1080 (12/2008) Quality of experience requirements for IPTV, Web: http://www.itu.int/rec/T-REC-G.1080/en.
[14] IEEE official website, BOLA: Near-optimal bitrate adaptation for online videos, Web: https://ieeexplore.ieee.org/document/7524428.
[15] IEEE official website, Study of user QoE improvement for dynamic adaptive streaming over HTTP (MPEG-DASH), Web: https://ieeexplore.ieee.org/document/7876191.
[16] A. Zabrovskiy, E. Petrov, E. Kuzmin and C. Timmerer "Automated Performance Evaluation of Adaptive HTML5 Player Deployments", Proceedings of the 21st Conference of Open Innovations Association FRUCT, Nov. 2017, pp. 532-536.
[17] Apple official website, HTTP Live Streaming, Web: https://developer.apple.com/streaming.
[18] Moving Picture Experts Group official website, MPEG-DASH, Web: https://mpeg.chiariglione.org/standards/mpeg-dash.
[19] Netflix official website, Internet Connection Speed Recommendations, Web: https://help.netflix.com/en/node/306.
[20] Nginx official website, Nginx HTTP server features, Web: https://nginx.org.
[21] GitHub official website, NGINX-based Media Streaming Server, Web: github.com/arut/nginx-rtmp-module.
[22] SourceForge official website, Wide Area Network Emulator, Web: https://sourceforge.net/projects/wanem.
[23] Encoding.com official website, 2018 Global Media Formats Report, Web: https://www.encoding.com/resources/#white-papers
[24] VideoHelp official website, Bitrate Viewer 2.3, Web: https://www.videohelp.com/software/Bitrate-Viewer-2.
[25] Google official website, Live encoder settings, bitrates, and resolutions, Web: https://support.google.com/youtube/answer/2853702?hl=en.