Chapter 9
Modeling of Packet Streaming Services in Information Communication Networks

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ABSTRACT
Application of the term video streaming in contemporary usage denotes compression techniques and data buffering, which can transmit video in real time over the network. There is currently a rapid growth and development of technologies using wireless broadband technology as a transport, which is a serious alternative to cellular communication systems. Adverse effect of the aggressive environment used in wireless networks transmission results in data packets undergoing serious distortions and often getting lost in transit. All existing research in this area investigate the known types of errors separately. At present there are no standard approaches to determining the effect of errors on transmission quality of services. Besides, the spate in popularity of multimedia applications has led to the need for optimization of bandwidth allocation and usage in telecommunication networks. Modern telecommunication networks should by their definition be able to maintain the quality of different applications with different Quality of Service (QoS) levels. QoS requirements are generally dependent on the parameters of network and application layers of the OSI model. At the application layer QoS depends on factors such as resolution, bit rate, frame rate, video type, audio codecs, and so on. At the network layer, distortions (such as delay, jitter, packet loss, etc.) are introduced.

INTRODUCTION
We present in this chapter simulation results of modeling video streaming over wireless broadband communications networks and the differences in spatial and time characteristics of the different subject groups during transmission over networks. Numerical results of the modeling and analysis of the effect of these parameters on quality of video streaming are presented and discussed. Also presented is the proposal of a completely new approach to modeling errors, based on a developed
Markov model with the use of actual statistics of errors in the channels of broadband wireless access networks. We show that discrete Markov processes with the necessary number of states describe the mechanism of transmission of video sufficiently well and an increase in the number of states of the Markov chain allows to observe less divergence between real and simulated data, but this increases the complexity of the model, analysis and processing of data. The chapter effectively summarizes the researches carried out to date by the authors in investigating the effects of video streaming errors on the performance of broadband wireless access networks.

In section 1, we present background information on the features of streaming services: their characteristics, quality parameters, and peculiarities of streaming H.264/AVC video over broadband wireless access networks. The second section presents the design and development of our streaming video software and its use in estimation of the quality of streamed video. In the third section of the chapter, we present our findings on investigating the effect of noise stability on the quality of streaming video. Each section of the chapter ends with a conclusion and relevant recommendations arising from the discussion of research findings.

1. PROPERTIES OF STREAMING SERVICES

1.1. Characteristics of Streaming Traffic and Quality Parameters Characterizing Continuity of Service

Streaming traffic—traffic type, which is characterized by viewing and (or) auditioning information as it becomes available to the user (terminal) equipment.

Traffic in modern computer networks can be divided into two large groups—elastic traffic, which generates the traditional services such as email, WWW, FTP, and real-time traffic, which generates multimedia services such as IP-telephony or video conferencing. The share of real-time traffic is gradually increasing, due to growing interest in services, which allow for sound and high-quality video to be transmitted over computer networks (with high-speed bit stream and high resolution), such as the Music on Demand (MoD), Video on Demand (VoD) and IP-Television (IPTV).

Transmission of Streaming services (audio and video) over various media (wireless access, Internet, etc) is becoming more popular. This rapid expansion defines a new challenge of maintaining quality of service for each stream. On the other hand, new mobile systems are anticipated that will offer wireless services to a wide variety of portable terminals, ranging from cell phones and personal digital assistants (PDAs) to small portable computers. All these devices are heterogeneous.

They have different processing power, display, memory, and possible data rate. Thus, the rate of decoded data and content resolution need to be adapted to the surrounding network and display device (terminal). This quality is necessary to transfer huge amount of data on heterogeneous networks, and at the same time should find applications where the above-mentioned terminals are not able to display the full image resolution or all of the picture properties. Despite the shift to higher speeds, overload conditions often arise when trying to run resource-intensive services such as IPTV, available to multiple users. As a result, service quality is low, which is especially critical for video streaming - it should be noted that even minor disruptions to the picture on the screen or desync of audio and video tracks will cause a negative viewer reaction.

However, the problem lies not in slow network speed, but rather in the characteristics of the traffic, and more precisely in the peculiarities of the interaction between elastic traffic flows and real-time data.
A characteristic feature of the QoS indicators for data services in broadband wireless access is that it takes into account the classes of data (traffic), as defined by the ETSI TS 123 107 standard: dialogue, streaming, interactive and background. The main difference between these classes is their sensitivity to time delay in the network transmitted data streams. Effect of packet delay in the network elements with respect to time manifested in the possibility (or otherwise) of man’s perception of a message fragment, for example, whole audio or video fragment. Thus, the subjective estimates of the messages for the maximum packet delay from a sender to a receiver should not exceed 400 ms. Delay constraint is stringent for some services, the inability to provide necessary delay in packets leads to unacceptable quality of service.

**The dialogue (speech) traffic class:** This is the only class in which delay is strictly determined by human perception and packet transmission is done real-time with extremely low latency. This traffic class corresponds to voice services and video telephony.

**Streaming traffic class:** The data is processed as a steady and continuous stream. This class corresponds to the reception/transmission of web-information and reception/transmission of information on request. This kind of application (transfer methods) are asymmetric and therefore able to withstand longer delay than symmetric dialogue systems, as well as allow for considerable variation, and change the values of the delay.

**The interactive traffic class:** the type of traffic, which is characterized by a direct interaction (dialogue) of communication service users or terminal equipment user. This traffic class is used to provide services for which the end-user (human or machine) engaged in a dialogue requests in real-time data from a remote source (e.g. servers). Examples of human interaction with a remote source are viewing Web-pages, searching for information in the databases, access to a remote server, interaction with remote sources - measurement data survey and automated queries sent to the database (telemetry system). The classical scheme of data transfer is interactive traffic, which in general is characterized by a form of “end-user query-response.” One of the key parameters is the delay associated with the acknowledgement of data reception. These services include: user location determination service, and computer games. However, depending on the nature of the game (i.e. how much data is being actively transmitted), this service can be classified by the degree of acceptable delay to the dialog class (data transfer in real time).

**Background traffic class:** This class of traffic (email delivery, SMS, download of databases, etc.) can be transmitted with a delay, as this information does not require immediate implementation of any action. The delay in this case may be seconds, tens of seconds or even minutes. At the initial stage of commercial operation of the UMTS traffic classes and streaming dialog are transmitted in real-time switching channels mode. Table 1 shows the main characteristics of data on 3G networks for the described classes of traffic. Each of which has its own specific features vis-a-vis QoS. Table 2 and 3 are the main quantitative indicators for certain services with traffic streaming and interactive classes, respectively.

As can be seen from these tables, the main characteristics of traffic streaming and interactive classes are data packets’ delay and Frame Error Probability (FEP), which characterizes the data quality.

Data quality can also be assessed based on the reliability information of “soft frame” (a soft decision decoder). Such information may include:

- Error probability of message elements (BER - bit error probability), calculated before the channel decoder - BER in the physical channel;
- Soft (current) information from the Viterbi decoder of the convolutional code;
Table 1. Quantitative indicators for services with traffic streaming and interactive classes

<table>
<thead>
<tr>
<th>Type of Service</th>
<th>Application</th>
<th>Degree of Symmetry</th>
<th>Data Transfer Rate (kbps)</th>
<th>Main Characteristics and Their Indicators</th>
</tr>
</thead>
<tbody>
<tr>
<td>Audio Transmission</td>
<td>High-quality direct audio playback</td>
<td>In general, one-sided</td>
<td>16-128</td>
<td>≤150, ≤1, FER &lt; 3%</td>
</tr>
<tr>
<td>Video Transmission</td>
<td>High-quality direct video playback</td>
<td>Unidirectional</td>
<td>32-384</td>
<td>≤150, FER &lt; 1%</td>
</tr>
<tr>
<td>Data Transmission</td>
<td>Array conversion data/search</td>
<td>In general, one-sided</td>
<td>&lt;250</td>
<td>Not applicable, 0</td>
</tr>
<tr>
<td>Data Transmission</td>
<td>Picture</td>
<td>Unidirectional</td>
<td>&lt;250</td>
<td>Not applicable, 0</td>
</tr>
<tr>
<td>Data Transmission</td>
<td>Telemetry-monitoring</td>
<td>Unidirectional</td>
<td>&lt;28.8</td>
<td>Not applicable, 0</td>
</tr>
</tbody>
</table>

Table 2. Quantitative indicators for voice and data streaming services

<table>
<thead>
<tr>
<th>Type of Service</th>
<th>Application</th>
<th>Degree of Symmetry</th>
<th>Data Transfer Rate (kbps)</th>
<th>Main Characteristics and Their Indicators</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice transmission</td>
<td>Voice messages</td>
<td>Mostly unidirectional</td>
<td>4.32.</td>
<td>playback &lt;1s, Record time &lt;2s, ≤1, FER &lt; 3%</td>
</tr>
<tr>
<td>Data Transmission</td>
<td>Web-search/Web-browsing</td>
<td>Mostly unidirectional</td>
<td>Preferably &lt;2s per page, Allowable &lt;4s per page, No, 0</td>
<td></td>
</tr>
<tr>
<td>Data Transmission</td>
<td>Transactions with high priority, E-commerce</td>
<td>Bidirectional</td>
<td>Preferably &lt;2s, Allowable &lt;4s, No, 0</td>
<td></td>
</tr>
<tr>
<td>Data Transmission</td>
<td>E-mail</td>
<td>Mostly unidirectional</td>
<td>Preferably &lt;2s, Allowable &lt;4s, No, 0</td>
<td></td>
</tr>
</tbody>
</table>

Table 3. Technical standards for average monthly network operation access to be controlled

<table>
<thead>
<tr>
<th>Indicator</th>
<th>Interactive</th>
<th>Interactive During Usage Satellite Communication Link</th>
<th>Streamed</th>
<th>Data Traffic, Except Interactive and Streaming Traffic</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet transmission average delay information, (ms)</td>
<td>≤ 100</td>
<td>≤ 400</td>
<td>≤ 400</td>
<td>≤ 1000</td>
</tr>
<tr>
<td>Deviation from the average information packet transmission delay (ms)</td>
<td>≤ 50</td>
<td>≤ 50</td>
<td>≤ 50</td>
<td></td>
</tr>
<tr>
<td>Error coefficient of information packets</td>
<td>≤ 10⁻³</td>
<td>≤ 10⁻³</td>
<td>≤ 10⁻³</td>
<td>≤ 10⁻³</td>
</tr>
<tr>
<td>Coefficient of information packet loss</td>
<td>≤ 10⁻⁴</td>
<td>≤ 10⁻⁴</td>
<td>≤ 10⁻⁴</td>
<td>≤ 10⁻⁴</td>
</tr>
</tbody>
</table>
• Soft information from turbo decoder such as the $E_b / N_0$ ratio in the channel.

However, the choice of using a particular method for determining the quality of the received data has its peculiarities. The quality of the data can be estimated through any of the three methods mentioned above, however, a hybrid (integrated) approach seems to be the most effective. The essence of the integrated approach would be to use the error information received from the decoder, and data on the reception quality of individual elements of the received signal (or the signal-to-noise ratio).

Multimedia stream: In this case, the streaming technology sends information through the network from the server to the user in real time. Tools are not loaded on the viewer hard disk. The media is viewed as long as the client accepts them (it is possible that the buffer is not used). Should the client desire to see the media again, the streaming process is repeated. The streaming system “from end to end” requires the availability of streaming software, a streaming server and a media player at the customer’s end. Clips are made with special programs that convert audio, video and animation to MPEG-4 format for streaming. Streaming servers, such as the Apple Inc. server or RealServer (from RealNetworks) can be used to transfer media clips to customers using MP4 Player, RealPlayer or QuickTime Player.


During the development and formalization of a new process of video compression the final format was arrived at via multiple organizations (ITU, ISO, etc.) and for this reason, the new standard has five different names: H.264, MPEG-4 Part 10, AVC, H.26L, JVT. Using the new coding standard is possible in many different digital video transmission systems known by several names: Internet TV, IPTV, streaming video, video over IP, IP-Video information system, including transmission of different resolution up to HDTV 1080i (1920x1080). The newest H.264 standard as well as earlier versions of MPEG-4, allow for high compression ratios by exploiting both spatial and temporal redundancy in video frames. Thus, only those elements of the image that has changed compared to the corresponding elements of the previous frame need be processed.

For ease of use in various applications and a variety of networks, the H.264 codec is divided into layers of video encoding VCL (Video Coding Layers) and network abstraction NAL (Network Adaptation Layer) (Figure 1). Layering allows for being independent of network transmission conditions (Wiegand, Sullivan, Bjøntegaard, Luthra, 2003). The VCL level consists of the main compression mechanism, and includes the syntactic level, known as Slice Macroblock (MB), and block (Lee and Kalva, 2008). The NAL H.264/AVC level defines the interface between the video codec itself and the outside world and is designed for the adaptation of bit sequence generated by the VCL for transmission over various networks (Marpe, Wiegand, Sullivan, 2006).

NAL encapsulation for the various transport networks such as H.320 (ITU-T H320, 1999), MPEG-TS systems (ISO-IEC 13818, 1994) and RTP/IP (Wenger, Stockhammer, Hannuksela, 2003) is outside the scope of H.264/AVC standardization. Moreover, all forms of NAL H.264 packaging are called models with a “simple packaging” in which one NAL is placed in one RTP packet (Wenger, 2003). Packaging rules for this method are really very simple: NAL (including its header, which also serves as the header for useful information) is placed in the RTP packet, header parameters are defined according to the specification in (Schulzrinne, Casner, Frederick, & Jacobson, 1996) and subsequently sent to the RTP packet transport layer UDP.
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Figure 3. An example of desynchronization of VLC

<table>
<thead>
<tr>
<th>Symbol</th>
<th>Codeword</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>0</td>
</tr>
<tr>
<td>B</td>
<td>10</td>
</tr>
<tr>
<td>C</td>
<td>100</td>
</tr>
<tr>
<td>D</td>
<td>101</td>
</tr>
<tr>
<td>E</td>
<td>110</td>
</tr>
<tr>
<td>F</td>
<td>111</td>
</tr>
</tbody>
</table>

The use of VLC results in information desynchronization, with the result that part of the information before the next code becomes ‘undecodable’. In some cases, even after synchronization restoration, the decoded signal cannot be used correctly, since the necessary additional information on how to use it, such as the type of frame, or the motion vector is lost. Distortion resulting from the impact of transmission errors and subsequent decoding, are defined by the following terms: tiling, blurring, color transmission errors, error blocks, jerkiness, mosquito noise, the quantization noise, blurring, smearing (ETSI TR 102 493; ITU-T P.910 P.920 P.930; Atayero et al., 2011b). In addition, the transmission of video in real time is fraught with other certain problems associated with visual quality control, speed control/delay, error control, and scalability (Lakshman, Ortega, & Reibman, 1998; Stuhlmuller, Farber, Link, & Girod, 2000; Wang, Ostermann, & Zhang, 2001).

2. STREAMING VIDEO QUALITY ESTIMATION SOFTWARE

Most modern communication systems provide real time video transmission services, where the issue of video quality is very important. There are many publications in the literature devoted to the mechanisms of ensuring the required quality of service (QoS) for example (Aguiar et al., 2003; Sanneck et al., 2002; etc), but only some of them were able to achieve satisfactory results of practical import (Hertrich, 2002 and Wolf, 1999), since the effect of QoS parameters on the quality of video cannot be uniquely determined because of the large variety of coding schemes and post-processing and error recovery methods.

In some studies reported in the literature, frame synchronization of the transmitting and the receiving side is often used to assess the quality of video, which implies the inability to assess quality in case of frame loss or decoding errors (Sarnoff Corporation, 2002; EURESCOM Project P905-PF, 2000). There are known methods of assessing the quality of video distorted during transmission, for example (Wu, D., Hou, Y.T., Zhu, W., Lee, H.-J., Chiang, T., Zhang, Y.-Q., & Chao, H.J., 2000; Wolf, S. & Pinson, M., 2002), but their software is not freely available.

2.1. The Hardware-Software Solution for Assessing the Quality of Streaming Video

The main objective of video streaming performance research in wireless networks is optimization of decoding using the coding Rate–Distortion criterion. This criterion is characterized by digital video transmission efficiency and the quality of decoding at the receiving end. Various decoding algorithms, which affect the quality of received signals in different ways are employed for correct decoding of video signals transmitted over imperfect data transmission media (Dai, Loguinov, 2005; Feng, 1997; Koutsakis & Paterakis, 2004; Krunz, Sass, & Hughes, 1995; Krunz & Tripathi, 1997).
Application Layer (i.e. in the codec): these mechanisms determine the quality of video sequences at the decoding stage. Various forms of frame segmentation (e.g. slice, block) were applied at the onset of coding standards such as H.261, H.263, MPEG-1 and MPEG-2. The properties of the specific decoder in use need be taken into consideration when decoding distorted video sequences. Not all of the MPEG decoders handle stream errors qualitatively (Atayero, Sheluhin, Ivanov, & Alatishe, 2011). Some decoders are capable of processing errors with high probability. Other decoders cannot decode streams with lots of errors effectively, but they will decode with high quality streams with little errors. There are decoders, which decode the streams with errors rather poorly, regardless of the errors and are simply not suitable for use with the particular type of stream. Data partitioning (DP), which is an effective way of increasing system error robustness is already employed in later versions of codecs such as H.263, MPEG-4 Part 2. As at this writing, some mechanisms for protection against errors e.g. a group of parameters such as SPS, PPS, FMO and others are already incorporated in the H.264/AVC standard (Wenger, 2003). The link layer uses changes in the packet size to reduce losses, ensure service discrimination, etc. Uneven error protection is employed at the physical layer.

Bit errors occur during transmission over wireless channels and can affect the quality of received (decoded) video in different ways, some of which are listed below (Romer, 2004):

- **Bit error in different parts of the bitstream:** Since the MPEG-4 compression mechanism employs the removal of redundancy in the sequence, a relatively low bit error level can significantly affect the quality of decoded video. Bit error values higher than that allowable may cause a drastic degradation of quality.

- **Bit error in the video stream header:** Title sequences include important information such as frame resolution, number of frames, and the quantization table.

- **Bit error in the image header:** If an error occurs in the image header, the decoder may fail to detect the beginning of a frame. In the worst case, the frame will be lost. In other cases, with time prediction, serious degradation of quality may occur.

- **Bit error in GOP frame group:** Error in either the GOP proper or in its header has little or no effect on the proper decoding of video.

- **Bit error in DCT coefficients:** (Richardson, 2003) informs that if part of the DCT coefficients is distorted, it may lead to inaccurate decoding of the variable length codes.

Since codecs process information in blocks, the minimum unit of video stream distortion when exposed to a single errors is a block (4x4 or 16x16) depending on the encoding. Another area of error propagation is macro-block and slice. Thus, a single error message can cause the spread of errors not only in the actual macro-block, but also in the slice and subsequently the frame. It was shown experimentally in that smaller size of the slice encoding significantly improves image quality in the presence of packet loss.

There are three possible sources of error propagation (Rodriguez, 2008):

1. **Spatial prediction:** A macroblock restored during decoding with distorted neighboring macroblocks will also be distorted.
2. **Temporal prediction:** If a frame is distorted, then subsequent frames using it as the original picture will also be distorted.
3. **Entropy coding:** Since VLCs are used, an error in the main code can affect the following codes if the boundaries of the main code are not accurately defined. Thus synchronization of subsequent codes is disturbed, leading to the inability of the decoder to distinguish between the key codes (see Figure 3).
model. This model assumes a small (non-zero) and independent probability of error $P_{\text{good}} > 0$, even in the “good” state. Thus, four parameters are required for a complete description of the GE model (Hohlfeld, 2008).

The Gilbert-Elliot model can be described by the matrix $P$. Let $s_n$ be a Markov process corresponding to $s_n = 0$ if the channel is in the “good” state in the $n$-th moment of time, and $s_n = 1$ otherwise. Thus the transition matrix would take the form presented in (1.1) (see Box 1).

Average loss probability and the mean error length is gotten from expression (1.2)

$$P = \frac{P_{01}}{P_{10} + P_{01}}, \quad L = \frac{1}{P_{10}} \quad (1.2)$$

Both the AWGN model and the Gilbert-Elliot model can be used both on the physical layer with respect to bits, and the link layer, in relation to transport packets. They are often used in the experimental evaluation of simulated channels (Baguda, Fisal, Syed, Yusof, Mohd SA, Mohd A., Zulkarmawan, 2008; Ebert and Willig, 1999), due to their different effects on video stream (Figure 2), but they cannot simulate all the errors.

The most realistic and accurate means of modeling error statistics at the data link and physical layers is by the use of probability data obtained from a real network. In cases of rare occurrence of errors, they cause significant problems in the visual quality of streaming video, while in other types of services they would have remained unnoticed.

1.4. Features of Streaming Video over Broadband Wireless Networks

Application of more sophisticated coding methods does not allow for the complete avoidance of the appearance of characteristic distortions in the real-time transmission of video streams. As a rule, in wired networks with abundant bandwidth, the transmission channel has a low probability of bit error occurrence. However, data transmission over the wireless channel has a number of peculiarities due to the unpredictability of transmission conditions. Specific error protection mechanisms are employed at each layer of the OSI model:

Figure 2. Models of bit errors and their effect on video stream
nate error effects, which in turn leads to loss of the transport packet. The use of special means of protection against interference entails a decrease in bandwidth and increases the cost of equipment.

Short-term changes in bandwidth: This is associated with a bad connection of dial-up equipment. QoS constraints necessarily affect the bandwidth limits, going beyond which limit may cause packet loss in the absence of adequate control. Strong jitter, or bursty traffic can overwhelm the buffer and subsequently affect the ability to handle a sequence of packets, resulting in either their loss or late reception and processing.

Equipment problems: This may be caused by malfunctioning devices, the parasitic coupling between components, and untimely processing of packets due to jitter. To simulate the errors, different models have been adopted with different effects on the transmitted information.

Additive White Gaussian Noise (AWGN) Model: Since wireless communication channels are characterized by randomly distributed and independent bit errors, the AWGN is often adopted in the simulation of wireless communication channels, in which a certain bit in a sequence of distorted (inverted) with an apriori probability. Adopted value describes the probability of occurrence of bit errors i.e. the Bit Error Rate (BER). The signal received in the AWGN channel can be represented as:

\[ r(t) = s(t) + n(t) \]

where \( s(t) \) - the transmitted signal; \( n(t) \) - a noise signal having a mean value 0 and noise power spectral density \( N_0 / 2WHz^{-1} \) (Telatar, 1999).

The AWGN model cannot simulate a fading channel. The attenuation of the transmitted signal results in packetization (clustering) of errors. Since most modern codecs use variable length code (VLC), differentiating between bit-error and error packets makes little sense, because in any case errors will be displayed as loss or distortion of whole groups of consecutive bits in the decoder (Atayero, Sheluhin, Ivanov, Alatishe, 2011). However, there is a difference in the impact of different types of errors on different parts of the video stream. For example, a single error in the header can cause a greater distortion than the group of errors in several blocks.

Gilbert Model: Another very popular error model is the Gilbert model (Gilbert, 1960). This model presents a channel in the form of two states: the “good” state, and the “bad” state. For the “good” state, bit or packet is received successfully, whereas it is lost for the “bad” state. Hence corresponding to the probability transition states of \( P_{01} \) and \( P_{10} \). For the “good” state, the error probability \( P_{\text{good}} = 0 \). While for the “bad” state error occurs with an independent probability \( P_{\text{bad}} \). Thus, to fully describe the Gilbert model three parameters are required: \( P_{01} \), \( P_{10} \) and \( P_{\text{bad}} \). However, what is often misunderstood about the Gilbert model is that the “bad” state corresponds to the error/loss state, that is, \( P_{\text{bad}} = 1 \) (Qi, Pei, Modestino, Tian, 2004). This corresponds to a simple two-state Markov model, which takes into account only single errors. It is for this singular reason that error groups or their packets cannot be modeled (Fantacci and Scardi, 1996; Wang and Moayeri, 1995).

Gilbert-Elliott (GE): The Gilbert model was supplemented by Elliot (Elliot, 1963), resulting in what is now known as the Gilbert-Elliott
The use of hierarchical coding allows for video distribution in complex networks with varying bandwidth allocation for individual segments. Ideally, VCL should not produce NAL of a size larger than the MTU, thereby avoiding fragmentation at the IP level. This is easily achieved by using slices. However, since the NAL is less than 64 kB, IP-level performs the fragmentation and recombination of fragmented packets.

IP networks are a quite convenient environment for streaming video, due to the delay and data rate requirements (Wenger, 2003). As a rule, standard IP networks do not depend on the implementation of the physical and link levels and can work with their different protocols. Therefore, these levels are not discussed. At the network level, IP-based networks use the Internet Protocol. The IP-packet header size is 20 bytes and is protected by a checksum. However, useful information when it is not secure. The maximum size of IP-packet is 64 kB, but this size is rarely used, since the maximum packet size of the network Maximum Transfer Unit (MTU) is limited.

At the transport layer, videodata is usually transmitted via the User Datagram Protocol (UDP) (Postel, 1980), which in contrast to the Transmission Control Protocol (TCP) (Postel, 1981), does not guarantee the successful delivery of packets. Nevertheless, it is widely used for streaming video and video telephony, due to the small delay in transmission. Usually, checksum calculation is used for detecting transmission errors on UDP. At the application level, streaming video transmission is achieved by means of the following core protocols (Dvorkovich, 2005):

- **Real-time Transmission protocol (RTP):** This is a packet-oriented data delivery real-time transport protocol, which includes both video and sound (Schulzrinne, Casner, Frederick, Jacobson, 1996).

- **Real-Time Control Protocol (RTCP):** This is the control protocol designed to work with RTP, it helps to synchronize video and audio, as well as provide quality of service (QoS).

- **Real-Time Streaming Protocol (RTSP):** Management protocol for the initialization and directing of streaming data from video server, realizing the possibility of “remote control” (Schulzrinne, Rao, Lanphier, 1998).

- **Resource Reservation Protocol (RSVP):** A protocol to establish and support the required level of quality of service (QoS), ensuring the availability of adequate network resources (such as sufficient bandwidth).

### 1.3. Broadband Wireless Access

#### Network Error Models

Error-rates arising from the transmission of digital streams over the physical layer of wireless networks can be divided into two types: Single-bit errors and Bit error packet.

A single-bit error is expressed as a bit inversion in the transmission, which leads to incorrect recognition bit sequences and bytes as a whole. Bit-error packet is any sequence of errors longer than two inverted bits in a data segment. Bit error packets occur more frequently than single-bit errors. The length of error packets is measured from the first to the last inverted bit.

Bit errors are the most common and easily avoidable distortion of the digital stream. However, in some cases they may lead to the loss of a data segment. Thus, not eliminating bit errors at the physical layer will result in a loss of information (transport) packets at the link layer known as information packet error. Figure 1 presents types of errors in the context of the OSI model.

The following are possible reasons for loss of data packets:

- **Analog and electromagnetic interference, impulse noise:** Usually occur because of external factors, including weather and close proximity to electrical equipment. Error correction mechanisms at the physical layer are not always able to elimi-
Generally, video data can be presented in several forms including the following:

- The actual encoded bit stream, usually large, copyrighted and requiring expertise in coding/decoding, making its distribution to users quite difficult;
- Videotrace containing encoded video information in the bit stream (but not the actual encoded information), and freely distributed to users;
- In the form of video traffic models based on videotrace with certain statistical properties, which limit the user’s choice of subject (e.g., sports, news).

Videotrace give the opportunity to explore the network without the use of expensive equipment and software. At the same time they are much smaller in volume than an encoded video, and can easily be used in the simulation. Easy videotrace integration into any transmission system forms the basis for the development of hardware and software complex (HSC) based on the Evalvid system for assessing the quality of video transmitted over real or simulated communication networks (Klaue, Rathke, & Wolish, 2003).

The HSC modular structure allows for changing the main transmission system as well as codecs at the discretion of the user. This provides a wide range of experimental possibilities. Along with the evaluation of such network QoS parameters as the rate of loss, delay and jitter, assessment of the quality of the video based on the calculation of indicators PSNR and MOS is also possible.

The block diagram of Evalvid HSC for assessing the quality of streaming video transmitted in a variety of telecommunication networks is shown in Figure 4. The diagram shows the interaction between modules in the process of transmission of digital video from a source through a network connection to the viewer.

In order to assess video quality, video file data prior to transmission over the network (on the transmit end) and after reception from the network (on the receiving end) are required. The necessary data on the transmit side are: the original unencoded video in YUV format, encoded MPEG-4 format video, as well as time and type of each packet sent to the network.

The following data must be obtained at the receiving end: the reception time and type of each packet received from the network, the encoded video (possibly distorted) in MPEG-4 and the

Figure 4. HSC structure: VP – video transmitter; OT – trace scores; BB – reconstructed video; PSNR – quality estimation
decoded video in YUV for display. Data evaluation is performed by comparing the transmitted information with the received. In practice, uncompressed video can be very large, for this reason, it is advisable to transfer only the additional information in a file with a record receipt time of each packet. It is more convenient than transmitting a full (distorted and converted) video file from the receiving side. Data processing is carried out in three stages as described below:

In the first stage, the time taken to send and receive each packet on both sides as well as the packet type are analyzed. This results in a record of the type of frame and the time elapsed between transmitted and received packets. The distorted video file at the receive end is restored using the originally encoded video file and information about lost packets. Subsequently, the video is decoded for playback to the viewer. Assessment of video quality is done at this stage. Video quality indicators always require a comparison of the received (possibly distorted) video frame and the corresponding source frame. In the case of a total loss of frame in transit, the necessary frame synchronization before and after transmission over the network becomes impossible.

In the second stage, the problem of quality assessment is resolved based on the analysis of information about frame losses. Substituting the last relayed frame for the lost frame restores frame synchronization. This methodology allows for subsequent frame-wise assessment of video quality. At the third stage, the assessment of the quality of decoded video is achieved by means of both the restored and source video files.

The HSC modules interact with the network by using traces containing all the necessary data listed above. Thus, for proper functioning, the HSC requires two traces, the source video and the decoder. The data network can be considered simply as a two-port black-box that introduces delay, packet loss, and possibly packet rearrangement. The network was simulated based on the aforementioned assumptions in the NS-2 environment (http://www.isi.edu/nsnam/ns/ns-documentation.html). A more detailed description of the functional modules of the HSC is presented in (Klaue, Rathke, & Wolish, 2003).

2.2. Representation of the Source Files and Video Format

Standard test video in YUV format can be used as initial test video sequences. However, these videos are limited in play time and do not allow for estimating the change in long-term quality of broadcast video, and getting a sufficient amount of experimental data. Thus, with the aid of special software, personal YUV format videos of 30min length and 640x480 resolution with frame rate of 25 frames/sec was recorded. Then the original video file was encoded in format H.264. The encoded video stream is then packaged in an MP4-container for onward transport over the network using the User Datagram Protocol (UDP). After encoding the source video, MP4-file is obtained.

Since it is necessary to evaluate the quality of video transmitted over the network, the need arises to create a spare decoded YUV file from the newly created MPEG layer-4 file, which serves as the control in evaluating the quality of video transmitted over the network, excluding the impact of the codec. It is thus possible to estimate the influence of a wireless network on the received visual video quality, while excluding encoding and decoding losses. For simulation purposes, it is necessary to create a video trace file that contains the following information: frame number, frame type, frame size, and the number of segments in which the frame is divided into packets. This video trace serves as the input to the simulator network, where the sending and reception of video data occurs. As a result of video transmission over the network, it is necessary to obtain transmission trace files and reception trace files, which contain the following packet data: the transmission/reception time, a unique identifier and trace file size. These two traces are used to determine lost packets in the
network. In the end, we obtain files of the sent and received packets containing detailed information about the time of sending from the transmitter and the time of reception by the receiver.

### 2.3. Investigation of Broadband Wireless Access for Streaming Video Using the NS-2 Software

When modeling the wireless communication channel it is necessary to consider many destabilizing factors in the transmission medium, such as the imposition of white Gaussian noise (AWGN), multipath propagation, fading, interference, and many others. One cannot imagine the signal at the receiver end as purely the source signal, undistorted by any effects of the transmission medium. The results and reliability of the model are directly dependent on the construction of the communication channel, hence the need for employing an extremely accurate model, that approximates the real conditions (IEEE 802.16 Broadband Wireless Access Working Group; Deb, S., Jaiswal, S., Nagaraj, K., 2008). An accurate model may be obtained on the example of an extensive network of WiMAX IEEE 802.16e standard with the base and subscriber stations; the general scheme of such connections is shown in Figure 5.

Network Simulator 2 (NS-2) can be used for modeling errors arising in the real network, it allows for providing the desired level of simulation and obtaining the required performance characteristics of the network being modeled. NS-2 is distributed as Free Open Source code Software (FOSS). This category of software is distributed for free of charge without usage restrictions on

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**Figure 5. Architecture of a WiMAX broadband network**

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modification and distribution by third parties. Thus, NS-2 is by far the leader in comparison with similar products cost wise. For this reason, its updates are distributed free of charge as well as new libraries, protocols, e.t.c.

Another no less remarkable property of a FOSS is the possibility of modifying the core software and flexibility of configuration in accordance with the requirements of a particular user. NS-2 satisfies the requirements of flexible process modeling, since the language incorporates the ready scripts and C++ programming language, with which modeling objects such as nodes, channels, background traffic generators, etc. are described, parameterized from actual measurements. Figure 6 presents the functional interaction diagram of the SHC with NS-2.

The NS-2 software with several incorporated agents is used to simulate the network. MyTrafficTrace agent is used to calculate the type and size of the video-trace frame. In addition, this agent breaks the video into packets and sends them to the UDP in due course, according to user defined simulation settings.

MyUDP: in essence is an agent of the UDP. This agent writes each packet transmission time, packet ID, size and payload to file sent by the trace. MyUDP agent function is similarly to those of TCP-dump or Win-dump on real networks.

MyUDPSink agent: this is the object that receives packets of fragmented frames sent by MyUDP. This agent records the time, packet ID, packet size and payload of each received packet in the receiver trace file.

As a result of video transmission over UDP transmission and reception trace files are obtained. These two traces are used to determine the lost packets on the network. In real network scenario, the receiver trace is formed using a network analysis tool such as TCP-dump (http://www.tcpdump.org) or Win-dump (http://www.windump.polito.it). If the network is simulated, the simulation receiver object creates this file. Trace files contain complete information on the transfer of video over a network, necessary for further evaluation SHC. The VI module allows for generating traces for different video with different packet sizes, which can then be transmitted to the network (or simulator). The networking introduces delay and possibly loss and reordering of packets. An example of wideband wireless access IEEE 802.16 standard simulation in NS-2 software package is shown in Figure 7.

A feature of the presented simulation of wireless networks is the evaluation of the transmission

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*Figure 6. Interaction of SHC with NS-2*
of video streams with (and without) the effect of background traffic. Two types of background traffic; FTP and CBR are employed. FTP traffic is transmitted over TCP protocol at a data transfer rate of 512 kbps. CBR traffic is sent over UDP protocol at a transfer rate of 256 kbps. Bandwidth between the video server and base station equals 2 Mbps with a delay of 10 ms.

The SHC components allow for simulation of the two main types of errors that occur in wireless networks:

1. **Bit Error Rate (BER) Simulation:** Simulated transmission over a wireless channel model with Additive White Gaussian Noise (AWGN) or Gilbert-Elliott (GE) is conducted. In the process of simulation, certain bits in the sequence are distorted (i.e. inverted) with a given probability. The probability value used is defined by the BER. The error generator produces a distortion of the transmit trace file with a given probability and error distribution model. The receiver trace file is thus generated indicating the lost packets.

2. **Packet Error Simulation:** The UDP packets can be manually deleted from the received trace file. This allows for the observation of codec functionality and analysis of change in visual quality in cases of packet loss. At the same time, both the received and undistorted files can be obtained during transmission over an “ideal” channel with unlimited bandwidth and no delay, with subsequent removal of some packets.

### 2.4. Video Quality Assessment

There are two major methods of estimating the quality of digital video, namely, the objective and subjective methods. Objective methods are defined by the International Telecommunications Union (ITU) in (ITU-R BT.500-11), ANSI in (T1.801.01/02, 1996) and the Motion Picture...
was missing. Information about the parameters of the video streams is presented in Figure 11. This illustration shows the distribution of RTP-packets in time, where the white areas (Figure 11b) represents the received packets, while the black region corresponds to lost packets. Real-time estimation of objective video quality obtained at the communication channel output was calculated using the equation (1) (shown on Figure 11a). The delay was calculated as the difference
Time Division Duplex TDD, the maximum transmit power – 23 dBm);
- DELL Latitude D430 laptop and a Logitech QuickCam Pro 9000 webcam.

The video broadcast lasted for 30 minutes with the following parameters:

- **Format**: mp4;
- **Codec**: H.264;
- **Constant bitrate**: 1150 kbps;
- **Frame rate**: 25fps;
- **Resolution of**: 640x480;
- **GOP Type**: IBBPBBPBB.

The streaming server *DarwinStreamingServer* was used for broadcasting together with Apple Inc. QuickTime Player (http://www.apple.com). Information about the transmitted and received network packets were processed using the *Wireshark* protocol analyzer (http://www.wireshark.org). The *VirtualDub* program (http://www.virtualdub.org) was used for synchronizing both the sent and received video sequences. The *Wireshark* software was used to compute the video protocols used during broadcast as well as their statistical parameters (see Figure 10).

A deterioration in the quality of transmitted video was observed as a result of interference. Quality assessment was performed by means of the quality indicators and specifically PSNR and MOS, calculated using the HSC tools. Processing of the RTP-packet files sent and received during video transmission over the WiMAX network showed that a total of 281,720 pieces of RTP-packets was exchanged, of which 2,590 (0.919%)
user experience for streaming video is authentic for different network simulations.

3. EFFECT OF NOISE STABILITY ON THE QUALITY OF H.264/AVC STANDARD STREAMING VIDEO

The most important QoS parameters for wireless networks include the probability of bit error occurrence i.e. Bit Error Rate (BER) and that of packet error occurrence Packet Error Rate (PER). Fading channels are not capable of simulating single packet loss as well as single-bit errors. Typically, error occurrence are often long-term in nature, since a high probability of packet loss occurs in specific periods of data transmission, such as in the case of poor propagation. Attenuation of the transmitted signal leads to clustering (grouping or packetization) of errors. A group of erroneous packets is a sequence of packets either totally lost in transit (i.e. not received) or received with transmission errors incurred over a communications channel in a stipulated time frame. In this regard, the notion of the length of error burst i.e. Burst Error Length (BEL) was introduced in (Cornaglia and Spini, 1996; Lemmon, 2002). The length of a group of errors is defined as the number of erroneous packets from the very first to the last (inclusively) in a particular group of errors.

Due to the constant changing in the location of sources of transmission and reception of broadband communication systems in urban settings, parts of the information sequence (packet groups) do not reach the subscribers, which leads to distortion of the video signal or any other information transmitted. Markov processes with the necessary number of states describe the mechanism of transmission of information sufficiently well. This is necessary for the analysis of network problems encountered in the process of packet transmission of video. The model parameters allow for the estimation of the quality of transmitted video and the statistical parameters of the network.

3.1. Experimental Studies of the Quality of H.264/AVC Video Standard

In order to assess the impact subscriber mobility, a WiMAX IEEE 802.16e standard network was deployed with the use of a base and subscriber stations. The Base Station (BS) was installed on the roof of a 23-story building. Cross-polarized antenna with a gain of 9 dBi and beam width 90 degrees were used. Antenna directional azimuth was 300°. The Mobile Subscriber Station (MSS) was installed in a car. Two omni-directional antennas with a gain of 2 dBi were installed on the roof of a car. Images were recorded from a camera carried in a moving car on a laptop and simultaneously broadcast to a remote computer through the WiMAX network. The network architecture of the WiMAX broadband network and the overall connectivity diagram is shown in Figure 8.

The average vehicle speed was 60 kmph. Maximum distance from the BC was equal to 950 m. In most of the route between BC and AC lacked direct line of sight, so the main causes of interference in the broadcast were a reflection, diffraction, scattering, etc. The route of the vehicle and the location of the BS shown in Figure 9.

The following equipments were used for setting up the WiMAX network and subsequent video broadcast:

- **The base station and its specifications:**
  *RuggedMAX™ WiN7000|Specifications:* operational frequency bands – 1350... 1400 MHz, OFDMA, compression method – Time Division Duplex TDD, the maximum transmit power – 36 dBm);

- **The user station and its specifications:**
  *RuggedMAX™ WiN5100|Specifications:* operational frequency bands – 1350... 1400 MHz, OFDMA, compression method –
The values of PSNR cannot be totally predicted even with the full knowledge of experimental conditions, under which measurements are made. One can only indicate the likelihood of having certain values or the interval of the probable occurrence. However, knowing the probability distribution of this quantity, one can draw conclusions on its properties and characteristics. Thus it is possible to calculate the distribution of PSNR values, so that the quality of transmitted images can be evaluated statistically. Along with the evaluation of other network QoS parameters such as the rate of loss, delay and jitter. The HSC allows us to analyze statistical data under different conditions of video transmission, such as the probability density function, the expectation and variance. The mathematical expectation of PSNR value is given by (3)

\[
M = \frac{1}{n} \sum_{i=1}^{n} PSNR_i
\]  

and the variance is gotten from (4)

\[
\sigma^2 = \frac{1}{n-1} \sum_{i=1}^{n} (PSNR_i - M)^2
\]  

where \( n \) - number of frames in video sequence.

The probability density is described by Equation (5)

\[
W(PSNR) = N(M,\sigma^2) + \Delta
\]  

The distribution function is given by (6)

\[
F(PSNR) = P(PSNR_i \leq X_{\text{thr}})
\]

2.4.2. MOS Assessment of Video Quality

The mean opinion score subjective assessment of video quality is evaluated in terms of PSNR. The video quality is determined by computing the average MOS, having a value in the range from 1 to 5 (ITU scale), where 1 corresponds to the worst, and 5 the best video quality according to Table 1. Evaluation of the subjective quality of the video in the HSC is done by the MOS module. As a result a file with a frame by frame MOS value is generated. It is possible to calculate the percentage of frames with MOS values worse than that of the original video based on PSNR estimates. Quality of user experience (user satisfaction) is calculated on the basis of MOS values obtained in the HSC. Thus, network performance can be expressed in terms of user perceived reception quality. For normalization of the PSNR curve across MOS scale, a scale factor \( a \) and a shift factor \( b \) are required. The optimal scale and shift are defined by Equations (7) and (8) respectively.

\[
a = \frac{Cov_{MOS,PSNR}}{\sigma^2_{PSNR}}
\]  

\[
b = \mu_{MOS} - a \mu_{PSNR}
\]

where \( Cov_{MOS,PSNR} \) - covariance between PSNR and MOS, \( \mu_{PSNR} \) and \( \mu_{MOS} \) - average PSNR and MOS, respectively, \( \sigma^2_{PSNR} \) - PSNR variance.

Thus, we can obtain an expression for the evaluation of MOS based on the PSNR (9):

\[
MOS_{PSNR} = a \ast PSNR + b
\]

Thus, the quality of streaming video transmitted over any random network is assessed based on the results of the software-hardware complex, by calculating the following quality indicators: PSNR, MSE, and MOS. Assessment of quality of
Subjective quality assessment is always based on viewer impression. It is extremely costly, very time consuming and requires specialized equipment. Traditionally, subjective video quality is determined by expert assessment and calculation of the average Mean opinion Score (MOS), which is assigned a value from 1 to 5 on the ITU scale as prescribed in (ITU P.800; Atayero, 2000), where 1 and 5 represent worst and best received video quality respectively (Table 4).

Traditional signal distortion measures for system quality determination employ the absolute difference between original and processed signal. Objective video quality is usually measured using mean-square error (MSE) and peak signal to noise ratio PSNR, which is calculated from the RMSE and is a logarithmic measure of its inverse. MSE and its derivative indicator PSNR are the traditional metrics that allow for comparing any two images. RMSE can be called a measure of “distortion” and PSNR that of “quality”. In comparison with other objective indicators, PSNR can be easily calculated and is best understood by most users. However, both measures do not correlate with the subjective quality of the reconstructed image and do not properly reflect the minute deteriorations in intensity. Equation (1) (see Box 2) is the definition of PSNR between the luminance component Y of source (S) and received (D) image.

Based on the evaluation of PSNR, MOS estimate can be calculated, and the percentage of frames with worse MOS than the that of the original video. Table 4 shows the correspondence of PSNR with the MOS scale.

2.4.1. Estimating PSNR Value

PSNR is the main indicator of quality, which cannot be calculated, if two images are equivalent, i.e. standard error is zero. This is resolved by computing the PSNR between the original and the received video files. This ensures that the difference between the images will always exist, since all modern video codecs are lossy. It is possible to use other indicators, and software modules instead of PSNR/MOS, for example (Berts and Persson, 1998). Evaluation of objective video in the HSC module is done by the PSNR module. The result will be the values of PSNR, obtained from formula (1) between the original and distorted frames. Standard deviation of quality on average PSNR value is gotten from (2)

<table>
<thead>
<tr>
<th>PSNR, dB</th>
<th>MOS,%</th>
<th>ITU Scale Quality</th>
<th>The Deterioration of the Image</th>
</tr>
</thead>
<tbody>
<tr>
<td>&gt; 37</td>
<td>5</td>
<td>Excellent</td>
<td>Not perceptible</td>
</tr>
<tr>
<td>31 - 37</td>
<td>4</td>
<td>Good</td>
<td>Perceptible, but not annoying</td>
</tr>
<tr>
<td>25 - 31</td>
<td>3</td>
<td>Satisfactory</td>
<td>Slightly irritating</td>
</tr>
<tr>
<td>20 - 25</td>
<td>2</td>
<td>Poor</td>
<td>Irritating</td>
</tr>
<tr>
<td>&lt; 20</td>
<td>A</td>
<td>Very poor</td>
<td>Annoying</td>
</tr>
</tbody>
</table>

Box 2.

$$PSNR = 20\log_{10}\left[\frac{V_{peak}}{\sqrt{\frac{1}{N_{col}N_{row}} \sum_{i=0}^{N_{col}} \sum_{j=0}^{N_{row}} (Y_{S}(n, i, j) - Y_{D}(n, i, j))^2}}\right] dB$$  (1)
between the time of sending and reception of a packet (see Figure 11).

Figures 12 through 14 show the values of objective and subjective quality indicators.

### 3.2. Effect of Bit Errors on the Visual Quality of Streaming Video

Wireless communication channels are characterized by randomly distributed and independent bit errors. To simulate a wireless link with this type of error, the “additive white Gaussian noise,” or AWGN model is often used, where certain bits in the sequence is distorted (inverted) with a given probability. The value used is described by probability of bit error occurrence BER. Different BER values have different effects on the quality of streaming video. When comparing the original and distorted video stream, it is possible to calculate the effect of bit error on the resultant quality of the video. In order to study the influence of bit error on the resulting video quality, simulation of the transmission of a 30-minute video via IEEE 802.16 standard broadband wireless network with random bit errors in the channel was carried out. The experimental setup is shown in Figure 15.

MP4 codec was used for encoding/decoding the original video sequence in the H.264 standard within the structure of the HSC. Simulation of wireless networks with random bit errors in the channel was carried out using the VCDemo program (http://www.ict.ewi.tudelft.nl/vcdemo). Then simulated transmission of video stream over the wireless network was carried out according to the Open System Interconnect (OSI) model at the known levels, namely: application level, transport level, network level, data link level and physical level. Encoding, decoding, and video packetizing occur at the application layer of the OSI model. The video stream is divided into packets of variable length of up to 1,500 bytes, while adding a 12-byte RTP header. When adding RTP header to the data, the MPEG bit stream is segmented so that the MPEG start codes are

![Figure 11. Information about the parameters of video streaming: a) change in PSNR indicator; b) distribution of received/lost packets, c) packet delay](image)

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**Figure 11. Information about the parameters of video streaming: a) change in PSNR indicator; b) distribution of received/lost packets, c) packet delay**
Modeling of Packet Streaming Services in Information Communication Networks

The UDP protocol is modeled at the transport layer, respectively, adding a header and a checksum (8 bytes). Subsequently, a 20-byte IP header is added at the network layer. IEEE 802.16 protocol is modeled at the link layer of communication. The channel transmission rate is set to 20 MB/s. Finally, the simulation of random bit error in the channel (Gaussian noise) with a given BER probability is done at the physical layer. For modeling purposes, the encoded video stream was split into RTP/UDP–packets using the HSC tools. Quality assessment was performed by means of PSNR and MOS indicators, calculated using the tools of the HSC. The standard deviation of quality on average PSNR value was gotten from Formula (2). Figure 16 shows the effect of BER indicator on the quality of streaming video.

BER ranges are shown, within which the received video quality is maximal, i.e. equal to the original, and minimal i.e. with maximum difference from the original. It is shown that at $BER \leq 3 \times 10^{-5}$ error does not affect the visual quality and can be easily eliminated using decoders and existing methods of protection against errors. Further changes in quality is of step (sequential) nature and decreases with increasing BER (as expected). At $BER = 4 \times 10^{-3}$ packet loss reaches its maximum value and represents more than 99.9% of the total.

Analyzing the results of streaming video over a simulated wireless network with different values of BER, one can draw the following conclusions:

1. Simulating a wireless channel using AWGN model, and additive, bit errors with a value of $BER \leq 3 \times 10^{-5}$ does not affect the quality of the video. However, when $BER \geq 4 \times 10^{-3}$ packet loss in the network reaches its maximum value $\geq 99.9%$.

2. Objectively, excellent quality of video transmission over a channel can be guaranteed for all bit error probabilities less than $1 \times 10^{-4}$, good quality is in the range of $10^{-4} to 4 \times 10^{-4}$, satisfactory quality is in the range of $4 \times 10^{-4} to 8 \times 10^{-4}$, poor qual-
Empirical BER values of transitions from acceptable to poor quality, according to the table of correspondence between PSNR and MOS are presented in Table 5.

However, the AWGN model does not allow for adequate simulation of a fading channel. Typically, errors are often long term, since high probability of bit loss occurs in specific periods of transmission, e.g. during poor propagation. Attenuation of the transmitted signal results in packetizing (grouping) of errors. Another cause of error grouping can be physical defects of, and failures inherent in the information storage system. When using VLC, bit error occurrence results in group errors or packetization of errors.

### 3.3. Effect of Packet Errors on the Visual Quality of Streaming Video

In order to study the effect of errors in data packets on received quality, simulation of the transmission of a 30-minute video over a wireless network with random errors in the channel packets was carried out. For the purposes of simulation, the encoded video stream was split into RTP/UDP-packets using the HSC tools. Simulation of packet errors during transmission over a wireless channel was done by removing packets from the receive trace.
Figure 16. Values of video sequence quality indicators for different values of wireless channel BER: a) PSNR value distribution histogram; b) PSNR value distribution histogram for certain values of BER; c) quality deviation from average PSNR value; d) quality gradation for MOS values.
file. This made it possible to investigate and analyze changes in visual quality during packet loss. At the same time the received and undistorted trace file was obtained during transmission over an “ideal” channel with unlimited bandwidth and no delay in the NS-2 software environment, and then randomly removing transport packets according to the parameters of PER and BEL. Quality assessment was performed by means of PSNR and MOS indicators, computed using the HSC tools.

Figure 17 shows the effect of PER on the quality of streaming video. PER ranges are shown, within which the received video quality is maximal, i.e. equal to the original, and minimal i.e. with maximum difference from the original. It is shown that at $PER \leq 1 \times 10^{-4}$ error does not affect the visual quality and can be easily eliminated using decoders and existing methods of protection against errors. Further changes in quality is of step (sequential) nature and decreases with increasing PER (as expected).

Analyzing the results of streaming video over a simulated wireless network with a given probability of packet loss, we safely conclude that:

1. In simulation of a wireless network, a value of $PER \leq 10^{-4}$ does not affect the video quality. When $PER \leq 10^{-3}$ impact of errors on video quality is not noticeable and does not irritate during viewing experience. When $PER \geq 0.1$, packet loss in the network has the worst effect on visual quality.

2. Objectively, excellent quality of video transmission over a channel can be guaranteed for all packet error probabilities less than $10^{-3}$, good quality is in the range of $10^{-3}$ to $3 \times 10^{-3}$, satisfactory quality is in the range of $3 \times 10^{-2}$ to $10^{-2}$, poor quality is in the range of $1 \times 10^{-2}$ to $5 \times 10^{-2}$, while very bad quality is for any $PER > 5 \times 10^{-2}$.

3. Histograms of the distribution of values of PSNR when $PER \leq 6 \times 10^{-4}$, in general, have a bimodal shape. One of the peaks characterizes the value of PSNR of video stream distorted due to packet loss. The second maximum characterizes deterioration in the PSNR of dependent frames. As the number of errors increases, one of the peaks increases due to a decrease in the other.

Empirical values of PER transitions from an acceptable quality to the poor, according to the relationship between PSNR and MOS, are presented in Table 6.

### 3.4. Effect of Errors Length on the Quality of Streaming Video

To study the effect of the length of error groups on resultant quality, the simulation of a 30-minute video transfer over a wireless network for the values of $PER$ of $10^{-3}$ to $5 \times 10^{-2}$ is repeated, since a visual change in video quality is observed at this range. The simulation of groups of error packets during transmission over a wireless chan-

### Table 5. Relationship between quality indicators and BER

<table>
<thead>
<tr>
<th>PSNR [dB]</th>
<th>MOS [%]</th>
<th>BER</th>
<th>ITU Quality Scale</th>
<th>Picture Degradation</th>
</tr>
</thead>
<tbody>
<tr>
<td>&gt; 37</td>
<td>81–100</td>
<td>$&lt; 1 \times 10^4$</td>
<td>5 Excellent</td>
<td>Noticeable</td>
</tr>
<tr>
<td>31–37</td>
<td>61–80</td>
<td>$1 \times 10^{-4} – 4 \times 10^{-4}$</td>
<td>4 Good</td>
<td>Noticeable, but not irritating</td>
</tr>
<tr>
<td>25–31</td>
<td>41–60</td>
<td>$4 \times 10^{-4} – 8 \times 10^{-4}$</td>
<td>3 Satisfactory</td>
<td>Slightly irritating</td>
</tr>
<tr>
<td>20–35</td>
<td>21–40</td>
<td>$8 \times 10^{-4} – 1 \times 10^{-3}$</td>
<td>2 Poor</td>
<td>Irritating</td>
</tr>
<tr>
<td>&lt; 20</td>
<td>0–20</td>
<td>$&gt; 1 \times 10^{-3}$</td>
<td>1 Very poor</td>
<td>very irritating</td>
</tr>
</tbody>
</table>
states sufficiently well describe the mechanism of transmission of information, the knowledge of which is necessary to analyze network problems during packet video transmission. The model parameters allow for determining the quality of transmitted video as well as the statistical parameters of the network. A model describing the length of error intervals and error-free reception for streaming video transmission was developed based on the experimental data obtained as a result of streaming video from a moving source on WiMAX network. Based on the graph of packet loss distribution over time (Figure 3.3, b), an array was formed in which the lost packet corresponds to a logic zero (0) and received packet corresponds to a logic unit (1). The original array was split into two, one of which contains information about the lost packets and the other contains information about the received packets. The formation of arrays was carried out in accordance with the procedure shown in Figure 20.

### 3.5.1. Markov Model for Describing the Experimental Data

In accordance with the method presented in (ITU P.800), the available raw data file was divided into two parts, each of which separately contains the duration of ON periods and OFF periods.

Variables \( y_n[n] \) fall under the ON periods, while variables \( y_n[n] \) fall under the OFF periods.

An approximation of the Distribution Function (DF) of real processes is done. Equation (10) is used for approximating the DF OFF function.

\[
F^\ast(k) = A \sum_{i=1}^{3} e^{-\alpha_i k}
\]

By using the method of least squares we find the unknown coefficients of the approximation for the expression (10) as presented in Table 7.

Substituting the coefficient values obtained and given in Table 7 into Equation (10), we obtain the approximation of the original additional distribution of the length of OFF periods as equation (11) (see Box 3).

Equation (12) is used for approximating the ON distribution function state.

\[
F^\ast(k) = B \sum_{i=1}^{6} e^{-\beta_i k}
\]

By using the method of least squares we find the unknown coefficients of the approximation for the expression (12) as presented in Table 8.

**Figure 20. Formation of arrays**

```
0 0 0 0 0 0 1 1 0 0 0 1 1 1 1 1 1 1 0 0 0 0 0 . . . 1 1 0 0 0 . . .
```

### Box 3.

\[
F^\ast(k) = 0.612086 \times e^{-0.072072 k} + 0.631933 \times e^{-0.040023 k} + 0.073586 \times e^{-0.040006 k}
\]
1. Increasing the length of error groupings leads to an increase in the average quality of the video sequence. This is due to the deterioration of a small section of video, where error groups are concentrated, whereas in the case of single bit errors deterioration in the quality of video may be observed across the whole sequence;

2. When the length of erroneous packets is $BE_L \leq 6$ the change in quality is minor and identical to the influence of single packet errors ($BE_L = 1$);

3. When $BE_L \geq 60$ the average quality is almost identical to the original ($PSNR < 90dB$). It is logical to assume that the value of $BE_L$ in the longer video sequences, with the same average quality may have a higher value;

4. The highest dynamics of change in $PSNR = 60dB$ is observed in two cases: a) for a fixed $PER = 10^{-3}$ and the variable values of $BE_L$; and b) at $BE_L \geq 80$ and the varying values of the $PER$. In other cases, the dynamics is not essential and minimal in the absence of clustering of errors ($BE_L = 1$);

5. With increasing $PER$, the effect of $BE_L$ on quality decreases due to increase in denseness of single errors;

6. Analysis of the results of $PER$ and $BE_L$ shows that for effective assessment of the impact of transmission errors on resultant quality it is necessary to analyze not only the likelihood of errors, but also their structure and length of their grouping. The most realistic and accurate method of modeling statistical errors in communication channels is the use of probability data obtained from real networks.

3.5. Investigation of Error Packetization Effect on the Quality of Streaming Video in Wireless Broadband Access Channels

A scientific problem of important consequence is a need to create realistic simulation and mathematical models of behavior of losses in the communication channels based on the apparatus of Markov chains for wireless access systems. Markov processes with the necessary number of
Figure 18. Values of video sequence quality indicators for $\text{PER} = 10^{-3}$ and varying values of wireless channel BEL: a) PSNR value distribution histogram; b) PSNR value and RTP/UDP packet distribution (black spaces correspond to lost packets) for certain values of BEL; c) quality deviation from average PSNR value; d) quality gradation for MOS values
nel was done by means of random deletion of packet groups from the receive trace file with a given BEL. For this particular example, \( BE_L = 100 \) implies that the total random number of consecutively deleted packets does not exceed 100. The total sum of erroneous (deleted) packets in the video sequence for the whole experiment given \( PER = \text{const.} \) remained the same, irrespective of the value of BEL.

Figure 18 shows the effect of BEL on the quality of streaming video for \( PER = 10^{-3} \).

Analyzing the results of streaming video over a simulated wireless network with a given group of erroneous packets we can draw the following conclusions:

1. For \( PER \leq 10^{-3} \) the effect of single packet errors on quality is insignificant and does not irritate the viewing experience.

2. Histograms of the distribution of values of PSNR have two maxima. One of the peaks characterizes the value of PSNR of video frames distorted due to the loss of packets. The second maximum characterizes the deterioration of PSNR of dependent frames. With increasing quantities BEL is one of the peaks decreases as the number of dependent frames are also reduced, whereas the second peak remains unchanged. This is explained by the fact that the single scattered throughout the video sequence error number of distorted frames is large due to error propagation to dependent frames.

3. An increase in the BEL value leads to a decrease in one of the maxima, since the number of dependent frames also decreases, while the second maximum remains the same. This is due to the fact that under singular errors spread across the whole video sequence, the number of distorted frames is large because of the distribution of errors on dependent frames.

4. Effect of error groups on the quality is more powerful because of the local concentration of errors. However, the average quality of the video sequence increases with increase in the length of the grouping for a given value of probability of occurrence of packet errors.

5. For \( BEL \geq 60 \) the average quality is almost identical to the original video.

### 3.4.1. Relationship between the PER and BEL Indicators

The average quality of the experimental 30-minute video sequence for different values of PER and BEL is shown in Figure 19.

Thus, it is shown that in assessing the impact of erroneous packets received on the quality it is necessary to analyze not only the probability of error occurrence, but also their structure and length of the grouping. In addition, the following conclusions can be drawn:

---

**Table 6. Relationship between quality indicators and PER**

<table>
<thead>
<tr>
<th>PSNR[dB]</th>
<th>MOS[%]</th>
<th>PER</th>
<th>ITU Quality Scale</th>
<th>Picture Degradation</th>
</tr>
</thead>
<tbody>
<tr>
<td>&gt; 37</td>
<td>81–100</td>
<td>&lt; 1x10^4</td>
<td>5-Excellent</td>
<td>Noticeable</td>
</tr>
<tr>
<td>31–37</td>
<td>61–80</td>
<td>1x10^-3–3x10^-3</td>
<td>4-Good</td>
<td>Noticeable, but not irritating</td>
</tr>
<tr>
<td>25–31</td>
<td>41–60</td>
<td>3x10^-3–1x10^-2</td>
<td>3-Satisfactory</td>
<td>Slightly irritating</td>
</tr>
<tr>
<td>20–35</td>
<td>21–40</td>
<td>1x10^-2–5x10^-2</td>
<td>2-Poor</td>
<td>Irritating</td>
</tr>
<tr>
<td>&lt; 20</td>
<td>0–20</td>
<td>&gt; 5x10^-2</td>
<td>1-Very poor</td>
<td>Very irritating</td>
</tr>
</tbody>
</table>
Figure 17. Values of video sequence quality indicators for different values of wireless channel PER: a) PSNR value distribution histogram; b) PSNR value distribution histogram for certain values of PER; c) quality deviation from average PSNR value; d) quality gradation for MOS values
Substituting the coefficient values obtained and given in Table 8 into Equation (12), we obtain the approximation of the original additional distribution of the length of ON periods as equation (13) (see Box 4).

The approximation of DF ON is shown in Figure 21.

After the normalization of obtained approximating expressions (11) (13) additional distributions of duration of ON-and OFF-processes, the matrix of transition probabilities is created, which is of the form presented in Figure 22.

Substituting the values of the coefficients found in Table 7 and 8 into the matrix of transition probabilities, we obtain the matrix of values (Figure 23).

### 3.5.2. Software for Error Packetization Simulation

With the probability transition matrix and vector of initial probabilities simulation of the transmission of streaming video traffic over a WiMAX network can be done. The choice of the initial state of the system was carried out using the condition that all states are equiprobable (i.e. $p = 1 / N$, where $N$-number of states the system can be in after DF approximation). Description of the block diagram of the simulation algorithm is as given below.

### Table 7. Approximation coefficients

<table>
<thead>
<tr>
<th>$A_1$</th>
<th>$\alpha_1$</th>
<th>$A_2$</th>
<th>$\alpha_2$</th>
<th>$A_3$</th>
<th>$\alpha_3$</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.612086</td>
<td>0.072672</td>
<td>0.631933</td>
<td>0.540023</td>
<td>0.073586</td>
<td>0.040006</td>
</tr>
</tbody>
</table>

### Table 8. Approximation coefficients

<table>
<thead>
<tr>
<th>$B_1$</th>
<th>$\beta_1$</th>
<th>$B_2$</th>
<th>$\beta_2$</th>
<th>$B_3$</th>
<th>$\beta_3$</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.065836</td>
<td>0.000643</td>
<td>0.107716</td>
<td>0.000708</td>
<td>0.33109</td>
<td>0.007203</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>$B_4$</th>
<th>$\beta_4$</th>
<th>$B_5$</th>
<th>$\beta_5$</th>
<th>$B_6$</th>
<th>$\beta_6$</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.057449</td>
<td>0.0000618</td>
<td>0.007203</td>
<td>0.291568</td>
<td>0.224767</td>
<td>0.0094038</td>
</tr>
</tbody>
</table>

### Box 4.

$$F^*(k) = 0.065836 \times e^{-0.000643k} + 0.107716 \times e^{-0.000708k} + 0.33109 \times e^{-0.007203k} + 0.057449 \times e^{-0.0000618k} + 0.007203 \times e^{-0.291568k} + 0.224767 \times e^{-0.0094038k} \quad (13)$$
3.5.3. Error Packetization Algorithm

**Step 1:** Start program (Description of the variables, functions, procedures and modules used).

**Step 2:** Enter two-dimensional array matrix of transition probabilities. In the developed software, this is matrix was given as an array of constants in the declarations section and named markov.

**Step 3:** Set state from which to begin modeling. Since a 9-state model was chosen, the state variable can take integer values on the interval (1 – 9). Also, at this stage of the algorithm the accumulated variables summa_on and summa_off, which reflect the duration of the ON periods and OFF periods, are reset to zero respectively.

**Step 4:** Begin cycle with parameter \( i \). The number of iterations equals the number of transitions in the simulated system.

**Step 5:** Instantiate the built-in generator of pseudorandom uniformly distributed sequence, generating a random value in the interval (0, 1). Assign the generated value to \( rnd \). At the moment of generating the variable \( rnd \), the system moves to the next state. The exact state into which it falls will be determined by the subsequent actions of the algorithm. The variable \( summa \) is reset to zero.

**Step 6:** Start the cycle with parameter \( k \). The number of iterations in the cycle equals the number of states of the system being modeled. For this case, the number of iterations is eight (8). This loop is used to determine the state into which the system has moved at the particular time of consideration.

**Step 7:** Check if the value of \( rnd \) fell in the \( k^k \) state of the Markov chain. At the same time the following variables are involved: \( summa \) - accumulates the probability of all states up to the \( k^k \); \( markov \{ state, k \} \) - a two-dimensional array, which contains the transition matrix.

If \( rnd \) falls within a range of probabilities corresponding to the \( k^k \) state, then goto step 8, otherwise goto step 9.

---

**Figure 21. DF ON approximation (2), DF ON experiment (1) (embedded graph - reduced scale of DF ON)**

**Figure 22. Transition probabilities’ matrix**

\[
\begin{array}{ccccccc}
A_1 & A_2 & A_3 & B_1 & B_2 & B_3 \\
\hline
A_1 & e^{-\alpha_1} & 0 & 0 & (1-e^{-\alpha_1})B_1 & (1-e^{-\alpha_1})B_2 & (1-e^{-\alpha_1})B_3 \\
A_2 & 0 & e^{-\alpha_2} & 0 & (1-e^{-\alpha_2})B_1 & (1-e^{-\alpha_2})B_2 & (1-e^{-\alpha_2})B_3 \\
A_3 & 0 & 0 & e^{-\alpha_3} & (1-e^{-\alpha_3})B_1 & (1-e^{-\alpha_3})B_2 & (1-e^{-\alpha_3})B_3 \\
B_1 & (1-e^{-\beta_1})A_1 & (1-e^{-\beta_1})A_2 & (1-e^{-\beta_1})A_3 & e^{-\beta_1} & 0 & 0 \\
B_2 & (1-e^{-\beta_2})A_1 & (1-e^{-\beta_2})A_2 & (1-e^{-\beta_2})A_3 & 0 & e^{-\beta_2} & 0 \\
B_3 & (1-e^{-\beta_3})A_1 & (1-e^{-\beta_3})A_2 & (1-e^{-\beta_3})A_3 & 0 & 0 & e^{-\beta_3} \\
\end{array}
\]
Modeling of Packet Streaming Services in Information Communication Networks

Figure 23. The matrix of values

\[ \Gamma' = \begin{bmatrix}
0.999 & 0 & 0 & 2.4 \times 10^{-3} & 8.69 \times 10^{-4} & 1.1 \times 10^{-4} \\
0 & 0.9944 & 0 & 1.344 \times 10^{-4} & 0.0049 & 6.16 \times 10^{-4} \\
0 & 0 & 0.965 & 8.4 \times 10^{-4} & 0.0304 & 0.0039 \\
1.8 \times 10^{-5} & 3.6 \times 10^{-4} & 3.6 \times 10^{-4} & 0.9991 & 0 & 0 \\
4.2 \times 10^{-5} & 8.4 \times 10^{-4} & 8.4 \times 10^{-4} & 0 & 0.9979 & 0 \\
2.58 \times 10^{-4} & 0.0052 & 0.0052 & 0 & 0 & 0.9871
\end{bmatrix} \]

Step 8: Check – in which state is the process currently? If in the active state, then goto step 10. If in passive state, then goto step 11.

Step 9: The summa variable is increased by the value of the probability of being in state \( k \). Then proceed to the next iteration of step 6.

Step 10: Check – was the last state of the matrix passive? If yes, goto step 12. Otherwise, goto step 16.

Step 11: Check – was the last state of the matrix active? If yes, goto step 13. Otherwise, goto step 17.

Step 12: Arrival at this step implies the end of OFF period. Therefore save or print to file summa_off.

Step 13: Arrival at this step implies the end of ON period. Therefore save or print to file summa_on.

Step 14: Since the OFF period as ended, reset the variable summa_off to zero in preparation for the record of fresh OFF-period information, when the process will be in the passive state.

Step 15: Since the ON period as ended, reset the variable summa_on to zero in preparation for the record of fresh ON-period information, when the process will be in the active state.

Step 16: Arrival at this step implies either the continuation of the previous ON period, or the start of a new ON period. So increment the variable summa_on and assign the value of cycle \( k \) to the state variable.

Step 17: Arrival at this step implies either the continuation of the previous OFF period, or the start of a new OFF period. So increment the variable summa_off and assign the value of cycle \( k \) to the state variable.

Figure 24. DF of simulated samples of the length of OFF (a) and ON (b) periods: curve 1 – experiment, curve 2 – simulation
Step 18: At this step of the algorithm, the system just transited to the next state, so turn to the next iteration of the parameter $i$.

Step 19: End program.

As a result, the amount of packets falling either in the received state or the lost state in a row is accumulated

$\text{\textit{summa}}_{\text{ON}} = \text{\textit{summa}}_{\text{ON}} + 1$.

4. SIMULATION RESULTS

DF of ON- and OFF-processes for both the simulated and experimental sequences obtained using the described Markov model, shown in Figure 24.

The numerical experiments have shown that increasing the number of states of the Markov model describing the packetization of errors, allows to obtain a satisfactory correspondence between the results of the experimental data and the data obtained by simulation.

Figure 25. Distribution of error-free and erroneous values for arrays Nº1 and Nº2

Figure 26. Distribution of errors in a group of errors for arrays Nº1 and Nº2
5. MARKOV MODEL OF PACKETIZATION OF ERRORS: SIMULATION RESULTS

Two independent data sets, each containing 300,000 values were generated with the aid of the developed Markov model. This amount of data will allow for comparing the number of RTP packets, according to results of the experiment conducted on the transmission of a 30-minute streaming video on a real WiMAX network, with the results of the experiment conducted using the HSC. Each value

Figure 27. Block diagram of the experiments Nº1 and Nº2

![Block diagram of the experiments Nº1 and Nº2](image)

Figure 28. Change in PSNR indicator from experiments Nº1 and Nº2

![Change in PSNR indicator from experiments Nº1 and Nº2](image)


Sanneck, H., Mohr, W., Le, L., & Hoene, C. (2002). Quality of service support for voice over IP over wireless. Wireless IP and Building the Mobile Internet.


Project P905-PF EURESCOM. (2000). *Aquavit-Assessment of quality for audio-visual signals over internet and UMTS.* EURESCOM.


ETSI TR 102 493. (n.d.). *Guidelines for the use of video quality algorithms for mobile applications*. ETSI.


reception traces were obtained in the transmission over an “ideal” channel with unlimited bandwidth and no delay in the NS-2 environment. Then each packet of the receive trace was matched with a corresponding value from the data set array (packet id = serial value of the data set array). All packages corresponding to 1 (erroneous) were deleted. This allowed for simulating sequence of errors that occur in the network and to effect corrective decoding of the video stream. Thus, two experiments were carried out: with arrays Nº1 and Nº2. Figure 27 shows a block diagram of the experiments.

The results of experimental quality indicators obtained are shown in Figures 28 through 30.

Analysis of the quality of received video sequence when simulating Markov model of error packetization shows that the average quality video sequences is slightly worse than during transmission over a real network. For example, in an experiment on streaming video over a real WiMAX network, the average quality of 31 dB was obtained, and for the simulation 26 dB and 28 dB respectively. The subjective MOS quality indicator also shows a difference in values: a real WiMAX network returned a mean value of 3.59 (corresponding to satisfactory), while the experiments returned values of 2.72 (corresponding to poor) and 3.01 (corresponding to satisfactory), respectively. This suggests that the Markov model of packetization of error obtained from a real network for streaming video can be used in the simulation of transmission of video across networks in the HSC structure.

The average quality of video sequences when simulating Markov model packetization of errors are similar to those obtained when simulating single packet errors with PER index in the range of $3 \times 10^{-3}$ to $1 \times 10^{-2}$. While the length of error group depending on the PER index of the specified range be attain values of $BEL \leq 10$.

**CONCLUSION**

We have presented in this chapter a detailed discussion of the fundamental concepts of video streaming. Characteristics of streaming traffic were enumerated and discussed and relevant parameters that characterize service continuity were mentioned. The peculiarities of transmitting H.264/AVC standard video over wireless access networks were highlighted. The broadband wireless access network error models were explained, serving as background information for the report on the research findings of investigating the effect of video streaming errors on the quality of transmission media, with particular emphasis on broadband wireless access conduit as presented in sections two and three of the chapter.

**REFERENCES**


in the array is represented by the numbers 0 or 1, where 0 means error-free value, and 1 - erroneous value. Figure 25 shows the distribution of data set values, where the white areas correspond to error-free values (0), and black - erroneous values (1).

The first array contained 2,743 (0.91%), and the second had 2,430 (0.81%) error values. The distribution of the number of errors in the same group of errors is presented in the form of histograms in Figure 26.

It is shown that the distribution of errors can not be approximated by an exponential function, which confirms the validity of the Markov model. Furthermore, in order to study the influence of Markov model of packetization of errors on the quality of video streaming, simulation of the transmission of a 30-minute video in the structure of the HSC was conducted. Simulation of a network with Markov model for packetization of errors entailed the following: transmission and

![Figure 29. Histogram and distribution function of the PSNR indicator in experiments Nº1 and Nº2](image)

![Figure 30. Quality Gradation in MOS value for video broadcast in experiments Nº1 and Nº2](image)

**KEY TERMS AND DEFINITIONS**

**Bit Error Rate (BER):** Error probability of binary message elements, calculated before the channel decoder.

**Markov Model:** A stochastic model with a finite number of states and probabilities of transition from state to state used for modeling real-life events.

**Model/Modeling:** A model is a representation of an object, system or concept in a form different from that in which it exists naturally. A model is thus essentially an instrument for forecasting the effect of input signals on a given object, while modeling is a method of improving the reasoning efficiency and intuitive capacity of specialists.

**Signal-to-Noise Ratio (SNR):** A quantitative measure of signal quality calculated as a ratio of transmitted signal power to the power of Nyquist noise in the transmission medium.