Emerging Topics and Questions in Infocommunication Technologies

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Edited by

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**Cambridge
Scholars**
Publishing

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This book first published 2020

Cambridge Scholars Publishing

Lady Stephenson Library, Newcastle upon Tyne, NE6 2PA, UK

British Library Cataloguing in Publication Data
A catalogue record for this book is available from the British Library

ISBN (10): 1-5275-5413-9
ISBN (13): 978-1-5275-5413-9

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PREFACE

This collection consists of selected papers from the IEEE FRUCT 2019 Conferences. The conference is a regular and significant open event organized by the Open Innovations Association FRUCT, which has a long and positive history of pioneering and assisting long-term cross-border pan-European academic and commercial cooperation in the field of Information and Communication Technology (ICT). In total, the conference received over 300 submissions; of these, less than 130 were accepted for publication as full papers (with an acceptance rate less than 40%). For this collection of essays, we selected 14 papers. These are extended versions of the best papers from the conference.

The main motivation for organizing regular FRUCT conferences is the distinctive features of modern IT and ICT industries, which are experiencing a dramatic shortening of the innovation cycle and a reduction of the period before technology becomes commercially viable. This is due to competition between market players pushing all manufacturers to accelerate innovations combined with a speeding up of technological progress caused by the growing expansion of intellectual resources invested in R&D and design activities. This trend is a key challenge for leading educational and research institutions around the globe. In FRUCT we believe that it is strategically important to develop pan-European cooperation and cultural exchange (and also keep Russian scientists in the loop). FRUCT aims to help regional teams in following up and adopting university research and education and aligning it to real industrial challenges. The primary mission of the conference is to strengthen cooperation between the Russian and Finnish academic communities, increase the visibility of research teams, and build up direct personal contacts between academic and industrial experts.

FRUCT conference activities are based around the idea that continuous development of the strategic partnership between industrial and academic research is a key factor in the success of the modern innovation ecosystem. There are a few success stories of such strategic partnership frameworks functioning in different parts of the world. These programs bring significant benefits to the parties involved and fuel their R&D. As we know, the fundamental science work undertaken by universities and other academic organizations should not be directly attached to existing industries, but industrial research can benefit greatly by early access to results and information about important trends and weaker pathways. At the same time, many universities are also active in applied research, but to be efficient they need timely feedback from industry. Another key driver for ensuring stronger connections between academia and industry is the shortening of the innovation cycle. An interesting new trend for addressing this need is the building of open innovation frameworks targeting the development of strategic partnerships between industrial and academic research. Such framework programs help potential research partners find each other and jointly incubate new competencies.

The FRUCT Association was established at the beginning of 2007, when the need for such a community was recognized by a group of enthusiasts and supported by Nokia's university cooperation program in Russia, Saint-Petersburg State University of Aerospace Instrumentation, and the University of Turku. Over the past few years, more than 30 research groups from top universities in Finland, Russia, Italy, Ukraine, Denmark and India, as well as the R&D institution of the Russian Academy of Science, have joined the FRUCT Association. Nowadays, FRUCT is one of the largest and most significant associations facilitating the development of research and collaboration between universities and ICT companies in the region. The FRUCT Association has good contacts with industry and events are attended by many representatives of industrial companies. FRUCT has a good history of cooperation with such giants as Nokia, Intel, Samsung, Dell EMC, MariaDB, and Microsoft. Formal membership of the FRUCT Association is by invitation, but FRUCT is an open society and we welcome all interested universities and companies to engage in cooperation and join our friendly community.

The main objective of FRUCT is to establish and develop contacts between academic and industrial research teams from different countries, especially in Russia, Finland, India, Italy, Ukraine, and other European countries. The association is developing processes to enable the early involvement of undergraduates and postgraduates in scientific activities, creating joint teams for solving challenging scientific problems with the use of knowledge intensive technologies and increasing the prestige of scientific and research work. FRUCT has developed many processes that support mutually beneficial cooperation and the development of strategic partnerships between academic and industrial research. The processes developed help overcome barriers in the path of innovation and illustrate how business can benefit from greater social responsibility and the support of long-term research and academic cooperation.

The essays in this collection were selected on the basis of reviewer comments and conference participant feedback. The contents has been organized into four chapters that best highlight the most popular conference sections addressing the hottest topics in ICT research and development.

The first chapter is titled Next Generation Networks. Today, most telecommunications companies use NGN technology for providing their subscribers with a Triple Service (Data, Voice, and Video). Providers have received good feedback from users in terms of the Quality of Experience (QoE) of data transmission, voice perception, and video streaming, all depending on the Quality of Service (QoS) parameters of the existing network. As video streaming now uses High Definition and Ultra-High Definition it needs 10 times the bandwidth of Standard Definition. Providing high resolution video with the required QoE for users with expanding bandwidth demand is becoming an urgent task. The global trend of modernization of NGNs involves software defined networks (SDNs), which can resolve the problem of ensuring the required quality for different types of media streaming. But SDNs must ensure an adequate level of reliability, which is no simple task for centrally controlled networks. The ever-increasing number of new applications for NGNs is a real trend and serves as an attractive point for new customers.

The second chapter discusses Innovative Knowledge-based Systems. This chapter focuses on the development problem of innovative knowledge-based systems. Exciting examples of such system classes were presented and discussed at the IEEE FRUCT 2019 Conferences. The problem domain topics include: augmented reality systems for industrial needs; management systems for the Internet of things; information spaces for smart factories; monitoring and analytics for agricultural data; mobile video and interactive applications for everyday use; ontology-oriented software development for smart spaces; and time-series analysis for recommendation systems. The discussed topics meet new research challenges and the authors define relevant problems, suggest possible formalizations and concept models, and illustrate the study with demonstration implementations.

The third chapter present research on Innovative Applications and Services. Innovative applications play a significant role in modern life.

Robotics, value added services, and intelligent mobile applications are now widespread in our daily lives. Nowadays, the functionality of the modern smartphone goes far beyond making calls. Smartphone devices allow the implementation of calculations and proactively provide the user with multi-modal information, as well as connecting him/her to the e-world in any physical location. Modern information and communication technologies provide a strong basis for research and are widely used by the research community. This chapter presents papers related to research methodology development, as well as prototypes for the implementation innovative application development.

The last chapter focuses on Innovations in Healthcare and eHealth. Visual analysis and interpretation still represent the gold standard in clinical practice. Therefore, the success of correct clinical evaluation, depends on an examiner's skills and is prone to subjective variability. In the age of personalized medicine, more objective methods are needed to improve diagnosis, characterize diseases, make prognoses, and monitor therapy. Image analysis and processing constitute essential tools for providing non-invasive or minimally invasive alternatives to improve, quantify, and maximize *in vivo* information obtainable in clinical practice, thus playing a fundamental role in clinical decision-making.

April 2020

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CHAPTER 1

NEXT GENERATION NETWORKS

CHAPTER 1.1

TCP ALTERNATIVES FOR VIDEO STREAMING VIA OTT

ALEXEY NESTERKIN, VLADIMIR DEART

Abstract. Video services using over-the-top (OTT) delivery are becoming extremely popular across the world, but video streaming through an unguaranteed Internet path can cause a lot of problems in terms of maintaining a suitable quality of experience (QoE) level. Previous work has addressed the influence that quality of service (QoS) parameters have on the re-buffering effect. Here we continue this research for video at higher resolutions of 1080p, 2K, or even 4K. The primary purpose of this research is to investigate how delay and packet loss influence the re-buffering effect in the HLS, RTMP, and MPEG-DASH streaming protocols during the delivery of high-resolution video.

Introduction

Over-the-Top (OTT) technology providing video-on-demand services is becoming more and more popular in countries all over the world. The number of users of streaming video services is increasing significantly from year to year. The biggest providers have already implemented this technology in their IP networks to achieve larger profits from greater utilization of the existing network infrastructure. This has all lead to an increase in competition between providers seeking to capture the market.

Forecast analysis of OTT market trends suggest a continuing increase in the volume of OTT video traffic and the use of smartphones. An analysis of the Internet up to 2022 by Cisco Systems [1] suggests that we will see a decrease in the use of personal computers to access the Internet (PC): PC activity is predicted to decrease from 49 % to 19 % (Fig. 1). At the same time, smartphones are becoming more and more ubiquitous Internet devices and will account for 50 % of total traffic on the Internet by 2022 (Fig. 1).

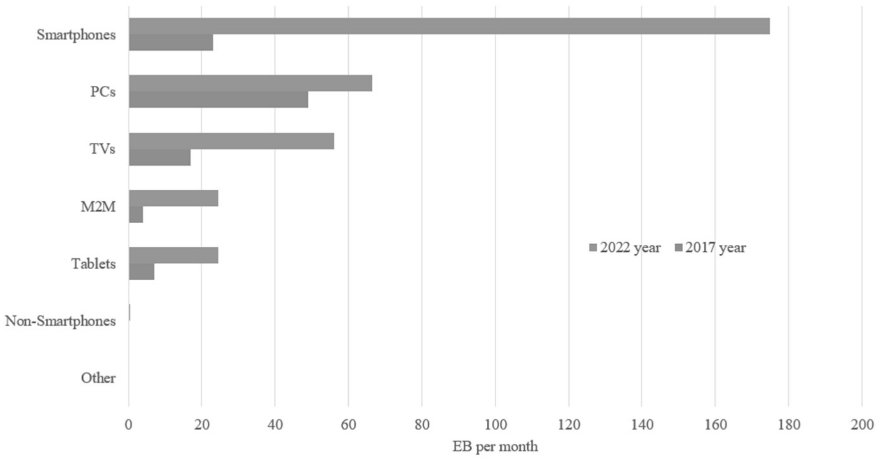

Fig. 1. Most active Internet devices

Currently, a primary use of smartphones and tablets is viewing video [2]. As a result, the volume of IP video traffic will also increase year by year. By 2022, IP video traffic is set to account for 82 % of traffic [1] (Fig. 2).

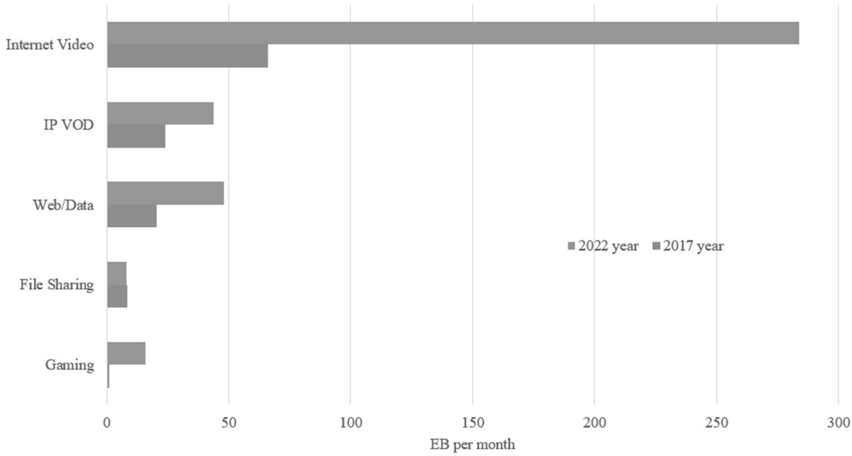

Fig. 2. Most active Internet application

Competition for customers is forcing OTT streaming video providers to offer new and varied services. Most of them offer video with 1280 × 720 HD (High-Definition) or 1920 × 1080 FHD (Full High Definition) resolution, but this is no longer enough. Consumers are seeking even better quality video at 3840 × 2160 UHD (Ultra-High Definition) 4K, or even 7680 × 4320 UHD 8K resolution. This trend has been identified in TV technology research [3].

This will increase Internet traffic because videos in UHD resolution require more than twice the bandwidth of videos in FHD, and more than ten times the bandwidth of those in SD (Standard Definition).

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

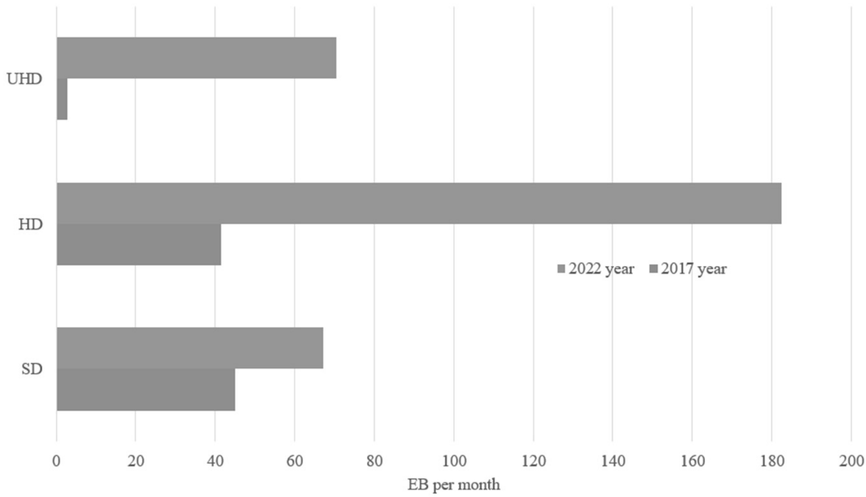

Fig. 3. Global IP video traffic

These trends in Internet traffic will change its overall pattern. The role of content delivery networks (CDNs) is set to increase significantly. According to the analysis in [1], these services will carry 72 % of total Internet traffic by 2022 (Fig. 4). As such, OTT services will be located closer to the end user, the consumer, decreasing delivery latency during OTT video streaming.

Some modern devices, such as smartphones, tablets, and TVs, support 7680 × 4320 8K UHD resolution. The forthcoming 5G network will allow all consumer devices to have a high-speed Internet connection. This new challenge creates demand for new services from OTT streaming service providers [3] who must be able to supply users with video at the maximum resolution of their devices: 3840 × 2160 4K or 7680 × 4320 8K.

Our previous research into video streaming with a resolution of 1280 x 720 and a mean bitrate value of 5.68 Mbit/s showed that the combination of delay and loss lead to significant degradation of the QoE [4]. The re-buffering effect appears with packet loss of 0.5 % and delays of 12-15 ms for different streaming protocols. Most client devices use wireless technology to access the Internet over the final mile with loss rates and delay.

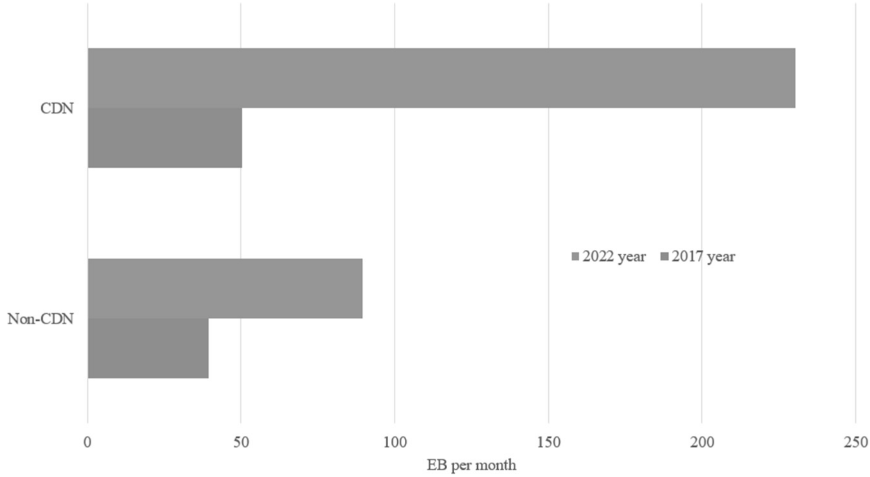

Fig. 4. The role of CDNs in content delivery

This research is devoted to the estimation of packet loss and delay thresholds in the re-buffering effect for the 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. An assessment was made of the significant influence of packet loss and delay on re-buffering. A proposal is suggested for acceptable levels of delay and packet loss for streaming video with 7680 × 4320 8K UHD resolution, excluding re-buffering.

This paper is structured as follows. In the following section, current research on the problem is reviewed. In the next section, the experimental platform is described and the characteristics of the video clips examined are presented. In the fourth section, the results are presented. In the fifth section, the influence of packet loss and packet delays on re-buffering for different video resolutions are analyzed and discussed. The sixth section describes different transport protocols, which can be used as alternatives to TCP. The seventh section offers some conclusions and the final section is devoted to future work.

Review of OTT Technology Research

A. Consumer satisfaction with OTT technologies

In the early days of OTT implementation, consumers were very tolerant of re-buffering and delayed starts to their videos. The extension of OTT video services has led to an increase in consumer expectations. Consumers now wish to watch video content anywhere and without any degradation: this new capability can be summed up as “I can watch what I want where I want”.

Here are some examples from consumer satisfaction research done by various 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 the 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].

Taking into account the user opinions, it is possible to highlight three common problems, emphasizing:

- The re-buffering effect;
- Delayed start;
- Reduced video bitrate.

We are also continuing to study consumer QoE. Year by year, customers become less tolerant of the re-buffering effect and variable video quality. They do not want to have any distortion while watching an OTT video stream. Based on this trend, we have set a new focus for our research of finding the threshold of QoS network parameters past which we will see the re-buffering effect for different video resolution and streaming technologies.

B. Results of previous research

Competition is always good for users, and, as such, OTT streaming video providers must deliver videos with the best QoE possible. For this reason, a lot of research has been done by authors to evaluate network performance factors affecting QoE. During the first phase of this research, experiments for subjective evaluation of QoE with different sets of QoS parameters (packet loss, delay, and throughput) were undertaken using an experimental platform. The results of these experiments [8, 9, 10] have shown that IPLR (IP packet loss ratio) and IPTD (IP packet transfer delay) have a significant impact on QoE. Unfortunately, these subjective measurements display great dispersion and are time-consuming to develop. For objective measurements, the authors suggest using factors like re-buffering, when the receiving buffer is empty and video playback is stopped. An estimation of video quality can be undertaken based on the frequency and mean duration of re-buffering [10].

Research was performed to discover thresholds for QoS parameters that radically influence QoE. The simultaneous influence of two QoS parameters, packet loss and delay, was shown to have a significant impact on re-buffering. In this work, three streaming technologies were compared: HDS (Adobe Flash HTTP Dynamic Streaming); RTMP; and pseudo-HTTP streaming. A video clip was chosen with a resolution of 1280 x 720 pixels; VBR (variable bit rate; the mean value of the bitrate was equal to 5.68 Mbit/s) and quick changes of scene with dynamic objects; and audio at 192 kbit/s. The experimental results for the influence of QoS parameters on re-buffering thresholds [4] are presented in Table 1.

The two top rows show the influence of these two distinct factors: packet loss and delay. The bottom rows show the simultaneous influence of these two QoS parameters.

Table 1. Re-buffering thresholds

	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 shows that the thresholds after which re-buffering starts are rather high when compared to the objectives for Class 4 IP networks [11]. In real networks, both factors exist simultaneously. With a combination of the above-mentioned parameters, the thresholds for delays were 4-5 times lower when compared to the thresholds for the separate factors, being in the range of 12-15 ms for packet loss of 0.5 % and 9-11 ms for packet loss of 1 %. These values are very close to measured QoS parameters on real IP networks.

Comparison of streaming technologies was undertaken by looking at the influence of both factors in combination. Best results in terms of resistance to packet loss and delays were shown by pseudo-HTTP, while HDS was the worst. RTMP showed intermediate results, but RTMP has a specific functionality as a decoder at low bitrates, leading to a slide show, and this mode also influences a user's perception [12].

As OTT becomes more popular among 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 the most important aspects for supporting QoE. For OTT services there are currently no such regulatory documents. Meanwhile, the mechanisms for supporting QoE of streaming over OTT are becoming vital for video service providers. For practical needs, providers need a recommendation for monitoring and controlling specific QoS parameters to exclude the influence of network performance parameters on the quality of user perception.

C. Other work by the authors

Other work by other authors were 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 sees slight modifications to the basic ABR technique. Most of this research has been targeted towards an open-source standard for ABR streaming: MPEG-DASH. It is usually assumed that a lower video bitrate is present. For example, in [14] a bitrate of 800 kbps for 720 × 576 SD and 2 Mbps for 1280 × 720 HD video resolution was assumed. This assumption does not suit our work. Netflix [19] gives the following recommendations:

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

The focus of this research was solely on the ABR algorithm and it did not consider current market demands and trends in consumer requests. Users want to watch video with FHD 1920 × 1080 resolution and without any video quality degradation. ABR technology mitigates the re-buffering effect by using video quality degradation, but this would be unacceptable to OTT streaming users.

Experimental Platform

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

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

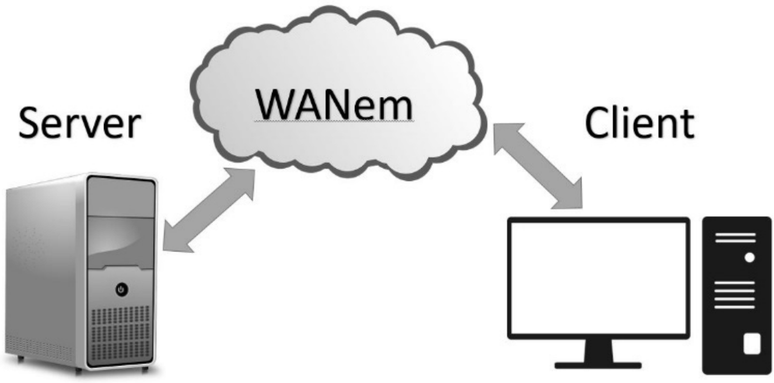

Fig. 5. Experimental platform

A full description of the experimental platform is given in Table 2.

Table 2. Detailed information about the platform

Name	Description
PC Hardware Platform	CPU - Intel Core i7 – 4770 RAM – 16 Gb 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 an 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 the experimental platform, drawing on the results of our previous research [4] and in line with new trends:

- The bandwidth of the path was not limited.
- We changed streaming technologies to HLS, RTMP, and MPEG-DASH, which are now more popular than others [23].

- The video clip was chosen with a resolution of 1920 x 1080 pixels for the next step in our research [4]; detail parameters are presented in Table 3.

Server, WANem, and client were virtualized on VirtualBox. They were all connected to the virtual switch. WANem was used to emulate internet cloud behaviour. Traffic between client and server was routed through the WANem virtual appliance.

Table 3. 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

Results

A. General research results

The first 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]. The new streaming technologies, HLS and MPEG-DASH, were added to the experimental platform, reflecting current trends in video streaming [23]. From the results of the study, we identified the beginning of re-buffering in the streaming protocols RTMP, HLS, and MPEG-DASH, as presented in Table 4.

Table 4. Streaming technology comparison (influence of a single parameter on the start of re-buffering)

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

We analyzed the results obtained in terms of the current requirements to ensure QoS for different traffic classes under ITU-T Recommendation Y.1541 (12/2011) [11]. Streaming video corresponds to class 4, for this class the maximum total allowable delay is 1,000 ms and packet loss must be less than 0.1 %. The means that delays are a more significant factor for re-buffering than packet loss. Moreover, exceeding the limits of delays indicated in Table 4 is guaranteed to result in re-buffering even if packet loss is zero. In the Recommendations, rather big delays are allowed; in reality, however, delays of 180-287 ms cause re-buffering. The main reason for this effect is that in the Recommendations for streaming video, transmission was supposed to use the UDP/IP stack; in OTT technology the TCP/IP stack is used. While using TCP, retransmission can occur increasing the delay of a single packet significantly and causing the buffer to empty and leading to re-buffering. This means that with mean delay values of 200-300 ms re-buffering can occur.

In modern computer networks, packet loss is less than 0.1 % and delays are less than 100 ms so that the thresholds for re-buffering decrease significantly.

Research results on the influence of two factors (IPTD and IPLR) on re-buffering are presented in Table 5. During research, we fixed the delay and changed the packet loss until re-buffering started.

Table 5. Streaming technology comparison (influence of the parameters in combination on re-buffering)

Delay, ms	**Packet loss, %**		
	RTMP	HLS	MPEG-DASH
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 results are in the range of QoS parameters of real modern data networks; QoS parameters should be monitored and controlled to enable video streaming transmission.

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

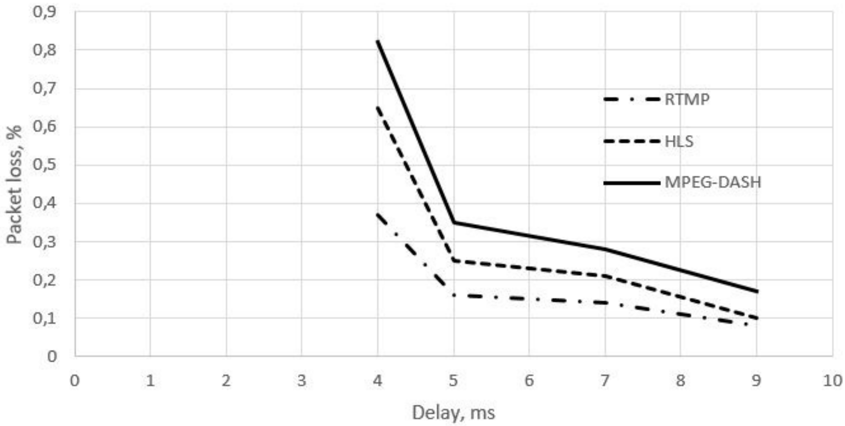

Fig. 6. Thresholds of re-buffering for HLS, RTMP and MPEG-DASH

The other meaningful factor that influences 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 use the formula for calculating the effective throughput of a TCP session presented in this Recommendation in the fifth section of this paper. We obtained a theoretical maximum of TCP throughput from packet loss and delay in using it. This will be the upper limit of video AVG (average) bitrate.

B. Re-buffering time-frame

During our research, we found that re-buffering effects occur only at a specific time-frame of video playback. We took three different clips with resolution at 1920 x 1080 FHD and analyzed their bitrate using the Bitrate Viewer [24] software:

- Video 1: Big Buck Bunny (Fig. 7), used in current research.
- Video 2: Sport (Fig. 8) used in previous research [4, 8-10].
- Video 3: Nature (Fig. 9) used in previous research [4, 8-10].

The results of this analysis showed that each tested video has a specific bitrate, but they all have variable bitrate (VBR). VBR is used by default for most video on the Internet because it gives much greater quality at the same AVG bitrates. It is an encoding method that enables the bitrate of a video to dynamically increase or decrease according to the complexity of video fragments. For more complex fragments, the bitrate will be higher while for less complex fragments the bitrate will be lower. Constant bitrate (CBR), in contrast, utilizes a set bitrate over the entire video clip leading to image quality being degraded during complex video segments.

Analysis of this video is presented in Table 6. The videos have nearly the same AVG bitrate, but extremely different MAX and MIN values at specific time-frames of playback. Based on this analysis of video bitrate, we can predict for which time-frames of video playback consumers will frequently see re-buffering. For Video 1, re-buffering can occur at the beginning, in the middle, and at the end of the video. For Video 2, re-buffering is likely to occur at the beginning and after the middle of the video. And in Video 3, re-buffering could occur at any time, because the compensation buffer increases the possibility of re-buffering throughout.

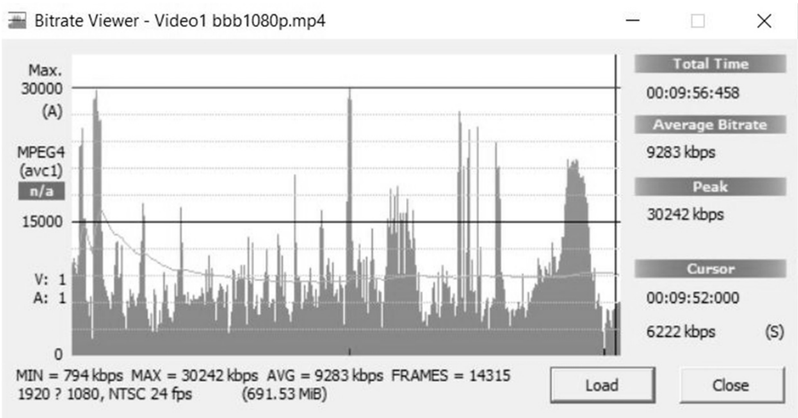

Fig. 7. Bitrate of Video 1

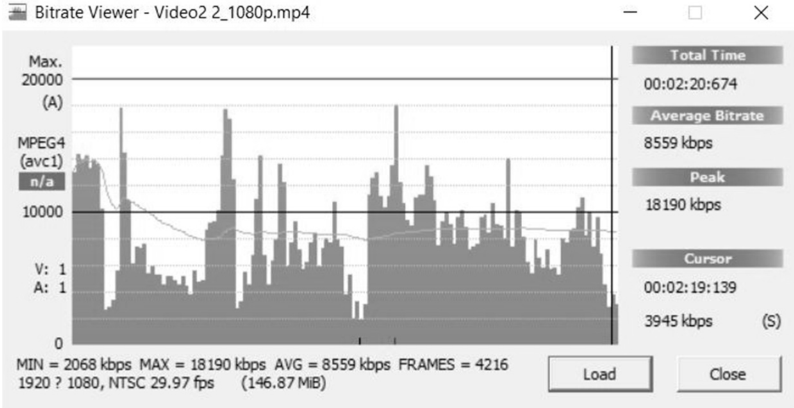

Fig. 8. Bitrate of Video 2

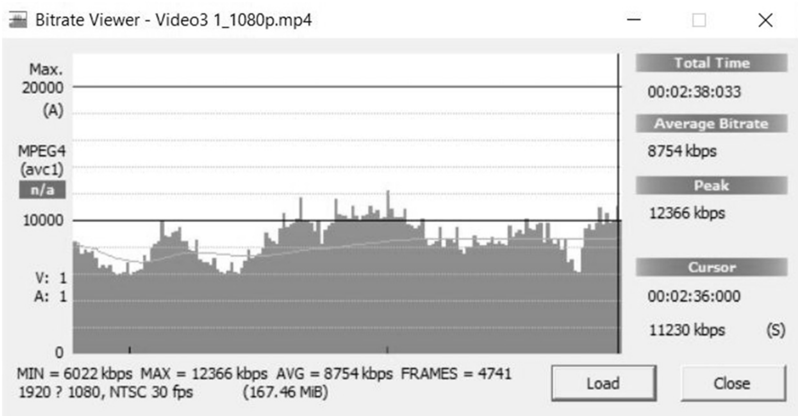

Fig. 9. Bitrate of Video 3

Table 6. Parameters of video bitrate for videos 1, 2, & 3

Parameter	Video 1	Video 2	Video 3
AVG bitrate, kbps	9283	8559	8754
Min bitrate, kbps	794	2068	6022
Max bitrate, kbps	30242	18190	12366

Research into QoS Parameters Thresholds

Analysis of the results showed that TCP throughput has a significant influence on re-buffering. Using formula [11] to calculate the effective speed of a TCP session, we can estimate the opportunity to transfer streaming video over the network with defined QoS parameters and without re-buffering.

It is not straightforward to compare the AVG of streaming video and throughput of a TCP session. It is necessary to consider additional headers and service packets of these streaming technologies. This can lead to a significant reduction in throughput during video transmission.

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

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

where W_{max} is the maximum window buffer size of a receiver (packets); RTT is the round trip time (sec.); b is the number of packets that are acknowledged by a received ACK; p is the probability that a packet is lost; and T_0 is the time-out for re-transmitting an unacknowledged (lost) packet (sec.).

We can use the following simplification for formula 1, because packet loss is less than 0.1 % in modern networks:

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

From formula (2) we can use the formula for TCP throughput:

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

where MSS is the maximum segment size.

Table 7 presents recommendations from Google [25] for uploading a video to YouTube. If we compare the recommendations in Table 7 and a 3D chart of the maximum TCP throughput in Fig. 10, we can see that it is possible to watch a video at 4K resolution with current standard network

QoS parameters if we use a stable cable connection with a bandwidth of more than 30 Mbps. However, for an 8K video, there is a high probability that we will see re-buffering. In modern telecommunication networks, it is very difficult to ensure a delay of less than 15 ms and a packet loss of less than 0.03 % continuously. As such, these QoS parameters may be guaranteed only by a CDN provider that has servers in each city. This restricts the possibility of expanding 4K and 8K services from OTT providers.

To watch 8K UHD video from an OTT service provider, the client must have a stable internet connection with a minimum of 50 Mbps speed and additional delay and packet loss on the client side.

Table 7. Video bitrates for different resolutions

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

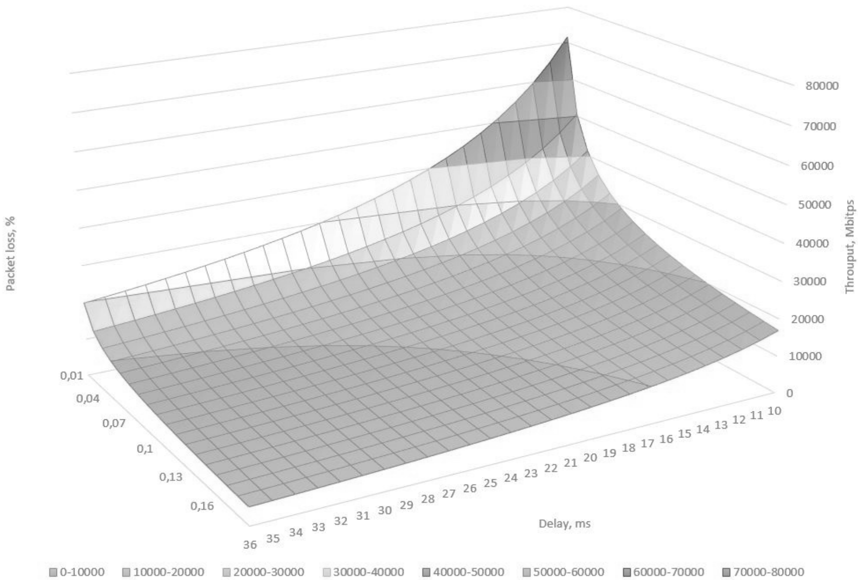

Fig. 10. Maximum TCP throughput with packet loss and delay

TCP Alternatives

The effective throughput of a TCP session depends on delay and packet loss and this is the primary cause of re-buffering. The calculated effective TCP throughput for a given delay and packet loss allows us to estimate the possibility of delivering a video stream at a definite bitrate. As such, TCP session throughput functions as a bottleneck for UHD video streaming via OTT.

We need an alternative transport protocol to overcome the limitations of TCP. This protocol must be as reliable as TCP and have throughput that does not depend on delay. In Table 8, we present some transport protocols [26].

Table 8. Comparison of transport layer protocols

Feature	UDP	TCP	RUDP	SCTP	MPTCP	QUIC
Connection-oriented	No	Yes	Yes	Yes	Yes	Yes
Reliable transport	No	Yes	Yes	Yes	Yes	Yes
Ordered delivery	No	Yes	Yes	Yes/ No	Yes	No
Multihoming	No	No	No	Yes	Yes	Yes

Connection-oriented protocols set up a connection before data delivery. For example, TCP uses a three-way handshake to set up a connection.

Reliable transport guarantees packet delivery over the network.

Order delivery ensures that packets are delivered in the same order that they were sent.

Multi-homing gives clients and servers the possibility of setting up multiple connections using different addresses. This may increase reliability/performance.

A. RUDP

The first draft of the Reliable User Datagram Protocol (RUDP) was published in 1999 [27], but this protocol has not yet been ratified. Cisco and Microsoft have both created versions of this protocol, but they are not