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THE OPTIMIZATION OF VIDEO TRANSMISSION QUALITY IN WIRELESS NETWORKS

Dinu Ţurcanu *, ORCID: 0000-0001-5540-4246,
Tatiana Ţurcanu, ORCID: 0000-0002-8972-8262

Technical University of Moldova, 168 Stefan cel Mare Blvd., Chisinau, Republic of Moldova
*Corresponding author: Dinu Ţurcanu, *dinu.turcanu@adm.utm.md*

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Abstract. The transmission of video stream over wireless networks is used by many applications and services, from the use of equipment in home communication networks to those carrying the video surveillance stream. Numerous technical challenges may arise when the unpredictable characteristics of the radio channel align or do not align with the stipulated requirements for transmission bandwidth and the necessary latency for transporting video data. All these challenges prevent users from experiencing smooth video streaming. Depending on the application scenarios, video service emphasis may vary according of *Quality of Experience - QoE* or parameter used traditionally in the networks, *Quality of Service - QoS*. The purpose of the research consists in optimization of the flow service, which ensures the adaptability of video communication to changes of conditions over the wireless computer network. As a result of a thorough analysis of the video playback characteristics related to the wireless channel, the most optimal approach in optimizing the quality of the video stream service over a wireless LAN requiring minimal modifications was identified, and namely system centric approach.

Keywords: *QoS, wireless network, video streaming, latency.*

Rezumat. Transmisia fluxului video prin rețele wireless este utilizată pentru multe aplicații şi servicii, de la utilizarea echipamentelor în rețelele de comunicații de la domiciliu la cele cu transport a fluxului de supraveghere video. Diverse provocări de ordin tehnic pot apărea în momentul când natura imprevizibilă a canalului radio îndeplinește sau nu cerințele fată de banda de transmisiune și față de latența necesară pentru transportul informației video. Toate aceste provocări împiedică utilizatorii să experimenteze fluxul video fără probleme. În funcție de scenariile aplicației, accentul serviciului video poate varia în funcție de *Calitatea Experienței - QoE* sau parametrul utilizat în mod tradițional în rețele, *Calitatea Serviciului - QoS*. Scopul cercetării constă în optimizarea serviciului de flux, care să asigure aplicațiilor video adaptabilitate la schimbările condițiilor peste rețelele locale fără fir. În rezultatul unei analize atente a caracteristicilor redare video referitoare la canalul fără fir, a fost identificată cea mai optimă abordare în optimizarea calității serviciului fluxului video pe o rețea fără fir LAN care necesită modificări minime, și anume - abordarea centrată pe sistem.

Cuvinte cheie: *QoS, rețea fără fir, video streaming, latență.*

1. Introduction

Video transmission through *Wireless Local Area Network - WLAN* is currently considered one of the most interesting applications of contemporaneity. As mobile data costs remain high and most users use non-wired communication, the video data transmitted over WLAN will grow. Video broadcast streaming and social video transmission encompass a category of data that is captured and delivered to end users in real-time [1].

Despite significant technological advances in recent decades, implementation advantages of mobility in the network, transmission of video data over WLAN faces multiples challenges. These challenges are distinguished by two main parameters: the wireless centric-network approach and the end-user-centric approach (based on streaming/video communication), Tabel 1 [2].

Table 1

Challenges of video streaming in wireless networks

Wireless network challenges	Challenges related to video communication/streaming
Fading	Video stream/transmission compression.
Limited bandwidth and its dynamic variation	For applications requiring real-time video communication:
Interference	The latency associated with encoding of the video data and decoding procedures.
	The computational demands of a video encoder (in terms of power consumption).

Source: Table made by the author based on [3].

In the first scenario, the QoS approach is employed to configure the network nodes in such a way that ensures the fulfillment of requirements related to data rate, latency, and packet binding. In the second scenario, end-user-centric approaches involve a collection of control techniques operating on the application layer, without requiring layer 3 OSI model QoS support [2,4].

Among the most widespread problems related to the wireless network we can mention:

Fading: this occurrence originates from multipath transmission, wherein the received signal consists of a series of weakened, phase-shifted and time-delayed duplicates of the transmitted signal.

Interference: this occurs when signals from other sources share the same frequency, typically leading to a degradation in the quality and capacity of the wireless link.

Limited bandwidth and its dynamic variation: at times, limited bandwidth and its dynamic fluctuations can occur, often due to protective mechanisms like binary exponential backoff adapted to protocol overhead and rate of transmission.

Considering that video data often consumes the lion's share of available bandwidth, especially when compared to other media types, the optimization of the video stream becomes paramount for the effective deployment of practical systems. Enabling fully scalable video streaming within the constraints of WLANs with stringent latency requirements is of considerable importance for various multimedia applications. In this context, we discuss about *Hybrid Coordination Function (HCF)* [4]. The most current challenges focused on video communication/streaming are:

Video stream compression. Uncompressed video streams require high bandwidth and generate high data rates. Reducing the redundancies of the video stream allows compression to be achieved, and thus high bandwidth is not required, but it makes the bitstream vulnerable to errors and channel distortion. As a result, it becomes essential to employ error-tolerant techniques when transmitting compressed video bitstreams across error-prone channels, including WLAN and IP networks.

The latency associated with encoding of video data and decoding processes is a critical factor in real-time data transmission, particularly in video conferencing. While encoders of video data such as H.264/AVC provide cutting-edge compression performance, their complexity and time-consuming nature increase when employing a significant number of optimized encoders.

The computational complexity of a video encoder (in terms of energy consumption) for portable devices requires low-complexity algorithms.

All these challenges/obstacles prevent users from experiencing smooth video streaming [5–7]. Depending on specific application scenarios, the priorities for video services can vary. For instance, in video streaming platforms like Netflix and YouTube, a slight delay at the beginning of playback or transmission (typically a few seconds) is acceptable. However, in the context of video conferencing applications like FaceTime and WebRTC, maintaining an even lower delay (often less than a few hundred milliseconds) becomes crucial. This strict requirement arises because, in real-time data systems, the frame must be delivered and Protocol Data Unit (PDU) decoded precisely according to its scheduled playback time. Any retransmitted packet, resulting from excessive delay, becomes impractical when it cannot meet the stringent deadlines for decoding and display [8].

Improving the quality of video transmissions, often quantified by metrics like *Peak Signal to Noise Ratio*, is of paramount importance [9]. As such, the research endeavors to optimize the flow service, thereby enhancing the ability of video applications to adapt to fluctuations in WLAN conditions [10].

2. Technical characteristics of video transmission in non-wired networks

The challenge to transmitting real-time data over wireless channels has undergone extensive scrutiny in recent years. Video applications demand a consistent network stream while imposing stringent requirements on delay, particularly in interactive communication scenarios. Conversely, wireless channels exhibit high dynamism, characterized by rapid fluctuations in *Bit Error Rate (BER)* that can vary significantly in under a second [11].

Numerous approaches have been suggested to address the demands of secure video communication over noisy channels. These solutions encompass techniques such as *Adaptive Modulation (AM)*, *Forward Error Correction (FEC), Automatic Repeat Request (ARQ), Joint Source-Channel Coding (JSCC)* for retransmission, robust source coding, adaptive channel-source coding, data partitioning, post-processing error concealment, and scalable transport-priority coding [12–15].

A significant portion of research concerning video streaming over wireless channels has concentrated on enhancing the performance of source encoders, with minimal adjustments to network-related aspects. Nevertheless, these investigations frequently overlook the dynamics associated with the playback/recording buffer, a critical component for ensuring uninterrupted video playback. Furthermore, certain approaches within this domain entail substantial computational demands, rendering them less practical for real-time operations.

Presently, researchers are employing a multi-layer architecture to address challenges in video streaming, including issues related to data transfer, latency, jitter, and packet loss.

These challenges typically emerge due to the substantial volume of video data being streamed and the inherent characteristics of WLAN [16]. The cross-layer concept was introduced in the context of video communication over WLAN's, aiming to enhance overall system performance [17].

Rate control plays a crucial role in multimedia video streaming applications by optimizing the utilization of congested links, ensuring adequate send rates. Additionally, it safeguards against congestion collapse by moderating send rates to prevent excessive aggression. Lastly, fair rate control promotes equity among users who share shared network links [18,19].

The aim of this approach is to establish a concept of cross-layer system facilitating communication various MAC layers, enabling seamless cross-layer mapping and QoS adaptation. This approach is centered on end-systems and strives to enhance the ability of video applications to adjust to evolving wireless network conditions [20,21].

A commonly used scheme for video transmission in wired networks is referred to as *TCP-Friendly Rate Control (TFRC)*. TFRC calculates a rate by considering factors such as RTT, packet loss rate, and message size to replicate the stable and uniform performance of the *Transmission Control Protocol*. However, in wireless networks, packet loss primarily arises from physical channel errors, rendering this rate control scheme inappropriate. Neither TCP nor TFRC can differentiate between loss of packet resulting from overflow of the buffer and loss due to errors at physical layer in wireless environments [19,22,23].

Numerous attempts have been made to enhance the performance of TFRC / TCP in WLAN settings. These approaches either shield end hosts from packet loss attributable to errors in the non-wired channel or empower end terminal equipment differentiate between loss of packet arising from congestion and resulting from radio channel issues.

These proposed solutions offer rate control mechanisms at the source-to-send end system, allowing for adjustments in video coding and encoder buffer parameters to accommodate changes in channel transmission conditions. An alternative rate control method involves region-based coding, utilizing a block segmentation technique to isolate regions of interest within the source video streams and quantitatively reduce the data to be transmitted [24].

Quality-related challenges in video streaming pertain to the implementation of a rate control system aimed at preventing degradation, as measured by PSNR, which might otherwise result in a reduction of the bit rate.

This model incorporates conditional resending and employs interleaving strategy with low-delay, utilizing the encoder buffer like an integral component of the interleaving memory. PSNR metric has traditionally served as a reference point for devising these metrics. Nonetheless, certain research indicates that PSNR may not be well-suited for accurately assessing visual quality [25].

Certain authors propose a rate control mechanism founded on a stochastic model, applied prior to both the source and the priority channel [21]. In alternative research, an H.264-bit allocation rate control scheme is suggested, wherein channel throughput is assessed at the frame level and the unit base level.

Moreover, this strategy takes into account the state of the encoder buffer and the prevailing channel conditions. [26,27].

Many of these investigations tend to overlook the dynamics associated with the playback buffer, an important factor to ensure uninterrupted video playback. Additionally, the high computational demands of the proposed frameworks often render them less appealing for real-time applications.

3. Improving video streaming quality for specific services in WLANs

The basic task of an information system is to transfer a PDU from one point to another while satisfying certain QoS requirements and resource constraints. QoS requirements may include BER, *Packet Error Rate (PER)*, data transfer rate, and latency [28].

Rate control assumes a significant role modern networks, particularly in real-time video communication scenarios. It stands as one of the fundamental technologies in video encoding. The absence of rate control in any video compression standard can severely constrain its practical application [29,30].

In a standard wireless network environment, source rate control involves the allocation of radio resources among a varying number of equipment's employing a mechanism - Control Contention-Based (CCB). This characteristic, coupled with the fluctuating traffic load and BER variations, results in a bursty channel throughput pattern, as experienced by each end-user who competes for channel access alongside other stations [31−33]. ARQ (Automatic Repeat Request) suggests the retransmission of data, although forwarding is not typically favored, especially in long-distance and wireless communications, such as satellite links. In FEC, redundancy is introduced to prevent errors, where additional bits are incorporated alongside data to create encoded data. Despite this augmentation increases transmission payload and is referred to as channel coding. To enhance reliability, a combination of FEC and ARQ, known as *Hybrid-ARQ (HARQ)*, can be utilized [34]. In this context, we consider ARQ policy named stop-and-wait. This choice is appropriate when the RTP latency is shorter than transmission packet time, as is typically the case in WLANs.

Architecture and basic hypothesis. The video data transmission system's structure comprises a mobile client station, establishing communication in WLAN with the video data server. The server may either be a mobile hardware equipment or a stationary system connected through a non-wired network to the *Access Point (AP)* of the wireless channel.

Typical scenarios related to video streaming service include:

- the video source;
- the display;
- the transceiver channel;
- the encoder and the decoder;
- the corresponding buffers.

A vital element introduced in the proposed algorithm, the Rate Control Algorithm (RCA), is the source rate scheme, managed by server level. This module assumes crucial task of calibrating the source rate to its optimal value, leveraging periodic feedback associated with both channel bandwidth and playback buffer occupancy.

We presume that the core encoder possesses the capability to adapt its encoder settings in order to attain the targeted bit rate as determined by the RCA. Encoding and rate of playback, denoted as R_f, is expressed in frames per second and remains constant throughout the video streaming session. In contrast, R_{in} represents the frame reception rate at the client's end, and it fluctuates in response to changing conditions of the radio channel.

The video transmission can be real-time generated or sourced from a data archive. Once a frame of video data undergoes encoding, it is divided into one or more pieces of data, subsequently directed to the layer 2 - Media Access Control, for transmission over the radio channel. Every packet comprises a total data of $T = L + H$, where L signifies inlet bit count and H represents the error bits correction number. It is assumed that value of L remains constant for all information throughout the video session, while the H value remains consistent for all packets associated with the same real-time data frame.

For a given coding scheme, such as the BCH code, and specific pair of parameters (*L, T*), it's straightforward to determine the Bit Errors maximum number that can be corrected within PDU ($E_{max} = E_{max}(L, T)$). Additionally, the ARQ rule allows receiver transmit either a positive (ACK) acknowledgment or a negative (NAK) acknowledgment through a data channel. Depending on this feedback, the received PDU is deemed correct or not correct. In the event of receiving a NAK message, the sender will initiate the retransmission of the packet. The period required to transmit a single PDU over the channel, as well as the retransmissions number necessary for successful packet delivery to the receiver, can fluctuate over time due to line contention and variations in the Bit Error Rate (BER).

Following the adopted model, ARQ resending are overseen by the ARQ architecture, which gathers positive / negative messages from the destination and can buffer data at the encoder when retransmission is necessary. By processing incoming acknowledgments, the ARQ system is also capable of determining interval needed for successful transmission of a PDU. Specifically, when the initial attempt to transmit a packet commences, the buffer encoder sends a signal to architecture ARQ, awaiting the appropriate ACK and computing the total time duration essential for successful packet delivery. Consequently, upon the receipt of each positive acknowledgment, a response is dispatched to main module, conveying the time elapsed for sending the most recent packet. It was assumed that the ARQ messages utilize the response channel from the decoder and temporarily occupy the buffer of playback. Given that monitoring packets are compact, they can be efficiently protected with FEC, ensuring transmission error-free of responses through the channel. Consequently, the ARQ system can accurately compute the period required for successful transmission of each PDU. This information, coupled with the playback occupancy buffer, is then relayed to the source rate of the module of control.

The main goal of the approach was to maintain uninterrupted playback of video by minimizing occurrence of events of starvation within the playback buffer. To count this, the buffer starvation probability was employed as a widely accepted metric, serving as an indicator of the likelihood of buffer depletion during video playback [35,36]. To mitigate the adverse consequences of buffer starvation, efforts are directed toward identifying an appropriate startup latency configuration strategy for video stream transmission.

To accomplish the intended goal, an approach based on time windows was implemented. This method involves adjusting the coding rate of source while considering its impact over a fixed-size entire window. This strategy allows the buffer playback to accumulate packets in interval of high bandwidth, preventing starvation during time of low throughput. Consequently, this approach balances channel throughput without introducing the undesirable "saw" effect often associated with rate control frame-based. In this scheme, the axis of the time is segmented into consistent size windows, denoted as T. Each window is identified by an index, represented as m. At the outset of each window time, the module of rate control calculates likelihood of a starvation event occurring in the buffer playback during the window subsequent. Based on buffer occupancy and channel behavior information, it then establishes the coding rate of source, denoted as $R_{s,\ m}$, to be used throughout the current window. The current rate is chosen to maximize the video quality expected, ensuring that buffer starvation probability remains below the threshold desired.

The buffer starvation probability represents an important performance measure for communication network protocol design and usually occurs when the buffer is empty. The starvation is undesirable, both in real-time voice and in video stream applications. Considering buffer occupancy $(Q_m)(\tau)$, during the window identified by an index, represented as "m", where $0 < \tau \leqslant T$, which develops according to the following equation:

$$Q_m(Q_m) = \frac{R_f}{R_{s,m}} Y_m(\tau) - R_f \tau, \;\; 0 < \tau \leq T, \quad (1)$$

here $Y_m(\tau)$ is defined as $X(\tau + mT) - X(mT)$, where the arrival process $X(t)$, characterized as a Markovian process that tallies the accumulation of traffic arrivals reaching the rendering buffer within the interval $(0, t)$ [37].

Buffer occupancy [38,39] is determined by count of PDU within rendering buffer at the initiation time (Q_m) of the window m-th. This count is incremented by the number of PDU transmitted during the interval τ and subsequently reduced by the number of PDU rendered within the same period (1). In simpler terms, it's the disparity between the incoming and outgoing frames within the considered interval. $R_{s,m}$ signifies the Rate Bit Source Encoding for the window m-th, and this value is dictated by the control module to encoder. The control module's goal is to compute $R_{s,m}$ such that it guarantees a lower probability of buffer starvation compared to the threshold desired. Once the requirement is met, the rate maximum value is selected [38,39].

If $\Phi_m(\tau)$ denotes the probability of buffer starvation experiencing, it is imperative:

$$\Phi_m(\tau) = Pr\left[Q_m + \frac{R_f}{R_{s,m}} y_m(\tau) \leq R_f \tau\right] \leq \varepsilon, 0 < \tau \leq T, \quad (2)$$

here $0 < \varepsilon << 1$ serves as a default threshold. This equation is established separately for each interval of time within the flow, each of which equals T.

To adhere to the constraints (1, 2), Q_m necessitates specific information regarding occupancy of the buffer playback and the traffic of network anticipated in the upcoming window. The initial occupancy data is directly supplied by ARQ response loop, as outlined in the section preceding. For the initial window, Q_0 is set to N, representing the number of pre-loaded frames before initiating playback. Conversely, details about the $Y_m(\tau)$ process remain unknown, but can be forecasted based on latency transmission observed in previously sent packets, with this latency information retrieved through the return channel. Once all the pertinent data is gathered, we assign the probability $P_\tau(n)$ to denote the likelihood of successfully transmitting precisely n frames within the τ interval. Calculating $P_\tau(n)$ considers the transmission of successful $n+1$ frames, time intervals t_i ($i = 1, ..., n + 1$), resulting in the following relationships: $\sum_{i=1}^{n} t_i \leq \tau \;\; e \; t_{n+1} \geq \tau - \sum_{i=1}^{n} t_i \cdot t_i$. These relationships signify the required time to effectively transmit packet i for the server. It denotes the duration between the first attempt to transmit packet i and the moment of receiving the positive message. It's worth noting that the same frame could be transmitted multiple times within interval t_i before achieving successful reception at the destination. Assuming that t_i is independently and identically distributed, we can simplify by omitting the index i and define $f_d(t)$, corresponding to the probability density function (*pdf*) associated with t. It will also be taken into account the known $f_d(t)$ and the information about this function will be directed to a specific section. Therefore, $P_\tau(n)$ could be calculated as follows:

$$P_\tau(n) = \int_0^\tau \int_0^{\tau - t1} \dots \int_0^{\tau - t_1 \dots t_n} \int_{\tau - t_1 \dots t_n}^{\infty} \prod_{i=1}^{n+1} f_d\,(t_i)\, dt_1 \dots dt_{n+1} \quad (3)$$

Using this probability as a foundation, we can calculate *pdf* $f_{y_{\tau,m}}(\cdot)$ for $Y_m(\tau)$ process using the following relationship:

$$f_{Y_{\tau,m}}(y) = \sum_{n=0}^{\infty} P_t\,(n) \pi_{L,n}(y), \quad (4)$$

where $\pi_{L,n}(.)$ is a rectangular function that equals one within the intervals *(n-1) L ÷ nL*, with *L* representing the PDU dimension. Additionally $f_{Y_{\tau,m}}$ (.) is a function that displays piecewise continuity [38,39].

Now, let's consider that for the present window *m*, the value of $R_{s,m}$ has been set as a constant. According to (1) it results that starvation will occur whenever we have:

$$\Phi_m(\tau) \leq Y_\tau^* = \left(R_f\tau - \mathcal{Q}_m \frac{R_{s,m}}{R_f}\right), \tag{5}$$

and the probability of starvation will be:

$$\Phi_m(\tau) = \int_0^{Y_\tau^*} f_{Y_{\tau,m}}(y)dy \tag{6}$$

In the frame proposed in reverse, a starvation probability value is imposed and it determines the corresponding value of $R_{s,m}$. Therefore, it is initially calculated Y_τ^*, which is obtained from relationship (6), that makes $\Phi_m(\tau) = \varepsilon$ for $0 < \tau \leqslant T$. Subsequently, based on Y_τ^*, we can calculate $R_{s,m}$ according to the relationship (5).

The suggested algorithm hinges on the *pdf* availability. To ensure the efficacy of the control rate scheme, it necessitates straightforward statistical approach to forecast the *pdf* of upcoming transmissions by drawing insights from observed outcomes. Furthermore, given the time-varying nature of network characteristics, the statistical pattern of latency also experiences fluctuations.

Hence, it becomes imperative to continually refresh this data to accommodate network variations. Recent advancements have yielded several techniques for handling statistical patterns derived from observed measurements. The complete aggregation method aggregates the data into a probability distribution curve, attributing equal weight to each sample-whether old or recent. This method is straightforward in its approach, treating recent and older data with equal significance concerning their impact on the probability distribution. However, it lacks the capability to promptly respond to shifts in network traffic.

Conversely, refresh method and flush and constructs the probability curve of distribution exclusively from most recent packets, discarding older data and incurring additional overhead expenses. Implementing periodic flushing entails the complete removal of historical information, which could potentially introduce boundary effects at the flush points.

There is a necessity for a suitable statistical approach capable of forecasting future network latency by analyzing both historical and present data. This method should facilitate the monitoring, maintenance, updating, and retention of statistical patterns related to network delay.

Aging techniques. Aging refers to the accumulation of damage over time, leading to an elevated vulnerability to collapse, deterioration, or eventual failure. This phenomenon occurs due to interdependencies among components, where the failure of one component can have negative repercussions on its dependents. Recent research has been dedicated to unraveling the dynamics of these processes, revealing temporal cascades involving scaling and failure. Remarkably, these dynamics are capable of reproducing empirical survival patterns observed in various biological organisms and technological devices [40–42]. Aging techniques are applied to improve informational efficiency and thus to predict delay.

An intermediary approach is proposed here, emphasizing information storage and tracking. In this method, old data isn't entirely discarded; instead, its influence on the statistical distribution is gradually diminished. Each value within the histogram is established by counting the occurrences of packets sent within a specific latency interval. It's worth noting that a smaller recycling bin or base unit for the time interval results in a more pronounced prediction of $f_d(t)$. Conversely, a smaller bin size escalates the numerical computational complexity of algorithm (3). To mitigate the impact of older samples, each histogram bin is periodically fine-tuned, diminishing its contribution through an aging factor denoted as *F*. This operation, commonly referred to as aging, is implemented at a predefined frequency marked as *f* [32].

4. Conclusions

The introduction of (next generation) electronic communication networks, based on the IP paradigm, allows increasing network productivity by optimising resource management and increasing the attractiveness of the services offered to the user. However, the introduction of new NGN platforms should be implemented in symbiosis with the use of appropriate Quality of Service management procedures. In fact, an NGN solution, compared to the described advantages, leads to an increase in the complexity of managing QoS control tools compared to traditional network solutions.

Based on research, it was identified that the main challenge in the context of QoS optimization of video streaming over WLANs consists in network optimization method, i.e., to design a rate control algorithm, which would maximize the quality of video streaming application and channel utilization.

As a result of a thorough analysis of the video playback characteristics related to the wireless channel, the most optimal approach in optimizing the quality of the video stream service over a WLAN requiring minimal modifications was identified, and namely the end-system centric approach.

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References

1. Qiu, Q.; Zuo, Y.; Zhang, M. Can Live Streaming Save the Tourism Industry from a Pandemic? A Study of Social Media. *IJGI* 2021, 10, 595, doi:10.3390/ijgi10090595.
2. Qian Zhang; Wenwu Zhu; Ya-qin Zhang End-to-End QoS for Video Delivery Over Wireless Internet. *Proc. IEEE* 2005, 93, pp. 123–134, doi:10.1109/JPROC.2004.839603.
3. Su, G.-M.; Su, X.; Bai, Y.; Wang, M.; Vasilakos, A.V.; Wang, H. QoE in Video Streaming over Wireless Networks: Perspectives and Research Challenges. *Wireless Netw* 2016, 22, pp. 1571–1593, doi:10.1007/s11276-015-1028-7.
4. Țurcanu, D. Quality of services in MPLS networks. *Journal of Engineering Science* 2020, 5, 27(3), pp. 102–110, https://doi.org/10.5281/zenodo.3949674.
5. Correa, G.; Assuncao, P.; Agostini, L.; Da Silva Cruz, L.A. Performance and Computational Complexity Assessment of High-Efficiency Video Encoders. IEEE Trans. *Circuits Syst. Video Technol.* 2012, 22, pp. 1899–1909, doi:10.1109/TCSVT.2012.2223411.
6. Morrison-Smith, S.; Ruiz, J. Challenges and Barriers in Virtual Teams: A Literature Review. *SN Appl. Sci.* 2020, 2, 1096, doi:10.1007/s42452-020-2801-5.
7. Suduc, A.-M.; Bizoi, M.; Filip, F.G. Status, Challenges and Trends in Videoconferencing Platforms. *Int. J. Comput. Commun. Control* 2023, 18, doi:10.15837/ijccc.2023.3.5465.
8. Wei, J.; Venkatakrishnan, S.B. DecVi: Adaptive Video Conferencing on Open Peer-to-Peer Networks 2022.
9. Tsurcanu, D.; Nistiriuk, A.; Chihai, A.; Nistiriuk, P.; Baxan, L.; Andronic, S.; Nistiriuk, P.; Russu, G.; Alexei, A; Iazlovetkii, M.; Tsurcanu, T. Evaluation of Bit Error Rate probability for radio communications and fiber-

optic communication systems. In: *24th International Crimean Conference Microwave & Telecommunication Technology*, 2014, pp. 230-233, https://doi.org/10.1109/CRMICO.2014.6959370.

10. Sethi, P.; Sarangi, S.R. Internet of Things: Architectures, Protocols, and Applications. *Journal of Electrical and Computer Engineering* 2017, 2017, pp. 1–25, doi:10.1155/2017/9324035.
11. Tsurcanu, D; Nistiriuk, P.; Alexei, A.; Tsurcanu, A.; Beregoi, E.; Tsurcanu, T.; Smokin, D. MPLS Network Hardware Reliability. In: 1*7th International Crimean Conference - Microwave & Telecommunication Technology*, 2007, pp. 233-235, https://doi.org/10.1109/CRMICO.2007.4368711.
12. Hassan, M.M.; Farooq, U. Adaptive and Ubiquitous Video Streaming over Wireless Mesh Networks. *Journal of King Saud University - Computer and Information Sciences* 2016, 28, pp. 432–446.
13. Ibrahim, S.K.; Khamiss, N.N. A New Wireless Generation Technology for Video Streaming. *Journal of Computer Networks and Communications* 2019, 2019, pp. 1–9, doi:10.1155/2019/3671826.
14. Morshedi, M.; Noll, J. Estimating PQoS of Video Streaming on Wi-Fi Networks Using Machine Learning. *Sensors* 2021, 21, 621, doi:10.3390/s21020621.
15. Švigelj, A.; Junuzović, M. Network Coding-Assisted Retransmission Scheme for Video- Streaming Services over Wireless Access Networks. In: *Broadband Communications Networks - Recent Advances and Lessons from Practice*; Edited by Abdelfatteh Haidine and Abdelhak Aqqal, 2018, 432 p. ISBN 978-1-78923-742-9, 10.5772/intechopen.69590.
16. Bin-Salem, A.; Wan, T.C. Survey of Cross-Layer Designs for Video Transmission Over Wireless Networks. *IETE Tech Rev* 2012, 29, 229, doi:10.4103/0256-4602.98865.
17. Chen, X.; Cao, B.; Ahmad, I. Lightweight Neural Network-Based Viewport Prediction for Live VR Streaming in Wireless Video Sensor Network. *Mobile Information Systems* 2021, 2021, pp. 1–12.
18. Haratcherev, I.; Taal, J.; Langendoen, K.; Lagendijk, R.; Sips, H. Automatic IEEE 802.11 Rate Control for Streaming Applications. *Wireless Communications* 2005, 5, pp. 421–437, doi:10.1002/wcm.301.
19. Minghua Chen; Zakhor, A. Rate Control for Streaming Video over Wireless. In: P*roceedings of the IEEE INFOCOM 2004*; IEEE: Hong Kong, China, 2004; 2, pp. 1181–1190.
20. Goudarzi, P.; Tadayon, M.; Mousavinejad, M. An Optimization Theoretic Framework for Video Transmission with Minimal Total Distortion over Wireless Networks. *J Wireless Com Network* 2009, 598063, doi:10.1155/2009/598063.
21. Zhu, X.; Setton, E.; Girod, B. Congestion–Distortion Optimized Video Transmission over Ad Hoc Networks. *Signal Processing: Image Communication* 2005, 20, pp. 773–783, doi:10.1016/j.image.2005.05.005.
22. Floyd, S.; Handley, M.; Padhye, J.; Widmer, J. Equation-Based Congestion Control for Unicast Applications. *SIGCOMM Comput. Commun. Rev.* 2000, 30, pp. 43–56, doi:10.1145/347057.347397.
23. Floyd, S.; Kohler, E. TCP Friendly Rate Control (TFRC): The Small-Packet (SP) Variant; RFC Editor, 2007; p. RFC4828.
24. Zhang, X.; Xu, Y.; Hu, H.; Liu, Y.; Guo, Z.; Wang, Y. Profiling Skype Video Calls: Rate Control and Video Quality. In: *Proceedings IEEE INFOCOM*; IEEE: Orlando, FL, USA, March 2012, pp. 621–629.
25. Bondzulic, B.P.; Pavlovic, B.Z.; Petrovic, V.S.; Andric, M.S. Performance of Peak Signal-to-noise Ratio Quality Assessment in Video Streaming with Packet Losses. *Electron. lett.* 2016, 52, pp. 454–456.
26. Gragido, W.; Pirc, J.; Selby, N.; Molina, D. Signal-to-Noise Ratio. In: *Blackhatonomics*; Elsevier, 2013, pp. 45–55, ISBN 978-1-59749-740-4.
27. Kwon, D.; Shen, M.; Kuo, C. -c. A Novel Two-Stage Rate Control Scheme for H.264. In: *Proceedings of the 2006 IEEE International Conference on Multimedia and Expo*; IEEE: Toronto, ON, Canada, July 2006, pp. 673–676.
28. Ahmed, T.; Mehaoua, A.; Boutaba, R.; Iraqi, Y. Adaptive Packet Video Streaming over IP Networks: A Cross-Layer Approach. *IEEE J. Select. Areas Commun.* 2005, 23, pp. 385–401, doi:10.1109/JSAC.2004.839425.
29. Santamaria, M.; Blasi, S.; Izquierdo, E.; Mrak, M. Analytic Simplification of Neural Network Based Intra-Prediction Modes For Video Compression. In: P*roceedings of the 2020 IEEE International Conference on Multimedia & Expo Workshops (ICMEW)*; IEEE: London, UK, July 2020, pp. 1–4.
30. Yan, T.; Ra, I.-H.; Zhang, Q.; Xu, H.; Huang, L. A Novel Rate Control Algorithm Based on ρ Model for Multiview High Efficiency Video Coding. *Electronics* 2020, 9, 166, doi:10.3390/electronics9010166.
31. Malik, A.; Qadir, J.; Ahmad, B.; Alvin Yau, K.-L.; Ullah, U. QoS in IEEE 802.11-Based Wireless Networks: A Contemporary Review. *Journal of Network and Computer Applications* 2015, 55, pp. 24–46, doi:10.1016/j.jnca.2015.04.016.

32. Sarkar, N. *Improving the Performance of Wireless LANs: A Practical Guide*; Chapman & Hall/CRC computer and information science series; CRC Press/Taylor & Francis Group: Boca Raton, Florida, 2014; ISBN 978-1-4665-6063-5.
33. Sarkar, N.I.; Ammann, R.; Zabir, S.M.S. Analyzing TCP Performance in High Bit Error Rate Using Simulation and Modeling. *Electronics* 2022, 11, 2254, doi:10.3390/electronics11142254.
34. Kunal, K.; Tripathi, R.C.; Singh, V. An HARQ Based Optimized Error Correction Technique. *IJCA* 2010, 9, pp. 1–5, doi:10.5120/1353-1826.
35. Xu, Y.; Xiao, Z.; Feng, H.; Yang, T.; Hu, B.; Zhou, Y. Modeling Buffer Starvations of Video Streaming in Cellular Networks with Large-Scale Measurement of User Behavior. *IEEE Trans. on Mobile Comput.* 2017, 16, pp. 2228–2245, doi:10.1109/TMC.2016.2616402.
36. Xu, Y.; Altman, E.; El-Azouzi, R.; Haddad, M.; Elayoubi, S.; Jimenez, T. Analysis of Buffer Starvation with Application to Objective QoE Optimization of Streaming Services. *IEEE Trans. Multimedia* 2014, 16, pp. 813–827, doi:10.1109/TMM.2014.2300041.
37. Atzori, L.; Floris, A.; Ginesu, G.; Giusto, D. Streaming Video over Wireless Channels: Exploiting Reduced-Reference Quality Estimation at the User-Side. *Signal Processing: Image Communication* 2012, 27, pp. 1049–1065, doi:10.1016/j.image.2012.09.005.
38. Chou, P.A.; Zhourong Miao Rate-Distortion Optimized Streaming of Packetized Media. *IEEE Trans. Multimedia* 2006, 8, pp. 390–404, doi:10.1109/TMM.2005.864313.
39. Huang, J.; Li, Z.; Chiang, M.; Katsaggelos, A.K. Joint Source Adaptation and Resource Allocation for Multi-User Wireless Video Streaming. *IEEE Trans. Circuits Syst. Video Technol.* 2008, 18, pp. 582–595, doi:10.1109/TCSVT.2008.919109.
40. Mitnitski, A.B.; Rutenberg, A.D.; Farrell, S.; Rockwood, K. Aging, Frailty and Complex Networks. *Biogerontology* 2017, 18, pp. 433–446, doi:10.1007/s10522-017-9684-x.
41. Stroustrup, N. Measuring and Modeling Interventions in Aging. *Current Opinion in Cell Biology* 2018, 55, pp. 129–138, doi:10.1016/j.ceb.2018.07.004.
42. Vural, D.C.; Morrison, G.; Mahadevan, L. Aging in Complex Interdependency Networks. *Phys. Rev. E* 2014, 89, 022811, doi:10.1103/PhysRevE.89.022811.

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