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IEEE COMSOC MMTC R-Letter

Message from R-Letter Co-Director

Welcome to the sixth issue of IEEE MMTC Review-Letter (R-Letter) in this year. The objective of the R-Letter is to introduce cutting-edge and promising new concepts and ideas in multimedia communication to MMTC members by rigorously selecting and reviewing high-impact and innovative papers from recent IEEE Communication Society and MMTC sponsored publications as well as other IEEE publications.

Building on the success of prior issues, we are pleased to introduce seven high quality papers in this issue. The first paper, from the *IEEE International Conference on Multimedia and Expo*, presents a unified framework for energy-efficient wireless multimedia communication, based on Markov Decision Processes and Reinforcement Learning. The second paper, published in the *IEEE Transactions on Multimedia*, develops optimization techniques that address important issues of in-network rate allocation and packet scheduling. The third paper, from the *IEEE Transactions on Circuits and Systems for Video Technology*, proposes a cross-layer content-aware retry limit adaptation scheme, which adaptively chooses the retry limit for each packet in wireless local area networks, based on its loss impact. The fourth paper, published in the *IEEE Transactions on Multimedia*, proposes a new image quality metric that can be used for efficient perceptual image coding. The fifth paper, published in the *IEEE Transactions on Multimedia*, proposes a new subjective quality assessment methodology for scalable videos using paired comparisons. The

sixth paper, from the *IEEE Network Magazine*, presents a Web-based platform for facilitating quality-of-experience assessments in network and multimedia studies, using paired comparisons. Finally, the seventh paper, published in the *IEEE International Conference on Communications*, proposes a probabilistic approach for detecting blocking attacks in RFID systems.

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: Carl James Debono, Christian Timmerer, Ai-Chun Pang, Tao Liu, Cheng-Hsin Hsu, Jong-Seok Lee, and Walid Saad. I also would like to thank the R-Letter Director Guan-Ming Su for all his efforts.



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An Energy-Efficient Solution for Wireless Multimedia Communications

A short review for "Reinforcement learning for power management in wireless multimedia communications"

Edited by Carl James Debono

N. Mastrorarde, and M. van der Schaar, "Reinforcement Learning for Power Management in Wireless Multimedia Communications", in Proc. of IEEE International Conference on Multimedia and Expo, July 2011.

Transmission of multimedia content over wireless channels presents a significant burden on the limited power supply of mobile devices. Furthermore, these systems operate in time-varying channel conditions that demand more energy per bit under low Signal-to-Noise Ratios (SNR) to guarantee data transmission within acceptable delays. This variable bit-rate will also translate into a dynamic traffic load scenario within the system. These dynamic issues make end-to-end multimedia energy-efficient wireless transmission a non-trivial problem requiring robust solutions that need to operate efficiently in unknown traffic and channel conditions whilst still maintaining end-to-end delay constraints.

In literature, energy-efficient solutions for wireless communications can be placed in two categories: physical (PHY) layer-centric solutions, which include power-control and Adaptive Modulation and Coding (AMC); and system-centric techniques, such as turning network cards to low-power states when not needed [1][2][3] in Dynamic Power Management (DPM) systems. Although these techniques differ significantly, they allow the system to find a tradeoff between delay and energy and increase the lifetime of mobile devices [1][4][5].

A number of PHY-centric schemes focus on optimal single-user power-control to minimize the transmission power subject to queuing delay constraints (e.g.[4]). However, most solutions require statistical knowledge of the underlying channel state and traffic distributions, data which is not available in practice. Without this data, only sub-optimal heuristic solutions can be found, thus the optimal power consumption whilst satisfying stringent delay constraints required by wireless multimedia applications cannot be obtained. Other PHY-centric solutions use adaptive modulation, adaptive coding, or AMC [6]. However, these solutions are generally used to tradeoff error-robustness and throughput in fading channels. Yet, they can also be exploited to tradeoff delay and energy [5].

While PHY-centric solutions are effective in minimizing transmission power, keeping the wireless transceivers on and ready to transmit consumes a lot of power. This results in a significant amount of wasted power even when there are no packets being transmitted. System-level solutions address this problem. These techniques rely on DPM to save power. However, existing online solutions to the DPM problem do not exploit the structure of the network, and therefore exhibit sub-optimal learning performance.

In their solution, the authors present a rigorous and unified framework which is based on Markov Decision Processes (MDPs) and Reinforcement Learning (RL). They combine the PHY-centric and system-level techniques in an attempt to minimize the energy consumption, under delay constraints. Thus, power-control, AMC, and DPM policies are adapted on currently known and predicted traffic and channel conditions.

The technique applies (offline) value iteration and (online) RL algorithms by splitting the system's dynamics into a priori known and predictable components. This is done by generalizing the concept of a *Post-Decision State* (PDS, or *afterstate* [7]), which is an intermediate state that occurs after known dynamics occur but before unknown dynamics take place. Similar decompositions were used for delay-constrained scheduling over fading channels (i.e. power-control) in [8][9]. However, the authors provide a more general formulation of the PDS concept than literature. Specifically, previous literature defined the PDS concept for very specific problems, where the PDS is a deterministic function of the current state and action, and where the cost function is known. In general, however, the PDS can be non-deterministic and can be used in any MDP in which the transition probability and cost functions have known and unknown components. The advantage of this PDS algorithm is that it exploits partial system

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information, requiring less training data than conventional RL algorithms. Furthermore, under certain conditions, it obviates the need for action exploration, which severely limits the adaptation speed and run-time performance of conventional RL algorithms.

The algorithm takes advantage of the fact that unknown dynamics are independent of certain components of the system's state (i.e. the buffer and power management state). Thus a batch update (referred to as virtual experience learning) can be done on multiple PDSs in each time slot. The presented experimental results show that the virtual experience improves learning speed over conventional PDS learning.

The framework proposed in the paper can be applied to many different network and system resource management problems that use controlled buffers. These include, but are not limited to, multi-user uplink and downlink transmission; delay-sensitive data transmission over multi-hop wired/wireless networks; DPM for system components such as processors and hard-disks; and job scheduling and dynamic-voltage-scaling for energy-efficient processing on multi-core processors and in data centers. Furthermore, traffic-sampling techniques and history can be exploited to provide better predictions and reduce state switching times further reducing the power consumption.

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In-Network Media Processing Enabling Content-Aware Networking

A short review for "In-network packet scheduling and rate allocation: a content delivery perspective"

Edited by Christian Timmerer

J. Chakareski, "In-Network Packet Scheduling and Rate Allocation: A Content Delivery Perspective," IEEE Transactions on Multimedia, vol.13, no.5, pp. 1092-1102, October 2011.

Today's computer networks are increasing in scalability and, thus, performance improvements are sought by taking advantage of local information along the network delivery path, e.g., like in Content Delivery Networks (CDN) and commercial overlay networks. In many scenarios, both the sender and the receiver are not fully aware of the context through which their packets traverse the network. In particular, performance can be improved by performing operations such as packet scheduling and rate allocation on-the-go, i.e., while the data is being sent over intermediate hops and, specifically, where packets are characterized with delivery deadlines and varying importance for the reconstruction of the overall stream.

This paper investigates the in-network rate allocation and packet scheduling, as an evolution of studies previously conducted in this area such as [1][2][3]. The scenario in question comprises multiple video streams sharing a common backbone network while being delivered to distinct clients through an intermediate proxy located at the junction of the backbone and the last hop links towards the clients. Such a setup is very typical today, e.g., in a CDN, where a backbone origin server supplies multiple clients through a common edge server (the proxy in our case). Therefore, it is important to properly allocate the network resources of the backbone and the last hops in order to maximize the efficiency of the end-to-end delivery while taking into account the video quality requirements for the media sessions of the clients.

In the setup, the proxy requests the streams from the server on behalf of the clients based on the optimal rate at which each of these streams should be sent on the backbone. These rates are provided by an optimization process that takes into account the backbones' data rate capacity, the last hop links' capacities, and corresponding packet loss and delay thereof. Additionally, the optimization considers the existence of minimum video quality requirements that the clients may have for their respective sessions. The proxy

applies an optimal trade-off of the distortion-rate characteristics of the streams using Lagrange multipliers in order to maximize the overall video quality for the given network resources while respecting the clients' video quality requirements. Finally, the proxy also caches packets before they are forwarded on the last hops and executes retransmissions due to packet loss based on the deployed packet scheduling policy.

The paper also studies whether performing hybrid receiver/sender-driven packet scheduling at more than one intermediate node in the network path provides additional benefits. In this case there are possibly $n-1$ active nodes between the sender and the receiver (with n denoting the number of links) at which scheduling of the packet transmissions can be performed locally. Each of the network links in the path is characterized by a random packet loss and delay profile, in each direction (forward and backward). Therefore, an optimization framework is designed that provides the active nodes with packet scheduling performed in a distributed, yet coordinated, fashion such that the end-to-end performance of the data delivery process is maximized.

One of the main contributions of this paper is the formulation of the quality-rate performance gains due to multi-agent scheduling as a function of the placement and number of active nodes along the network path. Furthermore, the performance of the multi-agent scheduling system is analytically modeled and a close correspondence with simulation-generated data is established.

Potential future work may include the integration of the proposed framework into (large-scale) real-world test beds (e.g., taking [4] as a starting point) and the evaluation thereof with respect to performance, scalability, and quality of service as perceived by the end user.

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Error Protection for Video Transport over WLANs

A short review for "Cross-layer packet retry limit adaptation for video transport over wireless LANs"

Edited by Ai-Chun Pang

C.-M. Chen, C.-W. Lin, and Y.-C. Chen, "Cross-Layer Packet Retry Limit Adaptation for Video Transport over Wireless LANs," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 20, no. 11, pp. 1448-1461, November 2010.

To overcome the packet loss problem encountered at the receiver side, video transport over wireless networks may deploy retransmissions for the lost packets. By doing so, the data to arrive at the receiver will experience the increased delay time. However, a video packet received later than its playback time will disturb the playback smoothness at the client side. Therefore, delay constraint often poses an important requirement in real-time applications and packet scheduling becomes an important mean of retransmission-based error control for wireless video streaming. A strategy based on the Early-Deadline-First (EDF) principle [1][2] gives priority to the lost packets with earliest play-out deadlines in transmission. However, the penalty of easy implementation may override regular packets by the retransmitted packets. The retransmission methodology can be enhanced by jointly addressing the wireless channel conditions, packet deadlines, and application layer importance of packets [3].

It has been shown that MAC-layer adaptive retry limit adaptation serves a promising scheme to achieve unequal error protection for video packets. This paper proposes a cross-layer Content-Aware Retry Limit Adaptation (CA-RLA) scheme that adaptively chooses the retry limit for each packet with consideration of its loss impact over WLANs. The proposed CA-RLA scheme involves two steps: off-line encoding based error estimation and on-line adaptation. In the off-line encoding stage, the encoder estimates the amount of expected error propagation when a packet transmitted through a lossy channel. The estimated error will be stored as the side information (in terms of metadata) in the streaming server. During the on-line streaming phase, the server will reference both the off-line side information associated with a video bitstream and the on-line estimated channel condition at the client side to derive the optimal retry limit of each packet. To minimize the overall distortion via unequal error protection, larger retry numbers are allocated to packets of higher loss impact. Besides, the backoff time for

each retry is analyzed to accurately estimate the transmission delay for retransmission scheduling.

To estimate the amount of error propagation caused by a packet loss, the proposed scheme characterizes the pixel-level loss-impact (LI) metric as the product of pixel reference count (PRC) and pixel-wise concealment error (PCE), where PCE denotes the norm of pixel-wise concealment error, and PRC represents the frequency of a pixel being referenced by pixels in the following frames within a GOP in the motion-compensated prediction. A macroblock-level loss-impact is calculated by summing up the LIs of pixels in the macroblock. As a result, all macroblock-level LIs in one packet are then summed up to estimate the packet-level error-propagation). This error estimation scheme is simple yet effective, although the proposed loss impact is not a strictly linear function of the MSE value.

To estimate backoff time for each retry in an IEEE 802.11 WLAN, the Markov chain model proposed in [4] is adopted that is an extension of the throughput analysis presented in [5]. In this paper, a closed-form formula is derived to accurately estimate the mean backoff time of a packet retry so as to obtain the mean transmission time of a packet for a given retry limit, which is an important parameter in delay-constrained video streaming applications. Under the constraint that the packets should arrive at the receiver before the presentation deadline, CA-RLA tends to reduce the retry limits of low-loss-impact packets to save time budget to increase the retry limits of high-loss-impact packets to achieve loss-impact-based unequal error protection of packets. A cross-layer approach is proposed to achieve optimal retry limit adaptation.

Extensive simulations were conducted using the OPNET v.8.3 network simulator to simulate an 11 Mbps 802.11b network. The simulation results show that the proposed model achieves

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good accuracy in back-off time estimation, which provides a good base to estimate the mean transmission time of a packet in the delay-constrained scheduling optimization problem. With to the accurate back-off time estimation and packet scheduling based timeout estimation, the proposed CA-RLA scheme can have better packet loss and visual quality performance when compared to the traditional static-retry-limit mechanism and the state-of-the-art time-based retry adaptation method.

In summary, this paper offers an insightful systematic analysis to derive for IEEE 802.11-like networks a closed-form formula to accurately estimate the mean transmission time of a packet for a given retry limit, which is an important parameter in delay-constrained video streaming applications. Besides, it addresses the retry limit adaptation problem under a constrained optimization framework by appropriately incorporating cross-layer parameters including the loss impact (application-layer) and estimated transmission time (MAC-layer) of each packet. Overall, this paper provides valuable analytic tools for researchers in designing a retransmission-based video streaming system.

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Towards Efficient Perceptual Image Encoding

A short review for "Practical image quality metric applied to image coding"

Edited by Tao Liu

F. Zhang, L. Ma, S. Li, and K. Ngan, "Practical Image Quality Metric Applied to Image Coding", *IEEE Transactions on Multimedia*, vol. 13, no. 4, pp. 615-624, August 2011.

An important goal of multimedia communication is to minimize the redundancies of the content to be delivered such that the transmission bandwidth can be minimized. The expense of this saving in bandwidth is the certain amount of quality degradation (or content distortion). The relationship between the encoding rate and distortion of the image data has been extensively studied by numerous researchers and international standardization bodies of digital image and video coding. Most of these existing studies and standardized image/video codecs use some traditional fidelity measures, e.g. Mean Squared Error (MSE), instead of actual human quality judgment, as tools to automatically assess the quality of the encoded image/video and to optimize the encoder. However, the recent tremendous progress of image and video quality modeling facilitates researchers to evaluate the encoded images and videos with more advanced and accurate quality metrics, which at the same time can benchmark the performance of the underlying codecs and ultimately helps to enhance the image and video compression and communication technologies.

Most of the works toward this goal choose the approach of applying standalone quality metrics to the target encoded contents; while the others address the quality measurement and perceptual optimization within the encoding loop. The former approach can be easily deployed in broad image/video QoE *measurement* or *monitoring* applications; the latter approach has more focus on quality *enhancement*, although it might be constrained to measuring the quality of encoded image/video by specific codecs. This paper is one of the few works addressing the image quality measurement and enhancement issues in the latter scenario.

In the perceptual encoding process, the concern of the trade-off between accuracy of the quality metric and the additional complexity it introduces needs to be carefully addressed.

Accurate quality metrics tend to require considering multiple quality-affecting features, while over complex models may cause too much additional computation burden of the encoder and they usually may not be easily integrated into the encoder's optimization process. This paper represents an exemplary solution to exploit the impact of perceptual (or psychovisual) redundancies on image quality during encoding. The authors first proposed a simple yet effective full-reference image quality metric, which incorporates two human visual system (HVS) properties. And then they demonstrated one of its use cases in perceptual coding by embedding it into JPEG image encoder.

The proposed image quality metric considers only two factors: the texture masking effect and contrast sensitivity function (CSF) in sub-band and local regions (an even simpler yet effective metric is designed in authors' previous work [1]), and then combine them with different weights. The proposed objective metric is validated on a total of eight publicly available subjectively-rated image databases and is proved to be able to accurately measure the typical image impairments in real-world applications. The performance gain was analysis for each factor. The authors kept the model simple and introduced a parsimony set of parameters to avoid over-fitting. The perceptual image coding scheme is also confirmed to perform well in the subjective tests performed by the authors.

With respect to the integration of the proposed quality metric into image encoding, the proposed image codec adopts adaptive quantization technique. The quantization step is selected to minimize the rate-distortion objective function, where the distortion is measured by the proposed metric. In order to avoid extra overhead, the quantization steps are not transmitted with the bit stream, but estimated at the decoder. The quantization residual might be recovered according to the prior knowledge that natural images have smooth gradients. Experiments confirm that the proposed codec achieves good

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tradeoff between the rate and perceptual distortion.

An interesting conclusion is that the typical impairments in common practices seem not that diverse, so a simple measurement might be more effective than expected. Another finding is that the learned CSF in DCT domain conforms to the common sense that human eyes are more sensitive to the low spatial-frequency sub-bands than the high frequency ones, but its shape is much steeper than other well-known CSFs, for example Watson's mask. The result happens to be similar with the CSF in Karhunen-Loève domain which was designed but not "learned" in the authors' another paper [2].

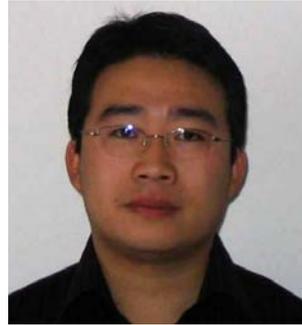
In summary, this paper investigates the relationships between perceptual distortion, bitrate and quantization during the process of image encoding. The proposed metric could be easily optimized, and applied to broader range of image communication applications.

The related source code is available at: <http://ivp.ee.cuhk.edu.hk/projects/demo/piqm/index.html>

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quality metric," IEEE Trans. Image Processing, vol. 20, no. 11, pp. 3207 – 3218, 2011.



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A New Approach to Quantifying the Benefits of Scalable Video Coding

A short review for "Subjective quality evaluation via paired comparison: application to scalable video coding"

Edited by Cheng-Hsin Hsu

J.-S. Lee, F. De Simone, and T. Ebrahimi, "Subjective Quality Evaluation via Paired Comparison: Application to Scalable Video Coding," *IEEE Transactions on Multimedia*, vol.13, no.5, pp.882-893, Oct. 2011.

Scalable Video Coding (SVC) enables multimedia systems to transmit a single coded stream to many receivers, while each receiver selectively decodes a part of the coded stream based on its capability, such as screen size, computational power, network bandwidth, and battery level. SVC also allows multimedia servers to adapt a coded stream against dynamic network conditions in real time without time-consuming transcoding. Modern SVC standards support three types of scalability: temporal scalability for various frame rates, spatial scalability for several resolutions, and quality scalability for multiple frame fidelities. While AVC-based SVC solution [1] was published as an amendment of the H.264/AVC standard a few years back, we haven't seen large-scale deployments of SVC-enabled multimedia systems. This is probably because common signal-based quality metrics, such as Peak-Signal-to-Noise Ratio (PSNR), are less applicable to scalable videos, and hence the benefits of scalable video coding are often underestimated.

The authors of this paper propose a new subjective quality assessment methodology for scalable videos using *paired comparisons*. In it, user subjects do not directly give scores to a stimulus; rather they only provide preference between each pair of stimuli. In contrast to single stimulus and double stimuli methods, paired comparison is simpler and more reliable when multiple modalities of scalability vary, such as frame rate, spatial resolution, and picture fidelities. This is because providing preference between two videos with diverse temporal, spatial, and fidelity scalability combinations is much easier than assigning scores to them.

The outcomes of pair comparisons are *winning frequency* matrixes between every pair of stimulus. The authors propose a systematic approach, called Paired Evaluation via Analysis of Reliability (PEAR), to convert the matrixes into quality scores of stimulus. More importantly,

the proposed approach also quantifies the confidence intervals, which are valuable for researchers to judge the reliability of the resulting quality scores in an intuitive way. The authors then use the PEAR approach to analyze the quality scores of various scalability combinations extracted from scalable video streams. The experimental results from subjective tests provide guidelines for a general stream adaptation strategy of scalable video coding under a given target bit rate constrained by the network and computing resources.

The proposed paired comparison methodology is fairly flexible. It allows the subjects to provide either categorical or continuous preferences. Examples of the categorical preferences include: {'better', 'worse'}, {'better', 'tie', 'worse'}, and {'much better', 'better', 'slight better', 'tie', 'slightly worse', 'worse', 'much worse'}; examples of continuous preferences include scores in [-100, 100]. While at first glance, there may be a huge number of paired comparisons to perform when the number of total stimulus is large. This is however not a big concern for stream adaptation applications, because for any given available bandwidth, there are only a few potential scalable combinations. It should be noted that this is partially because the authors only consider Coarse Grained Scalability (CGS), rather than Medium Grained Scalability (MGS) coded streams.

The PEAR approach uses the frequency of ties between compared pairs to compute the confidence intervals. This is based on an assumption that ties contain information of uncertainty between pairs of stimulus. In other words, more ties imply that the confidence intervals are likely to be larger, and the authors propose to use a parameter β between 0 and 1 to capture the portion of ties corresponds to uncertainty. For any given β , the confidence intervals are derived, which can be used to identify the occasion when the quality difference is not significant enough for a reliable

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dominance stimulus. The authors also present how to use ties to verify whether a subject can make reliable judgments by counting the number of circular triads among subjects' preferences, which can be used to filter out outliers.

The PEAR approach is used to conduct subjective tests on three high-definition (HD) video sequences coded by two scalable video coders: H.264/SVC [1] and Wavelet-Based Scalable Video Coding (WSVC) [2]. The three video sequences are in a maximum resolution of 1280 x 720 with a maximum frame rate 50 fps. The tests were conducted in a lab environment with a 30-inch HD LCD monitor. PEAR is compared against Single Stimulus Continuous Quality Scale (SSCQS) approach, in which a subject watches a stimulus and votes between 0 and 100 in 5 seconds. The score ranges are associated with adjective descriptions such as 'excellent', 'poor', and 'bad'. In PEAR, each pair of stimuli is presented to the subject side-by-side on the LCD monitor. There were 16 subjects (11 males and 5 females) in the subjective tests.

Several observations were drawn in the subjective tests. First, SSCQS and PEAR result in consistent preferences in almost all pairs of stimulus. The main difference between them is that PEAR has higher *discriminability* compared to SSCQS. Higher discriminability is crucial for scalable streams, as a diverse set of scalability combinations can be extracted from each coded stream, and thus PEAR is more suitable for scalable streams. Second, the pixel bit rate does not have direct relationship with the quality scores in both scalable coders. This reveals that a simple method of incrementally increasing the pixel bit rate for stream adaptation may not work efficiently, and demonstrates the importance of subjective tests.

Last, the authors provide a few guidelines for scalable stream adaptation. First, when the bandwidth is limited, a scalability combination of a larger resolution with acceptable fidelity level (i.e., no strong blurring effects) is preferred over another combination with high fidelity and low resolution. Second, when the bandwidth is high enough, most scalability combinations would have reasonable resolution and fidelity, and thus high frame rates are preferred. Third, both video sequence and coder characteristics have direct impacts on subject preferences and thus quality scores.

I found that the high discriminability of PEAR is a unique feature for conducting effective subjective tests on various scalability combinations, and may better demonstrate the actual benefits of employing scalable streams in multimedia systems. I believe this paper could stimulate more studies along the direction and eventually result in an objective quality metric suitable for scalable video streams.

Several extensions can be made in different directions. First, the subjective tests presented in this paper consider reasonable numbers of video sequences and subjects. But more conclusive guidelines may require even more video sequences and subjects. Nevertheless, designing economic subjective tests is fairly challenging [3]. Second, the subjective tests presented in this paper do not consider MGS video streams, which provide even more scalability combinations that can be extracted. Subjective tests with MGS streams are interesting as it better reflects the full potential of scalable coders. Last, subjective tests of scalable streams on resource-constrained mobile devices may be even more interesting as mobile devices are likely to need stream adaptations. We have seen some recent works [4] along this direction, although their experimental designs do not consider scalable video coders.

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Crowdsourcing for Quality Assessment

A short review for "Quadrant of euphoria: a crowdsourcing platform for QoE assessment"

Edited by Jong-Seok Lee

K.-T. Chen, C.-J. Chang, C.-C. Wu, Y.-C. Chang, and C.-L. Lei, "Quadrant of Euphoria: A Crowdsourcing Platform for QoE Assessment", IEEE Network, vol. 24, no. 2, pp. 28-35, March/April 2010.

Quality of experience (QoE) is a notion implying measurement of how users subjectively evaluate multimedia services. It is from the viewpoint of users, while the conventional quality of service (QoS) focuses more on service-related characteristics from the provider perspective. In general, QoE of multimedia contents can be assessed either subjectively or objectively. Subjective assessment is conducted by asking human subjects to rate the perceived quality of given stimuli on a pre-defined (categorical or continuous) scale. The final quality score for a stimulus, namely Mean Opinion Score (MOS), is obtained by taking the average of the ratings of multiple subjects.

The authors bring up the problem that, although subjective quality assessment is the most accurate way to measure perceived quality, it has following limitations. First, conducting subjective assessment experiments is usually time-consuming and expensive, because a sufficient number of subjects must be hired and they must carefully perform the task of assigning a numerical or categorical rating to each stimulus. Second, it may not be easy for a subject to map his/her perception on a scale. Moreover, the scale may be interpreted differently by different subjects. Also, the scale is in fact not linear, i.e. the cognitive distance between "bad (1)" and "poor (2)" may not be the same to that between "good (4)" and "excellent (5)". Thus, it is questionable to take the arithmetic average of individual scores to obtain the final quality score.

The paper tries to alleviate these problems by adopting two ideas. First, instead of categorical or numerical rating, paired comparison is used. Paired comparison significantly simplifies the subjects' task, i.e., they only need to provide preference between two stimuli presented. Second, in order to reduce personnel and time costs, crowdsourcing is employed, in which a lot of people are motivated to participate in subjective tests by much smaller rewards in comparison to conventional laboratory-based subjective quality tests. Thanks to the

technological advances of Internet access, there exist many participative and self-aware end-users who are good candidates for subjects of QoE experiments. In fact, the crowd is by whom multimedia contents with improved quality are consumed. The authors name their quality assessment platform Quadrant of Euphoria.

First, the authors introduce the basic principle of paired comparison and the Bradley-Terry-Luce model [1] for conversion of the paired comparison results into quality scores. Then, the proposed quality assessment platform is described in detail. In particular, the authors focus on the problem that the reliability of the ratings in crowdsourcing is usually lower than that of laboratory-based controlled tests. Due to the lack of supervision, participants may give erroneous ratings. Such results are from careless and perfunctory attitudes, or dishonesty and malign conduct.

In order to check whether a participant's ratings are reliable or not, the authors notice that preference must be transitive. In other words, if stimulus A is preferred to stimulus B, and stimulus B to stimulus C, then it is reasonable to expect that stimulus A is preferred to stimulus C, which satisfies the transitivity rule. Thus, the Transitivity Satisfaction Rate (TSR) is defined as the number of judgment triplets satisfying transitivity divided by the total number of triplets where transitivity may apply. As a rule of thumb, the authors consider a participant showing a TSR higher than 0.8 as fully attended. Readers may be interested in reference [2], in which a similar idea of using transitivity for checking inconsistent preferences was also given, but with more emphasis on the mathematical viewpoint.

Finally, the authors present two network-related case studies, quality assessment of VoIP under packet loss and quality comparison of IPTV loss concealment, along with those already presented in the authors' previous paper [3], quality assessment of MP3 for different bit rates and quality comparison of various video codecs.

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They were carried out on three different platforms, i.e., a physical laboratory, crowdsourcing using the Amazon Mechanical Turk (MTurk), and crowdsourcing in another populous Internet community. It is shown that the ratings from the three platforms were consistent. However, the cost per each run was able to be lowered in crowdsourcing in comparison to the laboratory platform (on average, 3¢, 1¢, and 0.07¢ for the three platforms, respectively). As expected, a significant number of ratings were filtered out while checking TSR values in the case of crowdsourcing.

As indicated in the last part of the paper, further research such as QoE assessment for interactive applications, multidimensional consistency quantification, and consideration of a neutral option for paired comparison and subsequent model changes would be desirable in the future based on the work in the paper.

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On Securing RFIDs against Blocking Attacks

A short review for "A probabilistic approach for detecting blocking attack in RFID systems"

Edited by Walid Saad

E. Vahedi, Rabab K. Ward, V. Shah-Mansouri, V. W.S. Wong, I. F. Blake, "A Probabilistic Approach for Detecting Blocking Attack in RFID Systems" in Proc. of IEEE International Conference on Communications, May 2010.

Radio-frequency identification (RFID) is an emerging wireless technology that allows objects to be identified automatically. An RFID system consists of small, inexpensive tags placed in objects and readers. Each tag is a small electronic device with an antenna and has a unique serial identification (ID) number. It replies to interrogations initiated by the readers. The use of RFID systems can simplify many applications and provide many benefits. However, the privacy of consumers that use items with RFID tags should be taken into account. A potential threat for the privacy of a user is that of illegitimate readers obtaining information about the tags in the system. Some consumers are concerned about being tracked by such readers when they are carrying items (such as clothes, medicine, or currency) embedded with RFID tags. In addition to tracking individuals, RFID tags may also be used to extract personal information such as the type of clothes somebody wears, the specific brands that an individual is interested in, or some medical information about the patient carrying an RFID-embedded container of medicine [1].

To cope with security issues in RFID systems, various schemes have been proposed. Among them, the use of a blocker tag is one of the most applicable solutions proposed [2]. A blocker tag can simulate all or a portion of tag IDs in the system. This prevents the illegitimate readers from identifying the tags and obtaining information from the system by hiding the tag IDs among many fake IDs. Although this solution is low cost and simple to implement, it can impose a serious threat to RFID systems when used as a malicious tool to attack the systems. A malicious blocker tag can deteriorate the performance of an RFID system by simulating fake tag IDs [3]. Prior to this work, the only way for detecting the blocking attack in an RFID system was counting the number of detected tags, which is a very time consuming and inefficient procedure.

In [3], we study the malicious use of blocker tags to prevent nearby legitimate readers from correctly receiving reply messages from the tags. We mathematically model the blocking attack for RFID systems whose operation is based on the Binary Tree Walking singulation mechanism. This singulation mechanism is described in [2] and [3]. After developing the analytical framework, we propose a probabilistic blocker tag detection (P-BTD) algorithm to detect the presence of an attacker in the RFID system. As mentioned above, in an RFID system, the reader interrogates the tags and collects their replies. The P-BTD algorithm can detect the existence of a blocker tag using the information extracted from the interrogations performed by the reader. Based on the different replies from the interrogated tags, three different events can be observed by the reader as explained in [3]. In the proposed P-BTD method, the probability of the observed event is calculated after each interrogation using two assumptions. The first is that there exists a blocker in the system, and the second assumption is that there is no blocker in the system. As a result, two conditional probabilities can be calculated for each observed event in the presence and absence of a blocker tag. If the probability of the event observed by the reader in the presence of a blocker is larger than the probability of this event in the absence of a blocker, the P-BTD algorithm announces that the system has been attacked by a blocker tag.

It should be noted that in addition to the Binary Tree Walking, there exists another singulation technique which is used by many RFID systems, called the framed slotted ALOHA. The RFID systems whose operation is based on the framed slotted ALOHA are mathematically modeled as well, and the P-BTD algorithm is modified for these systems in [4].

In the last part of the paper, extensive numerical simulations are presented to assess the performance of the proposed P-BTD algorithm, and to compare it with the solution proposed in

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[2]. This solution is called the threshold-based blocker detection algorithm. Simulation results show that the proposed P-BTD algorithm has a better performance than the threshold-based detection algorithm in terms of the smaller number of interrogations required for detecting the blocking attack.

It should be noted that the blocking attack is not limited to RFID systems that use the blocker approach for privacy protection. Any passive RFID system that uses a singulation mechanism is also vulnerable to this attack and needs to be well-prepared against malicious blocking. The importance of the proposed P-BTD algorithm can better be explained using a real example. Many large companies, such as Wal-Mart and Procter and Gamble, currently use RFID tags to improve inventory accuracy, on-shelf availability, and monitoring purposes. For instance, it has been estimated that in the U.S.A only, Wal-Mart tags approximately 250 million men's jeans and other items annually [4]. These tags are read frequently during working hours. Using this technique, the number of items is checked over time to prevent stealing them from the shelves. Moreover, the on-shelf monitoring system is notified if for example, a specific size or model of apparel is near to being sold out or is in high demand. This way, the store is able to keep the balance between consumers' demand and the number of items on shelves. These apparels are tagged, stored in the warehouse, and moved to the sales floor as the previous items are being sold. Meanwhile, an attacker may put a blocker tag in the store or warehouse so as to sabotage or adversely affect the operation of the legitimate reader. The attacker can put a blocker among the apparels of the sales floor to convince the reader that there are enough jeans of a specific size on the shelf, to steal some items or to prevent the system from keeping the balance between the demand and the apparels required on the shelf. This type of attack is very hard to detect but the store should know about it as soon as possible, otherwise, drastic financial loss could be inevitable. Based on the above, large organizations and suppliers are vulnerable to the blocking attack, and they are in need of efficient methods for detecting this attack. This is exactly where the proposed P-BTD algorithm can play a significant role as it can prevent millions of dollars in loss to supply companies that use RFID technology for supply chain management.

In summary, this paper provides a fresh look at the blocking attack problem in RFID systems and develops a probabilistic solution for detecting this attack. Unlike the conventional threshold-based algorithm that is very time consuming, the proposed method is capable of detecting the blocking attack using a few number of interrogations by the reader. Although the probabilistic framework is proposed for blocking attack detection, it can also be used for other purposes, such as tag estimation and protocol design.

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