MULTIMEDIA COMMUNICATIONS TECHNICAL COMMITTEE
IEEE COMMUNICATIONS SOCIETY
http://www.comsoc.org/~mmc

R-LETTER

Vol. 2, No. 3, June 2011

IEEE COMMUNICATIONS SOCIETY

CONTENTS

Message from R-Letter Director

Welcome to the third issue of IEEE MMTC Review-Letter (R-Letter) in 2011. Thank you for your continuing great support. In this issue, we are pleased to introduce six high quality papers. Three papers are from Comsoc-sponsored journals, one from MMTC-sponsored conferences, and two from multimedia communication related journals

The first paper, published in the *IEEE Journal on Selected Areas in Communication*, describes a cross-layer optimization for multi-user video streaming over wireless networks via exploring the multiuser diversity. The second paper, from the *IEEE Communication Magazine*, proposes a peer-to-peer multimedia transmission framework with the support of scalable video coding to achieve better end-user experience. The third paper, from the *IEEE Transactions on Multimedia*, introduces a multi-layer video multicasting solutions to tackle the resource allocation problem in hybrid 3G/ad hoc networks.

The fourth paper, published in the *2010 IEEE International Conference on Multimedia & Expo*, investigates the performance improvement for distributed video coding via rate control in the decoder side with aid of network working. The fifth paper, from the *IEEE Transactions on Broadcasting*, introduces a new methodology to evaluate subjective video quality assessment by encouraging viewers to watch the video in the same environment they normally watch television, which intents to bridge the gap between audiovisual quality measurement in research and real-world applications. The last paper, published in the *IEEE Transactions on Information Forensics and Security*, studies the biometric security from information theoretic point of view to jointly maximize achievable security and privacy levels.

Besides the aforementioned introduced papers, MMTC will hold the first annual Multimedia Communications Workshop in conjunction with the IEEE Globecom, Houston, Texas, Dec. 2011. Please see details on Page 24.

We hope that this issue will be both informative and a pleasure to read.

Finally, I would like thank all the editors of this issue for their great work: Hulya Seferoglu, Christian Timmerer, Cheng-Hsin Hsu, Vladimir Stanković, Tao Liu, and Man-On Pun.

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Wireless Video Streaming: Exploiting Multi-User Diversity

A short review for "Cross-layer optimization for streaming scalable video over fading wireless networks "

Edited by Hulya Seferoglu

H. Zhang, Y. Zheng, M. A. Khojastepour, and S. Rangarajan, "Cross-Layer Optimization for Streaming Scalable Video over Fading Wireless Networks," IEEE Journal on Selected Areas in Communications, vol. 28, no. 3, pages 344-353, Apr. 2010

Popularity of streaming video over wireless networks is increasing, and the significant progress in video compression techniques, wireless data communication, and cross layer design are continuously advancing the state-of-the art in wireless video [1], [2]. On one hand, the data transmission rates of wireless networks are steadily growing, *e.g.*, 1Gbps target rate for nomadic and 100Mbps for mobile users in 4G systems [3]. On the other hand, H.264/MPEG4-AVC [4] achieves more efficient video compression and the Scalable Video Coding (SVC) extension [5] of H.264/MPEG4-AVC obtains both high coding efficiency and high scalability. However, providing high quality video over wireless networks is still a challenging problem, because the wireless medium is often shared by many users.

When a wireless base station receives multiple requests for streaming service of different video sequences, it has to make the following decisions; (i) how much radio resources should be allocated to each user in order to maximize the overall video quality, (ii) how to achieve the desired radio resource allocation. These decisions are especially challenging for fading wireless channels.

This paper presents a cross-layer design of transmitting scalable video streams from a base station to multiple clients over a shared fading wireless network. In particular, rate adaptation and exploiting the multi-user diversity for video streaming using video coding over a shared fading wireless channel is jointly considered in a cross-layer optimization framework.

The authors formulate an optimization problem which maximizes the weighted sum of video quality of all users subject to the achievable long-term (ergodic) rate constraint under a fading wireless channel model. A long-term resource allocation algorithm is proposed as a solution to the optimization problem. The long-term resource allocation algorithm determines the wireless scheduling policy and the parameters used by the scheduling policy. Its key idea is to exploit multi-user diversity by scheduling users in relatively good wireless channel conditions and adapt the video transmission rate to the wireless channel capacity using SVC.

Based on the structure of the optimization solution (the long-term resource allocation algorithm), an online scheduling algorithm (*Static*) is proposed to meet real-time video traffic Quality of Service (QoS). An improved version of *Static* scheduling algorithm; *Dynamic* is proposed to meet the instantaneous rate, deadline requirements of video traffic and wireless channel conditions. Both scheduling algorithms answer the following questions; (i) at each time slot, which user should be scheduled, (ii) after a user is selected, which packets/frames of the selected user should be transmitted, and (iii) when does a base station need to drop frames and which frames should be dropped?

In the *Static* scheduling algorithm, user scheduling policy is determined according to the long term resource allocation algorithm. In particular, a user with the highest product of the channel capacity and the resource allocation value (determined by the long term resource allocation algorithm) is selected for transmission. Among the packets which are destined to be transmitted to the selected user, a packet with the earliest deadline is selected and transmitted by the base station to the selected user. In terms of packet dropping, two schemes are employed. The first scheme drops packets when their deadlines expire, while the second one employs early dropping scheme in which less important packets in terms of video quality and deadline are dropped. In the *Dynamic* scheduling algorithm, resource allocation value for each user is periodically updated to reflect the instantaneous rate and delay requirements of video streams.

The simulation results show that both *Static* and *Dynamic* scheduling algorithms achieve higher video quality in terms of average PSNR as compared to the baseline schemes such as the maximum capacity scheduling, proportional fairness scheduling [6], and modified largest weighted delay first scheduling [7]. As compared to the baseline schemes, the PSNR improvement of *Static* and *Dynamic* scheduling algorithms is as high as 10dB which is significant in terms of video quality improvement. It has also been shown that the proposed schemes are much more robust under weak wireless channel conditions, and they are robust to channel estimation errors.

The authors clearly demonstrate the great potential of rate adaptation and exploiting the multi-user diversity for video streaming over a shared wireless channel to improve video quality and design robust schemes for time varying channel conditions as well as time varying video rate and delay requirements. An interesting extension of this work would be to understand the potentials and limitations of rate adaptation and exploiting the multi-user diversity for smartphones whose demand for video applications are increasing rapidly over 3G/LTE links [8].

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Peer-to-Peer Streaming: Scalable Video Coding Approach

A short review for "Peer-to-peer streaming of scalable video in future Internet application"

Edited by Christian Timmerer

N. Ramzan, E. Quacchio, T. Zgaljic, S. Asioli, L. Celetto, E. Izquierdo, F. Rovati, "Peer-to-Peer Streaming of Scalable Video in Future Internet Application", IEEE Communication Magazine, vol. 49, no.3, pp.128-135, March 2011

Peer-to-peer (P2P) protocols have emerged in the years as an alternative solution to current and future multimedia Internet applications. A major advantage of P2P protocols is that each peer involved in the content delivery process contributes with its own resources to the actual delivery. As a consequence, there is an increase in the amount of overall resources in the network and the usual bottleneck problem of the client-server model can be overcome.

One of the most commonly used P2P protocols for file sharing is BitTorrent [1]. In order to make it suitable also for streaming applications, substantial research has been carried out recently. In BitTorrent, file chunks are downloaded in rarest-first fashion. This is an efficient strategy in file sharing since the availability of rare chunks is eventually increased, and higher downloads rates can be achieved. However, in video streaming this can result in an interruption of the video playback since chunks are not received sequentially. Therefore, special care needs to be given to those chunks that are close to the playback position. An example of an algorithm that takes into account these considerations is Give-to-Get (G2G) [2], implemented in Tribler [3]. In this algorithm chunks of compressed video are classified into three priority categories according to playback position: high, medium, and low. Chunks with *high* priority are downloaded in sequential order, while *medium*- and *low*-priority chunks are downloaded in standard rarest-first strategy.

Despite these efforts, several problems are still open and need to be addressed in order to guarantee high Quality of Service in particular for the delivery of high-resolution video. In fact, the bandwidth capacity of a P2P system is extremely varying, as it relies on heterogeneous peer connection speeds and directly depends on the number of connected peers. To cope with varying bandwidth capacities inherent to P2P systems, as well as to match features of wide range of high-end and mobile terminals, the underlying video coding/transmission technology needs to support bit-rate and spatio-temporal adaptation.

Scalable video coding (SVC) techniques [4][5] address these problems as they allow "encoding a sequence once and decoding it in many different versions." The scalable coded bitstreams can efficiently adapt to the application requirements, as adaptation is performed fully in the compressed domain, by directly editing parts of the bitstream.

In order to match flexibility of scalable contents with efficiency of distributed protocols, some modifications are necessary in the P2P streaming architecture. At the sender side, the compressed scalable bitstream can be further processed to make it more suitable for transmission and additional metadata can be added into the stream or in a torrent file. Such data may contain information about the organization of the video into scalable layers, the resolution and quality of each layer, and so on. At receiver side, given a scalable encoded video unit, a peer can initially decide to recover all or just a subset of available layers, depending on its capabilities. Besides such a static selection, a peer can dynamically retrieve just a subset of layers in order to react to a temporary narrowing of the bandwidth. Such dynamic adaptation can be achieved through a carefully designed piece-picking policy. Furthermore, peers which provide the base layer need to be carefully selected, in order to avoid interruption in the playback. Such issues have already been considered in several P2P architectures aiming to support the delivery of scalable media.

The MVV platform proposed in [6] extends Tribler, by modifying G2G algorithm and peer selection policy, and adding the support for scalable video compressed using the W-SVC encoder [5]. The compressed sequence consists of groups of pictures (GOPs) and scalable layers. Along with the bitstream, the description file is generated, which contains information on mapping of GOPs and layers into chunks and vice versa; each GOP consists of chunks that are independent from other GOPs. The description file is transmitted with the video sequence. It has the highest priority and therefore should be downloaded before the bitstream. At the beginning of the streaming session, information about GOPs and layers is extracted from the bitstream description file. At this point, a sliding window is defined, and the pre-buffering phase starts. Inside the window, chunks have different priorities. First, a peer will try to download the base layer (BL), then the first enhancement layer (EL1), and so on. Pieces from the BL are downloaded in sequential order, while all other pieces are downloaded rarest-first (within the same layer). Every time the window shifts, two operations are made. First, downloaded pieces are checked to evaluate which layers have been completely downloaded. Second, pending requests concerning pieces of the GOP located just before the window are dropped. Fully downloaded layers from that GOP are sent to a video player for playback. In order to avoid pauses in the playback, chunks belonging to the base layer are requested from good neighbors, i.e., those which provide the highest download rates. Each time the window shifts, download rates of all the neighbors are evaluated, and the peers are sorted in descending order. The pieces are then requested from peers providing download rates above a certain threshold.

Similar to MVV, the NextShare platform proposed in [7] is also based on Tribler and, thus, Bittorent protocol which has been modified to support adaptive delivery of scalable media. In NextShare the content is encoded using the scalable extension of advanced video coding (AVC) [4] and the bitstream generated is divided in several files, each one carrying a single scalability layer, i.e., within the torrent file, the layers are indexed as independent files. Additional meta-information describing content composition in term of scalable layers is added in torrent file in order to let client devices select the best layer according to local capabilities. The SVC content is encoded with a constant bitrate in a way that each chunk contains for each layer a constant number of GOPs and adaptation only occurs in correspondence of synchronization points between chunks. Clearly, even in this case content download starts from the file carrying the base layer. Once a sufficient amount of data has been collected (called safe buffer), the NextShare client tries to switch to a higher layer by retrieving chunks from the files carrying, e.g., a higher resolution. Download priorities are re-assigned dynamically to the higher layer according to the number of layers, playback position, buffer size, and also network performances. Peers are classified in good and bad according to their estimated transmission rate. Bad peers providing lower download speed are moved from the high-priority set to low-priority ones starting from the base layer.

A different approach to deliver scalable media over a P2P overlay is available through the SEACast streaming platform [8]. While MVV and NextShare are both based on a full mesh P2P topology created by the Bittorent protocol, in SEACast the scalable content is delivered over a multi-tree overlay. SEACast creates an overlay forest of independent trees, each one carrying a different part of the stream. A central entity (tracker/broker) is used for managing the overlay. Such an approach is typically push-based, i.e., a root node injects the content in the multi-tree architecture, and when a node receives a data packet, it also forwards copies of the packet to each of its children. In case of layered video, the root node injects each layer over a different sub-tree. The granularity introduced by splitting the content into multiple layers allows peers with limited upload bandwidth to contribute to the swarm by uploading data relative to a reduced version of the scalable bitstream. Intrinsic robustness to node failures is added as the loss of connections carrying an enhancement layer will cause temporary degradation of the overall quality. Clearly, in case of failure or congestion of a connection carrying the base layer, there is no possibility to easily recover the quality even if enhancement layers are received. In such a situation the performance may be improved by assigning priorities to each tree according to the importance of the layer carried, and injecting and forwarding the base layer over the most reliable path. A local probe technique is used in each node to select the best parent. In particular, a combination of round-trip-time, available bandwidth, and the number of already active connections is used. Once the parent node is selected for each layer, connections are established. The available bandwidth is checked periodically and in case of congestion or connection failure the peer contacts the broker again to rebuild the tree.

The platforms described offer a good example of some of the latest advancements in exploiting flexibility given by layered media in P2P networks. Results are encouraging, since scalable bitstreams can be easily adapted to tune resulting bitrate of the stream and to match both varying network bandwidth and the spatio/temporal capabilities of the terminals. It is clear that P2P streaming systems supporting SVC technologies will play an important role in future Internet architectures. Nevertheless, the performance can be still improved and there is still enough room for a wide range of research efforts, specifically in the areas of *SVC encoding configuration and optimization*, *peer selection*, *piece picking*, *adaptation strategies*, and *Quality of Service/Experience (QoS/QoE).*

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Harvesting Free Spectrum for Large-Scale Video Multicast Services

A short review for "Scalable video multicast in hybrid 3G/sd-hoc networks"

Edited by Cheng-Hsin Hsu

S. Hua, Y. Guo, Y. Liu, H. Liu, and S. Panwar, "Scalable Video Multicast in Hybrid 3G/Ad-Hoc Networks," IEEE Transactions on Multimedia, vol. 13, no. 2, April 2011, pp. 402~413

Fueled by the increasing popularity of touch-screen smartphones, more and more users consume multimedia content over cellular networks. Nowadays, most video multicast services employ unicast communications in 3G/4G cellular networks, and thus do not scale to a large number of receivers. For example, a field study reports that each High-Speed Downlink Packet Access (HSDPA) cell can only support 4 to 6 mobile receivers at 256 kbps [1], which results in degraded video quality in streaming services due to late packets. Hence, how to support many more video receivers without deploying additional network infrastructure becomes a critical issue to cellular service providers.

Broadcast/multicast extensions of cellular networks [2][3] have been standardized to mitigate the scalability issue. With these extensions, a base station can multicast a single video stream to all its receivers for better spectrum efficiency. However, sending a video stream to all receivers, regardless their location in the cell, may not be ideal because the base station has to apply conservative modulation scheme, high channel coding rate, and high transmission power to guarantee that receivers close to cell boundaries can still decode the video. This in turn slows down receivers closer to the base station and thus results in suboptimal overall video quality.

In this work, the authors design a video multicast system, called SV-BCMCS (Scalable Video-Broadcast and Multicast Services), that leverages on Scalable Video Coding (SVC) [4] to maximize the overall video quality. The main idea is multicasting each layer to the receivers within a certain distance from the base station, and sending more layers to receivers that are closer to the base station. To guarantee basic video quality for all receivers, the base-layer is multicast to the entire cell. Meanwhile, more aggressive modulation schemes and channel coding rates are employed to multicast the enhancement layers, so as to maximize overall streaming rate and video quality.

Multicasting different number layers to receivers based on their distances to the base station, however, is not fair to mobile receivers that are closer to the cell boundaries as they can only get basic video quality. In SV-BCMCS, the authors propose to leverage on a short-range ad-hoc network to boost the video quality of receivers that do not get all the enhancement layers via the cellular network. More precisely, upon realizing that it's outside the full-quality radius to the base station, a receiver seeks help from its neighboring receivers who received more layers than itself, and the helper relays the layers over the ad-hoc network to as many receivers as possible. Relaying enhancement layers over the ad-hoc network helps to reduce the quality variation among receivers and increase the overall video quality. Such a cellular and ad-hoc hybrid network allows service providers to offer large-scale and high-quality video multicast services *without* deploying new network infrastructure or purchasing additional wireless spectrum.

Using ad-hoc networks to enhance the performance of cellular networks has been proposed in the literature. For example, Luo *et al*. [5] design a hybrid network that can route cellular data via other mobile devices with higher cellular data rates using a WiFi ad hoc network. Bhatia *et al*. [6] formulate a problem of finding the relay forest to maximize the overall data rate, and they propose an approximation algorithm to solve it. While the earlier works were designed for bulky data transfer, this paper is the first rigorous study that focuses on delay sensitive live video multicast over a hybrid network.

The SV-BCMCS system consists of three components: (i) a helper discovery algorithm, (ii) an optimal resource allocation algorithm, and (iii) a multi-hop relay routing algorithm. The helper discovery algorithm is a greedy heuristic for a receiver to find the best helper from its neighbors on the ad-hoc network, which is likely to provide the most enhancement layers to the receiver. The optimal resource allocation algorithm is a dynamic programming algorithm that solves an air-time allocation problem among layers. This algorithm leverages on a mapping between modulation schemes (and channel coding rates) and number of receivers. It searches for the best modulation scheme (and channel coding rate) for each enhancement layer, so as to maximize the expected average video quality. Last, the multi-hop relay routing algorithm selects the layers each helper should broadcast to its neighbors over the ad-hoc network. Its design goal is to aggregate multiple unicast transmissions into one multicast transmission for lower ad-hoc network load.

Authors evaluate the SV-BCMCS system via simulations driven by three video sequences with diverse characteristics. The simulations were done using OPNET. Simulation results indicate that the cellular and ad-hoc hybrid network significantly improves the overall video quality compared to a cellular-only network. More precisely, SV-BCMCS increases the overall video quality by up to 1.7 dB when receivers are fixed, and by up to 0.80 dB when receivers are moving at a maximum speed of 5 m/s.

While this paper provide insightful discussion and interesting simulation results about video multicast over a cellular and ad-hoc hybrid network, several research problems remain open in such video multicast systems. For example: (i) handover mechanism among cells is required to provide smooth video streaming services, (ii) an admission algorithm that takes ad-hoc interference into consideration could prevent service interruptions, (iii) a video quality adaptation algorithm that fully utilizes the SVC features will result in higher video quality, (iv) user-incentives or micro-payment mechanisms could lead to wider and faster deployments, and (v) advanced routing algorithms including energy-aware and trajectory-aware routing would lead to better system performance. .I believe this paper is an excellent starting point and will stimulate many more research works along these directions.

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Network Coding for Effective Rate Control in Wyner-Ziv Video Coding

A short review for "Delay-efficient rate control for Wyner-Ziv video coding in wireless video sensor networks using network coding"

Edited by Vladimir Stanković

H. Zhang and H. Ma, "Delay-efficient Rate Control for Wyner-Ziv Video Coding in Wireless Video Sensor Networks using Network Coding," *ICME-2010 IEEE International Conference on Multimedia and Expo,* pp. 243-248, July 2010.

Wyner-Ziv (WZ) video coding or Distributed Video Coding (DVC) [1, 2] is a non-conventional approach to image/video compression where the computationally heavy motion compensation process is shifted from the encoder to the decoder side. Consequently, the key characteristic of DVC is light encoding and possibly heavy decoding, which suits well distributed setups with multiple encoders, such as wireless video surveillance.

Despite tremendous research efforts and significant developments over the past ten years, DVC still substantially lacks behind conventional video coding solutions, performance wise. In the previous issue of R-Letters, it is advocated that one of the main reasons for this is correlation estimation. Yet, another reason for the practical inefficiency of DVC is rate control, which is significantly more difficult than that in conventional video coding. Indeed, the conventional video encoder possesses all the information needed to estimate the rate-distortion performance for a given rate. Thus, the encoder drives the entire coding process and performs rate-distortion optimization to select the best coding parameters. In DVC, on the other hand, the encoder does not access previous frames, used as side information (SI) at the decoder, which imposes a need for a completely different rate control approach.

In DVC, the rate can be changed by varying quantization parameters and by changing the Slepian-Wolf (SW) [3, 4] coding rate. Quantization parameters determine the reconstruction fidelity and should be set based on user's QoS requirements. On the other hand, SW coding rate needs to be determined based on the correlation between the source and SI. By the SW theorem [3], the minimum amount of bits needed be sent to the decoder is equal to the conditional entropy of the source given SI, which is unknown at the encoder. Hence, the encoder can only try to estimate the required rate.

Ideally, the encoder should always send just enough bits for the decoder to recover the bitstream correctly. That is, the encoder should always send the SW compressed data at a rate slightly above the conditional entropy given SI. Since the scene changes dynamically, the amount of needed SW bits for successful decoding can significantly vary from frame to frame. Wrong estimation can lead either to decoding errors, or to a waste of communications resources.

Several encoder rate control mechanisms are proposed in the literature for effectively determining the encoding rate [2, 5, 6]. However, since the entire DVC process is driven by the decoder, it makes sense for the rate control to entirely sit at the decoder side. Indeed, the best performing DVC coders (see [7] and references therein) exploit a feedback channel with rate compatible channel codes for SW coding and decoder rate control.

The decoder cannot calculate the optimal SW coding rate due to absence of the source, but it can successively require more bits from the encoder until the SW decoding succeeds. This intuitive approach proposed in the first DVC implementation [1] ensures the best rate-distortion performance, and hence it remained part of many successive designs. The main advantage of the decoder rate control is improved performance over the encoder rate control solutions, as well as lower encoding complexity, which is the main concern in DVC. However, the decoder rate control has several drawbacks: (1) it requires a feedback channel; (2) rate compatible codes must be used, which are for some classes of channel codes difficult to design; (3) additional coding delay is introduced. This paper addresses the last problem above, i.e., the delay introduced by the decoder rate control in a wireless multi-hop network. In such a setup, with traditional DVC solutions, after each decoder request, additional parity/syndrome bits need to be sent in multiple hops from the source (a DVC encoder) to the destination (a DVC decoder), which can incur huge delays.

The paper assumes that information travels from a source node to a destination node over *N-1* intermediate nodes - relays. The key idea is to allow for relays to cleverly store some of the received punctured bits and send them after receiving decoder requests. That is, partial punctured bits are delivered to relays and the decoder requests these bits from the relays only.
To make this process rate efficient, network coding [8] is used together with rate-compatible low-density parity-check (LDPC) codes [9, 10] which have become very popular in DVC due to their competitive performance and simple rate adaptation.

The rate compatible code of [10] is a syndrome former solution in which each syndrome node is connected to exactly two adjacent check-nodes. Based on this, a distributed SW-network coding scheme is proposed, where each relay forwards some of the received syndrome bits together with modulo-2 sums of the others. In particular, the first m adjacent bits in the generation n are summed modulo-2, and the derived accumulated bit is taken as a new syndrome and forwarded to the next hop, where the generation refers to the number of network coding operations. If the decoder at the destination node fails to correctly perform SW decoding, it will request additional bits from its closest (*N-1*-th) relay, whose additional syndromes will provide a lower rate LDPC code. If the decoder is still unsuccessful, it will request syndromes stored in the *N-2*-th relay (i.e., the one that is two hopes away from the destination), and the process repeats either until the decoding is successful or until the requests reach the source. The simulation results demonstrate reduced delay compared to the traditional decoder rate control DVC as expected.

In summary, rate control is an important factor to consider in practical DVC deployment, and network coding can help reducing the delay in a wireless multi-hop scenario.

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He has co-authored over 110 research papers published in peer-reviewed journals and conference proceedings. He has been awarded or filed five patents in the area of distributed source-channel coding and video communications. He serves as an Associate Editor of the IEEE Commun. Letters. Recently, he has given research tutorials at ICC-2007, Eusipco-2008, and ICASSP-2009, and co-organized special sessions at MobiMedia-2009 and ICME-2010 on distributed video and multimedia over DVB, respectively.

New Trend of Subjective Video Quality Assessment

A short review for "Assessing quality of experience of IPTV and video on demand services in real-life environments"

Edited by Tao Liu

N. Staelens, S. Moens, W. Broeck, I. Marie n, B. Vermeulen, P. Lambert, R. Walle, and P. Demeester, "Assessing Quality of Experience of IPTV and Video on Demand Services in Real-Life Environments," IEEE Transactions on Broadcasting, vol. 56, no. 4, Page 458

Delivering high quality video services, e.g. Internet Protocol Television (IPTV) and Video on Demand (VoD), over existing best effort packet based IP networks can be a real challenge for service providers. Network impairments, such as packet loss, often result in severe degradations of the audiovisual quality as perceived by the end users, commonly referred to as Quality of Experience (QoE). As willingness to pay correlates positively with the offered video quality [1], users' satisfaction and overall acceptability decreases when visual quality drops below the acceptability thresholds. Therefore, in order to ensure end-users receive adequate QoE, service providers are getting more interested in monitoring their network and measuring the audiovisual quality of the different video streams.

Currently, video service providers rely on "golden eye" experts, capable of detecting even the slightest visual impairment, for evaluating the quality of the offered video channels. However, as QoE should be measured from end to end [2], golden eyes cannot guarantee that their perceived quality corresponds with the quality received by the end-users. Therefore, as an alternative method of assessing the actual perceived quality of end users, modeling the actual quality with objective mathematic metrics has received an increasing amount of attention. Prior to the development and validation of such metrics, however, subjective video quality experiments need to be conducted to provide the "ground truth" of how human observers rate the (audio)visual quality of a series of short video sequences.

There already exists a number of internationally standardized subjective video quality assessment methodologies, such as the ones described in ITU-T Recommendation P.910 and P.911 and ITU-R Recommendation BT.500. These methodologies describe in detail how subjective experiments must be conducted and pose stringent requirements on the controlled test environment, such as the viewing distance between the screen and the viewer, the illumination of the test room and the maximum duration of the test video sequences. Before the start of a subjective experiment, test subjects also receive specific instructions on how to evaluate the different sequences. As a result, subjects are primarily focused on (audio)visual quality evaluation.

However, watching television is known to be a typical lean-backward experience, where people watch television content in their living room, together with friends and family. It is clear that this environment is substantially different from the test environment imposed by the standardized subjective test methodologies. People watch entire television programs or movies primarily for their content and usually do not pay that much attention to the optimal viewing conditions as used during subjective quality assessment. As QoE includes the complete end-to-end system effects and can be influenced by user expectations and context, QoE assessment should ideally be performed in the most natural environment where the video services are consumed.

In this article, to investigate the difference between subjective quality assessment experiments using a standardized methodology and in real-life environments, a novel real-life style subjective video quality assessment methodology is first proposed. It is based on full length DVD movies, and hence encourages the subjects to watch the DVD in the same environment and under the same conditions as they normally watch television, without being focused on active visual quality evaluation.

In the first experiment [3], the influence of packet loss impairments and frame freezes on perceived quality is researched. Results show that impairments are significantly less detected during real-life QoE assessment, especially in the case of frame freezes. However, packet loss impairments are rated better quality during real-life compared to frame freezes, which shows that playback fluidity is an important visual aspect for viewers. The second subjective experiment [4] focused on assessing the perceptual influence of the difference between temporal and Signal-to-Noise Ratio (SNR) downscaling of video sequences. This experiment was conducted as part of a wider social-scientific user research concerning video-in-the-home. The results also showed that visual degradations caused by downscaling are more rapidly perceived when using a standardized methodology conducted in a controlled environment and that the detection threshold is content dependent. Furthermore, the test subjects also preferred downscaled versions of the videos which did not impact playback fluidity. Based on the subjective quality ratings, viewers indicated that up to two visual impairments during movie playback are still tolerable.

In general, this research showed that the proposed novel subjective video quality assessment methodology enables real-life QoE assessment, mimicking the typical lean-backward TV viewing experience. Results show that visual impairments are perceived and rated significantly different when using a standardized methodology. Therefore, results obtained using such standardized methodology do not always hold in the case of real-life QoE assessment.

As an example of numerous efforts of seeking improved subjective experiment methodologies that can bridge the gap between audiovisual quality measurement in research and real-world applications, the proposed real-life alike subjective test method in this paper provides a new perspective of thinking about the video quality assessment. And this new research trend will definitely bring a great advance in the field of audiovisual quality assessment in near future.

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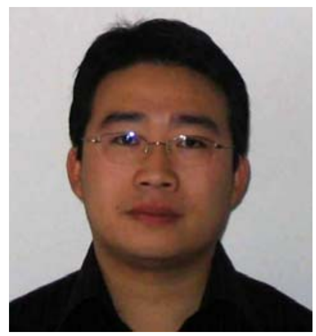

Tao Liu received his B.S. degree in Electrical Engineering from Beijing Institute of Technology in 2004, and received his M.S. and Ph.D. degrees in Electrical Engineering from Polytechnic Institute of New York University in 2008 and 2010, respectively. He has conducted research in the fields of image and video quality assessment, pattern recognition, and video analysis and retrieval for organizations including Technicolor, AT&T, and Dialogic. The content-based video copy detection system he jointed developed at AT&T research lab was evaluated and ranked among the top submissions at 2009 TREC Video Retrieval Evaluation (TRECVID). Currently, he is a research engineer at Dialogic Media Lab, NJ, and is the major contributor to the Dialogic's award-winning mobile video quality monitoring solution. He is an active participant in the Video Quality Experts Group (VQEG) and is a member of the IEEE. He is a member of MMTC and MMTC Review Board. He has been a TPC member or referee for various international conferences and journals in the field of image/video processing. He is a co-chair of 2011 IEEE Workshop on Multimedia Quality of Experience: Modeling, Evaluation, and Directions.

Privacy and Security: Achieving the Best of Both Worlds

A short review for "Privacy–security trade-offs in biometrics security systems—Part I: single use case"

Edited by Simon Pun

L. Lai, S.W. Ho and H.V. Poor, "Privacy–security trade-offs in biometrics security systems—Part I: Single use case," IEEE Transactions on Information Forensics and Security, vol. 6, no. 1, pages 122-139, Mar. 2011.~466, Dec. 2010.

Biometric security systems have been widely deployed in many civilian and military applications. In contrast to conventional password-protected security systems, biometric security systems exempt their users from the arduous challenges of selecting, memorizing and safeguarding long passwords by exploiting the fact that biometric characteristics are unique and do not vary significantly over time.

Typical biometric authentication systems operate in two stages, namely an enrollment stage and a release stage. In the enrollment stage, biometric characteristics X^n such as fingerprints or irises and so on are sampled. Here *n* is the sample size. Then, helper data V and key K are generated from the sampled data. The helper data V will be stored in a database to assist the recovery of the key during the release stage. In addition, a hash to the key *K* will also be stored. Now, we consider the release stage at the application site. The biometric characteristics are sampled again. The newly sampled biometric measurements Y^n and V are then employed to regenerate key $\hat{K}$. If the hash of $\hat{K}$ is the same as the hash of *K*, then the authentication process is successfully completed. Otherwise, the request for access will be denied.

Assuming the worst scenario in which the database V has been compromised by adversaries, most existing works in the literature [1-3] concentrate on the *security* issue of biometric authentication systems, i.e. preventing adversaries from deriving K based on V by maximizing the key size. However, these works have long overlooked the fact that the *privacy* design, i.e. safeguarding the biometric information X^n from being leaked from V is indeed equally, if not more, important than the security design. In contrast to conventional passwords, biometric characteristics cannot be easily modified even if we know the information has been stolen. To the editor's best knowledge, this work stands for the first of its kind to theoretically investigate the joint optimization of privacy and security design. More specifically, it rigorously addresses some of the most important questions in biometric authentication systems: Can we optimize both *security* and *privacy* at the same time? If yes, can the dynamics of security-privacy tradeoffs be rigorously characterized?

To answer these challenging questions, the authors have adopted an information theoretic approach to analyze the maximum achievable security and privacy levels. Unlike the complexity analysis commonly employed in the cryptography literature, the information theoretic approach results in fundamental limits on the maximum achievable security and privacy levels, regardless of adversaries' computational power or strategies.

Mathematically, the security and privacy levels of biometric authentication systems can be maximized by minimizing the mutual information between V and K denoted by $I(K;V)$, and the mutual information between V and X^n denoted by $I(X^n;V)$, respectively. Recall that zero mutual information between two random variables implies that the two random variables are statistically independent. In other words, if $I(K;V)=0$ and $I(X^n;V)=0$, no information about K and X^n can be ever inferred from V regardless of the attackers' strategies and computational power, which means that perfect security and privacy are attained.

For a special case of $X^n=Y^n$, perfect security and privacy can be attained by using the well-known one-time pad scheme. Unfortunately, due to practical constraints such as measurement noise, injuries, aging, etc., X^n and Y^n are usually not identical in practice, which makes the problem analytically challenging. Under such an assumption i.e. $X^n \neq Y^n$, the authors have considered two scenarios, namely attackers with side information (e.g. biometric characteristics stored in other databases and so on) and those without side information.

Through rigorous derivations, the authors have successfully identified the complete privacy–security region for both scenarios conditioned on either perfect privacy $I(X^n;V) \approx 0$ or perfect security $I(K;V) \approx 0$ for the case of attackers without side information. For the case of attackers with side information, inner and outer bounds on the privacy–security region have been derived. By investigating the amount of privacy leakage, the authors have analytically demonstrated that systems with higher security level entail higher privacy leakage of biometric information, which is an inevitable tradeoff between privacy and security. Finally, the authors have shown that biometric security systems with perfect privacy are possible if and only if common randomness can be generated from the two biometric measurements X^n and Y^n.

It is worth mentioning that this paper considers the design of single-use biometric security systems, e.g. single application site. In [4], the authors have extended the results reported in this paper to biometric security systems deployed in multiple locations.

Despite the fact that the results reported in this work remain rather theoretical, these theoretical results are expected to exert a substantial impact on the development of many new applications. In particular, these theoretical results enable us to optimally fine-tune system parameters for any given biometric security systems. At the same time, the results reported in this paper also shed lights on the optimal biometric security system structure to achieve the optimal security and privacy performance.

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Call for Papers: IEEE Communications Magazine

Special Issue on QoE management in emerging multimedia services

The realization of the paradigm of Internet anywhere, anytime and any-device and the diffusion of end-user multimedia devices with powerful and user-friendly capabilities such as smartphones, tablets pc, mobile gaming terminals and ebooks, are leading to the proliferation of a significant amount of emerging multimedia services: immersive environments, mobile online gaming, 3D virtual world, book/newspaper consumption, social networking, IPTV applications, just to cite a few. Some of these services have already reached a major market success, such as the case of newspaper/magazine mobile readers and smartphone multimedia apps. Their success could be achieved especially because a user-centered approach has been followed to design the whole process of content production, service activation, content consumption, service management and updating. Indeed, the quality of the user experience, the perceived simplicity of accessing and interacting with systems and services, and the effective and acceptable hiding of the complexity of underlying technologies are determining factors for success or failure of these novel services, as well as graceful degradation.

The management of the Quality of Experience (QoE) is then undoubtedly a crucial concept in the deployment of future successful services, and it is straightforward to be understood as well complex and stimulating to be implemented in real systems. The complexity is mainly due to the difficulty of its modeling, evaluation, and translation in what for more than a decade we have been mainly dealing with (partially in its substitution), that is the Quality of Services (QoS). Whereas QoS can be now easily measured, monitored and controlled at both the networking and application layers and at the end-system and network sides, the quality of experience is something that is still quite intricate to be managed. The practice in evaluating the QoS can be exploited in evaluating the QoE, but it is just a starting point for a complete QoE management procedure, which should encompass at least the following activities: monitoring of the experience of the user when consuming the service, adapting the provisioning of the content on the basis of the varying context conditions (e.g. network status, user behavior, user profile, environment), predicting potential experience level degradation, and masking quality degradation due to abrupt system changes. To have a complete control of the final user experience, all these tasks need to be performed in a coordinated way and their real effectiveness depends on the validity of the adopted user perception model.

Objectives

The purpose of this special issue is to present to the magazine's audience a concise, tutorial oriented reference of the state-of-the-art, current and future research challenges and trends on the management of QoE in emerging multimedia services. To achieve this goal the special issue seeks original research and review papers that survey and present new ideas, leading-edge research prototype development, trials and early deployment, and performance evaluations in the following areas:

- Definition of QoE (Quality of Experience) for emerging services
- Relationship between QoE and QoS
- Architectures for the management of the QoE in emerging multimedia services
- Offline and online prediction and evaluation of QoE
- QoE-oriented multimedia traffic management
- QoE-oriented multimedia source and channel coding
- Testbeds for performance evaluation of QoE-oriented systems
- Middleware solutions for QoE management
- Adaptive and self-configuring solutions for QoE management
- Advanced, scalable service-aware QoE-oriented traffic control and management
- QoE management in heterogeneous networks

Prospective authors should follow the IEEE Communications Magazine manuscript format described in the Authors Guidelines (http://dl.comsoc.org/livepubs/ci1/info/sub_guidelines.html). All articles to be considered for publication must be submitted through the IEEE Manuscript Central (http://commag-ieee.manuscriptcentral.com), according to the following timetable:

Submission Deadline:	**October 20, 2011**
Notification of Acceptance:	January 15, 2012
Final Manuscript Due:	March 10, 2012
Publication Date:	April 2012

Guest Editors:

- Luigi Atzori, Dept. of Electrical and Electronic Engineering, University of Cagliari, Italy
- Chang Wen Chen, Dept. of Computer Science and Engineering, University at Buffalo, NJ, USA
- Tasos Dagiuklas, Technological Educational Institute of Mesolonghi, Greece
- Hong Ren Wu, Royal Melbourne Institute of Technology, Australia

Call for Papers: IEEE Journal of Selected Topics in Signal Processing

Special Issue on Emerging Techniques in 3D: 3D Data Fusion, Motion Tracking in Multi-View Video, 3DTV Archives and 3D Content Protection

As a result of increasing consumer demand for 3D content, content creation associated with this new modality has increased significantly in conjunction with some recent standardization activities on this data type. Hence, a scientific revisit is required particularly to some challenging problems associated with the conventional video, considering the fact that multi-view video has many promising solutions to such problems. Moreover, any 3D representation also produces its own requirements to be dealt with. This special issue is an effort to compile and review the current advances in 3D multimedia and multimodal information analysis and processing. Particularly, it deals with emerging 3D techniques for 3D data integration and object analysis based on multi-views, as well as 3D content protection.

We would like to invite authors to submit their recent and original research results as well as experience reports. In the following, a non-exclusive list of related topics is suggested:

- Creation of 3D Content
  - 3D multi-view and multimodal data fusion
  - Calibration methods for 3D multi-camera system
- Tracking, Registering and Processing of 3D Content
  - 3D multi-view image/video processing
  - 3D registration of multi-view data
  - Motion tracking in stereo and multi-camera systems
- Archiving of 3D Content
  - Creation, compression in 3D digital archives
  - Indexing and retrieval of 3D content
- Security Issues for 3D Content
  - Methods for 3D content protection
  - 3D digital watermarking, fingerprinting and related security solutions
- 3D Objective Quality Measures
  - Artifacts Characterization
  - Full-reference and partial-reference measures
- Multimedia systems and applications using emerging techniques in 3D

Prospective authors should visit http://www.signalprocessingsociety.org/publications/periodicals/jstsp/ for information on paper submission. Manuscripts should be submitted using the Manuscript Central system at http://mc.manuscriptcentral.com/jstsp-ieee. Manuscripts will be peer reviewed according to the standard IEEE process.

Manuscript submission due:	**July 10, 2011**
First review completed:	October 20, 2011
Revised manuscript due:	November 20, 2011
Second review completed:	February 1, 2012
Final manuscript due:	March 1, 2012

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Call for Papers: IEEE Journal on Selected Areas in Communications

Special Issue on QoE-Aware Wireless Multimedia Systems

With the evolution towards new multimedia systems and services, user requirements are not limited to requirements on connectivity: users now expect services to be delivered according to their demands on quality. At the same time, audiovisual systems are becoming more and more complex and new possibilities of presenting content are available, including augmented reality and immersive environments. However, for wireless systems the possible limitations due to the characteristics of the transmission channel and of the devices can result in perceivable impairments, originated in the different steps of the value chain from content production to display techniques, that influence the user's perception of quality. In recent years, the concept of *quality of service* (QoS) has been extended to the new concept of *quality of experience* (QoE), as the first only focuses on the network performance (e.g. packet loss, delay and jitter) without a direct link to the perceived quality, whereas the QoE reflects the overall experience of the consumer accessing and using the provided service. Experience is user- and context-dependent (involving considerations about subjective multimedia quality and users' expectation based on the cost they paid for the service, on their location, on the type of service, on the convenience of using the service, etc.). Subjective QoE evaluation is however time consuming, costly and not suitable for use in closed loop adaptation, hence there is a growing demand for objective QoE evaluation and control: objective, rather than subjective, QoE evaluation enables user centric design of novel multimedia systems, including wireless systems based on recent standards, such as WiMAX and 3GPP LTE, through an optimal use of the available resources based on such objective utility index.

This special issue invites submissions on the latest research on QoE-aware wireless multimedia systems, including relevant applications in new areas. We particularly welcome papers reporting original research on QoE-aware systems exploiting and analyzing QoE information at the different layers of the communication protocol stack, addressing multiple and new media sources (audio, images, 2D/3D/multiview video,…), and proposing QoE-scalable transmission approaches. We seek original completed and unpublished work not currently under review by any other journal/magazine. Topics of interest include (but are not limited to):

(i) Architectures and Protocols for QoE-driven media streaming
- Wireless architectures for QoE-driven media streaming
- Protocols and mechanisms for wireless media streaming
- In-network media stream management and processing
- Cross-layer signalling for QoE-aware wireless communications
- QoE-aware security and rights management in wireless media streaming

(ii) System optimization based on QoE criteria
- QoE-aware error control
- QoE-aware cross-layer design
- QoE-aware MAC layer strategies
- QoE-driven adaptation and control mechanisms for existing and next generation wireless systems/devices
- Media synchronization, playback, and buffer management
- QoS to QoE mapping

(iii) QoE assessment and monitoring methodologies
- Objective QoE metrics for wireless image, graphics, video, animation and audiovisual transmission
- No reference or reduced reference QoE models for mobile multimedia applications/services
- Online and offline QoE monitoring schemes and QoE measurement from live mobile/wireless networks
- QoE metrics for 3D Video streaming and multimodal applications

- QoE metrics and system design for novel applications (e.g., augmented reality, cloud-based online gaming).

Prospective authors should prepare their submissions in accordance with the rules specified in the 'Information for Authors' section of the JSAC guidelines (http://www.jsac.ucsd.edu/Guidelines/info.html). Papers should be submitted through EDAS (http://www.edas.info). Prior to submitting their papers for review, authors should make sure that they understand and agree to adhere to the over-length page charge policy presented in the JSAC guidelines.

Manuscript submission deadline:	**August 7, 2011**
Second review complete/acceptance:	March 10, 2012
Preliminary review results:	December 15, 2011
Expected publication date:	3rd Quarter 2012

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Call for Papers: IEEE Communication Surveys and Tutorials

Special Issue on Energy-Efficient Multimedia Communication

Background

Multimedia has gained immense popularity in a variety of applications related to education, entertainment, business, and location-based services. Recent advances in networking and display technologies have enabled the dissemination of multimedia to a variety of devices, from cellular telephones to tablet PCs to wall-size screens. The proliferation of media hosting services and social networks have allowed users to easily share multimedia content with a much wider audience. Digital cameras and camcorders have replaced films and tapes, making it simpler to generate multimedia. Users can easily view, process, analyze, publish, retrieve, or modify multimedia on these devices. However, energy consumption is still a major challenge in the dissemination of multimedia. Energy is consumed during various stages - processing, communication, and storage - of multimedia. In addition, data centers where media services are hosted have also seen a rapid increase in energy consumption in recent times. This trend is not sustainable. Significant progress must be made to save energy and slow down the rate of energy consumption in all these stages. This special issue aims to provide researchers and professionals in the communication, networking, multimedia, and computing communities with insightful papers that present an overview of new approaches to making multimedia communications more energy-efficient.

Scope

Topics of interest include (but are not limited to):

- Energy-efficient network/communication protocols for multimedia data transmission
- Energy-efficient multimedia communication architectures
- Low-power hardware, software, or both for different stages of multimedia processing, such as acquisition, coding, compression, storage, transmission, and reception
- Energy-efficient techniques for content analysis, indexing, searching, and retrieval in resource-constrained (such as mobile and embedded) systems
- System-level energy-efficient design and implementation for multimedia communication
- Energy conservation for multimedia on mobile devices
- Tools for measuring and analyzing energy consumed during multimedia communication

Manuscript Submission

IEEE Communications Surveys & Tutorials is a ComSoc publication. It is an ideal venue for researchers and other communications professionals to publish tutorials and surveys reachable to a large global audience. Articles should be written in a style comprehensible and appealing to readers outside the specialty of the article. Authors are encouraged to visit the "Call for Papers" and "Information for Authors" pages at the IEEE Surveys and Tutorials web site at http://dl.comsoc.org/surveys/. Please submit manuscripts via the ManuscriptCentral website at http://mc.manuscriptcentral.com/comst-ieee (the entry name: Special Issue: Energy Efficient Multimedia Communication)

Important dates

Manuscript due:	**September 30, 2011**
Acceptance Notification:	January 31, 2012
Publication date:	Q2, 2012

Guest Editors:

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A Full Day Workshop on Enabling Green Wireless Multimedia Communications

IEEE MMTC´s Inaugural International Annual Workshop
to be held at IEEE Globecom, Houston, Texas, USA
http://www.ieee-mmcom.org

TPC Chairs:

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Workshop Dates: *December 5 or 9, 2011*

Seamless access from multiple devices to an open set of multimedia communication and information services is driving the accelerating demand for open, interconnected, secure, and energy efficient multimedia communication infrastructures, devices, and services, and raising many technical challenges, ranging from legacy migration, interworking, security, quality of service, charging, energy efficiency, network neutrality and business models.

We are pleased to announce the inaugural IEEE Multimedia Communications Workshop (MMCOM'11) to be held as a full-day workshop at the 2011 IEEE Globecom. It will provide a premium forum for discussions among experts from the academia and industry about hot topics of research and development in the emerging field of next generation "green" wireless multimedia communications, characterized by its efficiency, seamlessness, accessibility, quality of service and security. Areas of interests include, but are not limited to, next generation network infrastructures, multimedia communication over hostile networks, video coding and communications, wireless communications systems, and green computing and communications systems. The Workshop will consist of two keynote presentations by world renowned experts from the academia and industry respectively, as well as presentations of technical papers in morning and afternoon sessions. Interactive discussions will be facilitated via the open and interactive environment at the workshop.

We cordially invite you to contribute to and participate in this first in what is poised to become an important series of workshops in these research and development areas of increasing importance.

Important Dates:

- Paper Submission:	**July 7, 2011**
- Acceptance Notification:	August 15, 2011
- Camera-Ready Paper:	August 31, 2011

The official call for paper can be found **www.ieee-mmcom.org**

Paper Nomination Policy

IEEE MMTC R-letter welcomes review paper nomination. Any paper published in an IEEE ComSoc journal/magazine or in the MMTC sponsored proceedings: IEEE GLOBECOM, ICC and ICME, in the two years preceding the next award board's election, is eligible.

The paper nomination is always open. Paper nominations have to be sent to the IEEE MMTC Review Board Director by email. The nomination should include the complete reference of the paper, author information, a brief supporting statement (maximum one page), the nominator information, and an electronic copy of the paper when possible. Only papers published in the two years preceding the nomination will be considered.

Each nominated paper will be reviewed by two members of the IEEE MMTC Review Board, according to the area of expertise, and avoiding any potential conflict of interest. The reviewer names will be kept confidential. If both members agree that the paper is of award quality, they will recommend publishing the review of the paper (partially based on the nomination supporting document) in the IEEE MMTC Review Letter.

For more details, please refer to http://committees.comsoc.org/mmc/awards.asp