User-Centric Network-Application Interaction for Live HD Video Streaming

Thomas Zinner[1], Dominik Klein[1] and Tobias Hossfeld[1]

Chair of Communication Networks, University of Wuerzburg
{zinner|dklein|hossfeld}@informatik.uni-wuerzburg.de

Abstract. Applications and resource allocation within the network become more and more flexible in supporting divergent user demands. This is reflected by state-of-the-art video codecs like H.264/SVC which allow a dynamic adaptation of the video quality and therewith the required network resources. Within the network, techniques like network virtualization and multipath data transport allow flexible allocation of network resources based on application demands. This paper outlines the potential of the interaction between application and network with an example of a scalable video streaming service and a multipath transport network.

1 Introduction

Internet applications are becoming more and more flexible and support divergent user demands and network conditions. This is reflected by technical concepts, which provide new adaptation mechanisms to allow fine grained adjustments of the application quality.

To overcome network resource limitations, a service has to adjust its demands to the available network resources in order to provide the best possible user perceived quality, denoted to as *Quality-of-Experience* (QoE) [7]. In particular, this is of high importance for applications with stringent requirements like high-quality video streaming or cloud gaming. Accordingly, such applications have to use the available resources efficiently with respect to the user perceived quality. This requires a continuous QoE management of the video streaming service including the monitoring of the current network state as well as control mechanisms to dynamically adapt the video system to deliver the optimal QoE. Typically, the required bandwidth of a video stream depends on the frame rate, the video quality, and its resolution. The dynamic adaptation of these parameters at the involved devices and within the network can be achieved for instance by the

The work has been conducted in the project G-Lab, funded by the German Ministry of Educations and Research (Förderkennzeichen 01 BK 0800, G-Lab) and within the G-Lab project COMCON, funded by the German Ministry of Educations and Research (F¨oerderkennzeichen 01BK0918, G-Lab). The authors alone are responsible for the content of the paper.

scalable extension of the state of the art video codec H.264/AVC. Applications which monitor the network and adapt themselves to the current state are called *network-aware applications*.

Within the network and transport domain, new technologies have evolved during the last years providing a more flexible and efficient usage of data transport and network resources. The most promising technologies are *Network Virtualization* (NV), which is seen as an enabler to overcome the ossification of the Internet stack [5], and multipath transport, i.e., the simultaneous data transfer via multiple available transport paths.

NV promises to overcome the limitations of the current Internet and its protocols [11] by providing means to simultaneously operate multiple logical networks on a single physical substrate. Physical resources can be easily added to virtual networks to cope with high demands or removed to reduce operational expenditures during times of low workloads. This enables a flexible and efficient allocation of network resources based on the demands of specific virtual networks.

Multipath transport allows the concurrent usage of network resources and thus provides higher capacities for end devices. Accordingly, this technique allows faster downloads or video streaming with higher quality as compared to the utilization of only a single network resource. Both techniques allow the network to react on application demands, denoted as *application-aware networks*.

The combination of both approaches may be beneficial for both sides, the network and the application. On the one hand, network resources can be utilized more efficiently and on the other hand, applications can easily adapt their demands to the available resources. In this work, we detail the management of application and network interaction with the example of multipath video streaming using the scalable video codec H.264/SVC and show the potential of such an interaction.

The remainder of this paper is structured as follows: In Section 2 we provide background and requirements for scalable video streaming. This includes an introduction to QoE assessment. The benefit of interaction between network and application for multipath video streaming based on H.264/SVC is evaluated in Section 3. Section 4 discusses future work and in Section 5, the paper is concluded.

2 Scalable Live HD Video Streaming

In this section we briefly outline QoE for video streaming before we discuss user, network, and application requirements.

2.1 Background on QoE for Video Streaming

First we summarize different objective and subjective methods to evaluate QoE for video streaming. After that we detail the Provisioning Delivery Hysteresis which describes fundamental relationships between Quality-of-Service measures and QoE.

Objective and Subjective QoE QoE is in its nature a subjective measure of the quality a user experiences when using a service such as Voice-over-IP, video streaming, or browsing the web. It combines non-technical parameters such as user perception, experience, and expectations with technical parameters on application and network level.

The QoE of an application is a characteristic that is highly subjective to a user's perception, experience and expectations. Thus, the obvious way to measure it is to vary the network Quality of Service (QoS) and investigate its impact on the user perception by asking the users themselves. Based on the user ratings, the QoE can be calculated as, e.g., the average user rating, the Mean Opinion Score (MOS). The QoE assessment based on user surveys is called *subjective* QoE (sQoE) throughout this work.

On the other hand, it is possible to compare the original video clip with the received video clip based on signal and frame processing techniques. Thus, the distortion of the received video clip can be evaluated and the received video quality can be calculated. Based on the video quality, an approximation of the user perception can be estimated. Such an approximation of the user-perceived quality based on computational metrics is called *objective* QoE (oQoE) throughout this work.

The perceived quality can be investigated in subjective tests, where presented stimuli—such as impaired video sequences—are rated by subjects under controlled conditions. However, the assessment of results from these tests is difficult, since the individual scoring depends on daily form and mood of the human subject, as well as on generally overly pessimistic or overly optimistic users. For the purpose of having comparable and uniform tests, the International Telecommunication Union has created a recommendation for standardized user tests concerning video and image quality in ITU-R BT.500-11 "Methodology for the subjective assessment of the quality of television pictures" [8]. The recommendation gives guidelines to normalize the viewing conditions for a subjective study, in terms of room illumination, display position and parameters (resolution, contrast, brightness), hardware to use, the selection of test materials and the length of test sessions. Typical methods for assessing the sQoE are Double-Stimulus Continuous Quality Scale (DSCQS), Double-Stimulus Impairment Scale (DSIS) and Single-Stimulus Continuous Scale Quality Evaluation (SSCQE), as discussed in detail in [8].

The grades of the scale are mapped for instance to numerical values from 1 (bad quality) to 5 (excellent) or 1 to 100, and the mean of the scores, the MOS value, is obtained for each test condition. The obtained rating expresses the subjective Quality of Experience (sQoE).

The presented methods differ in their objectives. Whereas DSCQS is used to measure the remaining quality of an impaired video relative to the reference video, DSIS focuses on the distortions and rates impairments. SSCQE measures the overall satisfaction of the user with his experience regardless of a reference. Another criterion for the selection of a specific method is available equipment, budget, and time.

The results of such surveys reflect the user's perception and thus have a high significance. However, due to different quality judgment of human observers, multiple subjects are required to participate in a subjective study [13]. According to [8], at least 15 observers should asses stimuli in order to gain significant results. Tests are conducted manually in a controlled environment which is time-consuming and costly. Thus, it should be used as base data for objective video quality algorithms which automatically predict the visual quality of a video clip.

Objective video quality metrics can be classified into three categories by the required amount of reference information [15]: *Full-Reference* (FR) metrics are based on frame-by-frame comparison between a reference video and the video to be evaluated; *No-Reference* (NR) metrics have to make assumptions about the video content and distortions, e.g. by evaluating the blockiness of a frame, as a common artifact in block-based compression algorithms such as MPEG; *Reduced-Reference* (RR) metrics evaluate the test video based on a subset of features previously extracted from the reference video. Based on the complex nature of cognitive aspects and the human visual system, objective quality metrics do not capture its entire complexity and focus on aspects, which have been shown to correlate well with human perception in subjective tests. For our studies we use publicly available full reference metrics, i.e., the Peak Signal to Noise Ratio (PSNR), Structural Similarity Index Metric (SSIM), and Video Quality Metric (VQM). These mechanisms range in their complexity and their correlation with human perception.

To evaluate the impact of video distortion on the user experience, we use the SSIM metric and the SSIM to MOS mapping function presented in [4]. This exponential fitting function $f(x) = 13.91 \cdot e^{1.715 \cdot x}$ is intended to map SSIM values, ranging from 0 to 1, to a mean opinion score ranging from 0 to 100. It has to be noted that the used mapping function allows for MOS values $MOS \in [f(0); f(1)] = [13.91; 77.29]$. Accordingly, the results are presented on a scale of 1 to 100 instead of 1 to 5.

Provisioning Delivery Hysteresis The QoE-Provisioning Delivery Hysteresis (PDH) is introduced in [6] and describes fundamental relationships between QoE and Quality of Service (QoS) parameters. The PDH reveals that the impact of a controlled reduction of application demands and thus an adaptation to the available network resources outperform the uncontrolled adaptation to insufficient network resource, e.g., like packet loss. For both types of degradation we consider the goodput as joint parameter. For the controlled degradation or the *provisioning*, the relative goodput ratio is defined as the ratio between the current capacity compared to the capacity required for the optimal quality. For the uncontrolled degradation, the *delivery*, the goodput is defined as $1 - p_l$ where p_l is the packet loss rate.

Figure 1 sketches the hysteresis as a set of functions of the goodput ratio. While specific relationships between QoE and goodput ratio depend amongst others on application and context, we observe two fundamentally different areas. Controllable quality distortion allows to keep the QoE rather high in view of considerable savings, i.e. goodput ratios much smaller than one. Significant decreases

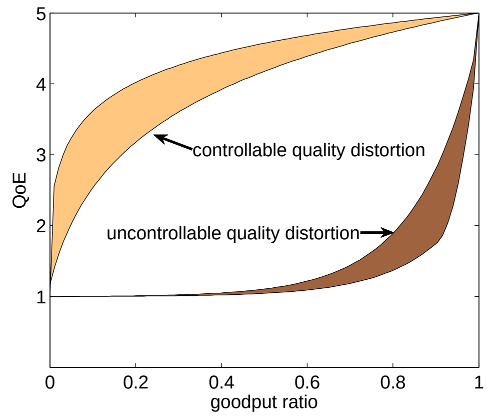


Fig. 1. Illustration of the QoE Provisoning-Delivery Hysteresis.

in QoE are observed for rather small goodput ratios. Uncontrollable quality distortion, however, yields a completely different behaviour. Small decreases in the goodput ratio imply large decreases in the QoE values, while that decrease flattens out at the lower edge of the QoE scale as the goodput ratio sinks. This implies that, in order to ensure good QoE, it is better to apply controlled actions than to suffer from problems that appear in an uncontrolled way.

2.2 Requirements for Scalable Live HD Video Streaming

First, we discuss the user requirements for the video streaming use case and derive requirements to the application which can be mapped to network requirements.

User Requirements The presented use case considers live streaming services. The user expects high video quality and a high resolution for watching e.g. a soccer match in HD resolution at home. Furthermore, the contents are to be delivered in real-time to guarantee the live experience during watching. Based on these user requirements, the network and application requirements are derived.

Requirements to the Network Due to the real-time nature of TV, only a small video buffer is available at the application which is in the order of a few seconds only. Hence, the network has to deliver the data with low delay and also low jitter that the video buffer can cope with. If data packets in the network get lost, retransmissions are often not possible without violating the real-time constraint. Typically, connectionless transport protocols are used therefore. However, data packets still should arrive in order at the video client so that the video player on application layer does not need to take care about the order of frames. Thus, in-order transmission and reception of frames is a network requirement.

As the user demands high video quality and high resolution, the network has to be capable of providing sufficient bandwidth to carry the video contents. Since video streaming is prone to packet loss and small packet loss rates already lead to a strong impairment of QoE [6], the packet loss ratio should be fairly low. Such insufficient network resources, e.g., too little bandwidth or packet loss, result in a strong impairment of the video service in form of video decoding errors and frame drops. Controlled quality adaptation e.g. by reducing image quality/resolution or frame rate allows considerable bandwidth savings while having only minor impact on user perceived quality [6, 17].

Requirements to the Application Typically, a video clip can be encoded in different qualities with respect to the frame rate, the resolution and the image quality. In case of limited network resources, these properties should be adapted so that the user still perceives a good quality. At least a minimum resolution, quality and frame rate, which also depends on actual usage context, has to be provided. Otherwise the user will not accept the video service. Higher quality-related layers will increase the QoE further, but also require a higher bandwidth.

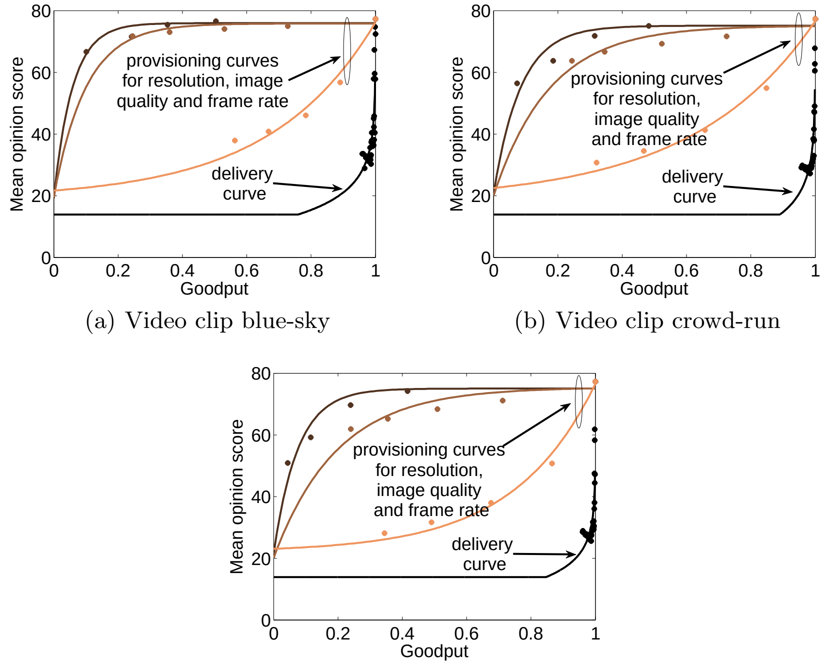
Current research on the impact of packet loss on user perceived quality indicates a strong impairment of the video already at small packet loss rates of less than 2 % [18]. This motivates mechanisms able to protect a video stream against packet loss, such as Forward Error Correction (FEC) methods. These techniques provide means to correct corrupted or lost packets and thus allow an exchange of bandwidth against packet loss protection.

3 Benefit of Interaction for Multipath Video Streaming using H.264/SVC

At first, this section highlights the impact of controlled quality degradation on the user perceived quality. Then, the effect of increasing the network capacity by using multipath transport is discussed. Finally, it presents the benefit of an interaction of both adaptation mechanisms, one on application and the other on network layer.

3.1 Adaptation to Network Resources on Application Layer Using H.264/SVC

In order to optimize the Quality of Experience (QoE) for live video streaming by real-time adaptation of video bitrate and thus video quality to the current network situation (i.e., available bandwidth or unreliable links with packet losses), it is necessary to know the bandwidth requirements of the different layers and their impact on QoE. Therefore, we conducted an intensive measurement study with different network conditions. For evaluating the user perceived quality, we use a full-reference video quality metric SSIM [2] and map it to subjective mean opinion scores (MOS). For our investigations we rely on H.264/SVC, the scalable extension of H.264/AVC [9], which provides an effective way to reduce the required bandwidth and adjust the video quality within the network.



(a) Video clip blue-sky

(b) Video clip crowd-run

(c) Video clip park-joy

Fig. 2. Provision-Delivery Hysteresis for the video clips blue-sky, crowd-run and park-joy.

This extension enables the encoding of a video file at different qualities within the same layered bit-stream. Besides of different resolutions, this also includes different frequencies (frames displayed per second) and image qualities with respect to Signal-to-Noise Ratio (SNR), which is referred to as spatial, temporal and quality scalability. In particular we investigate the resolution 1920x1080, 1280x720, 960x540, 640x360, and 480x270, different image qualities according to the quantization parameter ranging from 30 to 38 in steps of 2, and the frame rates 30, 15, 7.5 and 3.75 The impact of a bandwidth reduction, either by packet loss or by a graceful reduction of the video quality is shown for three different video clips in Figure 2. The x-axis denotes the *goodput ratio*, i.e. the relative capacity perceived on application layer as compared to the capacity for which the QoE is maximal. The y-axis denotes the user perceived video quality, respectively. As can be seen, a reduction of the quality in order to adapt the video bandwidth to insufficient network resources in terms of end-to-end throughput results in still a high good QoE. As indicated by the results, resolution and quality adaptation outperform temporal adaptation in terms of user perceived quality. However, as soon as packet loss appears, the video quality is strongly impaired. In order to cope with packet loss, additional mechanisms have to be provided, e.g., forward error correction (FEC) mechanisms. In order to evaluate such a mechanism, we implemented Luby codes [12] and evaluated the interaction of packet loss protection bandwidth reduction. In the investigated scenario, packet loss protection is implemented by adding additional redundancy while bandwidth reduction is performed by reducing the video resolution. More details can be found in [20].

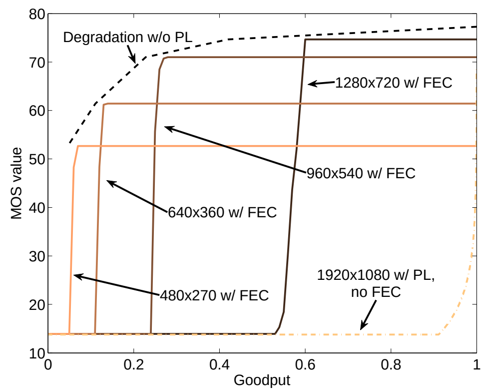


Fig. 3. Guideline for selecting the resolution of the SVC video stream depending on the available network capacity. In case of packet loss due to unreliable transmission, forward error correction is suggested to achieve high QoE, which increases the need for goodput.

The results of our study are depicted in Figure 3 with the user perceived quality in terms of MOS on the y-axis. The x-axis denotes the relative goodput, i.e. the application-perceived throughput, which is equal to one if the maximum quality can be streamed. In case of packet loss p_l the resulting goodput is $1 - p_l$. For streaming a video in 1920×1080 resolution with no FEC, the QoE decreases rapidly and becomes minimal for packet loss rates larger than 2 %, i.e. a goodput of 98 %. This is shown by the lower dashed line. In case of controlled quality reduction (lower resolutions) but no packet loss, QoE decreases much slower and maintains a tolerable value even for small resolutions at 480×270 pixels. This is illustrated by the upper dashed line. In that case, only 15 % of the bandwidth compared to the maximum resolution is required. Hence, in case of bandwidth shortage, SVC is a powerful mechanism for reducing the QoS requirements of the application without compromising QoE too much.

The other cases in Figure 3 denote controlled service degradation with FEC mechanisms. As an example, streaming a video with a resolution of 1280×720 with FEC is considered. Switching to this resolution reduces the required goodput by 60%. We add a FEC mechanism which adds an overhead of 50 %, resulting in a relative goodput of 60 %. By transmitting more symbols, it is now possible to cope with packet loss rates up to 40 % and still ensuring a very good quality of the video stream and not allocating more bandwidth than before. For lower video stream qualities, more bandwidth can be used to protect the video stream. The results in Figure 3 provide clear guidelines how to optimize QoE for live TV streaming according to the current network situation.

3.2 Adaptation of the Network Capacity to Application Demands using Multipath Transport

The split-up of a data flow on multiple paths towards a common sink has recently attracted a lot of attention since it allows to utilize different access networks, e.g., 3G and WLAN which are usually available on today's smart phones. Techniques like transport virtualization [14] or multipath transport protocols like Multipath-TCP [10] or SCTP-CMT [16] allow a flexible usage of these parallel network resources. A multi-path transmission is initiated by a *splitting* component that splits up the data on disjoint paths. In contrast to Equal-Cost-Multi-Path (ECMP), the splitter is not bound to simple packet-wise or flow-wise load balancing but might be able to make use of knowledge on the structure of the transmitted data, i.e., a scalable video stream. Finally, the multiple transmission paths end in an *assembling* component. Thereby, the assembler not only has to join but also needs to synchronize corresponding data streams by means of buffering and has to prevent reordering of packets.

3.3 Benefit of Mutual Adaptation and Interaction

If network and application are flexible and can adapt to application demands or network state, the question arises how such an adaption should be done. In the following, we discuss this question for the example of a video streaming service based on H.264/SVC and an exemplary network setup with two available paths. A discussion how to implement such a mechanism within a virtual network can be found in [3]. Each path is able to provide resources for a number of x video streams, e.g., for the equivalent of 1.5 video streams. We assume that the capacity of both paths can be bundled by a multipath transmission mechanism. We study four different policies, (1) no service or network interaction in case of insufficient resources, (2) the network reacts to the bottleneck by increasing the capacity with a concurrent transmission via a second path, (3) the service reacts by reducing the quality of the video stream, (4) first the network reacts by providing additional capacity, then the service decreases its demands.

For this study we use the clip *crowd-run* as reference as illustrated in Section 3.1. For the delivery curve, i.e., for the case of packet loss we use the illustrated fitting function, for the provisioning part we use the fitting function for the quality degradation by reducing the resolution. The results of the study are depicted in Figure 4. The x-axis denotes the number of customers using the video streaming service, the y-axis the average quality per customer. For the case of no interaction between applications and network, the application quality is sufficient for one customer, but already for two customers, the quality per customer drops rapidly. The network resources are insufficient and neither the

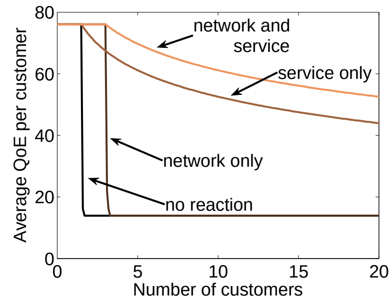


Fig. 4. Average QoE per customer for a varying number of customers and different adaptation mechanisms.

application nor the network adapt to the congestion state. Thus, a huge amount of packets are lost, resulting in low service quality. If the network reacts to the congestion by providing additional resources, the application can be provided to more customers without impairments. However, for more than three customers, the service quality drops rapidly. Again, the available network resources are not sufficient, congestion and packet loss occur resulting in a low service quality per customer. Otherwise, the service can adapt to the congestion by reducing the service requirements, i.e., reducing the quality of the video stream. Thus, the network can be relieved and recover from the congestion. Accordingly, no packets are lost and the huge impact of packet loss on video streaming can be avoided. However, in case of service quality adaptation, the quality of the video stream is reduced due to the adaption to the available resources. As illustrated by Figure 4, the best quality for customers can be achieved if both, service and network, adapt to the rising number of customers. In this case the network first reacts to the congestion and ensures that the service quality can be kept to the maximum. However, if further customers join the system no additional network resources can additionally be provided. The network may signal this to the service which then reduces its demands by reducing the service quality. Thus, the average service quality per customer can be maximized.

The results indicate a high potential to increase the user perceived quality for the interaction between networks and applications. To show the capabilities of multipath video streaming based on H.264/SVC, we implemented a software framework enabling multipath functionality either at the edge or within the network. We integrated the implemented multipath streaming framework into the demonstration of the COMCON [1] project at the GLab Status Meeting during the Euroview 2012 event [19]. During the course of the demonstration, the capabilities of a dynamic Network Virtualization Infrastructure based on GMPLS and of an elaborated service provisioning architecture, the multipath SVC streaming architecture, were shown.

4 Future Work

Future work in the area of service and network interaction may investigate the following issues:

– Monitoring: In order to be able to adjust network capabilities and application demands an appropriate monitoring of the varying application and network conditions is necessary. This includes a discussion of the trade-off of monitoring accuracy and the resulting monitoring and signaling overhead. Further, appropriate sampling techniques might be necessary if the data rates are too high.
– Control: How shall application and network react to changing conditions? On the one hand, the video quality should be maximized with respect to the current network resources, on the other hand frequent changes of the video quality also disturb the user experience.
– Decision Component: A detailed investigation where to place the decision entity including communication protocols between these entities as well as a discussion of the benefits for the application and network are required.
– Flow of Information: The necessary information, their impact on the user perceived quality as well as the information exchange between network and application have to be discussed. Taking video streaming into account, the impact of network distortion on the user perceived quality typically varies for each video scene. Thus, the question arises if a MOS value of a layer for the whole video is enough? Or if it is necessary to provide the MOS value for each scene of a layer?

5 Conclusions

This paper investigated the potential of network and service interaction for multipath transport in the network and service adaptation for H.264/SVC. It derives guidelines how the video streaming service should react in case of insufficient network resources.

In principle, the network can adapt to a congestion state by providing additional resources and thus support more customers. However, if no additional resources can be provided, congestion and packet loss occur, resulting in a strong impairment of the service quality. If the service adapts to the congestion in the network by reducing its requirements, the service quality is reduced gracefully. Thus, the perceived quality of a customer is reduced compared to the maximum available service quality, but still is much higher than in case of delivery failures and packet loss. Further, we have seen that the best service quality can be provided if both, network and service react together.

Current work in the area of future networks discusses a tighter interaction of applications and networks by providing unified interfaces between applications and networks. The design as well as the operation of such interfaces are objectives for future research.

References

1. "COMCON - COntrol and Management of COexisting Networks," 2011, information available at http://www.german-lab.de/phase-2/comcon//.
2. D. Brunet, E. R. Vrscay, and Z. Wang, "A class of image metrics based on the structural similarity quality index," in *ICIAR (1)*, ser. Lecture Notes in Computer Science, M. Kamel and A. C. Campilho, Eds., vol. 6753. Springer, 2011, pp. 100–110.
3. M. Duelli, S. Meier, D. Wagner, T. Zinner, M. Schmid, M. Hoffmann, and W. Kiess, "Experimental Demonstration of Network Virtualization and Resource Flexibility in the COMCON Project," in *8th International ICST Conference on Testbeds and Research Infrastructures for the Development of Networks and Communities*, Thessaloniki, Jun. 2012.
4. U. Engelke, M. Kusuma, H.-J. Zepernick, and M. Caldera, "Reduced-reference metric design for objective perceptual quality assessment in wireless imaging," *Signal Processing-Image Communication*, pp. 525–547, 2009.
5. A. Feldmann, M. Kind, O. Maennel, G. Schaffrath, and C. Werle, "Network virtualization - an enabler for overcoming ossification," *ERCIM News*, no. 77, pp. 21–22, April 2009, invited article; Special theme: Future Internet Technology. [Online]. Available: http://ercim-news.ercim.org/content/view/574/763/
6. T. Hoßfeld, M. Fiedler, and T. Zinner, "The QoE Provisioning-Delivery-Hysteresis and Its Importance for Service Provisioning in the Future Internet," in *Proceedings of the 7th Conference on Next Generation Internet Networks (NGI)*, Kaiserslautern, Germany, Jun. 2011.
7. T. Hoßfeld, D. Hock, K. Tutschku, and M. Fiedler, "Testing the IQX Hypothesis for Exponential Interdependency between QoS and QoE for Voice Codecs iLBC and G.711," in *18th ITC Specialist Seminar on QoE*, Karlskrona, Sweden, May 2008.
8. ITU-R Rec. BT.500, "Recommendation 500-10: Methodology for the subjective assessment of the quality of television pictures," 2000.
9. ITU-T Recommendation, "H.264 : Advanced video coding for generic audiovisual services," 2010.
10. J. Iyengar, P. Amer, and R. Stewart, "Concurrent multipath transfer using SCTP multihoming over independent end-to-end paths," vol. 14, no. 5, pp. 951–964, 2006.
11. D. Kofman and J.-P. Lefvre, "First update of the Euro-NF vision regarding the network of the future - D.SEA.10.1.1," Available at http://www.euronf.org, 2009.
12. M. Luby, "LT Codes," in *43rd Annual IEEE Symposium on Foundations of Computer Science*, 2002.
13. K. Seshadrinathan, R. Soundararajan, A. C. Bovik, and L. K. Cormack, "Study of subjective and objective quality assessment of video," *IEEE Transactions on Image Processing*, vol. 19, no. 6, pp. 1427–1441, Jun. 2010. [Online]. Available: http://dx.doi.org/10.1109/TIP.2010.2042111
14. K. Tutschku, T. Zinner, A. Nakao, and P. Tran-Gia, "Network virtualization: Implementation steps towards the future internet," in *Electronic Communications of the EASST, Volume 17: Kommunikation in Verteilten Systemen 2009*, Kassel, Germany, March 2009.
15. S. Winkler, "Video Quality and Beyond," in *Proc. European Signal Processing Conference*, 2007.
16. D. Wischik, M. Handley, and M. B. Braun, "The resource pooling principle," *ACM SIGCOMM Computer Communication Review*, vol. 38, no. 5, pp. 47–52, 2008.

17. K. Yamagishi and T. Hayashi, “Parametric Packet-Layer Model for Monitoring Video Quality of IPTV Services,” in *ICC ’08. IEEE International Conference on Communications*, 2008, pp. 110–114.
18. T. Zinner, O. Abboud, O. Hohlfeld, T. Hoßfeld, and P. Tran-Gia, “Towards QoE Management for Scalable Video Streaming,” in *21th ITC Specialist Seminar on Multimedia Applications - Traffic, Performance and QoE*, Miyazaki, Jap, Mar. 2010.
19. T. Zinner, D. Klein, S. Meier, D. Wagner, M. Hoffmann, W. Kiess, V. Singorzan, and M. Schmid, “Dynamic Topology Adaptation enabled by Network Virtualization: A Use-Case for the Future Internet.” 12th Wuerzburg Workshop on IP: ITG Workshop ”Visions of Future Generation Networks” (EuroView2012), Jul. 2012.
20. T. Zinner, F. Liers, T. Hoßfeld, D. Rauscher, B. Reuther, D. Günther, T. Volkert, and M. Fiedler, “Prospects for Realizing User-Centric Network Orchestration: FEC-protected SVC Streaming,” in *7. GI/ITG KuVS Fachgesprch ’Future Internet’*, Mnchen, Jan. 2012.