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CONTENTS

SPECIAL ISSUE ON "QUALITY OF EXPERIENCE FOR MULTIMEDIA COMMUNICATIONS"

Quality of Experience for Multimedia Communications

Periklis Chatzimisios, Alexander TEI of Thessaloniki, Greece
Martin Reisslein, Arizona State University, Tempe, AZ, USA
pchatzimisios@ieee.org, reisslein@asu.edu

The problem of understanding and enhancing Quality of Experience (QoE) in complex, distributed and diverse environments is continuing to be the subject of intense research investigations. Despite the efforts devoted to QoE studies, managing and controlling user QoE is still a largely open issue.

This Special Issue gathers a collection of six selected papers that present a variety of concerns and latest advances relating to Quality of Experience for Multimedia Communications.

The first article "Quality of Experience for Multimedia Communications: Network Coding Strategies" presents certain novel network approaches based on Network Coding (NC) to provide end users with the QoE that they require. In particular, the authors discuss how NC can address the main network impairments to user satisfaction, such as slow delivery over wireless networks, and optimize QoE metrics as well as trade-offs between rate and per packet delivery delays.

With the enhanced communication capability and the fast mobility of wireless devices, it is very important to enable the smooth video delivery over challenged wireless networks. The second article "Quality of Experience Oriented Video Streaming in Challenged Wireless Networks: Analysis, Protocol Design and Case Study" focuses on the design of QoE oriented video streaming system by first developing an analytical framework to characterize the QoE of users. Subsequently the authors formulate the video streaming as a cross-layer design problem and showcase the implementation of the proposed cross-layer framework in vehicular networks and cognitive radio networks.

The third article "Online QoE Computation for Efficient Video Delivery over Cellular Networks" presents a new concept for the online computation of QoE, where possible sets of metrics impacting the quality perceived by the user from the Core Network (CN) and from the Radio Access Network (RAN) are considered. The key idea is to combine metrics that specify a typical Content Delivery Network (CDN) environment with wireless metrics taking into account issues related to the entire video delivery chain, from a video source (cache) to the mobile user (terminal).

The fourth article "Quality of Experience for Multimedia in Clouds" first introduces the main groups of requirements related to the QoE support for multimedia applications based on the example of cloud-based services. The article then focuses on three important challenges for offering multi-cloud service realization: inter-cloud communications, global service mobility, and network access selection.

The fifth article "Adaptive Control based on Quality of Experience for Multimedia Communications" presents the concept of Quality of Experience, highlights the main differences with Quality of Service (QoS) and proposes a classification of QoE methods into three different approaches; usability metric, hedonistic concept, and buzzword extension. Subsequently, the authors present their user-centric research that aims at optimizing end-user QoE by tackling the problems of routing, meta-routing, and knowledge dissemination.

The last paper "Inferring and Improving Internet Video QoE" proposes MintMOS an accurate, lightweight, no-reference framework that can capture and offer suggestions to improve QoE inside the network core. The authors also suggest a simple, scalable, and efficient path selection strategy called Source Initiated Frame Restoration (SIFR) that applications can use to improve Internet video QoE.

We would like to thank all the authors for their contribution and hope these articles will stimulate further research on multimedia Quality of Experience and help by providing an up-to-date sketch of currently hot topics on Quality of Experience.

Finally, we want to express our gratitude to the Co-Director of IEEE MMTC E-letter, Dr. Chonggang Wang, for his invaluable support in coordinating this Special Issue.

Enjoy your reading!

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Periklis Chatzimisios is an Assistant Professor with the Department of Informatics at the Alexander TEI of Thessaloniki. He holds a PhD degree in Computer Engineering from Bournemouth University. He serves as a Member for the IEEE Communication Society (ComSoc) Standards Board. He also serves as an organising member for numerous conferences and he holds editorial positions for many IEEE/non-IEEE journals. His current research activities are mainly focused on wireless communications, multimedia communications, and security.

Martin Reisslein is a Professor in the School of Electrical, Computer, and Energy Engineering at Arizona State University (ASU), Tempe. He received his Ph.D. in systems engineering from the University of Pennsylvania in 1998. He currently serves as Associate Editor for the IEEE/ACM Transactions on Networking. He maintains an extensive library of video traces for network performance evaluation, including frame size traces of MPEG-4 and H.264 encoded video, at http://trace.eas.asu.edu. His research interests are in the areas of multimedia networking, optical access networks, and engineering education.

Quality of Experience for Multimedia Communications: Network Coding Strategies

Muriel Médard, Minji Kim, Ali ParandehGheibi, Weifei Zeng, Marie-José Montpetit,
Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA, USA 02139
{medard,minjikim,parandeh,weifei,mariejo}@mit.edu

1. Introduction

As the year 2012 is taking off, multimedia-viewing behaviors are changing [1]0, catalyzed by the ubiquitous availability of smartphones and tablet computers. The old paradigm of a single source serving a monolithic network and delivering content to a single device is more and more obsolete. The single delivery flow has evolved into a diffusive model of communicating nodes and converging nodes Media streaming is fast becoming the dominant application on the Internet. In this new world of immersive media viewing, new challenges are emerging for the network researchers. The behavior and reaction of the content consumers, their *Quality of Experience* (QoE) needs to be taken into account when designing the delivery mechanisms for the current generation of video-rich services.

In this paper, we present novel network approaches based on Network Coding (NC) to address the main network impairments to user satisfaction in multimedia networks: slow delivery over wireless networks (Section 2), QoE metrics such as buffering length and service interruptions (Sections 3 and 4) and trade-off between rate and per packet delivery delays (Section 5). Because NC considers the bits disseminated in Internet Protocol (IP) networks as algebraic entities, it allows creating powerful structures to provide resilience and efficiency in the network of networks that is the reality of the multimedia delivery today.

2. TCP/NC: Network Coding for efficient wireless transport

The Transmission Control Protocol (TCP) is one of the core protocols of today's Internet Protocol Suite. Many Internet applications, such as the Web, email, file transfer, and multimedia streaming, rely on TCP's promise to deliver correctly the data stream without losses or duplications. Furthermore, TCP plays an essential role in end-to-end flow control and congestion control.

TCP was designed for reliable transmission over wired networks, in which losses are generally an indication of congestion. This is not the case in wireless networks, where losses are often due to fading, interference, and other physical phenomena. TCP often incorrectly assumes that there is congestion within the network and unnecessarily reduces its transmission rate.

We propose to use NC to alleviate these problems. The work presented in [2] proposes a new protocol called TCP/NC that modifies TCP's acknowledgment (ACK) scheme such that it acknowledges degrees of freedom instead of individual packets. This is done so by using the concept of "seen" packets, in which the number of degrees of freedom received is translated to the number of consecutive packets received.

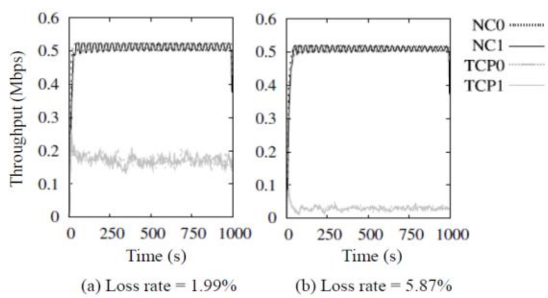

Figure 1: Throughput of TCP/NC and TCP with varying loss rates

This approach has analytically shown throughput gains of TCP/NC over standard TCP [3] further validated by simulation and experiment. The analysis and simulation results show very close concordance and support that TCP/NC is robust against erasures and failures. TCP/NC is not only able to increase its window size faster but also maintain a large window size despite the random losses, whereas TCP experiences window closing because losses are mistakenly attributed to congestion. TCP/NC still reacts to congestion; as a result, when there are correlated losses, TCP/NC also closes its window. Therefore, network coding naturally distinguishes congestion from random losses and TCP/NC is well suited for lossy wireless networks as shown in Figure 1. TCP/NC inserts a coding shim between TCP and IP layers, allowing the use of network coding to mask random losses from the congestion control mechanisms and significantly improve throughput in wireless.

3. QoE metrics and trade-offs for media streaming applications

In video streaming, the user may tolerate some initial buffering delay *before* the media playback, so that he or she has a seamless experience *throughout* the playback: there is a trade-off between the initial waiting time and likelihood of playback interruptions. These are the two of the key QoE metrics that we consider in this paper. The waiting time captures the *delay* aspect of the user experience, and the interruption probability captures the *reliability* aspect of the experience. Hence, we consider the problem of streaming a finite media file to a single receiver, over an unreliable channel, modeled via a Poisson process of rate R or other variations of it. We assume that the receiver buffers D packets, and then starts the media playback that consumes packets at unit rate. We are interested in characterizing the probability of the interruption event occurring before the file download is complete. Figure 2 illustrates the queue-length dynamics at the receiver.

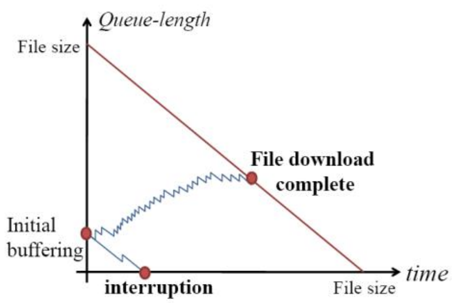

Figure 2 Dynamics of the receiver's buffer

If the arrival rate is slightly larger than the play rate, the minimum initial buffering for a given interruption probability remains bounded as the file size grows. Our reliability metric (interruption probability) is traded off with the delay metric (initial buffering). We also provide a refined trade-off characterization for finite file streams. In particular, for the case where the arrival rate of packets and their playback rate match, we observe that the required initial buffering to maintain a reasonable interruption probability grows as the square root of the file size [4].

4. QoE-Aware Streaming In Cost-Heterogeneous Networks

The streaming delay-sensitive information from multiple servers to a single receiver (user) is essential to meet the requirements of video over wireless. We consider a model that the communication link between the receiver and each server is unreliable, and hence, it takes a random period of time for each packet to arrive at the receiver from the time that the packet is requested from a particular server. One of the major difficulties with such multi-server systems is the packet tracking and duplicate packet reception problem, i.e., the receiver need to keep track of the index of the packets it is requesting from each server to avoid requesting duplicate packets. Since the requested information is delay sensitive, if a requested packet does not arrive within some time interval, the receiver need to request the packet from another server. This may eventually result in receiving duplicate packets and waste of the resources. We address this issue and show that using Random Linear Network Coding (RLNC) [5] across packets within each block of the media file we can alleviate this issue. This technique assures us that, with high probability, no redundant information will be delivered to the receiver.

NC allows is to greatly simplify the communication models and focus on end-user metrics and trade-offs. When there are multiple networks that can be used to access a particular piece of content each device must take decisions on associating with one or more such access networks. However, the cost of different access methods might be different e.g. 3G vs. Wi-Fi. and the cost of communication might be mitigated by the initial amount of buffering before playback. Figure 3 illustrates a conceptual three-dimensional cost-delay-reliability trade-off curve.

Hence, we consider a system with two classes of access methods: costly and free. Our objective is to understand the trade-off between initial waiting time, and the usage cost for attaining a target probability of interruption. Moreover, we would like to develop control policies that switch between the free and the costly servers in order to attain the target QoE metrics at the lowest cost.

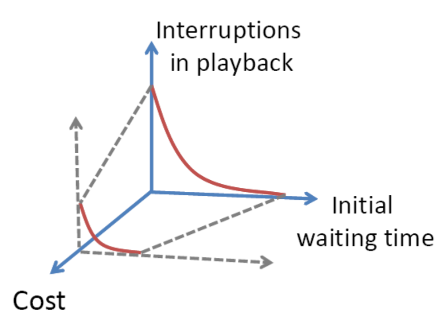

Figure 3 Trade-off between the achievable QoE metrics and cost of communication

5. Optimizing coding and scheduling for different delay sensitivities

The growing diversity of networking applications adds new challenges to the design of systems: file downloading aims solely at average rate maximization, real-time video applications seek to minimize the inter-packet delays for continuously meet successive packet delivery deadlines. And progressive video downloading, has stronger delay constraints than file downloading, but is typically less delay sensitive than real-time application.

We investigate a unified framework to characterize the rate-delay requirements for wide range of applications [6]0. In short, we define a class of delay metric $d(p)$, based on the p-norm of the in-order packet inter-arrival times. By varying p from 1 to infinity, we allow $d(p)$ to measure the some delay cost function that is increasing biased toward the large inter-arrival times. Consequently, for different values of p, optimizing $d(p)$ yields different coding block size and thus different trade-off point between $d(1)$, the average packet delay and d(¥) the maximum in-order inter-packet delay. Figure 4 illustrates the optimal trade-off curves. Based on this framework, we optimize the coding parameters and time-division scheduling parameters jointly, for efficient allocation of network resource: we assigns longer coding block sizes to relatively delay sensitive receivers for rate gains, which allow their rate requirements to be satisfied with shorter service time. As a result, more time can be allocated to serve delay sensitive receivers and increase the utility of the system.

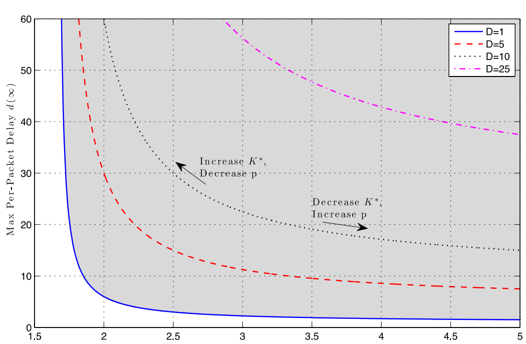

Figure 4 Trade-off curves for different feedback delay

6. Conclusions and future work

This short position paper has presented how NC can provide the current multimedia-rich networks with the mechanisms necessary to provide end users with the QoE that they require. The move to distributed consumption of content is not reversible, more and more devices are becoming multimedia-enabled including traditional television sets. We aim at creating strategies for operators and end-users alike that will provide the best experience knowing the network or networks that are available and implement these strategies across the device ecosystem but also inside the *cloud* and in particular the *mobile cloud* and the Internet of Things and Content.

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Muriel Médard is a Professor in the Electrical Engineering and Computer Science at MIT. Professor Médard received B.S. degrees in EECS and in Mathematics in 1989, a B.S. degree in Humanities in 1990, a M.S. degree in EE 1991, and a Sc D. degree in EE in 1995, all from the MIT. Her research interests are in the areas of network coding and reliable communications, particularly for optical and wireless networks. She was named a 2007 Gilbreth Lecturer by the National Academy of Engineering. She is a Fellow of the IEEE and member of the Board of Governors of the IEEE Information Theory Society.

MinJi Kim will receive her Ph.D. degree in Electrical Engineering and Computer Science at the Massachusetts Institute of Technology (MIT), Cambridge, USA in 2012. Her research interests include wireless communications and networks, security and reliability of networks, algorithms and optimization on networks, and network coding.

Ali ParandehGheibi will receive his Ph.D. degree in Electrical Engineering and Computer Science at the Massachusetts Institute of Technology (MIT), Cambridge, USA. His research interests include network optimization and control, network coding, video streaming, Peer-to-Peer networks, distributed optimization, gossip algorithms and social networks.

Weifei Zeng is a PhD student at the Massachusetts Institute of Technology. His current research interests include coding design and optimization over various networks. He held a Presidential Fellowhsip at MIT in 2009-10.

Marie-José Montpetit is a research scientist in the Research Laboratory of Electronics at MIT focusing on network coding for video transmission. Her research interests include converged video applications, social and multi-screen media dissemination and wireless networks. She was a recipient of the MIT Technology Review TR10 in 2010. Dr. Montpetit is a Senior Member of the IEEE.

Quality of Experience Oriented Video Streaming in Challenged Wireless Networks: Analysis, Protocol Design and Case Study

Tom H. Luan, Sanying Li, Mahdi Asefi, Xuemin (Sherman) Shen (Fellow, IEEE),
Department of Electrical and Computer Engineering, University of Waterloo, Waterloo, ON N2L 3G1, Canada
{hluan,s68li,masefi,xshen}@bbcr.uwaterloo.ca

1. Introduction

The networked video streaming has achieved tremendous success in the past decade. It has already become the killer Internet application. As reported in [1], 183 million U.S. Internet users watched 40.9 billion online videos in one month. Youtube, the most popular video sharing site, features over 40 million videos and attracts around 20 million subscriptions per month. On the other hand, the last decade has witnessed the equally exciting evolution and explosive adoption of various mobile portable devices, such as smartphones, tablet PCs and laptops. This makes efficient wireless video streaming to the heterogeneous and mobile devices ever more important and demanding.

With the limited network connectivity and high mobility, mobile devises are often connected through the challenged wireless networks, such as Delay Tolerant Networks (DTNs), Mobile Ad-hoc Networks (MANETs) and cognitive radio networks. These networks are characterized by the *network heterogeneity*, *frequent network partition*, and *dramatically changing networking conditions*. Video streaming in such environments inevitably suffers from the intensively changing throughput, long packet delay and severe packet losses, making traditional video systems and protocols operate poorly. In this paper we focus on the design of Quality of Experience (QoE) oriented video streaming system over the challenged wireless networks. To this end, we first develop an analytical framework to characterize the QoE of users, represented by the network performance metrics. Based on the developed model, we introduce a cross-layer design framework to build the QoE-oriented video streaming system over the challenged wireless networks. Lastly, we showcase the implementation of the proposed cross-layer framework in the vehicular networks and cognitive radio networks.

2. QoE-oriented Video Streaming

We consider a typical packetized video streaming as shown in Fig. 1. The encoded video clips are cached at media servers and streamed to remote mobile users using the UDP/IP protocol stack through the challenged wireless networks, such as DTN and MANET. Churned by the dynamic and uncertain network connectivity and performance, video playback tends to suffer from frequent playback interruptions and annoying delays once the playback halts. We thus model the QoE of users from the perspective of video playback smoothness and experience. In specific, we evaluate the QoE of users through two metrics: (1) start-up delay, i.e., delay when user subscribes to watch the video until the video playback starts, and (2) probability of playback frozen, i.e., the probability that playback halts during the video playout.

We evaluate the two QoE metrics by analyzing the playout buffer at the receivers. Specifically, to combat the network dynamics, a typical way is by deploying a playout buffer at the receiver, as shown in Fig.1. To eliminate the effects of variable arrival delays (or delay jitters), the playout buffer postpones the start of video playback by a short period (start-up delay), and buffers the downloaded video packets in a local cache until a certain threshold is reached.

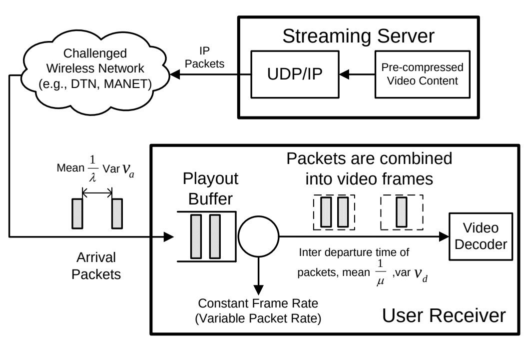

Figure1 Playout Buffer at the Receiver

Therefore, as long as the playout buffer is kept nonempty during the video presentation, the playback can always sustain. In other words, provided the network download performance, the QoE of end users is closely related to the evolution of the playout buffer over time.

Let λ and v_a denote the mean value and variance of the packet download rate, respectively. Let μ and v_d denote the mean value and variance of the video playback rate, respectively. Let *D* denote the start-up delay, which accounts the duration starting when the playback halts until the playback restarts when the number of the buffered packets reaches a threshold *b*. Let *P* denote the probability of playback frozen once the playback initiates. We model the playback buffer as a G/G/1 queue. As described in [1], assuming that $\lambda \le \mu$ and the playout buffer size is infinite, the start-up delay can be represented by the cumulative density function as

$$\Pr(D \le t) = \Phi\left(\frac{x-\lambda t}{\sqrt{\lambda^3 v_a t}}\right) - \exp\left(\frac{2b}{\lambda^2 v_a}\right)\Phi\left(\frac{x-2b-\lambda t}{\sqrt{\lambda^3 v_a t}}\right), \quad (1)$$

where $\Phi(\cdot)$ is the standard normal distribution. The probability of playback frozen *P* is

$$P = \exp\left(-\frac{2b}{\lambda^3 v_a + \mu^3 v_d}(\lambda - \mu)\right). \quad (2)$$

From (1) and (2), decreasing the network variations v_a would monotonically reduce the average start-up delay and probability of playback frozen. Increasing the playback threshold *b* will reduce the probability of playback frozen but enlarge the average start-up delay.

Based on relationship between the network performance metrics (mean value and variations of download rate) and the QoE metrics, as characterized by (1) and (2), the goal of the QoE-oriented video streaming is to optimize the network formation to attain the best user perceived video quality,

$$\max \sum_i U_i(D_i, P_i) \quad (3)$$
$$\text{Subject to } (\lambda_i, v_i) \sim \Omega$$

In (3), the utility $U_i(\cdot)$ represents the satisfaction of user *i*; it is a decreasing function of the start-up delay D_i and probability of playback frozen P_i of user *i*. The objective of (3) is thus to maximize the integrated user satisfaction in the system. The decision variables in (3) are λ_i and v_i, i.e., the mean value and variance of the download rate of each user. Ω denotes the feasible network solutions to enable user *i* to download at (λ_i, v_i). In practice, Ω is embodied by the constraint of networks, like the physical constraints of transmission rate, flow conservation and network resource allocation, etc. In what follows, we show the implementation of (3) in vehicular networks and cognitive radio networks, respectively.

3. QoE-oriented Video Streaming in Vehicular Networks

The newly emerged vehicular networks [3, 4] enable vehicles on the road to communicate among each other in proximity, namely Vehicle-to-Vehicle (V2V) communication, and to access the Internet through roadside infrastructure, namely Vehicle-to-Infrastructure (V2I) communication. Due to the limited coverage of infrastructure, Internet video streaming to the highly mobile vehicles typically involves both the V2I communication and the multi-hop V2V relays from the gateway to the destination vehicles. The intermittent connectivity of the video streaming path paired by the severe interference among vehicles make the smooth video streaming a very challenging task.

In [5], we have developed a QoE-oriented video stream routing protocol steaming from the cross-layer design framework in (3). Given the video playback rate (μ and v_d) and playback threshold *b*, we design the optimal packet retransmissions to attain the best QoE. In specific, due to the volatile wireless channel, coupled with the interference and contentions among vehicles in proximity, packet delivery to vehicles may suffer from severe packet losses. The retransmissions are used to correct the errors. This, however, prolongs the packet delivery and may lead to the underflow of playout buffer, which results in the frozen of video playback. In [5], we model the impact of packet retransmissions on the video download rate (λ_i, v_i) to each user and the resultant QoE (*D*, *P*) of users according to (1) and (2). The optimal retransmissions are designed based on (3) with the constraints subject to the tolerable QoE of users, i.e., upper bounded start-up delay and probability of frozen.

4. Smooth Video Delivery in Cognitive Radio Networks

The Cognitive Radio (CR) networks allow a group of CR users to dynamically access the idle spectrums when spectrums are not used by the licensed users; the CR users are dictated to vacate the channels instantaneously once the licensed users are online [6]. By doing so, the wasted spectrum can be recycled to improve the spectrum utilization. However, as CR users need to keep switching channels to avoid the possible interference to the licensed users, paired with mutual contention among CR users, the download of CR users tend to be turbulent and unstable, which poses significant challenges to the high-quality video streaming in CR networks [7].

To provide smooth video delivery to CR users in the dynamic system, we propose an adaptive channel spectrum allocation scheme in [8] based on the cross-layer framework (3). In specific, we consider two groups of users coexisting in the system, video users and best effort users. The former downloads the inelastic video traffic from the network, and the latter downloads elastic data traffic. Therefore, the two groups of users have distinct QoS requirements. The video users demand relatively static download rate to support the smooth video playback characterized by QoE; whereas the best effort users require lower bounded download rate to enable on-top applications. Based on the instantaneous channel status and different QoS requirements of users, we adaptively allocate the channel spectrums to users, which affects their download rates. Therefore, video users are rendered with different QoE which can be evaluated by (1) and (2). By feeding this to (3), the spectrum allocation is configured to maximize the integrated utility of all CR users. Fig. 2 plots the resultant video frozen probability of video users when the proposed algorithm is applied, compared with the random and greedy channel allocations. As we can see, the proposed scheme can achieve much lower video frozen probability compared to traditional heuristics.

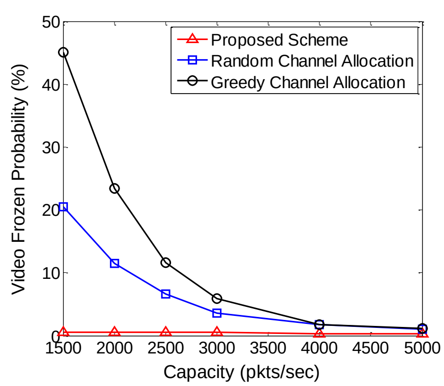

Figure 2 Video frozen probability with different channel allocation schemes in CR network

5. Conclusion

With the enhanced communication capability and fast mobility of wireless devices, the intermittent wireless connectivity yet high-rate during connection will occur more frequently in the near future. To enable the smooth video delivery over such challenged wireless networks needs to address the network dynamics. In this paper, we have provided a QoE-oriented video streaming framework as an effort on this issue. We first develop an analytical framework to characterize the QoE of users, represented by the network performance metrics. We then formulate the video streaming as a cross-layer design problem. Using the video streaming in vehicular networks and cognitive radio networks as examples, respectively, we have shown how the proposed framework can be used in the real-world design. In the future, we intend to test the proposed algorithms in the real-world environment.

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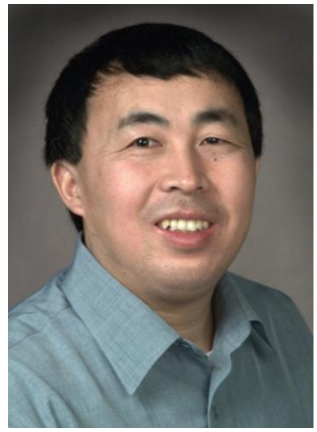
Tom H. Luan received the B.E. degree from Xi'an Jiaotong University, Xi'an, China, in 2004 and the M.Phil. degree in electronic engineering from the Hong Kong University of Science and Technology, Kowloon, Hong Kong, in 2007. He is now pursuing the Ph.D. degree at the University of Waterloo, Waterloo, ON, Canada. His current research interests focus on wired and wireless multimedia streaming, QoS routing in multihop wireless networks, peer-to-peer streaming, and vehicular network design.

Sanying Li received the B.E. degree from Beijing University of Posts and Telecommunications, Beijing, China, in 2008 and the Master degree in University of Waterloo, ON, Canada, in 2010. She is now associated with Sandvine, Inc. Her research interests include video streaming, peer-to-peer networking and cognitive radio networks.

Mahdi Asefi received BSc. in Electrical Engineering from Sharif University of Technology, Iran, in 2004, and MSc. And Ph.D. in Electrical and Computer Engineering from the University of Waterloo, ON, Canada, in 2006 and 2011, respectively. He is currently working in Met-Scan Canada LTD, Toronto, Canada. His research interests include cross-layer optimization protocols for video communications over mesh overlays and mobile ad-hoc networks, specially for vehicular communication systems.



Xuemin (Sherman) Shen [M'97, SM'02, F'09] received a B.Sc. (1982) degree from Dalian Maritime University, China, and M.Sc. (1987) and Ph.D. degrees (1990) from Rutgers University, New Jersey, all in electrical engineering. He is a professor and University Research Chair, Department of Electrical and Computer Engineering, University of Waterloo. His research focuses on mobility and resource management, UWB wireless networks, wireless network security, and vehicular ad hoc and sensor networks. He served as an Area Editor for IEEE Transactions on Wireless Communications and Editor-in-Chief for Peer-to-Peer Networks and Applications. He is a Fellow of Engineering Institute of Canada, a registered Professional Engineer of Ontario, Canada, and a Distinguished Lecturer of the IEEE Communications Society and Vehicular Technology Society.

Online QoE Computation for Efficient Video Delivery over Cellular Networks

Telemaco Melia, Alcatel-Lucent laboratories, Villarceaux, France
Daniele Munaretto, Leonardo Badia, Michele Zorzi, Department of Information Engineering, University of Padova, Padova, Italy; Consorzio Ferrara Ricerche (CFR), Ferrara, Italy
telemaco.melia@alcaltel-lucent.com, {munaretto, badia, zorzi}@dei.unipd.it

1. Introduction

The rise of mobile video streaming has attained unpredictable scores in the past few years. The availability of smartphones combined with broadband wireless access technologies provides to mobile users great always-on experience making content consumption as easy as sending an SMS. When a user either points her browser to a video sharing website, such as YouTube, or opens an HTTP adaptive streaming session, she just enjoys the video stream while in the background optimized procedures select the optimal location (cache, storage) from where to download the content.

Commonly deployed solutions such as GeoDNS allow a requesting client to resolve a specific fully qualified domain name and to be redirected to a suitable cache depending on its geographical location. While these approaches work to a great extent over the public Internet, they suffer from shortcomings when deployed in a mobile environment, since the mobile network infrastructure is exploited as a pure Internet pipe. While this is obviously an advantage for over-the-top players such as YouTube, Google, and Facebook, it becomes a great challenge for mobile service providers. The disconnect between Average Revenue Per Unit (ARPU) and cost per bit requires the development of new solutions to solve network capacity issues while addressing new business models. Mobile service providers need in fact to enter the video distribution value chain to better satisfy their customers and potentially generate new sources of revenue.

The integration of Content Delivery Network (CDN) technology in the mobile service provider network is a promising path. Making online content directly available from the operator network opens the door to a whole new range of optimizations taking into account traditional network metrics [1] as well as wireless specific metrics, since the last radio hop represents the real bottleneck in many broadband networks. The optimization of these parameters is not trivial. The task of defining a combination through a proper weighting and fine tuning may be formidably complex, thus limiting a viable optimization only to locally efficient operating points, solutions only valid for a given time interval, and heuristic approaches. Thus, the evaluation and optimization of the perceived Quality of Experience (QoE) is far from being a solved problem. In the literature, offline solutions exist but they do not take into account the dynamic nature of wireless channels. In our work we demonstrate the efficiency of online tools with respect to offline tools and we highlight the difficult points that need to be addressed.

2. Offline vs. online QoE computation

QoE is typically measured offline in subjective experiments, such as the Mean Opinion Score (MOS), which is the average over all viewers' rating for a given video. Subjective experiments have the drawback of requiring a large set of viewers in a reasonably short amount of time. Hence, this is not appealing for real-time monitoring or streaming applications.

In the last decade objective and subjective quality metrics have been designed to characterize and predict the viewer MOS. Starting from the offline objective metrics, the Mean Squared Error (MSE) and Peak Signal-to-Noise Ratio (PSNR) are objective measurements of the video quality. They are used in Quality of Service (QoS) based video assessment systems. The main advantage of these metrics is that they are computationally simple and straightforward, while the drawback is that comparing PSNR values with the perceived video quality at the user side, significant discrepancies may appear due to their distortion and content-agnostic nature.

Better visual quality metrics have been designed to take into account the impact of distortions and of the video content on the QoE. The Structural SIMilarity (SSIM) [2] index, for instance, and its adaptation for video, VSSIM [3], compute a distortion map from measurements performed on small patches of a video frame. Offline full-reference metrics, computationally light and simple to implement, such as MSE and PSNR, perform a comparison frame-by-frame between the source and the received videos.

Opposite to the full-reference metrics, the no-reference metrics analyze only the received videos, thus they make it possible to design online methods for the measurement of the QoE. As a direct consequence of their nature, the no-reference metrics are more flexible than the full-reference metrics, but they work based on theoretic assumptions about video content and distortions. The no-reference metrics are thus appealing for real-time services, while the full-reference metrics are more suitable for offline applications, but they require more computational efforts.

Finally, we remark that most approaches for evaluating video quality are not suitable for wireless access networks, since they mostly focus on instantaneous evaluations of stationary quality values, without taking any memory effect into account. In reality, not only is the radio channel highly variable over time, but also video traffic is particularly sensitive to error correlation, which is fairly frequent in wireless scenarios. As a result, error propagation phenomena may occur, in which case the perceived video QoE rapidly drops [4]. An analysis of how to counteract it by properly designing error-control methodologies is far from being trivial [5]. In this paper, we simplify this problem by assuming the availability of sufficiently frequent updates on the perceived QoE. Our research focuses on the design of a method for online QoE estimation based on a no-reference approach. We aim at taking into account the features of the radio access channels, with particular emphasis on their time-varying and unreliable character.

Our key idea is that in a mobile network a video application can be served through different paths, resulting in different QoE, measured at the end user, and fed back to the network, possibly with tight granularity so as to track user mobility and channel variations. Thus, the mobile operator is in charge of guiding the service to keep a target QoE and to optimize the network resource usage.

3. A novel cross-layer approach at the terminal side

In this work we propose a new concept of QoE online computation where we take into account metrics impacting the quality perceived by the user from the Core Network (CN), i.e., involving CDN-related metrics, and from the Radio Access Network (RAN), i.e., related to wireless channel metrics. The combination of these two sets of metrics opens the door to a promising research avenue taking into account issues related to the whole video delivery chain. Moreover, we intend to develop, based on these metrics, an online no-reference QoE computation method suitable for mobile devices. The key idea is to combine metrics that specify a typical CDN environment [1] with wireless metrics. The mobile operator has the unique possibility of combining both worlds and of proposing fast adaptive algorithms to optimize the perceived QoE. Figure 1 depicts a simplified vision of the Evolved Packet Core (EPC) enhanced for optimal CDN integration [4].

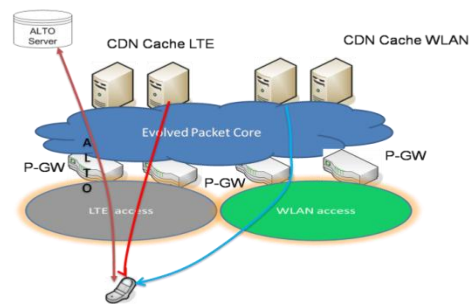


Figure 1. Mobile Network Architecture.

CDN caches are integrated on top of the existing Packet Data Network (PDN) gateways (P-GWs, giving Internet access to mobile devices) and take into account the diversity of both cellular and Wireless Local Area Network (WLAN) access. Latest extensions of the 3GPP specifications propose the deployment of several P-GWs with both local and global scope. This way, the same content can be potentially downloaded from several sources, each having different properties with respect to the underlying wireless technology and round trip time delay. To this end, the Application Layer Transport Optimization framework (ALTO) [5] proposes extensions accounting for the specific metrics of a mobile wireless deployment. It should be further noted that at the time of this writing the standardization process has just begun, leaving to researchers the great opportunity to impact the next release of the specifications.

The foreseen metrics in our approach are as follows. CDN-related metrics consider the distance of a specific End Point (EP), i.e., CDN cache, expressed in number of hops between the mobile device and the EP, and the computational load of an EP, expressed in terms of memory occupancy. Wireless metrics consider the Signal-to-Noise

Ratio (SNR) and the channel rate of both cellular and WLAN access, combined to allow sequential or simultaneous use of both technologies (i.e., accounting for multi-homed mobile devices).

Regarding the wireless side of the online QoE computation approach, we give an example of scenario to best highlight our innovative step. Consider a mobile user walking in an Long Term Evolution (LTE) micro-cell, with 20 MHz of bandwidth, and an area coverage of up to 2 km. We further deploy 3 WiFi 802.11n spots, with bandwidth of 20 MHz, at regular intervals of 0.5 km from the LTE base station. By running in Matlab two own-developed LTE and WiFi modules, we obtain values of average SNR with respect to the distance from the LTE base station as plotted in Fig. 2. It is evident that, once the user is in range of a WiFi hot spot (blue, red and black peaks in Fig. 2), the exploitation of this additional access technique, which should happen simultaneously on both LTE and WiFi channels, may be beneficial to both the user and the network operator. For the latter, it will result in additional available capacity to redistribute to the users that are not under coverage of any WiFi hotspot. For the former, video quality can be highly increased. Finally, assuming the delivery of a scalable video, i.e., encoded with the video compression standard H.264-SVC (Scalable Video Coding) [6], we might choose to deliver the base video quality through the LTE channel, which is always available, leaving the enhancement video layers to the WiFi channel, when available.

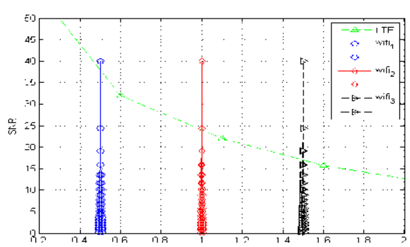


Figure 2. Wireless Scenario: single user' SNR vs. distance.

4. Cost for network operators vs. user's QoE

In the previous section we presented a possible set of metrics that we intend to consider in our online QoE computation approach. Our idea is to build vectors reflecting each possible path of the whole video delivery chain, from a video source (cache) to the mobile user (terminal).

The goal of our work is to find the optimal vector of values for both CDN and wireless related metrics. Optimality here has a different meaning whether we consider the network operator's or the user's point of view. We aim at optimizing the set of vectors based on two different criteria reflecting the requirements of service providers and users. The common set of solutions is the optimal set taking into account both network operator and user's sides. It is likely that the two solution spaces do not coincide, since the cheapest solution for a network operator in terms of resource usage unlikely gives the best QoE to the end user. Enhancing the perceived quality of a video comes at a cost, which usually results in higher bit rates to be provided by the network operator.

We foresee the design of a framework, to be implemented on the mobile terminal, which runs a computationally light optimization algorithm in real-time to best select the path for the video delivery, ensuring a target user's QoE level under the constraints of the available network resources, i.e., network operator's costs.

5. Conclusions

In this work we propose a new challenge for the online computation of the video quality perceived by a mobile user. The design of a heuristic algorithm for the optimization of the network and quality metrics that affect the QoE of a user at the mobile terminal is currently work in progress.

6. Acknowledgement

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Telemaco Melia received his Informatics Engineering degree in 2002 from the Polytechnic of Turin, Italy and his PhD in Mobile Communications from University of Göttingen in April 2007. From June 2002 to December 2007 he worked at NEC Europe Ltd. in Heidelberg, Germany in the Mobile Internet Group. He worked on IPv6 based Mobile Communication focusing on IP mobility support across heterogeneous networks and resource optimization control. In September 2008 he joined Alcatel Lucent Bell Labs. He is currently working on inter-working architectures spanning 3GPP, Broadband forum and IETF standardization bodies. His main research interests include wireless networking and next generation networks. He authored more than 20 publications and he actively contributes to the IETF. He is project co-ordinator of the EU funded MEDIEVAL project (www.ict-medieval.eu).

Daniele Munaretto received the BS and MS degrees (with honors) in Telecommunication Engineering in 2004 and 2007, respectively, at the University of Padova, Italy. After an internship for his MS thesis at DOCOMO Euro-Labs, in Munich, Germany (2006–2007), he joined Euro-Labs from 2007 to 2010 as R&D Engineer in the Ubiquitous Networking Research Group. He is currently a first year Ph.D. student at the Department of Information Engineering at the University of Padova, under the supervision of Prof. M. Zorzi. His research interests mainly cover wireless networks, network coding, video streaming applications and P2P networks. He authored several scientific papers and EU patents.

Leonardo Badia received the Laurea Degree (with honors) in Electrical Engineering and the Ph.D. in Information Engineering from the University of Ferrara, Italy, in 2000 and 2004, respectively. During 2002 and 2003 he was on leave at the Radio System Technology Labs (now Wireless@KTH), Royal Institute of Technology of Stockholm, Sweden. After having been with the University di Ferrara and the IMT Advanced Studies Institute in Lucca, Italy, he joined in 2011 the University of Padova, where he is currently an Assistant Professor. His research interests include protocol design for wireless networks, transmission protocol modeling, optimization of radio communication, and applications of game theory to radio resource management. He authored about 100 scientific papers and serves on the Editorial Board of the Wiley Journal of Wireless Communications and Mobile Computing.

Michele Zorzi received his Laurea degree and Ph.D. in electrical engineering from the University of Padova, Italy, in 1990 and 1994, respectively. During academic year 1992/93, he was on leave at the University of California, San Diego (UCSD) attending graduate courses and doing research on multiple access in mobile radio networks. In 1993, he joined the faculty of the Dipartimento di Elettronica e Informazione, Politecnico di Milano, Italy. After spending three years with the Center for Wireless Communications at UCSD, in 1998 he joined the School of Engineering of the University of Ferrara, Italy, and in 2003 joined the Department of Information Engineering of the University of Padova, Italy, where he is currently a Professor. His present research interests include performance evaluation in mobile communications systems, random access in mobile radio networks, ad hoc and sensor networks, energy constrained communications protocols, cognitive networks, and underwater communications and networking. He was Editor-in-Chief of the IEEE Wireless Communications magazine from 2003 to 2005 and Editor-in-Chief of the IEEE Transactions on Communications from 2008 to 2011, and serves on the Editorial Board of the Wiley Journal of Wireless Communications and Mobile Computing. He was also guest editor for special issues in IEEE Personal Communications and IEEE Journal on Selected Areas in Communications. He served as a Member-at-Large of the Board of Governors of the IEEE Communications Society from 2009 to 2011. He is a Fellow of the IEEE.

Quality of Experience for Multimedia in Clouds

Rafal Stankiewicz, Andrzej Jajszczyk, AGH University of Science and Technology, Krakow, Poland
{rstankie, jajszcz}@agh.edu.pl

1. Introduction

In recent years a rapid development of multimedia services is observed. Due to the growing end-user requirements and availability of more sophisticated technologies, implementation of such services is more and more challenging. Efficient multimedia communications, Quality of Experience (QoE) awareness, combining several technologies, network convergence, and support for mobility are often required. This article briefly introduces the main groups of requirements and then focuses on challenges related to the QoE support for multimedia applications based on the example of cloud-based services.

2. Quality of Experience

Quality of Experience (QoE), defined by ITU-T as "*the overall acceptability of an application or service, as perceived subjectively by the end-user*" [1], is currently the most meaningful criterion used for evaluation and comparison of services. Assurance of a high level of QoE is an important challenge faced by all stakeholders involved in delivering services over current heterogeneous networks [2]. It especially adheres to multimedia services.

A necessary condition for achieving a high level of QoE is assurance of an appropriate level of intrinsic characteristics of resources involved in service delivery classified as Quality of Service (QoS), Grade of Service (GoS), and Quality of Resilience (QoR) [3]. QoS is related to all phenomena occurring while the traffic is transported over the network, GoS describes the process of connection setup, release and maintenance, while QoR encompasses all aspects of network survivability, dependability as well as service availability. QoE strongly depends on those intrinsic network features and performance, although there is no simple mapping between the QoS/GoS/QoR parameters and QoE. These relations are better understood only in the context of particular applications. However, there are ongoing efforts towards finding mathematical relationships between intrinsic network parameters and QoE, usually expressed quantitatively by the Mean Opinion Score (MOS) value [2]. This is in fact a very challenging task since QoE also depends on many orthogonal factors (Fig. 1) such as the end-user terminal's capabilities, the environment in which the service is received, the type of service, pricing policy as well as sociological and psychological aspects. Those factors are very important for QoE assessment of multimedia services.

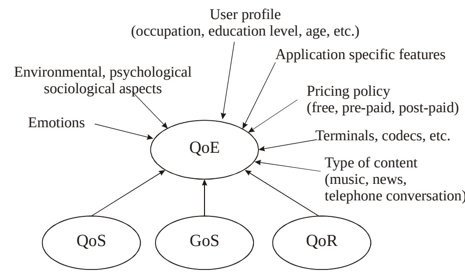

Figure 1. Factors influencing QoE [2].

3. Multimedia services in future converged networks

End-user QoE requirements are related not only to directly perceived service quality such as video quality received at TV at home. From a more general point of view end-users require high quality services regardless of localization and time constraints, that is, they want to receive any service, anytime, anywhere, and on any device [2].

Those requirements generate several challenges to heterogeneous networks involved in service delivery. Intrinsic end-to-end QoS, GoS and QoR should be provided over any media and networking technology and by any operator. More information about those challenges can be found in [2].

4. The world of clouds

Service providers more and more often do not invest in building their own physical infrastructure since it is costly and is subject to lots of constraints, especially if a service is intended to have a global range. Instead, service providers buy resources from other companies.

In recent years a rapid development of the cloud delivery model is observed. Cloud providers have arisen as new important stakeholders involved in delivering a service from its provider to end-user, especially in multimedia communications. In this model, at least three main stakeholders are distinguished: service (content) providers, cloud providers and network providers.

Clouds may offer various types of resources such as processors, storage, applications, etc. In general, three levels of cloud service offerings can be distinguished: Software as a Service (SaaS), Platform as a Service (PaaS) and Infrastructure as a Service (IaaS) [4].

5. QoE for multimedia applications in cloud delivery model

Delivering multimedia services to end-users with a stable and satisfactory QoE level using clouds requires solving several problems.

New solutions are needed to support realization of services using several clouds offered by different providers and/or integrating cloud services of various levels (SaaS, PaaS or IaaS) [5]. Such a scenario raises some new challenges.

Multi-cloud service realization requires *inter-cloud communications*. Consider a service provider that uses several clouds offering various services (e.g., storage, CPU & RAM, and software) and physically connected to different network operators. First of all, mechanisms integrating various types of services supported by distinct clouds are needed. The exchange of data between storage and computational resources is performed by the network and may pass through many intermediate network operators. While setting the connection, QoS/QoE requirements should be taken into account. The connection cost depends on the selected route. Some services require ongoing dynamic allocation of resources. This is due to migration of virtual machines within the cloud from one physical location to another. Changing the location may require a new connection between the clouds. The algorithm for selection of a new physical location of services or resources in the cloud should take into account the following criteria:

- data transfer cost,
- QoS/QoE assurance,
- service (MPLS, L2VPN, L3VPN, VPLS, PBB) availability,
- service setup cost,
- network resource availability, and
- data transfer security.

The migration of a service to the cloud offering the same type of resources but managed by a distinct cloud provider should also be addressed by such an algorithm.

The second challenge is related to the requirement of *global service mobility*. In future converged networks, a service should follow its recipient, that is, it should be available anytime and anywhere. Consider an end-user who roams around the world and uses a popular service based on cloud services. For example, the end-user uses the service on a notebook or tablet and wants to receive a high QoE regardless where he/she connects to the network: at home in Europe, in a train, airport, hotel on other continent, etc. It raises a challenge not only to network solutions and ISPs, but also to the service providers and, finally, to the clouds used by service providers to realize the service. To enable access to the service with the same level of QoE anywhere in the world, the service provider may need to use services of some number of clouds offering their resources in different parts of the global Internet. Additionally, different access network providers are used by end-users at different locations. Therefore, a complex optimization problem arises. One may need to decide which resources in the cloud or which cloud to use to provide the service to the end-user taking into account its location in the network, cost of traffic transfer, cost of access to a given cloud, cost of usage of a given cloud, cost of migration of data or resources, or virtual machines between clouds used to realize the service. The user mobility resulting in changing the access network may also result in the change of Point-of-Presence (PoP). Finally, if the service must be provided via a different PoP, the need to change the cloud used to provide the service may arise.

The third challenge related to multimedia services realized with cloud services is *network access selection*. Assume that some service needs an instantaneous access to the network. The service requires a stable level of QoE. The service is received by the end-user on a mobile terminal and the access network uses radio technology such as UMTS or LTE. The end-user is moving and handovers take place between base stations (cells). It may also be assumed that a mobile operator is changed while the service is being provided. Also a radio access technology may be changed (including LTE, UMTS, WiFi, or WiMAX) when the end-user moves, that is, a vertical handover may be triggered. At the network level a Media Independent Handover (MIH) may be used (IEEE 802.21). In turn, the service provider uses two or more cloud providers to realize the service. The clouds' resources may be connected to a mobile provider backbone network or via a fixed network (other ISP). Since the end-user is moving and quite often changes the access point to the network, it may be necessary to appropriately select cloud resources or even change the cloud to maintain the continuous service provision. The challenge is to optimize selection of the cloud and resources, the network path between cloud resources, and the end-user terminal, taking into account the traffic transfer cost and QoS, available network technologies and radio channel quality. It may also happen that a given handover results in the need to migrate a virtual machine between two clouds used by the service provider to realize the service. Such a migration may be triggered not only by the handover but also by network resource scarcity.

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Rafal Stankiewicz is an Assistant professor at the Department of Telecommunications of AGH University of Science and Technology, Krakow, Poland. He received the M.Sc. and Ph.D. degrees in Telecommunications from AGH University of Science and Technology in 1999 and 2007, respectively. Currently, he teaches courses on data base systems, operating systems as well as P2P and overlay networks. His research interests focus on network quality and reliability, optical transport networks, performance evaluation and analytical modeling, statistical analysis of telecommunications traffic, as well as traffic management concepts for peer-to-peer networking. He is an author of many conference and journal papers and co-author of one book. He was involved in several European research projects including BTI, LION, NOBEL, EuroNGI, EuroFGI, EuroNF, SmoothIT. He also served as a reviewer of papers submitted to journals (e.g., *IEEE Communications Magazine*, *Computer Networks*, *Performance Evaluation*) as well as top scientific conferences (e.g., *IEEE Globecom* and *ICC*). He is a member of IEEE.

Andrzej Jajszczyk is a Professor at AGH University of Science and Technology in Krakow, Poland. He received M.S., Ph.D., and Dr. Hab. degrees from Poznan University of Technology in 1974, 1979 and 1986, respectively. He is the author or co-author of seven books and more than 270 research papers, as well as 19 patents in the areas of high-speed networking, telecommunications switching, network management, and reliability. He has been a consultant to industry, government agencies, and telecommunications operators in Australia, Canada, France, Germany, India, Poland, and the USA. He was editor-in-chief of *IEEE Communications Magazine* and editor of *IEEE Transactions on Communications*. He is a member of advisory or editorial boards of several journals and magazines, including *Annals of Telecommunications*, *China Communications*, and *Security and Communication Networks*. He served as Vice-President and Director of Magazines of IEEE Communications Society. He is a Fellow of the IEEE.

Adaptive Control based on Quality of Experience for Multimedia Communications

Abdelhamid Mellouk, Said Hoceini, Brice Augustin, Nadjib Ait Saadi, Hai Anh Tran, Sami Souihi, University of Paris-Est Creteil Val de Marne (UPEC), France; Dept N&T – IUT Créteil/Vitry & Image, Signal and Intelligent Systems Lab (LiSSi) ; Transport Infrastructure and Network Control for E2E Services (TINCS)
{lastname}@u-pec.fr

1. The notion of Quality of Experience (QoE)

Nowadays, network providers pay an increasing attention to multimedia communication whereby the available server capacity and network bandwidth become overloaded by the evolution of high quality multimedia services (e.g. IPTV, online gaming, social networking, etc.). What is missing is a user-centered approach that is represented by the notion of Quality of Experience (QoE) [1]. This subjective measure relates to how end-users perceive the quality of a network service and includes the complete end-to-end system effects. It is expressed by human feelings like "excellent", "good", "bad", etc. Designing a network system based on QoE is today not only a solution to improve network quality but also the trend for competition between network providers. As a combination of user perception, experience, satisfaction and expectations, QoE is an important metric for the design of systems and engineering processes that can only be measured dynamically at the end of any transmission activity. QoE management takes into account the needs and the desires of the end-users when using network services, while the traditional concept of Quality of Service (QoS) [2] only attempts technical measurements of the delivered service.

Regarding the QoS/QoE relationship, QoE covers the QoS concept, whereas there is a common belief that QoE is a part of QoS. Fig. 1 illustrates this theoretical point. The quality of a network service including core network and access network is determined by *QoS Access* and *QoS Backbone*. The quality perceived by end-users when using end-devices is called

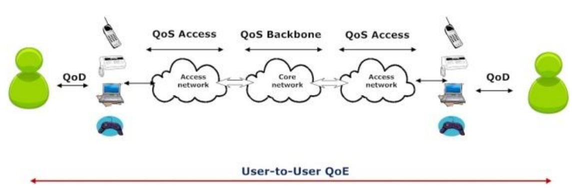

Figure 5: Quality chain of an e2e service.

QoD (Quality of Design). Therefore, QoE is satisfied only if both QoD and QoS are satisfied. So we can see that the end-to-end QoE is a conjunction of QoD and QoS.

2. Classification of QoE methods

There are today various methods for QoE assessment. We propose to classify QoE methods in three different approaches [18]: *usability metric*, *hedonistic concept* and *buzzword extension*. These three approaches can cover the whole QoE notion. Indeed, what makes us interested in considering the user perception concept are the three following issues:

- **Usability metric -** *The ease of use and learnability of a service*: This approach includes measuring and comparing usability metrics. Such metrics is of vital importance for user satisfaction. In fact, it is of decisive importance that users actually are satisfied with the network service or they will simply quit this service. Such metrics are useful for assessing long-term progress on a system. Usability metric is all about how easy a network service is to use. The purpose of usability is to set the design direction of a network system. For this approach, Soldani et al. propose a QoE management method in [3].

- **Buzzword extension -** *An extension of the known QoS concept* [5]: From a historical point of view, the QoS metric came first to designate a set of techniques to ensure the routing of network traffic such as voice or sensitive multimedia applications. Since then, QoS highlights the performance improvement of network systems. QoE appeared much more recently and directly affects end-users. As a comprehensive approach to quality (measured end-to-end), the QoE is considered as an extension of QoS. For this approach, S.Winkler et al. propose a subjective testbed for evaluating video quality of streaming applications [4].

- **Hedonistic concept -** *The pleasure and satisfaction of end-users when using a service*: QoE represents the overall level of end-user satisfaction with a service and expresses user satisfaction both objectively and subjectively. While not always numerically quantifiable, QoE is the most important single factor to assess the user experience. In other words, QoE is considered as a hedonistic concept. Some methods for this approach are described in [6, 7].

3. QoE in a Cloud CDN environment

Cloud Service [8] has become today a key IT technology. It allows us to access data, programs and other multimedia services from a Web browser via the Internet hosted by network service providers. This technology allows end-users to use applications without installation and access their personal files at any computer with internet access. However, the requirement of high bandwidth connection is an issue in Cloud Service deployment. Recently, we proposed an approach to integrate cloud services into a QoE-based Content Distribution Network.

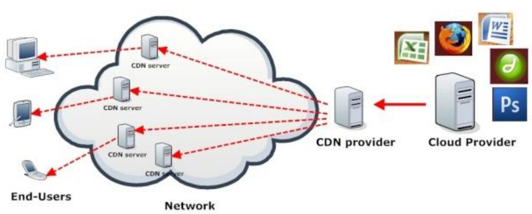

Figure 6: Cloud Content Distribution Network

Content Distribution Network (CDN) [9] are deployed to improve network quality and to optimize resource utilization. The main idea of CDN operation is to move the content from the origin server to servers that are close to end-users, namely *replica servers.* In our model (Fig. 2), the Cloud provider gives the cloud data and services to a CDN provider. The latter serves end-users using distributed replica servers in the edge network.

We focus on two layers of our CDN model: a *routing layer* and a *meta-routing layer*. In the routing layer, we propose a routing protocol, named QQAR [10] (QoE Q-Learning-based Adaptive Routing protocol), which is based on a Q-learning algorithm [11]. In the meta-routing layer, we propose a server selection method that we formalize as a Multi-armed bandit problem [12]. Both of our approaches are based on QoE-feedbacks from end-users.

4. Knowledge dissemination for QoE

In order to make appropriate decisions, the network needs to integrate the QoE parameter on each node. QoE is one of the components of the knowledge plane. This plane is responsible for knowledge management and the strategies selection. It must be created autonomously, continuously and dynamically. However, the knowledge dissemination in the network remains an open problem [13]. While a greedy approach suggests distributing knowledge over all network elements, this idea is impractical in reality.

An improvement of this assumption is using a diffusion region model [13]. This kind of diffusion model solves the overhead problem but knowledge is still partial in some cases. Thus, in [14] we propose a new knowledge dissemination mechanism based on selection of some nodes called “master nodes” in charge of knowledge management. This problem is a multi-constraint optimization problem. Looking for a solution to this problem, we use the concept of Pareto dominance by performing a comparison between different possible solutions with multiple criteria. Moreover, our proposition uses dedicated overlay networks, which consist of dividing the knowledge plane into many sub-planes, each of them representing a view of the knowledge plane corresponding to a given service (or application).
Finally, we support the idea of a backbone implementing intelligent agents over routers, such as programmable overlay routers [16].

5. Current development for QoE Testbeds

While a plethora of tools have been proposed for automated quality assessment of video and audio, there is a lack of public datasets to evaluate and validate them on a common ground. In our research, we propose two complementary approaches to build reliable QoE assessment datasets. The first approach is a testbed, developed in the framework of European Celtic IPNQSIS Project [17], in which each parameter affecting QoE (at the network level, but also at the video and user levels) can be precisely tweaked. The platform consists of a video server whose flows are disturbed by a customizable network emulator. Users are asked to rate a series of short videos on various devices (laptop, smartphone, TV, etc).

The second approach is a crowd-sourced experiment enabling the collection of realistic QoE data at a large-scale. It consists in a plugin for the Firefox Web browser that detects the presence of a video in a page, and automatically inserts a sober user interface enabling the user to rate the video. At the same time, network-level metrics are recorded, as well as other useful information on the device and user. These datasets will help us in devising an accurate and generic model for automated correlation of QoS and QoE.

6. Conclusion

The future Internet will be user-centric and a ferocious competition between providers will be based on Quality of Experience. Consequently, any new architecture and protocol must take into consideration the QoE of clients.

In this paper, we defined the concept of Quality of Experience, which quantifies the satisfaction of end-users. We then highlighted the main differences with QoS and proposed a classification of QoE methods. Afterward, we presented our user-centric research works that aims at optimizing end-user QoE by tackling the problems of routing, meta-routing and knowledge dissemination.

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Abdelhamid Mellouk is full Professor in Networks and Telecommunication (N&T) Department of IUT Créteil/Vitry, Paris-Est University (UPEC), France, His general area of research is in adaptive command/control for Qoe and QoS network based on bio-inspired artificial intelligence approaches.

Said Hoceini received the Ph.D. Degree in computer Networks from UPEC in 2004. His research focuses on Routing Algorithms, Quality of Service (QoS), Quality of Experience (QoE), and wireless sensor networks.

Brice Augustin received the Ph.D. degree in computer science from UPMC Sorbonne Universités, Paris, France. His research focuses on internet topology and traffic measurements, Quality of Experience (QoE), and wireless sensor networks.

Nadjib Ait Saadi is Associate Professor of computer science at University of Paris-Est Créteil Val de Marne - IUT Créteil-Vitry. He obtained the PhD in computer sciences with honors from LIP6 - Pierre & Marie Curie university (Paris 6) in France.

Hai Anh Tran is a doctoral student at UPEC. He obtained in 2009 a Master of Research in Computer Systems, at University Paris-Sud 11, Orsay, after an Engineer Diploma in Computer Science, in 2008 at Hanoi University of Technology, Vietnam. His research focuses on adaptive control in dynamic complex large scale networks used in content distribution.

Sami Souihi is a doctoral student at UPEC. He obtained a Master of Research at UPMC Sorbonne university in 2010. His research interests are focused on autonomic networking at knowledge management.

Inferring and Improving Internet Video QoE

Mukundan Venkataraman and Mainak Chatterjee
Department of EECS, Univerity of Central Florida, Orlando, FL 32826, USA
mukundan, mainak @eecs.ucf.edu

1. Introduction

Today, consumer generated video content alone accounts for one third of all consumer Internet traffic. The sum total of all Internet video traffic, including Internet Television (IPTV), Video on Demand (VoD) and Peer-to-Peer (P2P) sharing, will amount to 91% of all consumer traffic by 2014 [1]. Multimedia content is poised to *dominate* all Internet traffic in the coming decade. More people today opt for video-conferencing, live Internet television and video-on-demand services than they did a decade ago. Internet has evolved from being a platform for hosting web pages to be a playground for multimedia content.

2. Multimedia technology to enable low-cost capturing and delivery

As customers spend more and more time watching videos online, they are increasingly becoming unsatisfied by low bitrate videos and are embracing High-Definition (HD) streaming services. Providing high quality video streaming services over a best-effort and shared infrastructure such as the Internet, however, is non-trivial. It is increasingly observed that existing Internet Quality of Service (QoS) is insufficient at ensuring consistent consumer experience. As Internet-based multimedia competes with traditional cable-based streaming, an ever increasing load is placed on network elements on the Internet to deliver streaming content with high *perceptual* quality; including Content Delivery Networks (CDNs), overlay networks, VoD and IPTV infrastructures. To succeed, network service providers need to infer, predict and improve Interner perceptual video quality.

Service providers are hence trying to characterize a video stream in terms of Quality of Experience (QoE) [2] rather than QoS. Internet QoS has long attempted to assure statistical service guarantees for parameters like bandwidth, delay, loss and jitter [3]. However, QoS lacks an important element in characterizing video streams: that of human perception. For a given loss rate, the perceptual degradation caused by a network outage can vary dramatically depending on the type of frame impacted, the motion complexity inherent in the clip, and the encoding bitrate of the clip, to name a few. For example, a 1% loss on an MPEG-2 transport stream can either result in a minor glitch that is barely noticeable, or can severely degrade playout for an entire second depending on the aforesaid factors. Inferring perceptual quality of a video stream continues to be an open problem.

Existing perceptual quality evaluation frameworks are often complex, computationally intensive, or require specialized information. Perceptual quality is often expressed in terms of a Mean Opinion Score (MOS). A common way of inferring MOS is by comparing video frames before and after network transmission to check for degradations. MOS calculations are often hard (if not impossible) to perform inside the network core. This is because: (i) it is cumbersome to deploy computationally intensive software at arbitrary nodes/routers, and (ii) QoE evaluation is often infeasible at arbitrary routers/nodes because the original frames are unavailable for reference.

Inferring QoE apart, understanding present day Internet QoE and improving it are even harder. Internet architectures and protocols are highly optimized for elastic content like http, ftp and e-mail. For these applications, time-to-deliver is less important than message integrity. Given the diversity and size of the Internet, degradations at the link level and end-to-end path level that effect video QoE are not well understood. Further, there are few architectures and protocols that can efficiently infer and improve video QoE on the Internet.

3. Quality of Experience: A Primer

Quality of Experience describes how well a service performs in meeting user expectations. It is a rating of performance from the users' perspective. For Internet based streaming to compete with existing cable based infrastructure, QoE delivered by streaming services has to match or outperform QoE from cable based streaming. This section looks at what QoE is, why it is important, and the areas of QoE that we address in this article.

3.1 Why QoE?

Traditionally, Internet QoS was aimed at enabling streaming services. The Internet Engineering Task Force (IETF) standardized IntServ and DiffServ router mechanisms to improve quality of streaming content, which required changes to every router in the Internet. Given the scale of the Internet, as well as the diversity of various Autonomous Systems (AS) that comprise it, these changes could not be completely co-ordinated. As a result, even after years of slow adoption there have been no significant performance enhancements in terms of video quality delivery, as the Internet continues to operate on a "best-effort" delivery model. QoS mechanisms operate with a notion of providing service guarantees to enhance application performance. However, service guarantees alone are not sufficient to raise perceptual quality. QoS based quality assessments have often found to be grossly inaccurate at predicting user experience, and as such are not applicable in evaluating video quality [4, 5].

To understand why QoS guarantees do not promise perceptual quality and why objective quality evaluations often misrepresent quality, we consider two example scenarios. Consider a snapshot of a playout shown in Fig. 1 and Fig. 2. Both clips were subject to the same loss rate (QoS), however, the perceptual quality of both clips is very different. This is because the perceptual degradation depends not only on the loss rate, but the *type* of frame impacted as well. As a result, statistical service guarantees (like QoS) are insufficient in assuring perceptual quality. Fig.1 and Fig. 2 also show why objective functions misrepresent quality by representing two instances of a video sequence that suffer a similar network loss rate of 10%. However, clip-1 experiences losses in key frames whereas clip-2 suffers losses in non-key frames. As per QoS, both clips fare the same. It is not hard to see the huge difference in perceptual quality though.

Clearly, there is a strong need to diverge from objective QoS based quality evaluation approaches towards QoE based quality evaluations. Monitoring and improving QoE seems to be the only way by which service providers can prevent churn and raise revenue. Service providers apart, QoE as a concept has been the driving factor for evaluating customer satisfaction in a wide variety of domains: from retails, airlines, food-services to customer support.

Over the past few decades, QoE has been the most significant measure of human satisfaction in these domains. Understanding and improving QoE has had a great impact on the long term success of vendors in all of these domains.

Figure 1: A video clip that experiences a loss in a key frame, with a net 10% loss rate (QoS)

Figure 2: The same video clip as Figure 1 with a 10% loss rate, with key frames intact

4. Long Terms Goals of this project

The current article makes contributions to the following pertinent problems: (i) design a lightweight, no-reference tool called MintMOS [5] which can infer the QoE of thousands of video streams in transit at arbitrary Internet nodes, (ii) provide the first empirical characterization of Internet *link-level* degradations and their impact on video-QoE, (iii) provide the first large-scale characterization of *end-to-end* Internet paths in assuring video-QoE, (iv) investigate one-hop Internet redirections in large, unstructured overlays that can scale to support millions of users, and (iv) propose a simple, scalable, and efficient path selection strategy called Source Initiated Frame

Restoration (SIFR) that applications can use to improve Internet video QoE.

4.1 Inferring Video QoE

Inferring the perceptual quality of a video stream in transit at arbitrary nodes/routers in the Internet, where the original frame is not available for reference, continues to be an open problem. We present MintMOS: a loadable kernel module that is an accurate, lightweight, no-reference framework for capturing and offering suggestions to improve QoE inside the network core [6]. MintMOS can accommodate an arbitrary number of parameters to base quality inference decisions, and encompasses both network dependent and independent parameters. MintMOS internally consists of an Inference Engine (IE) to infer QoE, a Suggestions Engine (SE) to offer hints to improve QoE, a network sniffer to snoop traffic, and a QoE space. A QoE space is a known characterization of perceptual quality for various parameters that affect it. For any *k*-parameters that affect video quality, we begin by creating a *k*-dimensional QoE space, where each axis represents a parameter on which QoE is dependent. Hence, each point in the space is characterized by a *k*-tuple vector. For a given set of *k* parameters, we could get *N* "reference points" for MOS in the QoE space. To do this, we construct *N* versions of a given video by transporting the original video over a controlled environment i.e., for known values of the *k* parameters. The *N* video samples thus created are shown to a diverse population of human subjects who assign a MOS to each of the samples. Given the *N* reference points in the *k*-dimensional QoE space, we can infer the MOS for a new set of parameters by calculating the least distortion between the new values and the reference points in QoE space. MintMOS's modular organization allows every component to evolve independently.

We instrument an actual QoE space with 54 partitions using four parameters: loss, encoding bitrate, motion complexity, and type of frame impacted. We generated 54 video samples using one low motion and one high motion clip, and requested 77 human subjects in a lab environment and 143 users online to assign a perceptual quality score to the samples. Their feedback was used to create our QoE space. We deployed MintMOS with this QoE space on a 22-node wide-area measurement overlay on PlanetLab. We streamed IP-traces of various clips from every node to every other node for one week and used MintMOS to predict quality and detect outages.

Figure 3: Degradations by taking the default IP path

Figure 4: SIFR prevents many perceptual degradations that the default-IP cannot

4.2 Improving Internet Video QoE

We investigate ways of improving Internet video-QoE using one-hop redirections. Perceptual quality can be raised by path selection strategies which preserve application specific policies. To make our results more generally applicable, we seek path selection strategies that *do not* require background monitoring of alternative routes or any apriori path quality information. We analyze a large number of Internet path measurements from five different overlays built using PlanetLab. Our datasets include weeklong measurements taken from overlays of: (i) 21 nodes in United States, (ii) 19 nodes in Europe, (iii) 22 nodes in Asia, and (iv) two different overlays (16 and 32 nodes each) spread across the globe. Using these datasets, we compare the performance of the ``default" Internet path and other alternate paths derived by synthetically combining path metrics of disjoints nodes. Similar in spirit to randomized load allocation [6], we show that attempting to route key frames following a degradation using a random subset of 5 nodes is sufficient to recover from up to 90% of failures. We argue that our results are robust across datasets.

We further analyze the efficiency of a randomized path selection in large, unstructured overlays that can scale to service millions of users. Using weeklong measurements from 500+ vantage points in the Internet, we show that it is sufficient to reroute using k *randomly chosen* intermediate nodes for an overlay with N nodes. We show that the value of k is bounded by $O(\ln N)$; which implies that k equals 8 for an overlay with 1000 nodes, and k is just 14 for an overlay with one-million nodes. We also observed that for random-k to be effective, the subset k should be *uniformly representative* of the N nodes in the participating overlay.

Finally, we design and implement a prototype forwarding module in PlanetLab called SIFR. We evaluate the effectiveness of SIFR in improving video-QoE against the default IP-path. We show that we can minimize and recover quickly from perceptual degradations, thereby raising perceptual quality on top of the best effort Internet. SIFR requires no modification to the Internet core, and can be seamlessly integrated into any source-destination pair that wishes to exchange multimedia content.

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Mukundan Venkataraman obtained his Ph.D. from the Department of Electrical Engineering and Computer Science (EECS) at the University of Central Florida (UCF) in Aug 2009. He received a B.E. in Computer Science from VTU in June 2003. He is currently a post-doctoral researcher at the Multimedia and Security Labs, Stevens Institute of Technology, Hoboken, NJ.

Mainak Chatterjee is an Associate Professor in the School of Electrical Engineering and Computer Science at the University of Central Florida. He received his Ph.D. from the Department of Computer Science and Engineering at the University of Texas at Arlington in May 2002. He received his M.E. degree in Electrical Communication Engineering from the Indian Institute of Science in 1998. Prior to that, he did his B.Sc. in Physics from Presidency College, University of Calcutta. His research interest is in the broad area of computer communications and wireless networking and is associated with the Networking and Mobile Computing Lab (NetMoC) at UCF. He is a recipient of the AFOSR Young Investigator Program (YIP) award.

CALL FOR PAPERS

Wiley Transactions on Emerging Telecommunications Technologies (ETT) Special Issue on Quality of Experience in Wireless Multimedia Systems

Aim and Scope

One important objective of new broadband wireless communication networks is to enable people to enjoy multimedia services anywhere at any time. During the recent decade, a variety of techniques have been developed that address quality of service (QoS) for multimedia applications in wireless networks.

Good QoS is necessary, but not necessarily sufficient for good user perception, enjoyment, and acceptance of a service. Thus, quality of experience (QoE) / quality of perception has emerged as an important concept, covering subjective and objective, qualitative and quantitative measures of ultimate importance for users and thus, for service providers and operators.

QoE issues have been creating a new assessment and management paradigm in multimedia systems, and they are gaining special attention in wireless communication networks, as the latter have proven to be quite hostile environments for multimedia streaming because of volatile radio conditions and lacks of capacity. QoE metrics are considered as important metrics to measure the quality level of multimedia contents from the users' perspective. QoE-oriented approaches aim to overcome the limitations of current QoS-aware schemes, not only emerging as a new trend towards objective measures but also providing necessary links to human perception.

The use of such metrics is expected to turn the optimization of wireless networks more efficient in terms of user satisfaction than traditional techniques that focus solely on objective QoS metrics such as throughput, delay, jitter, throughput-based fairness, etc. QoE models that include user perception and user behavior are of essential importance for QoE-aware optimization of any communication system. In particular, novel QoE-aware transmission approaches are encouraged to be proposed to improve the efficiency of wireless networks for multimedia services.

The aim of this feature topic issue is to encourage researchers to submit their work related to QoE-aware wireless multimedia networks. Authors with recent unpublished work on QoE modeling, measurements, analysis, control, and optimization are particularly encouraged to submit their original contribution to this special issue. This feature topic also aims at bringing together the state-of-the-art research results of QoE issues for wireless multimedia networks.

Topics of Interest

The topics relevant to this special issue include but are not limited to:

- Relationships between traffic patterns, QoS and QoE parameters
- QoE models, their applications and use cases
- QoE inference from user behavior
- QoE measurement methodologies and metrics (subjective, objective, pseudo-subjective testing, etc)

- QoE for new wireless multimedia applications (e.g. 3D video)
- QoE-based routing and resource management
- QoE-aware transmission in wireless systems
- QoE-based network management
- QoE-aware cross-layer design
- QoE-driven adaptation and control mechanisms for wireless systems and devices
- QoE-based optimization in wireless environments
- Testbed for QoE performance evaluation
- Media synchronization, playback, and buffer management

Papers must strictly focus on QoE issues and thus significantly go beyond current QoS approaches. The editors maintain the right to reject papers they deem to be out of scope of this special issue. Only originally unpublished contributions and invited articles will be considered for the issue. The papers should be formatted according to the ETT guidelines (http://onlinelibrary.wiley.com/journal/10.1002/(ISSN)1541-8251/homepage/ForAuthors.html).
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