Ministry of Education and Science of Ukraine
Ternopil Ivan Puluj National Technical University

Faculty of Computer Information System and Software Engineering
(full name of faculty)
Department of Computer Science
(full name of department)

QUALIFYING PAPER

For the degree of

Bachelor
(degree name)

topic: Studying technologies to improve the quality for use video content

Submitted by: fourth year student _____ , group ICH-42
specialty 122 Computer science
(code and name of specialty)

Adekane Kehinde Douglas
(signature) (surname and initials)

Supervisor Zolotyi R. Z.
(signature) (surname and initials)

Standards verified by Matsiuk O.V.
(signature) (surname and initials)

Head of Department Bodnarchuk I.O.
(signature) (surname and initials)

Reviewer
(signature) (surname and initials)

Ternopil
2023

Ministry of Education and Science of Ukraine
Ternopil Ivan Puluj National Technical University

Faculty Faculty of Computer Information System and Software Engineering
(full name of faculty)

Department Department of Computer Science
(full name of department)

APPROVED BY
Head of Department
__________ Bodnarchuk I.O.
(signature) (surname and initials)
« » 20___

ASSIGNMENT

for QUALIFYING PAPER

for the degree of Bachelor
(degree name)

specialty 122 Computer science
(code and name of the specialty)

student Adekane Kehinde Douglas
(surname, name, patronymic)

1. Paper topic Studying technologies to improve the quality for use video content

Paper supervisor Zolotyi R.Z., PhD
(surname, name, patronymic, scientific degree, academic rank)

Approved by university order as of « 28 » 02 2023 № 4/7-233

2. Student's paper submission deadline 10.07.2023

3. Initial data for the paper Literature sources about architecture, principles of operation and development of Web-resource.

4. Paper contents (list of issues to be developed)

6. Advisors of paper chapters

Chapter	Advisor's surname, initials and position	Signature, date	
		assignment was given by	assignment was received by
Life safety,			
basics of labor protection			

7. Date of receiving the assignment 10.03.2023

TIME SCHEDULE

LN	Paper stages	Paper stages deadlines	Notes
1	Analysis of the task for qualifying work. Selection		*Completed*
	and work with literary sources.		
2	Writing chapter 1		*Completed*
3	Writing chapter 2		*Completed*
4	Writing chapter 3		*Completed*
5	Writing chapter 4		*Completed*
6	Standartization control		*Completed*
7	Plagiarism check		*Completed*
8	Preliminary defense of qualifying paper		*Completed*
9	Defense of qualifying paper		

Student ______________ Adekane Kehinde Douglas
(signature) (surname and initials)

Paper supervisor ______________ Zolotyi R.Z.
(signature) (surname and initials)

ANNOTATION

Studying technologies to improve the quality for use video content // Adekane Kehinde Douglas // Ternopil’ Ivan Puluj National Technical University, Faculty of Computer Information System and Software Engineering, Department of Computer Science // Ternopil', 2023 // P. __, Fig. – __, Tables – __, Annexes – __, References – __.

Keywords: streaming video protocol technology network wireless access, video content.

In the bachelor's thesis, a study of modern video content delivery technologies and their development prospects was carried out. The conducted research allows us to conclude that WiMAX and LTE data transmission networks are the most adapted to modern conditions, however, the potential capacity of LTE networks, due to higher speed in the downstream direction, speaks of its advantages.

The transmission of video streams in a wireless network is also considered, including in the presence of background web traffic, and the work of two algorithms that use different approaches when choosing the distribution capacity (quality) of video content is investigated.

LIST OF SYMBOLS, UNITS, ABBREVIATIONS AND TERMS

DB – database.

IS - information system.

ISUP - project management information system.

IT - information technology.

OS – operating system.

PC – personal computer.

DBMS – database management system.

CONTENTS

INTRODUCTION

Since the introduction of commercial products on the Internet in 1995, has there been a rapid growth in multimedia data services. The share main of resources during transmission is occupied by data streaming. The advantage, as rule, of streaming data is that materials video and audio visually present information in an easily perceived form. This, in turn, is effective in advertising, marketing, training, and coaching. For companies, this means increased efficiency, greater profitability, and costs lower for delivery information. The traditional method of delivering files from networks using downloads is not convenient enough for video and audio, even with the fact that client connections are getting faster. Often, transfer speeds are not sufficient to download video files, sizes of which could be tens of megabytes, even gigabytes. The solution to the problem of large files, however, has been around for several years now: streaming technology. With streaming media players such Microsoft's Movies and TV Media Player, or RealNetworks' RealPlayer and even the discontinued Apple’s QuickTime, it is possible to show audio and/or video data a few seconds after the first, as rule, bits of stream arrives to computer.

According to a report by InterDigital, in 2022, 82% of the internet global traffic came from streaming video and downloads alone and expected to continue as people consume more contents video online. Monthly increase of this service is calculated by millions of hours of backplay and hundreds of thousands of new media servers. This growth of video traffic and content over the internet has been driven by several factors including an increasing availability of high-speed internet which has made it easy for people to online contents stream video. The advent of popular social media platforms like Facebook, Instagram, WhatsApp and other platforms fot streaming video like YouTube, Snapchat and TikTok among others has gathered increasing popularity, and these platforms has made it easier for users across to create and share video content with one another. In addition, this service is regularly used by more than 250 million registered subscribers, which is replenished daily by more than 200,000 users. The global pandemic of 2019 as well meant a global quarantine which sent offices and employees into their homes providing for a new wave of over business conducting the internet with synchronous and asynchronous communication video becoming the main source of communication that has persisted in several industries even with the end of the pandemic in 2021. This was a huge point turning for the industry streaming, with people stuck indoors, watching videos, and streaming provided a much-needed escape and has remained popular.

This rapid increase has created a new challenge in maintaining a proper quality of service. Even with increasing internet speed, a regular video content would take forever to get completely from a service network to a user's network due to their huge storage size but with new technologies, even large videos are able to move networks across through streaming video. Nowadays, "streaming video" the term defines compression data and technologies buffering for transmission video real-time over networks communication. Compression of video is a component critical in the delivery of contents video across networks different. This of course also brings up the issue of balance between efficiency compression and quality video as size bit, this means the priority more placed on quality video, the less the compression efficiency and more spent bits and vice-versa. likewise, if you were to compromise on compression efficiency, thereby creating larger files, you'll have to spend more on content distribution network delivery costs.

Recent years have seen the growth rapid and development of wireless broadband technologies access, which are a alternative to systems communication cellular. This increase has also created a new challenge in quality maintaining of service for video streaming. Main disadvantage of transmission video over networks wireless is the lack of sufficient synchronization the original sequence between the decoding on the receiving side by the copy. Because of the aggressive transmission environment, data packets can be seriously distorted or even lost.

No standard approaches to determining the impact of a complex of errors on the transmission quality of the services provided. These problems, combined with commercial success, force research aimed at effective, stable encoding and transmission of video over unstable networks, which include wireless broadband access technologies.

Obviously, video contents play a big role in today’s world which isn’t unusual, according to statistics from statista.com, the average daily time spent on media including, but not limited to media social, television, on demand video, and several services streaming in the US alone is over eight hours. In the future, we can expect traffic video and over the internet content to continue to grow and the big takeaway is that these days, powers video everything, for everyone, everywhere as more people gain access to internet high-speed connections, we more can expect people to consume more video content. Thus, the topic of a bachelor's thesis devoted to the research of technologies for improving video content access is relevant.

1 METHODS OF STREAMING TRANSMISSION VIDEO IN MODERN COMMUNICATION NETWORKS

1.1 Characteristics streaming of video

When was initially created website, they made up of simple pages of texts, maybe one or two images could also be added but today anyone with internet high-speed can watch a high-definition movie or content or even make online video call. This is possible advice to the technology “Streaming”. STR are traffic of type that is characterized by listening and (or) viewing to information as it enters the user (end) equipment, this means a viewing method video content without the files media downloading. The main streaming part of traffic is audio and video streaming broadcasting.

For real-time video transmission is a used term video stream. The main feature video streaming is that when video transferred, the user does not have to wait for completely downloaded for viewing for the file to. The broadcast is video in a stream continuous in the form of compressed a sequence packets in format special and does not require a download full of the file data video containing. Viewing begins when there is sufficient data buffering, while ensuring an even display of data. To some extent, the video stream is analogous to traditional television broadcasting with continuous reception and display of video frames. For comparison, when watching a video in the DVD standard, a full download of the video file is required.

Main are two ways of video streaming, these are progressive real-time streaming and streaming. Streaming progressive is a delivering method of content video to users in a allows way for video watching the while it downloaded. The video delivered in chunks small and the user can video watching start the as soon as the chunk first o is downloaded, for example, watching on YouTube a online video. When stream video transmitting through streaming progressive is done, the image quality is better always since the video is played from a server or carrier data where it has been previously encoded and pre-recorded and than stored on. The disadvantage of progressive transmission method is the viewing episodes impossibility and the possibility of overflowing the information carrier on the receiving side. Therefore, progressive video streaming is only used for short clips.

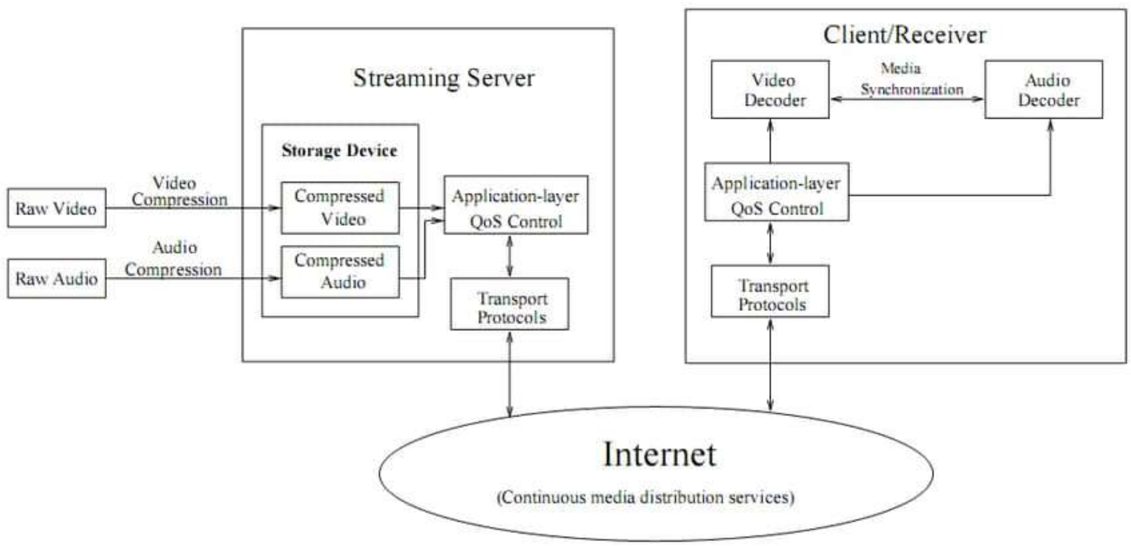

Figure 1.1 - Architecture of a video streaming system broadcasting

A Video streaming system has four main subsystems (Figure 1.1):

1. An encoding device.
2. Media Server

It usually consists of three components:

- A broadcast mechanism (such as transport protocols),
- An operating system and,
- A storage system.

3. Transport network: Transport network broadcasts packets from the media server to client device using specially developed and standardized protocols. Protocols provide such communication services as network addressing, transport, and

communication session control. According to their functionality, protocols are classified into three categories:

- Network layer protocol,
- Transport protocol,
- communication session control protocol.

4. Application layer: A application client plays and decodes the multimedia stream has means of control and management. In addition, a mechanism for audio and video synchronization is possible, which distinguishes streaming multimedia applications from other traditional applications. A video streaming client typically uses error detection and masking techniques to reduce the impact of lost packets on overall quality.

For stable operation, these components must be designed and optimally coordinated.

The volume of streaming traffic growing is, despite the presence of destabilizing factors in the transmission of a video stream over the network, such as changes in bandwidth (the presence of narrow channels), delays and packet losses. Streaming requires a stable connection, since some communication networks cannot provide direct data transfer between the sender and the receiver, the flow may be interrupted or stopped for some time.

Since the content of a video stream is extremely sensitive to available bandwidth and QoS, the quality of playback, from the viewer's point of view, is closely related to such network characteristics as performance, delay variation, and the number of lost packets.

With the advent of wireless client devices, successful video streaming means:

1. Continuous provision of services even under the condition of user movement,

2. Adaptation characteristics of according video transmitted to the features of the networks communication [1].

1.2 Video Compression

Compression video is the reducing process of the bits number needed to video re-present a without its visual quality compromising, it is the turning large method, raw streams video into files smaller. It is done by filtering out addition and data video redundant.

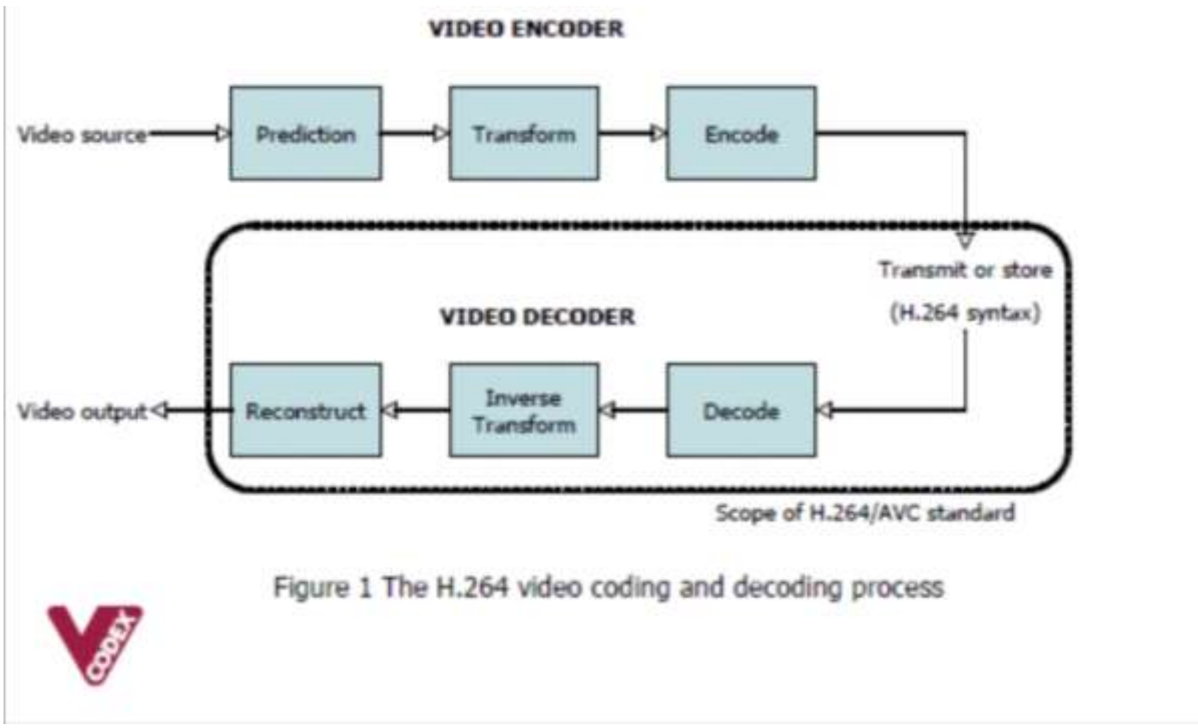

Figure 1.2 - Decoding and encoding process as provided from source vcodex

Basic streaming video protocols

Thanks to compression increased, the coding standard video H.264/AVC allows in low-speed networks video streaming without quality degradation noticeable, which the use of this standard for allows applications video in networks wireless [1].

A very convenient environment for video streaming is networks IP, due to the [3] for delay requirements and transfer speed data [2].

At the layer application, streaming video is provided by the following main protocols [2]:

(RTP) Real-time Transport Protocol.

Real-time Control Protocol RTCP: This is a control protocol created for joint work with RTP, it helps to synchronize video and sound, ensure service of quality.

Real-time Streaming Protocol RTSP.

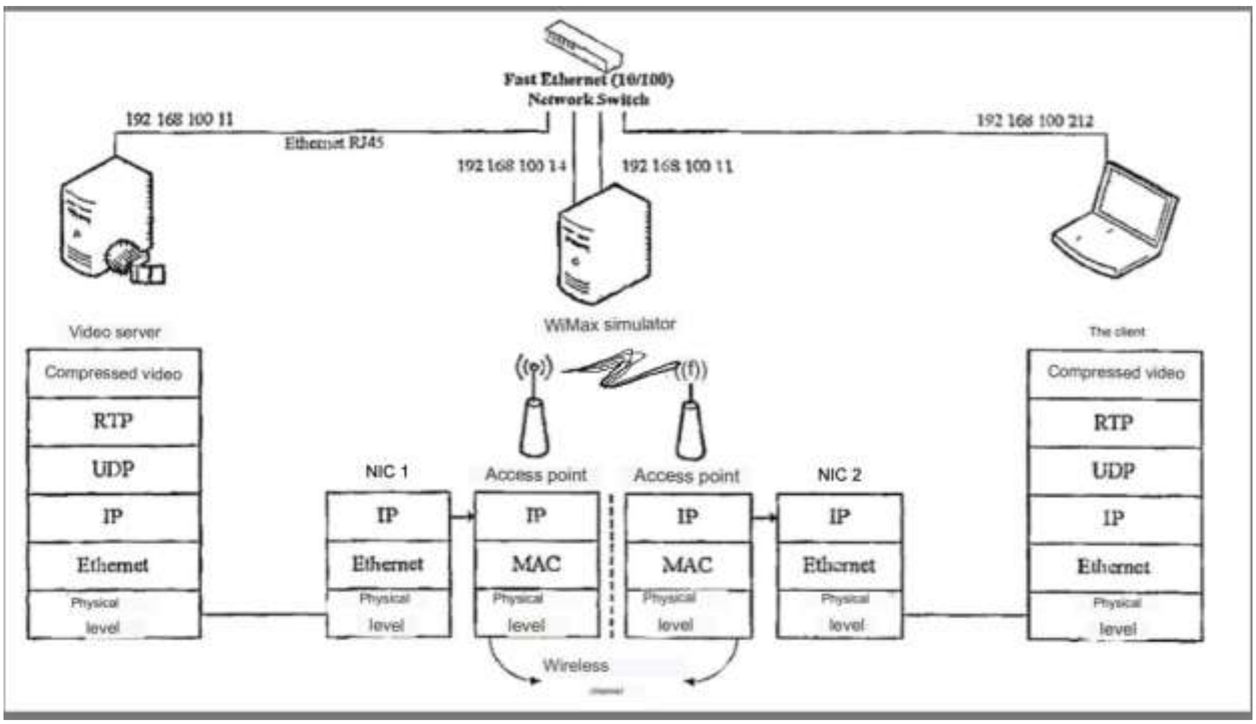

Figure 1.3 – Physical and protocol streaming video structure using WiMAX network wireless as an example.

Resource reservation protocol (RSVP).

Fig. 1.3 presents the physical and protocol structure broadcast of video over a network wireless in the model OSI context [2].

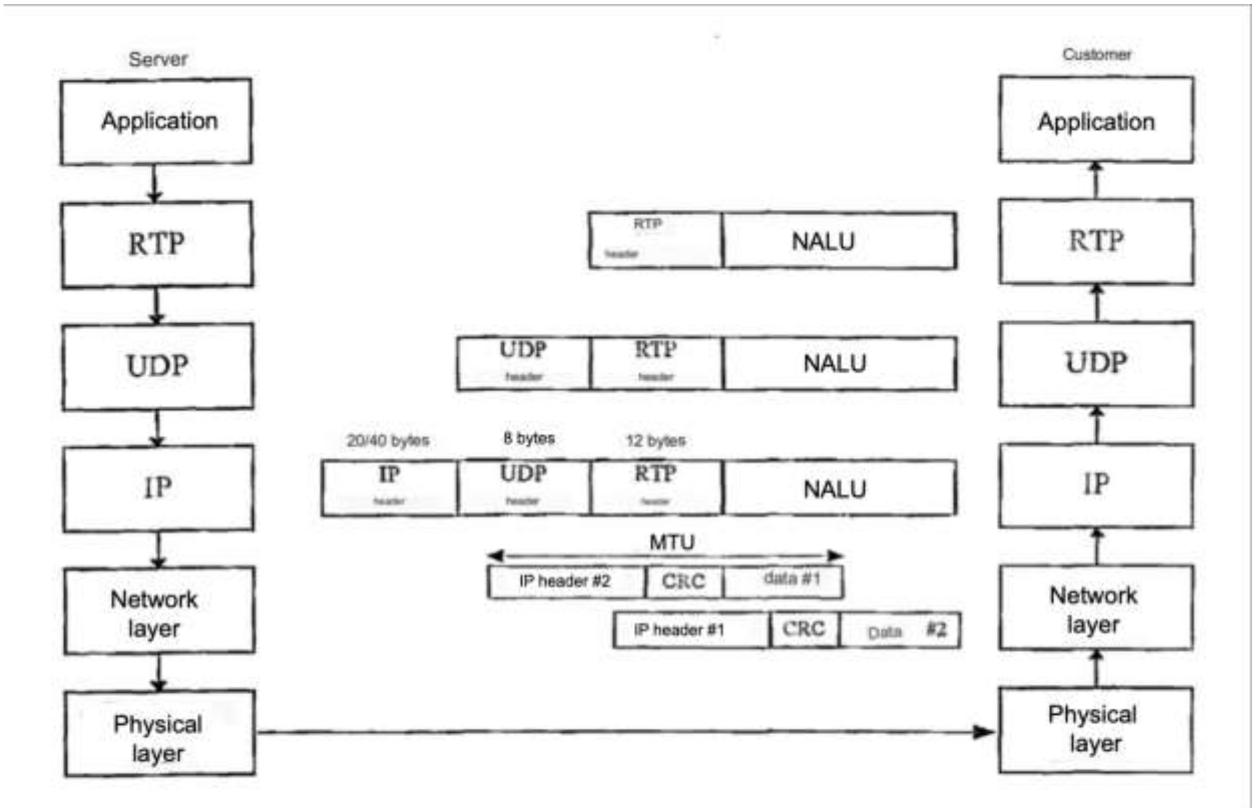

Figure 1.4 – Encapsulation of data video in the protocol slack

In fig. 1.4 presents the structure of the stack protocol during streaming multimedia.

The use of hierarchical coding allows distribution of video provide to information in complex networks with bandwidths different of individual segments.

The use of advanced more coding methods does not completely allow to avoid the appearance of characteristic video distortions in real time broadcasting when applications. Generally, in wired networks in which the bandwidth is sufficient, the transmission channel has a very low probability of erroneous bits. However, data transmission over a wireless channel has several peculiarities due to unpredictability of transmission conditions.

Accordingly, [5] not all decoders of the MPEG standard process the stream with errors qualitatively. Some decoders can handle errors with high probability. Other decoders cannot decode a stream with a high number of errors, but a stream with a low number of errors is decoded with good quality. There are decoders that stream decode with errors quite poorly, regardless of the errors, and are generally not suitable for handling this type of stream. Later versions of such codecs as Halready used partitioning data, which is an effective way to increase error tolerance. Now, H.264 standard such error includes protection mechanisms as a parameters set (SPS, PPS), FMO, etc. [6]. At the network level, packet size change is used to reduce losses, differentiated service, etc. At the physical level, uneven error protection is used [7].

Bit errors arising during transmission over networks wireless can quality affect of video decoded in ways different [8]:

Bit error in different parts of the bit stream.

Bit error in video sequence header.

Bit error in image header.

Bit error in frame group GOP.

Bit error in DCT coefficients.

Since codecs process in blocks information, the minimum unit of stream video distortion under the influence of a single error block is (4*4 or 16*16 encoding depending). There are three possible sources of error propagation [7]:

Spatial prediction: A macroblock decoded whose neighboring macroblocks are distorted will also be distorted.

Temporal prediction: If a frame is distorted, subsequent frames use the distorted frame as source will also be distorted.

Entropy Coding: Since VLC used code, an error in a key code can affect subsequent codes if the key code boundaries are not defined correctly. In this way, the synchronization of the codes following is disrupted, which entails the inability of the decoder to distinguish the key codes [6].

Using VLC results in desynchronization of information decoded, causing information some to the code next to undecoded become. In cases some, even after restoring synchronization, the signal decoded cannot be used correctly, because additional information about how it is used, such as the frame type or vector motion, is lost. Distortions arising because of errors during transmission and subsequent decoding are defined by the following terms:

Image blockiness (tiling effect)

Vagueness (blurring)

Color rendering errors (color errors)

False blocks (error block)

Tremors (jerkiness)

The effect of "mosquitoes" (mosquito noise),

Noise quantization (noise quantization),

Blurring (smearing).

In addition, when transmitting a video stream in real time, there are certain problems associated with visual quality control, speed / delay control, error control and scalability.

Error models in broadband access networks wireless

Errors in the stream digital that during occur transmission at the physical level of networks wireless can be conventionally divided into two types:

Single bit errors.

a packet of bit errors.

A single bit error is expressed as a bit inversion during transmission, which causes incorrect recognition of the sequence and the byte.

A bit error packet is a sequence of errors longer than two inverted bits in a data segment. Batch bit errors occur more often than single ones. The length of the error packet is measured from the first to the last inverted bit.

Bit errors are the simplest and easily overcome distortions of the digital stream. However, in some cases, they can lead to the loss of an entire segment of data. Thus, failure to eliminate bit errors at the physical level will cause the loss of the information (transport) packet at the channel level and will be called an information packet error.

The reasons for the information loss packets can be the following:

Analog and electromagnetic interference, impulse interference. As a rule, they arise due to external factors, including weather conditions and the proximity of electrical equipment. Error correction mechanisms at the physical level are not able always to consequences eliminate of the action, which in turn leads to the loss of the transport packet. The use of special methods of protection against interference causes a decrease in bandwidth and an increase in the cost of equipment.

Short-term bandwidth changes. Associated with poor connection of switched equipment. QoS constraints necessarily affect bandwidth limits, exceeding which can cause packet loss in the absence of control. Severe jitter or pulsating traffic can overwhelm the buffer and, consequently, the ability to process a sequence of packets, leading to loss or late reception and processing.

Equipment problems. Can be caused by improper operation of devices, parasitic communication between components, and untimely processing of packets due to jitter.

Different models are used to simulate these errors, which have different effects on the transmitted information.

AWGN model. Since wireless communication channels are characterized by randomly distributed and independent bit errors when simulating a wireless channel, the model of "adaptive white Gaussian noise" or AWGN is often used, in which a certain bit in the sequence is distorted (inverted) with given probability [7]. The value used is described by the probability of the appearance of erroneous bits BER (Bit Error Rate). The signal received in the AWGN channel can be represented as

$$r(t) = s(t) + n(t)$$

where s(t) is the transmitted signal.

n(t) is a noise signal having an average value of 0 and a noise power spectral density of N0/ 2, W / Hz.

The AWGN model is not capable of simulating a channel prone to fading. Attenuation of the transmitted signal leads to packetization (grouping) of errors. Since most modern codecs use VLC, it does not make much sense to separate bit and group errors, because decoder, the errors appear in the distortion or loss of a group whole of bits consecutive [7]. For example, a single error in a header can cause more distortion than a cluster of errors in multiple blocks.

GE model. Another well-known error model is the Gilbert model [7]. This model represents the channel in the form of two states: "good" and "bad". In a good state, a bit or packet is received successfully, while in a bad state, it is lost. At the same time, the transition probabilities between states correspond to p01 and r10. In good condition, probability of error / loss of rgood is equal to zero. In a bad state, an error / loss occurs with an independent probability rbad. Thus, to fully describe the Hilbert model, three parameters are needed: p01, r10 and rbad. However, it is often misunderstood that in Hilbert's model, a bad state corresponds to an error / loss state, i.e., pbad = 1.

This corresponds to a simple two-state Markov model that considers only single errors/losses. Therefore, groups of errors or their packaging cannot be modeled [8].

The Gilbert model was supplemented by Eliot, which today is called the Gilbert-Eliot (GE) model. This model assumes a small and independent probability of error / loss other than zero rgood > 0 even in good condition. Thus, four parameters are required to fully describe the model.

The Gilbert-Elliott model can be described by the matrix R. Let sn be a Markov process and corresponds to sn = 0 if the channel is in the "good" state in nth time, and sn = 1 otherwise sn = 0.

Thus, the transition matrix will look like

$$P = \begin{pmatrix} P[sn = 0 \mid sn - 1 = 0] & P[sn = 1 \mid sn - 1 = 0 \\ P[sn = 0 \mid sn - 1 = 1] & P[sn = 1 \mid sn - 1 = 1 \end{pmatrix} = (00^{p}_{p10}\ {}^{p01}_{p11})$$

The average loss probability and the average error length are calculated as

$$P_{losses} = \frac{p01}{p10 - p01},\ L_{wrong} = \frac{1}{p10}$$

AWGN and Gilbert-Elliott models can be used both at the level physical, to bits applying, and at the level channel, applying to transport packets. They are often used in the experimental evaluation of simulated communication channels due to different effects on the video stream (Fig. 1.5), but they are not capable of simulating all errors.

The most realistic and accurate way to model error statistics at the channel and physical level is to use probability data obtained from a real network.

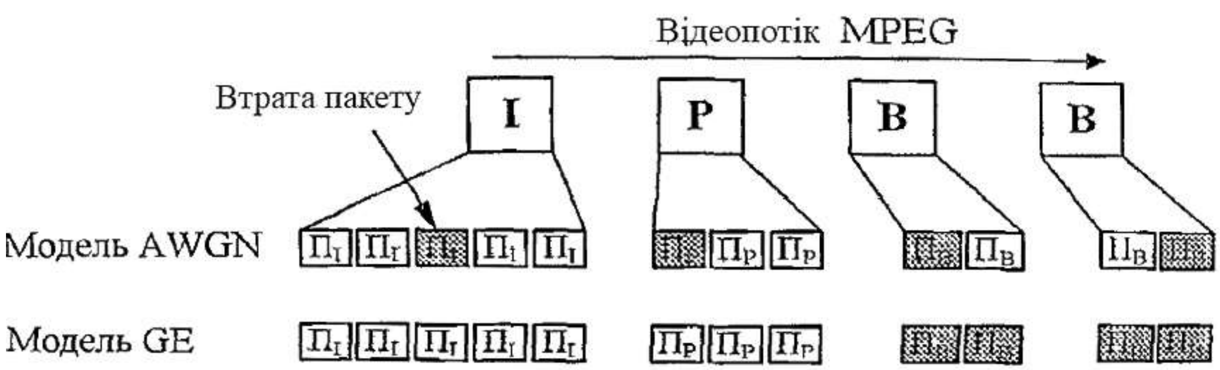

Figure 1.4 - Bit error models and their effect on the stream video

Even if the occurrence of errors is rare, they cause significant problems in the quality visual of video streaming, while in other types of services they will remain invisible.

2 METHODS OF EVALUATING STREAMING VIDEO QUALITY

For the correct decoding of a video stream that has passed through an unreliable data transmission medium, various methods of error correction are used, which affect the visual quality in different ways.

Since the video stream (in most cases) is an audiovisual medium, additional problems related to sound quality and synchronization with the video may arise [9]. When assessing quality, it is necessary to consider various components:

Video quality - an indicator of the visual quality of the image.

Audio quality is an indicator of sound perception.

Audiovisual (multimedia) quality - an indicator that compares quality picture, sound quality and their synchronization.

In addition to these indicators of perception, it is necessary to consider the quality of transmission - an indicator of communication networks, its ability to reliably transport video. This indicator reflects the network's ability to serve, not the quality of a specific video stream, and is independent of the type of codec used.

There are various methods that allow you to evaluate the characteristics of encoded video quantitatively and qualitatively. Video quality indicators can be obtained using subjective or objective methods.

One of the biggest technical problems when moving to voice and video transmission over packet-switched networks is to ensure the guaranteed quality of service (QoS), which allows you to receive sound and images without distortions and interference. Most existing packet-switched networks are designed for tasks and applications that are not particularly sensitive to signal delay. Voice and video on the other hand are very demanding on data transfer rates, and a packet delay greater than 200ms means that this packet is no longer needed because the data has already aged.

Therefore, networks for the transmission of voice and video must be designed, built, and operated in such a way as to maximize the efficiency of passing packets in real time.

Video traffic is by far the biggest contributor to bandwidth usage. With a known approximation, one channel of broadcast television or video on demand (VoD) today requires a transmission speed of about 4 Mbit/s. The situation will noticeably improve when the transition to the MPEG - 4 standard is made, but in any case, to obtain a high-quality image for video traffic, about 2 Mbit/s will need to be reserved.

The problems multiply when it is necessary to maintain the quality of the signal transmitted over a wide area network (WAN). Common speeds for local area networks (10, 100 Mbps and even 1 Gbps) are not used for WAN access due to high cost and therefore typical WAN access speeds are around 1.45 Mbps and below, creating a bottleneck at the LAN/WAN interface. Although this causes certain delays for e-mail and other types of data exchange, they are not critical. For voice and video transmission, it is necessary to reserve part of the bandwidth, otherwise the meaning of receiving services will be completely lost.

Quality of service (QoS) is output main for the multimedia implementation services traffic.

The main characteristics for QoS are defined as follows:

Package delivery delay. This parameter plays a role mainly when transmission of voice and video messages.

jitter - changes in delays during package delivery. Jitter can be measured by several methods. The jitter calculation is defined in the following recommendations:

IETF RFC 3550 RTP: A Transport Protocol for Real - Time Applications.

IETF RFC 3611 RTP Control Protocol Extended Reports (RTCP XR).

packet loss - when overloaded, the network is forced to discard individual packets. One of those parameters that plays a significant role in the transmission of voice and video messages.

2.1 Analysis of quality-of-service requirements for the voice transmission messages

The QoS requirements for the transmission of voice messages are softer than those of the transmission of video messages. Analysis of the recommendations of the ITU-T and the IETF (Internet Engineering Task Force) allows us to generalize the requirements for QoS characteristics when implementing the transmission of voice messages.

Voice traffic must be marked as DSCP EF, according to RFC 3246.

Signaling must be marked as CS3 (during migration it is possible use AF31).

Packet losses in trunks designed to provide VoIP service of high quality should not exceed 0.25 percent.

One-way delay should not exceed 150ms, according to MC3 - TG.114.

Delay fluctuations (jitter) should be less than 10ms. Maximum

jitter should be less than the network delay budget minus the minimum network delay. This typical value of the delay fluctuation for VoIP is due to the so-called "mouth-to-ear" delay budget of 100ms. This is a conservative budget compared to G.114, which recommends less than 150ms jitter. Subtract the trunk propagation time (30ms) and the codec delay (35ms) from this value, giving a jitter budget of 35ms. This 35ms is broken down into 30ms on access (15ms in/out) and 5ms on trunk. That is, in the worst case, for adaptive jitter buffers, delay fluctuations should be less than 10ms.

For each conversation (depending on the quantization frequency, codec, and header second level) requires 21-106 Kbps of bandwidth priority guaranteed.

Signaling traffic requires 150 bps (plus the header of the second level) of guaranteed bandwidth.

One of the important factors of effective use of channel capacity is the choice of the optimal algorithm for encoding/decoding language information - a codec.

All types of language codecs existing today can be divided into three main groups based on the principle of operation.

Codecs with pulse code modulation and adaptive differential pulse code modulation, which appeared in the late 50s and are used today in traditional telephony systems. In most cases, it is a combination of ADC/DAC.

Codecs with vocoder conversion of the speech signal arose in mobile communication systems to reduce the requirements for the bandwidth of the radio path. This group of codecs uses harmonic signal synthesis based on information, its vocal components - phonemes. In most cases, such codecs are implemented as analog devices.

Combined (hybrid) codecs combine vocoder conversion/speech synthesis technology, but already operate with a digital signal using specialized DSPs.

Table 2.1 – Assessment of voice quality using different codec.

Voice codec	Speed kbit/s	MOS assessment
G.711	64	4.10
G.726	32	3.85
G.728	16	3.61
G.729	8	3.92
G.729a	8	3.70
a723.1	6.3	3.9

Table 2.1 shows data on voice quality assessment when using different codecs (under ideal conditions).

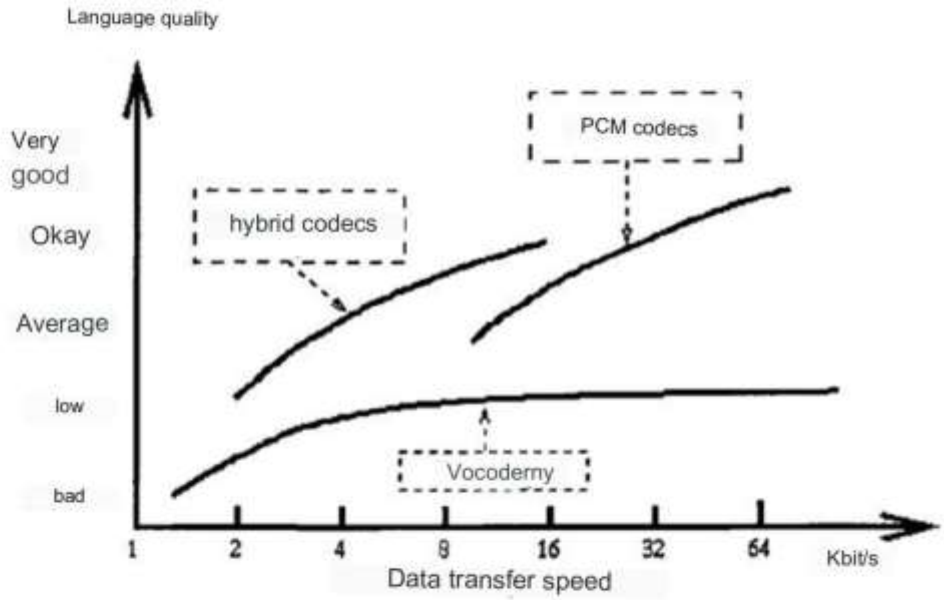

Figure 2.1 - The average rating of speech coding quality for the codec types listed above.

Most of the codecs used in IP telephone are described by the recommendations of the "G" family of the H.323 standard (table 2.2) [7].

Table 2.2 – Characteristics of codecs of the H.323 family

Voice codec	Codec type	Speed coding	Delay at coding
G.711	ADIKM	64 kbit/s	0.75ms
G.726	LD-CELP	32 kbit/s	1ms
G.728	CS-ACELP	16 kbit/s	From 3 to 5ms
G.729	CS-ACELP	8 kbit/s	10ms
G.729a	MP-MLQ	8 kbit/s	10ms
a723.1	ACELP	6.3 kbit/s	30ms

Considering regulatory documents, where it is recommended to ensure acceptable message quality and minimum delays during encoding/ decoding in the SGP (voice message service) equipment, the ADIKM adaptive differential pulse-code modulation method with a speed of 32 kbit/s should be used. This coding method should be considered basic.

On the market, most telecommunication operators use equipment that meets the ITU-T standards for fixed communication networks, which provides the speed of one voice connection of 8 Kbit/s and 16 Kbit/s, as well as for mobile networks of the ETSI standard GSM Full Rate, providing a speed of 13 Kbit/s. Based on these values, it was concluded that the speed of transmission of voice messages of 16 Kbit/s is sufficient for both fixed and mobile networks.

2.2 Analysis of quality-of-service requirements transmission data

In the future, Internet data transmission means the transmission of data messages, except for voice and video traffic. For Internet data transfer, you need consider network software requirements.

GSM networks are expected to evolve according to the latest versions of ETSI standards. At the first stage, the HSCSD (High Speed Circuit Switched Data) technology can be used, which provides data transfer speeds of up to 57.6 Kbit/s in channel-switched mode. A more radical way to increase throughput and expand the services provided, is to use GPRS technology, based on the packet switching mode when transmitting at a speed of up to 115 Kbit/s. This mode will provide more efficient data transfer with charging only for the volume of data transferred. As the next step on the path of evolutionary development, EDGE technology should be used, which provides data transmission at speeds of up to 384 Kbit/s with the implementation of a full set of third-generation communication network services.

To meet the requirements for the quality of Internet data transmission service, the following is required:

Consider software requirements for the network.

Plan the loading of production capacities to be sure of the adequacy of the main throughput.

Use no more than four main classes of traffic.

locally - defined critical class (for critical applications) - transactional and interactive applications with a high business priority.

Transactional / interactive class - client applications - service, applications for transmission of messages.

Volumetric class (Bulk) - transfer of large files, synchronization and database replication, e-mail

Class if possible (Best Effort) - the default class for everything not assigned traffic.

Locally defined critical class of data traffic - the term locally defined is used to emphasize the purpose of this class - that is, to have the highest class of service for each customer for a selected set of their interactive and transactional applications, the highest business priority for them. It is recommended to assign as few applications as possible to this class.

The Transactional/Interactive class is a combination of two similar application types: client-server transactional applications and interactive messaging applications. Response time requirements distinguish transactional from conventional client-server applications.

The data class is bulk (Bulk) - intended for non-interactive applications not critical to packet loss, which usually work in the background. Such applications include FTP, synchronization and replication of databases, distribution of video content or other types of applications in which the user does not have to wait for the results of operations.

The advantage of allocating bandwidth to a volume class of traffic (instead of throttling them) is that applications can dynamically use the free bandwidth and thus increase their performance during quiet periods, which in turn reduces the likelihood the impact of overloads on them.

Best Effort class is the default class for all Internet data traffic. Only if an application has been selected for special treatment will it be removed from the default class. Since many enterprise customers use hundreds, if not thousands, of data applications on their networks (most of which will remain in this class), it is required to allocate adequate bandwidth for the default class. Otherwise, applications that fall into this class will be suppressed. It is recommended to allocate at least 25 bandwidth percent to traffic class support whenever possible.

2.3 Analysis of quality-of-service transmission requirements forvideo message

The integration of television and services associated with it in a multi-service network leads to several problems for service providers. Video (in particular, video on demand), broadcast multichannel television and HDTV require more network resources than voice and data [5].

Video has more diverse QoS requirements than data. Even the most popular Internet data transmission applications can cope with delays (jitters) and some percentage of packet loss. However, video over IP (ATM) has a clear requirement for minimum packet loss, in the range of 10-9, which in practice means that packets can only be dropped because of bit errors and network congestion.

We have two main types of applications video: interactive video and streaming video.

Based on ITU-T and IETF analysis recommendations, we will summarize the main requirements for QoS characteristics when implementing the transmission of video messages.

Requirements for interactive traffic video.

When setting up the interactive, the video (video conferences) recommended:

interactive traffic video must be AF41marked.

Losses no more than 1%.

Unidirectional delay no more 150ms.

Delay in fluctuations no more 30ms.

Minimum bandwidth guaranteed (LLQ) should be equal to the video conference size plus 20 percent (384 Kbit/s video conference session requires a 460 Kbit/s guaranteed priority traffic bandwidth setting).

Since video conferencing includes the audio G.711 codec for speech, it also has requirements for loss, delay, and delay fluctuations corresponding to voice traffic.

Requirements for traffic video streaming.

When setting up video streaming, recommended:

video Streaming (single address or multicast) must be CS4marked.

Losses less 2%.

The delay less 4-5 seconds.

Requirements for guarantees bandwidth depend on the coding format.

Streaming Video is usually unidirectional and therefore, in branch remote offices, routers may not configured to support branch-to-hub streaming video.

2.4 Methods of assessment quality subjective

The perceived video quality is measured using subjective scales methods. A condition for such measurements is the presence of meaning, that is, that there is a between relationship of characteristics physical and the "impact". The final choice of of these methods one for a particular application depends on several factors, such as the content, purpose, and where in the test execution process it is performed.

The following most popular methods [2]:

Double Stimulus Impairment Scale.

Scale of continuous quality assessment DSCQS (English Double Stimulus Continuous Quality Scale).

Video evaluation using the SCACJ comparative scale (Stimulus Comparison Adjective Categorical Judgment).

The subjective method of evaluating SAMVIQ video quality (Subjective Assessment Method for Video Quality evaluation).

MSUCQE continuous quality evaluation (MSU Continuous Quality Evaluation).

Traditionally, subjective video quality is determined by expert evaluation and calculation of the mean MOS score (Mean Opinion Score) from 1 to 5 (ITU scale), where 1 is the worst and 5 is the best received video quality [7]. This approach, however, requires some expert skills and is therefore not used in automated systems.

The following is required for testing subjective:

Choose a sequence video for testing (usually a video of about 8 - 10 s is used to prevent distraction of the experts' attention and reduce the total time of the experiments).

Select the settings of the compared video processing systems.

Choose a testing method.

Invite enough experts (at least 15 are recommended).

Get final grades based on the opinion of experts.

2.5 Objective methods assessment

Methods of assessment objective of video digital quality are divided into three categories. First category, the quality assessment occurs when comparing the decoded video sequence with the original one. The objectivity methods is that there is no direct human intervention. The calculation of the distortion is carried out automatically.

The second category includes methods that compare the characteristics of the original and decoded video sequences.

The methods of the third category evaluate only the decoded video.

Quality video is measured by root square mean error and peak to-noise signal ratio between outputs I(x, y) both coded and subsequently decoded Ĩ(x, y) frames video. The greater the between difference I(x, y) Ĩ(x, y), which is expressed in decibels, according to the visual logarithmic sensitivity of the human eye. PSNR is usually calculated for luminance since human eyes are most sensitive to this component.

Consider a video sequence consisting of N frames with a resolution of DxxDy pixels (for example, for CIF (Common Intermediate Format) x = 288, y = 352, for QCIF (Quarter CIF) x = 144, y = 176). Let I(n, x, y), n = 0, ... N − 1, x = 1, ... Dx, y = 1, . . . Dy denotes the brightness (Y component) of a pixel with coordinates (x, y) in video frame n. At the same time, there will be fair formulas:

$$MSE_n = \frac{1}{D_x D_y} \sum_{x=1}^{D_x} \sum_{y=1}^{D_y} [\mathrm{I}(\mathrm{n}, \mathrm{x}, \mathrm{y}) - \tilde{\mathrm{I}}(\mathrm{x}, \mathrm{y})]$$

$$PSNR_1 = 10 log_{10} \frac{255}{MSE_n}$$

The values of MSE and PSNR are indicators reference of quality video.

They are both needed to determine the quality video of the video source frames in addition to the decoded frames video. At the same time, these indicators consider the evaluation of video quality without the actual stream of binary signals. In the table 2.3 presents the correspondence of subjective (MOS) and objective (PSNR) assessment.

However, both indicators do not correspond to the quality subjective of the restored image and do not properly reflect small differences in intensity degradation.

Calculation of distortion in motion estimation, based on the pixel error rate, does not fully correspond to visual human perception. It has been proven that SSIM can provide a better display of the value of image distortion than currently used PSNR or MSE.

Table 2.3 - Correspondence of MOS and PSNR

PSNR (dB)	MOS (%)	Quality according to the ITU scale	Image degradation
More than 37	81-100	5 - Excellent	Not noticeable
31-37	61-80	4 - Good	Noticeable, but not annoying
25 - 31	41 - 60	3 - Satisfied	Somewhat annoying
20 - 25	21 - 40	2 - Bad	Annoying
Less than 20	0 - 20	1 - Very bad	Very annoying

The index structural similarity of SSIM is the closest to perception human of the video sequence received. Its application uses visually perceived structural distortion, while most other proposed methods are error sensitivity based. The index SSIM determines the similarity of three image components: brightness, contrast, and structural similarity [11].

2.6 Image metrics quality and indicators

Several requirements can be made to the metric:

The relevance of the metric. The subjectively "better" video fragments should correspond to the "better" metric value. This characteristic can be measured quantitatively, for example, using the correlation Pearson coefficient.

Metric consistency. The "deviation" of values from values predicted based on subjective metrics should not be large. It is calculated as follows. First, several evaluations subjective of the video fragment are performed. The statistically results are processed and the MSE of the estimates is found. Then the values of the metrics objective are calculated, and their number is found, which are more than twice the MSE value from the subjective estimates.

There are six classes of objective image quality metrics:

Pixel.

Correlational.

Contour.

Spectral.

Contextual.

Consider the peculiarities of human vision.

2.7 Quality methodology assessment

The use of the described metrics and indicators for quality evaluating reproduction video is difficult in practice due to the lack of a priori information on many related factors.

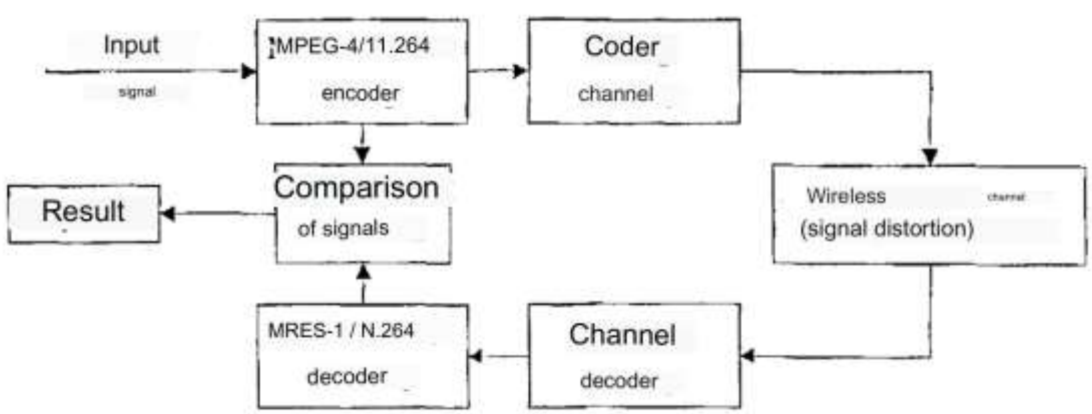

Figure 2.2 Structural diagram of the evaluation algorithm

To quality evaluate of decoding when transmitting video in networks wireless, it is suggested to use the methodology:

The output video sequence is coded by a codec.

The video stream will turn into a transport stream (modulated).

The traffic flow is transmitted through a wireless network.

The distorted stream is decoded in the receiver.

The decoded video stream is evaluated.

The structural diagram of the video evaluation algorithm during transmission over a wireless network, which implements the described methodology, is presented in Fig. 2.2.

In the simulation, publicly available traces of one-level and two-level coded video can be used with the use of temporal and spatial scalability [5].

The wireless network simulator can be implemented in the MATLAB Simulink software environment, WiMAX with mandatory implementation of all OFDM elements of the physical level defined in the IEEE 802.16-2004 standard.

3 COMPARATIVE ANALYSIS OF DELIVERY TECHNOLOGY VIDEO CONTENT

An urgent problem for telecommunications companies is the outflow of subscribers to competitors. It can be solved in several ways. One of them is quality improve the of services already provided by the operator, however, these actions require large financial costs, and the return from them may not be so significant. Another way is to put into operation a service unavailable to subscribers of another operator [2].

Most of the large companies in the telecommunications market provide services in the form of a package of services - data, voice, and video transmission, called Triple Play. When mobility support is added, the package offer is called Quadruple Play, which ultimately gives the operator the opportunity to implement single service packages in wired and wireless networks.

Now, large operators are operating heterogeneous networks [1] with the integration of high-speed switching systems. The network Internet itself can serve as the basis for the services described above (except for mobility), such services were named OTTV (Over the Top Television) [3]. At the same time, the communication operator acts only as an intermediary, providing an environment for data transmission, which threatens to turn it into a "bit pipe", depriving it of a significant part of the profit.

For purpose, the following tasks will be solved:

Possible strategies for solving the problem are determined, including the choice of delivery technology to the subscriber's device.

Advantages and disadvantages are considered for each scenario.

The choice of the most optimal scenario from the operator's point of view is substantiated. Let's list the possible scenarios.

The operator can organize a separate network, the purpose of which will be only the provision of video transmission services. Considering the support of subscriber mobility, DMB (Digital Media Broadband), MediaFlo (Media Forward Link Only) and DVB-H (Digital Video Broadcasting - Handled) technologies can be used for this. The last of them is the most popular in implementation [2].

It is possible to provide all services from the package in one wireless access network created based on 3GPP LTE (3rd Generation Partnership Project Long Term Evolution) or WiMAX (Worldwide Interoperability for Microwave Access) technology.

The operator himself can act as an operator of OTTV services.

3.1 DVB-H Technology

The main aspect that determines the efficiency of the transmission of moving images for mobile terminals is energy saving. It is the core of DVB-H technology. The ETSI EN 300 192 specification defines the main technical solutions for achieving effective energy saving parameters:

Time quantization: data in DVB-H is transmitted discretely, in packets with high transmission speed. At the same time, in pauses between packet transmissions, the mobile terminal switches to standby mode, thanks to which energy savings are up to 90%. Time quantization provides a smooth transition to another cell during the waiting period.

Reed-Solomon (RS) coding - protection against data loss is provided by the method of direct error correction (FEC), which allows you to preserve the quality of the signal even with the loss of many packets.

Service Addressing Table (INT) - service data about relevant services are transferred in the Service Addressing Table (INT). Based on the information in this table, the distribution of receivers is carried out and the delivery of the service for its playback on the corresponding mobile terminal is ensured.

4K modulation mode - in addition to 2k and 8k modulation, 4k modulation is installed for the DVB-H standard, which has optimal characteristics for planning mobile networks. It has the most appropriate parameters for effective reception at high speeds.

TPS Bits - Transport channel parameter signaling or TPS bits accelerates the detection of the video image transmission service by the receiver.

DVB-H is part of the DVB (Digital Video Broadcasting) family of standards. Therefore, the receiver of DVB-H signals can receive DVB-T signals. This is convenient for the subscriber, but it will not bring income to the operator since the subscriber will be able to use free-to-air digital television services at any time.

Support for mobility in the DVB-H standard is expressed in the use of MPE-FEC (Multi-Protocol Encapsulation Forward Error Correction) coding, as well as in the transmission of signals in time-varying packets. Depending on the type of modulation, simultaneous transmission of 25 streams video with a resolution of 240x320 mm is possible, which is unacceptable quality terms today's market.

This technology does not initially support interactivity, that is, the possibility of two-way data transmission. This problem is solved thanks to the addition of IPDC (Internet Protocol Data Cast), which allows the use of data transmission network channels (for example. UMTS (Universal Mobile Telecommunications System)) to exchange service information.

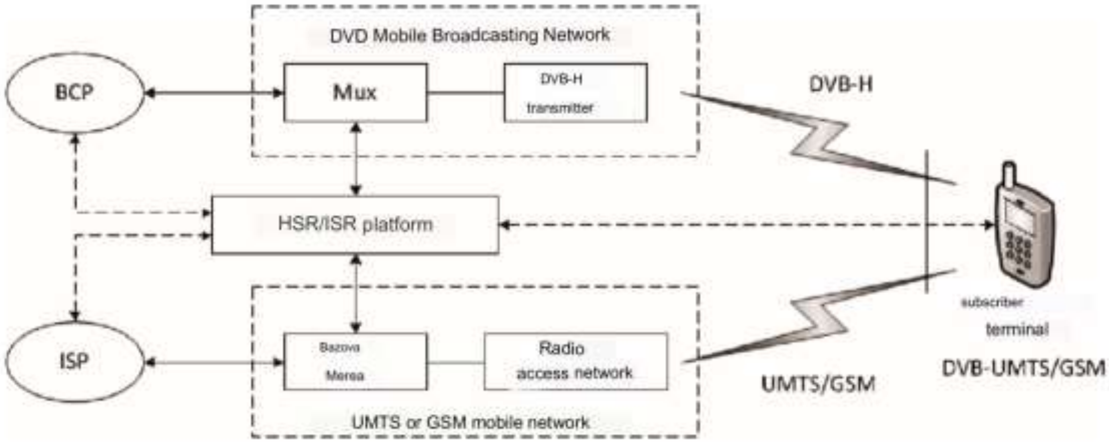

Figure 3.1 – BCP ISP network operation scheme

BSR (Broadcast Content Provider) - broadcasting operator.

ISP (Service Internet Provider) - Internet access network operator (same as HRV).

BSP / ISP-platform - a platform for coordinating the BSR and ISP to provide multimedia broadcasting.

MUX - multiplexer [5,6].

The main drawback of this technology for the subscriber is the limitation of models of subscriber terminals that support DVB-H, for the operator - the need to use two networks to support interactivity. The logical option in such a case is the use of one network - a wireless data transmission network.

3.2 Technology WiMAX

Essentially two standards - IEEE 802.16d-2004 and 802.16e-2005, for fixed and mobile access with a maximum access speed of 75 and 40 Mbit / s, respectively. The organization WiMAX Forum [5] deals with the promotion and standardization of WiMAX.

MBS (Multicast and Broadcast Service) is used in WiMAX to organize broadcast and group services, including video content delivery.

Thanks to this, low power consumption, fast channel switching and support for networks with one carrier frequency SFN (Single-Frequency Network) are supported when transmitting the same content using several base stations at the same time, which ensures reception in the entire coverage area. Figure 3.2 shows the operation of the WiMAX network with the integration of the MBS service.

The diagram shows the following elements:

ASP (Application Service Provider) - application service provider.

CSN (Connectivity Sendee Network) - service provision network.

ASN-GW (Access Sendee Network Gateway) - a node in which traffic and signaling messages from base stations are combined; - MBS-controller - functional unit of implementation of MBS service.

BS (Base Station) - base station.

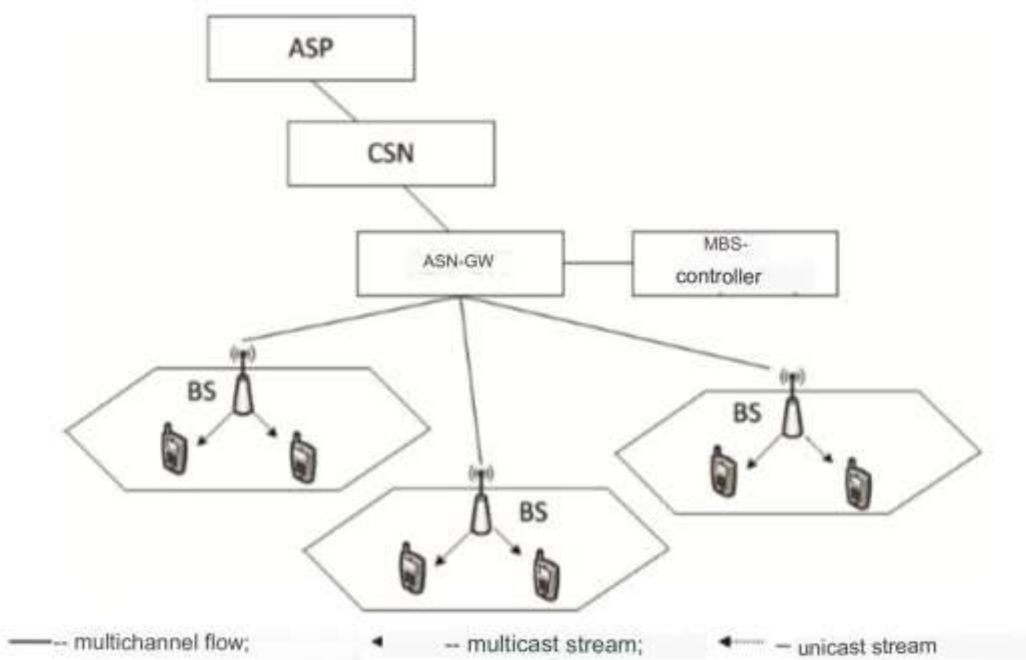

Figure 3.2 – WiMAX network operation scheme with MBS service integration

3.3 Technology LTE

LTE technology as a standard officially began at the end of 2004 (Fig. 3.3). The main goal of research at the initial stage was the choice of a physical layer technology that would be able to provide a high data transfer rate. Two options were proposed as the main ones: the development of the existing W-CDMA radio interface (used in HSPA) and the creation of a new one based on OFDM technology. As a result of the research, the only worthy technology turned out to be OFDM, and in May 2006, 3GPP created the first specification for the Evolved UMTS Terrestrial Radio Access (E-UTRA) radio interface. The first, preliminary LTE specifications were created as part of the so-called 3GPP Release 7. In December 2008, a version of the 3GPP standards (Release 8) was approved, which fixes the architectural and functional requirements for LTE systems. In the middle of 2009, the first experimental systems based on LTE appeared, and in 2010 - the first commercial networks.

Compared to previously developed 3G systems, the LTE radio interface will provide improved technical characteristics. In LTE, the bandwidth can vary from 1.4 to 20 MHz (according to earlier sources - from 1.25 MHz), which will allow meeting the needs of different communication operators with different bandwidths. At the same time, LTE equipment must simultaneously support at least 200 active connections (that is, 200 phone calls) for each 5-MHz center.

It is also expected that LTE will improve the efficiency of the use of the radio frequency spectrum, that is, the amount of data transmitted in each frequency range will increase. LTE will make it possible to achieve significant aggregate data transfer speeds - up to 50 Mbit / s for the uplink (from the subscriber to the base station) and up to 100 Mbit / s for the downlink (from the base station to the subscriber) (in the 20 MHz band).

At the same time, support for connections for subscribers moving at a speed of up to 350 km/h must be ensured. The coverage area of one BS is up to 30 km in normal mode, but it is possible to work with centers with a radius of more than 100 km. Multiple antenna MIMO systems are supported.

From a point technical of view, LTE is a further development of the UMTS standard networks (which, in part, explains their popularity), while the technology is constantly being improved and the transition to the next release, which is called LTE - Advanced, is already planned. The maximum data rate in the downward direction, from the base station, for LTE Rel. 8 is 100 Mbit/s, in LTE Advanced the fulfillment of the requirements of IMT-Advanced (International Mobile Telecommunications), i.e., a speed of 1 Gbit/s [5] is included.

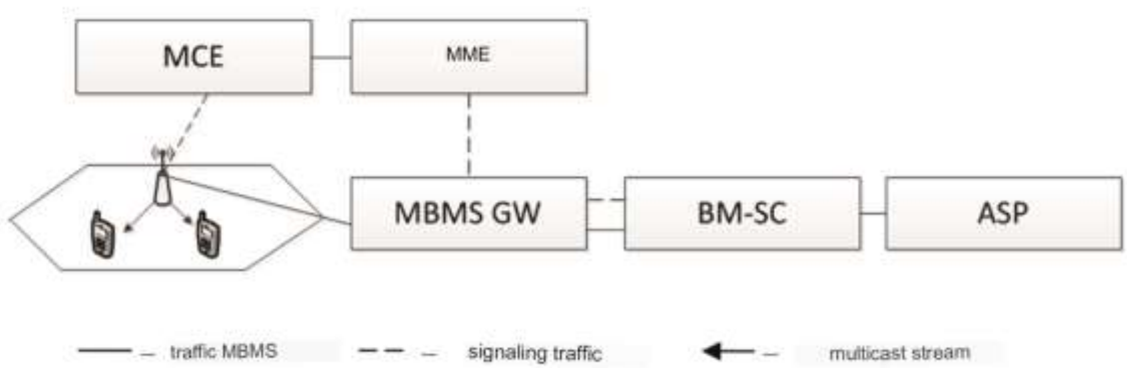

Figure 3.3 – Operation of the LTE network by the MVMS service

The service for video delivering content in networks LTE is called MBMS (Multimedia Broadcast and Multicast Sendee), which is functionally like the MBS service in WiMAX. BM-SC (Broadcast Multicast Sendee Center) acts as an architectural element of MBMS implementation. Figure 3.3 shows the operation of the LTE network with the integrated MBMS service.

The diagram shows the following elements:

ASP (Application Sendee Provider) - application service provider.

BM-SC - language services center.

MBMS GW (Multimedia Broadcast and Multicast Sendee GateWay) - gateway MBMS, which combines traffic and signaling messages.

MME (Mobility Management Entity) - mobility management unit.

MCE (Multi-cell / multicast Coordination Entity) - there are many coordination nodes cellular / multicast transmission.

It should be noted that the MBMS service, like MBS, can be used for

other purposes related to broadcast or bulk data delivery, for example, for software updates or emergency notification [8].

3.4 Internet television systems

Since neither the use of second-generation television technologies, nor attempts to implement fourth-generation telecommunication standards (LTE), do not solve the main task of modern telecommunications - the creation of a fully interactive video delivery content system. Therefore, one of the most appropriate options for solving the problem is the use and development of technology IPTV, which combines the technologies of digital television and the Internet in the sphere media. Undisputed IPTV include advantages:

The presence of full feedback with the subscriber, i.e., the both use group (broadcast / multicast) and single (unicast) broadcasting modes (the user can independently choose the content and the time of its viewing).

The possibility of flexible regulation of the volumes of transferred data.

The possibility of transferring information to both stationary and mobile devices.

But traditional IPTV also has several disadvantages, for example:

The need to allocate channels for different types of users in the general traffic low.

The dynamic lack of adaptation technology, which allows the user to continuously video content deliver (movies, files) with resolutions different and in containers different to terminals different, i.e., the lack of the ability to quickly quality change (bitrate, signal delays, etc.) of the content depending transmitted on the capabilities technical of the devices receiving. To solve these shortcomings, a new technology of broadcasting IPTV - OTTV (Over the Top Television).

OTTV Services

The main between differences technology OTTV and IPTV "traditional" technologies are:

The presence of technologies that allow adapting the content transmitted to the network bandwidth (Adaptive Bit Rating - ABR), based on "Adaptive HTTP Streaming" (Adaptive HTTP Streaming), which determines the bandwidth and switches the to the optimal flow broadcast.

The possibility of delivering content video to all currently existing subscriber devices, including and support for polyscreens.

When using cellular communication networks as CAD, there is no strict binding of the subscriber to the limited content of the operator's network.

The possibility of quick delivery of content video to small settlements with acceptable cost and quality services.

The scheme of providing services OTT includes the following components (Fig. 3.4):

The transcoding system transforms the original video content into suitable for over network transmission.

CA / DRM.

VOD.

Network scaling to serve any number of subscribers by adding nodes.

Reduction of the waiting time before the start of playback content.

Savings on trunk traffic for the operator [9].

All the above-mentioned disadvantages do not allow us to use advantages all of technology OTTV in full for the delivery of modern content video. In countries with developed communication infrastructure, for example, in the USA, OTTV technology is very popular.

The delivery of content video in the format of Ultra HDTV Spatial Imaging 3DTV will probably be based on the foundation of UPT technology: universal data transmission network (UDN) based on the general hybrid HBBTV transport (DVBT2 / C2 / S2 / H + Wi-Fi + LTE + OTTV) c general coding of video content. For this purpose, the so-called "Worldwide Broadcast Roaming" - (WBR), i.e., a base for accessing content in different environments and broadcasting zones that use different standards, between universal subscriber terminals, is already being formed.

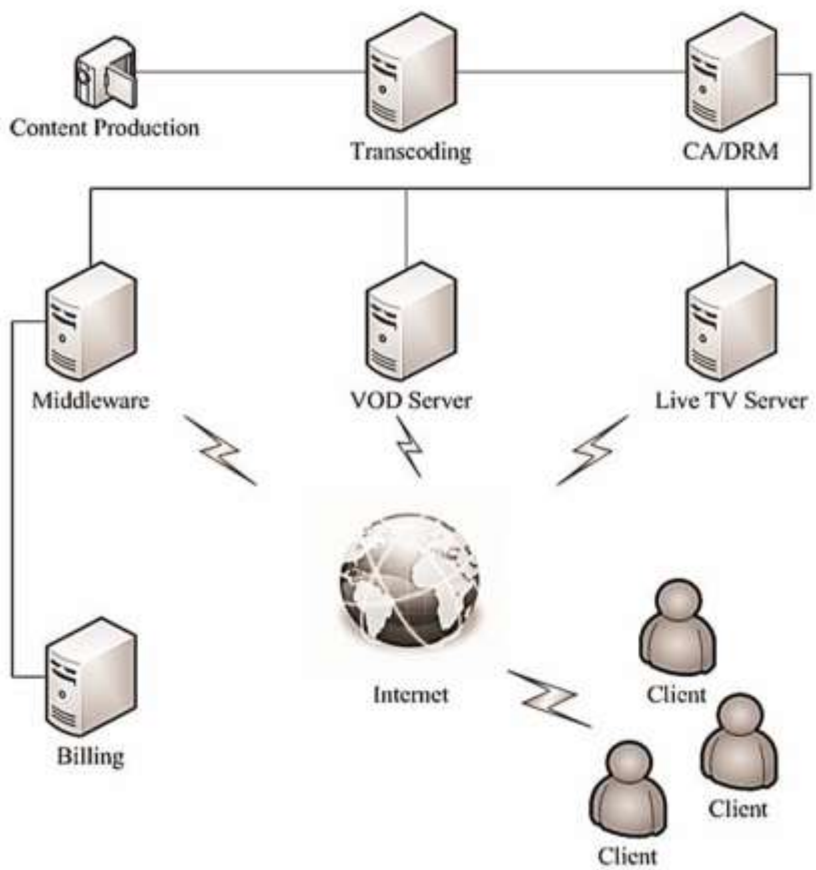

Figure 3.4 – Scheme of provision of OTTV services

Let's note another technology, like CDN, designed to save trunk traffic, called Torrent-TV or BitTorrent Live. It works in p2p (peer-to-peer) networks, which are called peering or peering. Each node in such a network act as both a client and a server. Such an organization allows maintaining the system's performance with any number of nodes.

Speaking about the transmission of content video, by sending a request, the subscriber device content receives simultaneously from several sources that the request sent to it, and time same at the these sources receive content from the subscriber device. Accordingly, for the system to work as effectively as possible, as many viewers as possible are needed. However, the disadvantage of such a system is obvious - the operator cannot guarantee the quality of the service, that is, the optimal number of viewers' devices [10]. Thus, both Torrent-TV and CDN are most suitable for the transmission of video content on demand, i.e., VOD services.

Active informatization of society has led to explosive growth in sales of content-consuming devices using wireless networks. Even personal computers were not as personal for users as smartphones and tablets have become. A modern subscriber wants to receive all possible services on his device.

The analysis of the architecture of the considered technologies allows us to conclude that LTE WiMAX transmission data networks are adapted most to conditions modern. Already during creation, the possibility of effective delivery of content video is built into them, but the potential capacity of LTE networks, due to speed higher in the downstream direction, speaks of their advantages. At the time same, technologies like DVB-H are currently unpromising.

However, one cannot fail to notice the potential of OTTV services, and especially their convenience for the subscriber. For the operator, they can become either a threat or a competitive advantage.

Thus, the analysis of modern video content delivery technologies and their development prospects showed:

A steady trend of increasing the resolution of information display means, and, therefore, the density of the transmitted data stream, up to holographic technology at the level of at least "16K". At the time same, the complex task of delivering content video of this resolution is impossible without the introduction of fundamentally new algorithms for compression (coding) of information and the creation of new networks for delivering this content.

Today, on the services market for realistic content delivering to the user (HDTV level), television technologies (DVB S2 / T2 / C2 / H / SH) are more common, but in a few years the leading technologies will be telecommunications (LTE) and Internet technologies (OTTV).

The urgency of integrating video cameras into television terminals in order to develop means of communication between subscribers, as well as contactless management of terminals.

When forming high-resolution color three-dimensional images, it is necessary to solve the problem of reducing the energy consumption of devices and their operational characteristics. One of the most effective ways to solve this problem is the introduction of flexible LED matrices based on organic LEDs OLED, PLED, etc.

3.5 Research of the transmission of short streams video in ip networks

Today, the share of video traffic on the Internet is about 82% and, according to the forecasts of Cisco Systems, this could only keep growing further.

In addition, the share of mobile traffic and the share of HD and Ultra HD video formats is growing. A distinctive feature of mobile (wireless) devices is that data transfer rates can change significantly over time. This is due, firstly, to the properties of the transmission medium: the change in signal strength during the movement of the device, the presence of random interference; and secondly, a variable number of devices competing for a wireless channel. At the same time, when transmitting video streams, it is necessary to maintain a data transfer rate of a large, specified threshold.

The standard describes the client-server interaction protocol but does not define a specific decision-making algorithm for the client, giving developers the opportunity to choose the algorithm most suitable for solving a specific task. In this regard, since the beginning of the development of the standard, many different approaches to the implementation of this algorithm have been presented in the literature.

In addition, software manufacturers present several proprietary algorithms, including those implemented in multimedia players Microsoft Smooth Streaming, Apple HTTP Live Streaming (HLS), and in the open player from Adobe Systems: Open-Source Media Framework (OSMF).

This section considers the transmission of video streams in a wireless network, including in the presence of background web traffic, and investigates the operation of two algorithms that use different approaches when choosing the distribution capacity (quality).

The choice of wireless network technology is because today in networks using this technology, about 40% of traffic is generated and this share continues to grow [2].

The section examines in detail the scenario in which the DASH client does not watch the entire video file but looks for fragments of interest to it. In other words, it regularly interrupts playback and requests a new video fragment that has not yet been downloaded. Such customer behavior is quite common. For example, according to research by Visible Measures, up to 20% of all video file views by users have a duration of less than 10 seconds [5].

In addition, services such as Snapchat, TikTok and Video on Instagram have become widespread, allowing users to share short video files on social network.

standard MPEG-DASH

The MPEG-DASH working group includes dozens of industry representatives, including Microsoft, Adobe, Google, Sony, Netflix, Qualcomm, Ericsson, Samsung and other companies.

Bitrate selection algorithm

The decision-making algorithm on the client side should be aimed at achieving the maximum quality of the video image viewed by the user (Quality of User Experience, QoE). The following factors affect QoE:

Waiting for the video fragment to start playing.

The number and duration of pauses when playing a video fragment.

The resolution of the video fragment being viewed.

The frequency of changes in the distribution capacity and their amplitude.

Examples of two algorithms using different approaches to bitrate selection are considered in the bachelor thesis. The main difference between these algorithms is whether information about the current state of the buffer is used when deciding on the choice of bitrate.

Instant algorithm

One of the simplest bitrate selection algorithms is the Instant algorithm. This algorithm selects the bitrate of each subsequent segment slightly less than the bandwidth of the connection measured over a period. The algorithm uses 3 parameters:

The minimum volume of the bmin buffer (in seconds), at which the playback of the video fragment begins.

Sensitivity coefficient of the algorithm β ($\beta \leq 1$), which determines the ratio between the measured bandwidth and the selected bitrate.

The duration of the time interval Δ, for which the average bandwidth of the connection with the server is calculated.

The essence of the algorithm is as follows. After the download of the i-th segment is completed, the average bandwidth of the connection when downloading this segment is estimated:

$$\rho_s = \frac{r(i)\tau}{t_f(i) - t_s(i)}$$

where r(i) is the bitrate of the i-th segment, and tf (i) and ts(i) are the start and end times of segment download, respectively.

The average bandwidth of the connection in the interval from t1 to t2 is determined as

$$\rho(t_1, t_2) = \frac{\Sigma_i \rho_s(i) |[t_f(i), t_s(i)] \cap [t_1, t_2]|}{\Sigma_i |[t_f(i), t_s(i)] \cap [t_1, t_2]|}$$

where $|[x_1, x_2]| = x_2 - x_1$ is the duration of the interval $[x_1, x_2]$.

The decision to select the bitrate of segment i+1 is made as follows. Playback of the video fragment on the client side does not start until the buffer volume is less than bmin. To reduce the delay of the start of playback, when the buffer size b < bmin, the bitrate of segment i+1 is chosen to be equal to the minimum possible bitrate:

$$r(i+1) = r_{min} = min\, r_{r \in \Re}$$

where R is the set of bitrates available on the server according to MPD.

In all other cases, the maximum possible bitrate is selected, which does not exceed the estimated bandwidth ρ(t−Δρ, t) multiplied by the coefficient β:

$$r(i+1) = max\, r, r < \beta\rho p(t - \Delta\rho, t) \quad r \in \Re$$

Miller’s algorithm

This algorithm aims to maintain the volume of the buffer within the specified limits. The algorithm reduces the bitrate if the buffer volume has become less than the lower limit of blow and increases it if the accumulated buffer is of a certain volume and the buffer volume grows quite quickly. This algorithm changes the bitrate in steps, i.e.

$$r(i+1) \in \{r^{\uparrow}(i), r(i), r^{\downarrow}(i)\}$$

where $r\uparrow(i)$ and $r\downarrow(i)$ are the next bitrate from R that is greater (less) than the current bitrate r(i), if any.

In addition, we denote the maximum and minimum bitrate with R as rmax and rmin, respectively.

The algorithm uses the following set of parameters:

Buffer volume threshold values bmin≤ blow < bhigh.

Sensitivity coefficients of the algorithm α1, ..., α5.

Duration Δ of the bandwidth smoothing interval.

When choosing next requested segment bitrate, the algorithm receives on the input of the dynamics of the change in the volume of the buffer b(t) and the statistics of the bandwidth of the connection when loading the previous segments ρs(i), which is determined by formula (4.1). The work of the algorithm is divided into 2 phases:

Buffer initial loading phase.

The phase of normal operation.

The main task of the phase of the initial loading of the buffer is to provide the balance between reducing the start delay of playback, accumulating enough buffer volume to avoid playback pauses, and matching the selected bitrate of the video stream with the available bandwidth of the connection. The choice of bitrate in the initial phase depends on the current buffer size b(t).

If $b(t) \leq bmin$ and $r\uparrow(i) \leq \alpha 2\rho(t-\Delta, t)$, then $r(i+1)= r\uparrow(i)$.

If bmin < b(t) ≤ blow and r↑(i) ≤ α3ρ(t−Δ, t), then r(i+1) = r↑(i).

If b(t) > blow and r↑(i) ≤ α4ρ(t−Δ, t), then r(i+1)= r↑(i).

If b(t) ≥ bhigh, then the next segment request is postponed until the buffer size is less than bhigh. The reduction of the buffer volume will occur because the user will watch part of the video fragment.

In all other cases, the algorithm does not change the bitrate r(i+1) = r(i).

Coefficients α2, α3, α4 are selected so that with an increase in the volume of the buffer, the aggressiveness of the algorithm in relation to the choice of bitrate increases (Table 3.1). The initial phase lasts as long as all the following conditions are met:

The current bitrate is less than the average bandwidth r(i)≤α1ρ(t−Δρ,t).

The growth of the buffer volume b(t2) ≥ b(t1) continues, when t2 > t1.

The maximum available bitrate rmax has not been reached.

Table 3.1 - Parameters of bitrate selection algorithms

Instant	P	0.95
	D, c	10
Algorithm Miller	b_{low}, C	10
	b_{highi}, C	50
	a_1	0.75
	a_2	0.33
	a_3	0.50
	a_4	0.75
	a_5	0.90
	Δ, c	10

The task of the normal operation phase is to maintain the buffer at the level bopt = (b_low + b_high)/2 and change the bitrate according to the current buffer size.

If b(t) < bmin, then the smallest available bitrate is selected, r(i+1) = rmin.

If bmin < b(t) ≤ blow and r(i) ≥ ρs(i), then r(i+1) =, r↓(i).

If b(t) > bopt, but r↑(i) > α5ρ(t−Δρ,t), then the client does not request a new segment until b(t) > bopt.

If $b(t) > bhigh$ and $r\uparrow(i) \leq \alpha 5\rho(t-\Delta\rho,t)$, then $r(i+1) = r\uparrow(i)$.

In all other cases, the algorithm does not change the bitrate $r(i + 1) = r(i)$.

Experimental research

We will conduct a study of the bitrate selection algorithms discussed above using the simulation environment NS-3 [19].

NS3 is free software distributed under the GNU GPLv2 license and is intended for research as well as educational use. NS3 source codes are open for research, modification, and use.

NS3 is a very flexible and at the same time powerful modeling tool by using C++ as the built-in model description language. Similarly, in addition to C++, Python can be used. Both languages in the simulator are equal and are accepted for describing models of telecommunication systems.

Thanks to a very large and flexible API, as well as due to the completeness of the documentation of the software interfaces, the model developer is practically not limited by anything. He is given the opportunity to build his own models of any complexity, and, thanks to the GNU GPLv2 license used, to change and supplement already existing models that are included in the software package.

In NS3, models of wireless types of networks have been developed, which allow modeling even with moving objects in three-dimensional space. Developed by FrameWork called FlowMonitor, which provides very flexible methods of collecting information from simulated active network devices and communication channels. The simulator does not have its own graphical interface, but NetAnimator and PyViz projects are used for model visualization.

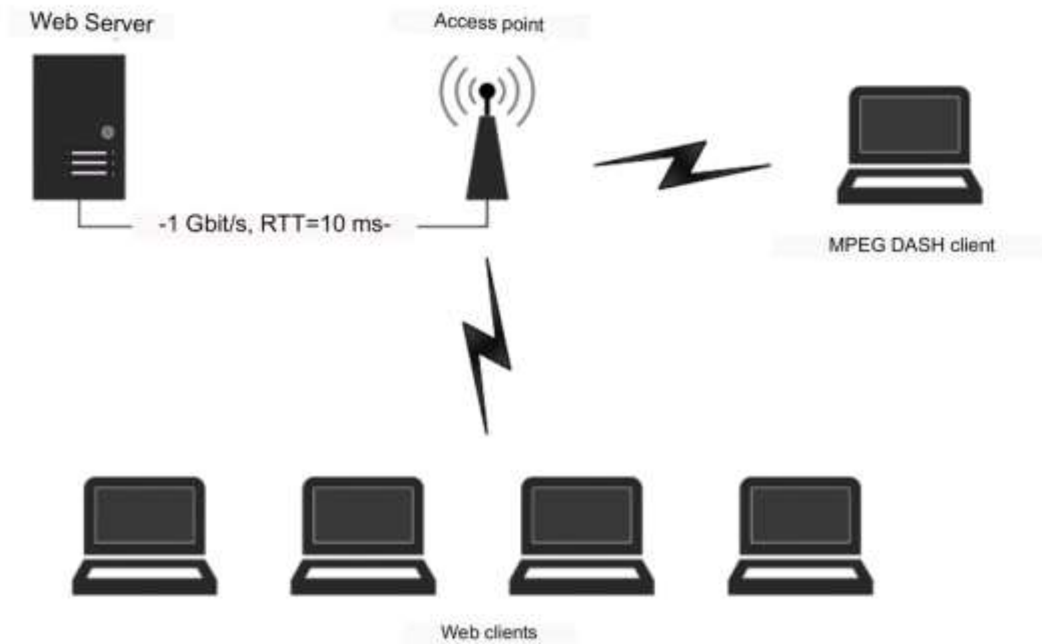

Figure 3.5 – Topology of the experimental network

Consider the scenario (Fig 3.5), the web server is connected to the access point using a wired connection with a bandwidth of 1 Gbit/s and a signal propagation delay of RTT = 10ms. The Wi-Fi network access point and client devices at the physical level use the IEEE 802.11n protocol. The Minstrel algorithm is used as the transmission speed control algorithm. Client devices are located 36 meters from the access point. As previous experiments have shown, with this arrangement of devices, the total throughput at the application level is S0 = 12.7 Mbit/s.

Customers generate the following traffic. One device is a DASH client viewing video files from the server. The remaining Nweb = 4 client devices download web pages from the server via an exponentially distributed time with a given average TinterReq.

The size of the web pages to be loaded is chosen in such a way that, on average, Nweb devices have a fixed share of the bandwidth of the Load channel:

$$L = \frac{s_0 + T_{interReq} + Load}{N_{web}}$$

The DASH client views video files as follows. After a random time with a given distribution, the playback of the current video fragment ends, and the download of a new video fragment begins. Thus, instead of watching a video from the beginning to the end, the client searches for the most interesting episodes. In this work, the time between viewing two consecutive video fragments has an exponential distribution with a mean TinterClick, which is bounded from below by the duration of one segment τ.

To minimize the delay when switching to a new video fragment and not to download packets from the previous video fragment into the buffer, the TCP connection with the server is forcibly closed, and a new connection is opened, with which the download of the new video fragment begins.

The duration of the experiment Tsim = 1500 s. For each experiment, 5 independent runs of the simulation model were conducted. Other parameters of the experiment are presented in table. 3.2.

Table 3.2 - Experiment parameters

$T_{interReq}$, c	30
$\Re$, Mbit/s	0.3, 0.5, 1.0, 1.8, 2.5, 5.0, 8.0, 16.0, 24.0, 40.0
Load, %	28, 65
τ, p	2

Analysis of research results

Figures 3.6–3.12 present the results of simulation modeling for the two studied algorithms at the values of the parameter bmin = {2; 5} and load values generated by web clients, Load = 28%.

The dashed line in Fig. 3.6 shows the total share of the bandwidth of the channel, which is accounted for by web clients. The value measured in the experiment turned out to be close to the initially specified load of 28%.

The solid line in the same figure shows the share of bandwidth that was allocated to the DASH client.

From the obtained results, it follows that the DASH client is unable to use the remaining bandwidth of the channel. The reasons for this effect are as follows. First, for both algorithms, the bandwidth of the connection may be unused due to the delay in determining the available bandwidth of the connection by the TCP protocol, namely, the operation of the TCP Slow Start mechanism. The influence of this factor is especially significant with small TinterClick values (in this case, the DASH client often establishes a new connection) and a small load generated by web clients.

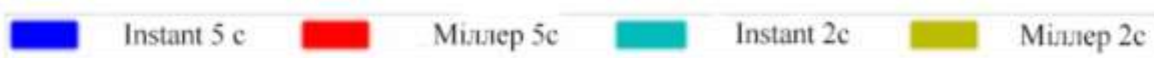

Figure 3.5 – Notations for 3.6 – 3.12

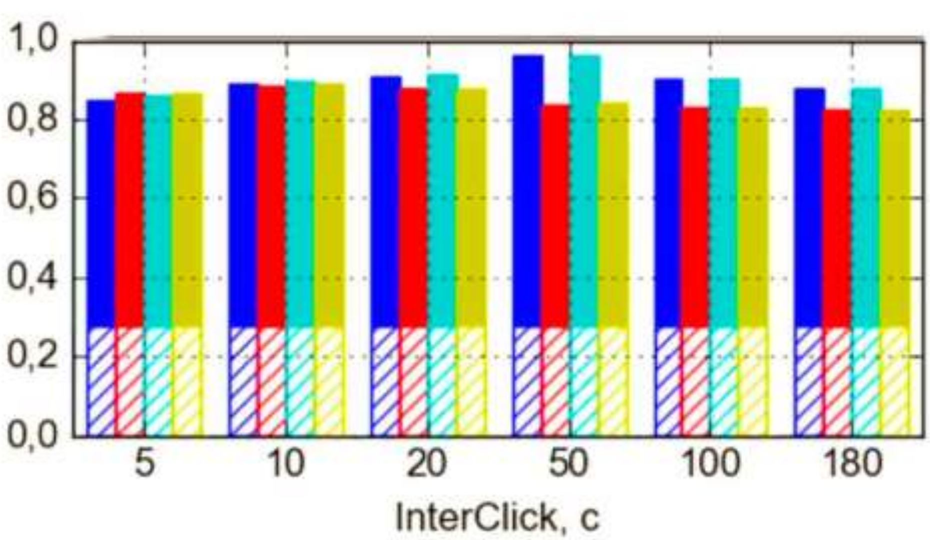

Figure 3.6 – The share of the bandwidth of the channel, which falls on the DASH client for Load = 28%

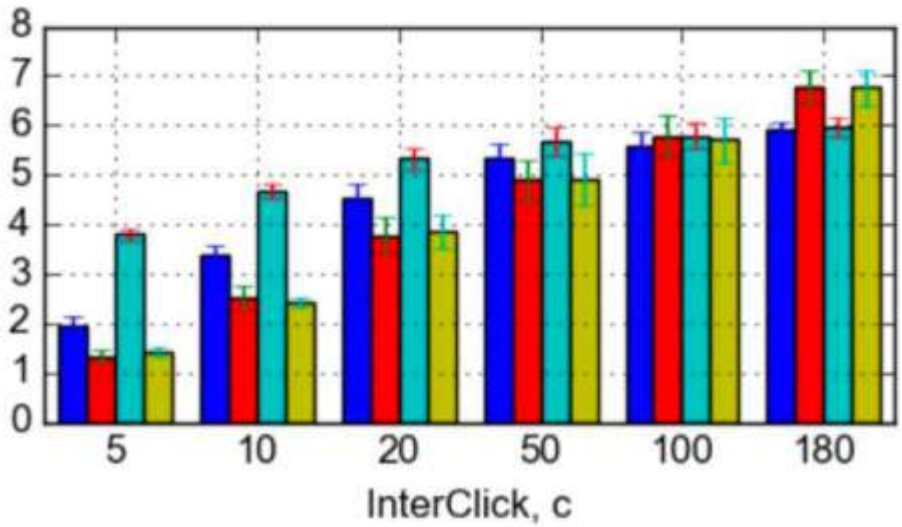

Figure 3.7 – Average bitrate of a video stream viewed by a DASH client for Load = 28%, Mbps

Secondly, the insufficient use of bandwidth by Miller's algorithm is explained by the discreteness of the available bitrate set and the fact that the algorithm stops downloading segments after reaching a certain buffer size. From the graph of the average bitrate of the played video fragments, on short video fragments (at low TinterClick values) the average bitrate is much lower than the channel bandwidth available to the DASH client.

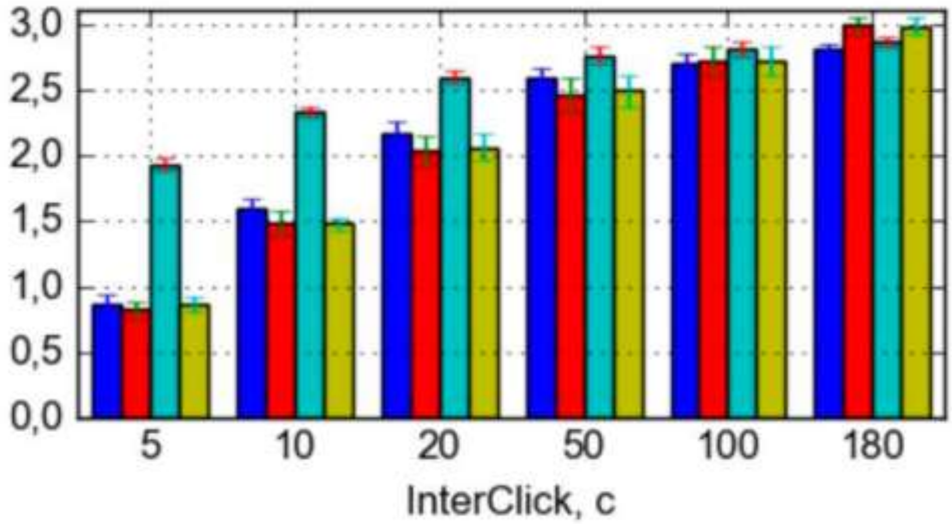

Figure 3.8 – Average bitrate logarithm for Load = 28%

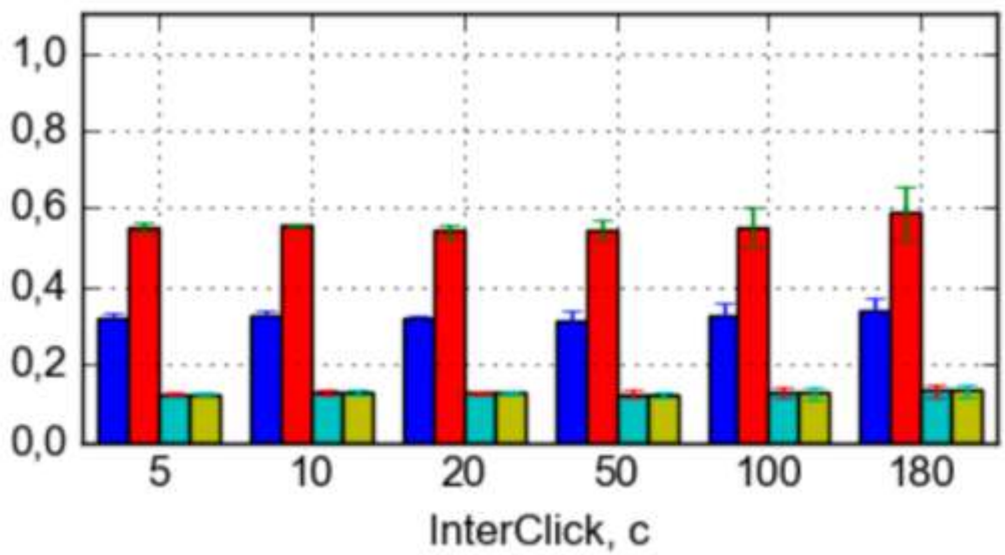

Figure 3.9 – The average delay of the start of playback of a new video fragment for Load = 28%, p

The main reason for this effect is the lack of information about the state of the channel at the beginning of the video fragment transmission. If you compare the two bitrate selection algorithms, you can see that the average bitrate is higher in the Instant algorithm. This is because this algorithm selects the bitrate of the next segment according to the measured bandwidth of the ports, regardless of the bitrate of the previous segment being loaded or the current buffer state. While Miller's algorithm with a small buffer volume increases the bitrate in steps and only when the measured bandwidth of the channel is significantly greater than the next bitrate.

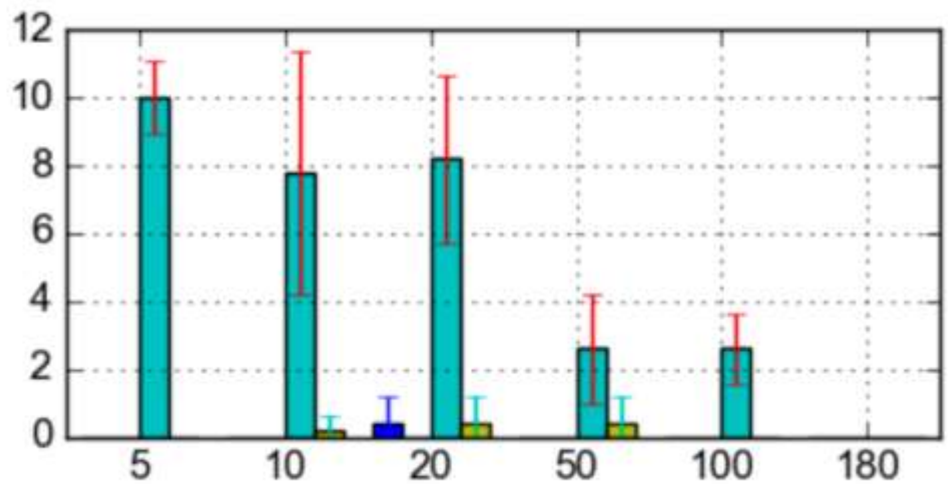

Figure 3.10 – The number of pauses for the entire time of the experiment for Load = 28%

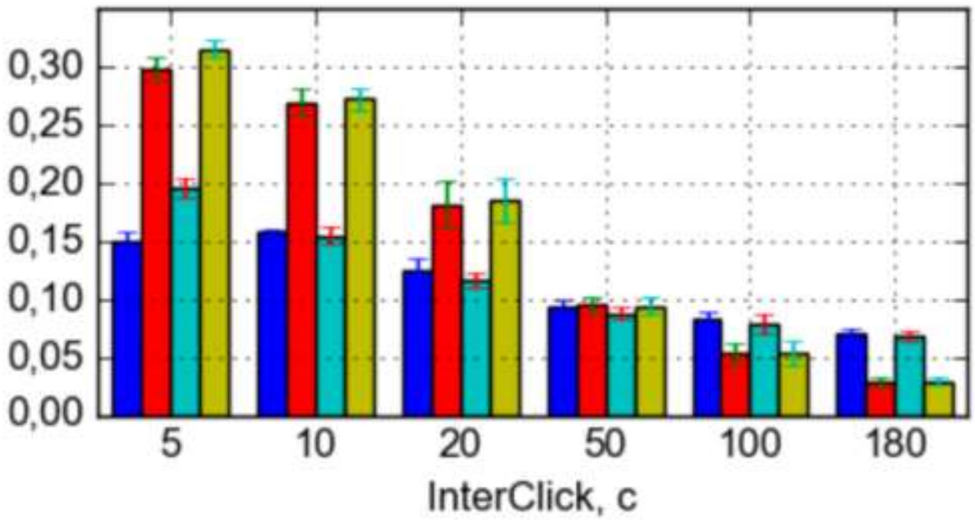

Figure 3.11 – The average number of bitrate changes per 1 second of the viewed video fragment to Load = 28%

In addition, from the graphs presented, reducing the value of bmin allows you to increase the average bitrate, since in this case the algorithms begin to increase the bitrate of the requested segments earlier.

From the start delay graphs, you can see that using bmin = 2 also allows you to reduce the start delay of the video fragment. This is because with a smaller value of bmin, playback starts after loading a smaller number of segments. On the other hand, the graph of the total duration and number of pauses shows the drawback of the Instant algorithm when bmin = 2 s. It is due to the aggressive policy of increasing the bitrate, which is implemented at the initial stage of playing a video fragment, which is particularly vulnerable to fluctuations in the bandwidth of the connection.

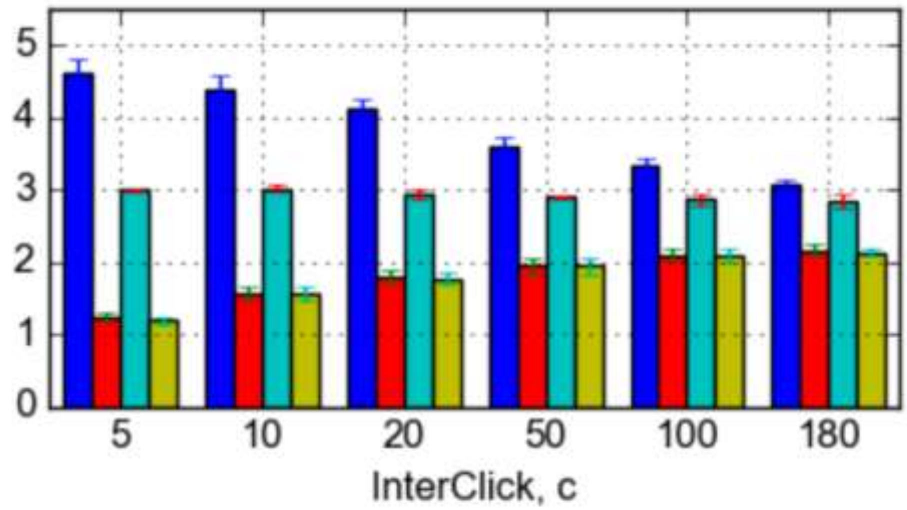

Figure 3.12 – Bitrate change module for Load = 28%, Mbp/s

Since at the beginning of playback the volume of the buffer b=bmin, there is a high probability of emptying the buffer and causing pauses during playback. Miller's algorithm, in turn, allows pauses in playback when the network load from web clients increases and the average duration of a video fragment increases. This is because, unlike the Instant algorithm, this algorithm supports a limited buffer volume, which in case of a significant and long-term decrease in the bandwidth of the connection leads to emptying of the buffer and the onset of pauses. The higher number of pauses in the Miller algorithm at bmin = 2 s, compared to bmin = 5, is caused by too late switching to the minimum available bitrate.

Also, on the graphs presented, you can see the differences between the algorithms in the frequency and amplitude of bitrate changes. On short video fragments, Instant changes the bitrate less often and almost always by a larger value, while Miller's algorithm changes it more often, but by a smaller value. The reason for this effect is that the Miller algorithm changes the bitrate in steps, and Instant immediately selects the bitrate that satisfies the current bandwidth of the connection. At the same time, on long video fragments, the Miller algorithm shows better results on both indicators: it changes the bitrate less often and changes it to a smaller value. Thus, Miller's algorithm is more stable on long video fragments.

The following approaches can be used to solve the problems described in this section.

1. Use bandwidth statistics collected during the transmission of previous video fragments.

2. Implement a cross-level protocol that allows you to transfer information about the bandwidth of the connection from the channel level to the application level.

However, as the research conducted in the bachelor's thesis shows, increasing the bitrate at the beginning of the video stream transmission can lead to the following problems:

- increasing the bitrate of segments that are loaded before the start of playback of a video fragment leads to an increase in the delay of the start of playback.

- a more aggressive policy for choosing the bitrate of segments with a small buffer size can lead to pauses in playback.

These problems can be solved by channel-level prioritization of traffic for DASH clients that start watching video files or clients with a small buffer volume.

4 LIFE SAFETY, BASICS OF LABOR PROTECTION

4.1 First aid for shock

Let's consider the procedure that determines the mechanism of providing pre-medical care in case of suspected shock by non-medical workers.

Shock is a state between life and death; a general severe disorder of the body's vital functions, caused by a violation of the nervous regulation of vital processes; characterized by disorders of hemodynamics, breathing, and metabolism.

Signs of shock in the victim:

- pale, cold and moist skin;
- weakness;
- restlessness;
- dry mouth, feeling thirsty;
- frequent breathing (more than 20 breaths per minute);
- disturbance of consciousness; faint.
- Causes of shock can be:
- external bleeding;
- internal bleeding;
- injuries of various genesis;
- burns;
- heart attack, etc.

The sequence of actions in the provision of pre-medical assistance to victims of suspected shock by non-medical workers:

1) make sure there is no danger;

2) conduct an examination of the victim, determine the presence of consciousness, breathing;

3) call an emergency (ambulance) medical team;

4) if the victim is not breathing, start cardiopulmonary resuscitation;

5) eliminate the cause of shock: stop bleeding, immobilize the fracture, etc.;

6) provide the victim with an anti-shock position:

a) transfer the victim to a horizontal position;

b) put a box, a roll of clothes, etc. under the feet of the victim in such a way that the feet are at the level of his chin;

c) put clothes/pillow under the victim's head;

d) cover the victim with a thermal cover/blanket;

7) ensure constant supervision of the victim until the arrival of the emergency (ambulance) medical assistance team;

8) if the victim's condition worsens, call the emergency medical dispatcher again before the emergency (ambulance) medical team arrives.

4.2 Development, design of a room for psychological relief of employees

The tense rhythm of the life of school workers, the intensification of their work against the background of low motor activity create a well-known dissonance between the demands placed on the intellect, the emotional sphere, and the relatively small physical load. The work of the nervous system in this mode often leads to increased tension, the inability to relax, get out of a tense state, and find mental balance. In most cases, people prone to "diseases of the century" - neuroses, hypertension and ischemic heart disease - can record increased muscle tension, loss of the ability to voluntarily relax muscles. In addition, intensive study of some subjects causes the need to relieve mental tension. All this puts before the psychological service of the school the urgent task of creating a psychological relief office (CPR).

The office of psychological relief at the school works in five modes:

- Psychological relaxation of employees and schoolchildren after hard work at the end of the working (school) day or at a specially designated time.

- The psychological mood (mobilization) of those employees and schoolchildren who find it difficult to get into the busy rhythm of work at the beginning of the working day, learning the skills of mobilization under stress (control, exam, etc.).

- Removing the psychological burden of teachers and schoolchildren according to the course prescribed by the psychotherapist.

- Psychoprophylactic work with practically healthy teachers and schoolchildren (teaching relaxation methods, meditation, autogenic training, conflict-free communication skills, communication training, etc.).

- Ensuring the process of intensive training, including the methods of suggestopedia, relaxopedia, hypnopedia, as well as the use of the psychological unloading room as an experimental base for the development of new training methods.

The question of the possibility and necessity of attending sessions of psychological relief is decided by employees of the psychological service on the basis of psychodiagnostic data, depending on the nature of the impact. For individual work, 5 to 30 minutes are allotted for one person, 60 minutes for a group. If there are 12-15 places in the CPR, its capacity is 60-80 people per shift, and up to 200 people can receive course treatment at the same time, since classes are held two or three times a week. When conducting intensive training classes, the passing possibility of CPR decreases, however, persons who undergo an intensive course simultaneously experience a psycho-prophylactic effect.

There are certain technical requirements for the arrangement of the CPR. The office should consist of two interconnected rooms. The first room is simultaneously the office of the psychological service. All the equipment used for psychotherapy sessions and intensive training classes is brought here. In addition, from the operator's room through a special mirror glass on one side, it is possible to observe the behavior of visitors in the psychotherapy room. Such a hall is equipped with 10-15 soft chairs with high headrests and built-in connectors for connecting individual headphones. The area of the hall should be at least 40

CONCLUSION

The main results of the bachelor work are as follows:

The analysis of modern video content delivery technologies and their development prospects showed:

- Today, on the market of services for delivering realistic content to the user (HDTV level), television technologies (DVB S2 / T2 / C2 / H / SH) are more common, but in a few years the leading technologies will be telecommunications (LTE) and Internet technologies (OTTV).
- The study of the architecture of the considered technologies allows us to conclude that WiMAX and LTE data transmission networks are most adapted to modern conditions, but the potential capacity of LTE networks, due to higher speed in the downstream direction, speaks of its advantages. At the same time, technologies like DVB-H are currently considered unpromising. However, one cannot fail to notice the potential of OTTV services, and especially their convenience for the subscriber. For the operator, they can become either a threat or a competitive advantage.

The transmission of video streams in a wireless network was considered, including in the presence of background web traffic, and the work of two algorithms that use different approaches when choosing the distribution capacity (quality) was investigated.

To solve the known problems that arise when transmitting video streams, it is suggested to use the following approaches:

- Use bandwidth statistics collected during the transfer of previous video fragments.
- Implement a cross-level protocol that allows you to transfer information about the bandwidth of the connection from the channel level to the application level.

Increasing the bitrate at the beginning of the video stream transmission can lead to the following problems.

- Increasing the bitrate of segments that are loaded before the start of playback of a video fragment leads to an increase in the delay of the start of playback.
- A more aggressive policy for choosing the bitrate of segments with a small buffer size can lead to pauses in playback.

These problems can be solved by channel-level prioritization of traffic for DASH clients that start watching video files or clients with a small buffer volume.

REFERENCES

1 A. V.M., Digital compression of video information and sound: Study guide, A. M. Y. Shelukhin O. I., Ed., Moscow: Dashkov and Co, 2003.

2 D. V.Yu., Multi-service communication networks:, Ed. In.zv., 2007, p. 166.

3 D. S, Trends in the modern development of cable networks television and Broadcasting, 2007, pp. 22 - 23.

4 M. A. McColkin, Methods of assessing the quality of video transmission in communication networks: textbook, SPBGUT,, 2012.

5 V. D. Kulikov D.L., Evaluation of the quality of work of video decoders of the MPEG-2 standard when working in an unreliable data transmission environment, Tr. conf. Graphicon, 2006, pp. 367 -370.

6 B. I. A.A. Ivakin, "Science and Education," in The III international research and practice conference, Munich, 2013.

7 I. A. A. Dmitriev VN, Analysis of providing multimedia services to mobile terminals, Vols. Upravlenie, vychislitel'naia tekhnika i informatika, 2014, pp. 105-111.

8 T. A.S., Wetherall DJ Computer Networks, P. Hall, Ed., 2010, p. 960.

9 MPEG-DASH, 2nd Edition Specification (ISO/IEC 23009-1:2014), vol. 2nd Edition Specification, 2014.

10 M. M. B. R. C. Timmerer, Which Adaptation Logic? An Objective and Subjective Performance Evaluation of HTTP-based Adaptive Media Streaming Systems.

11 A. C. B. a. C. D. Saamer Akhshabi, "An experimental evaluation of rate adaptation algorithms in adaptive streaming over HTTP," in In Proceedings of the second annual ACM conference on Multimedia systems (MMSys '11), New York, 2011.

12 E. Q. G. G. a. A. W. K. Miller, Adaptation Algorithm for Adaptive Streaming over HTTP. In Packet Video Workshop (PV), vol. 19th International, 2012, pp. 173-178.

13 L. Romero, "A Dynamic Adaptive HTTP Streaming Video Service for Google Android," Stockholm, 2011.

14 N. s. NS-3, "Network simulator NS-3," [Online]. Available: https://www.nsnam.org/.