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CONTENTS

Message from R-Letter Co-Director

Welcome to the second 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 five high quality papers in this issue.

The introduced papers span two main broad areas: video delivery and video coding. The first paper, published in the *IEEE Journal on Selected Areas in Communication*, introduces a systematic framework for coordinating a network of wireless users seeking to stream video data so as to share efficiently the network while accounting for the dynamically changing network conditions. The second paper, from the *IEEE Transactions on Multimedia*, describes a complete prototype for a live P2P streaming system using layered video coding. The third paper, also from the *IEEE Transactions on Multimedia*, tackles the resource allocation problem in layer-encoded IPTV multicasting in WiMAX wireless networks. The fourth paper, published in the *2010 IEEE International Conference on Multimedia & Expo*, studies the performance of scalable video coding over a DVB-SH satellite link. The last paper, published in the *IEEE Transactions on Circuits and Systems for Video Technology*, proposes and analyzes a transform-domain adaptive correlation estimation for Wyner-Ziv video coding.

In addition to the aforementioned introduced papers, this issue of the R-Letter includes Call for Papers for four special issues of IEEE magazines and journals that are currently being organized by MMTC interest groups.

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: Walid Saad, Cheng-Hsin Hsu, Ai-Chun Pang, Christian Timmerer, and Vladimir Stanković. I also would like to thank the R-Letter Director Guan-Ming Su for all his efforts.

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Multi-user Decomposition for Dynamic Video Streaming

A short review for "A systematic framework for dynamically optimizing multi-user video transmission"

Edited by Walid Saad

F. Fu and M. van der Schaar, "A Systematic Framework for Dynamically Optimizing Multi-User Video Transmission," IEEE Journal on Selected. Areas in Communications, vol. 28, no. 3, pages 308-320, Apr. 2010.

The increasing proliferation of resource-demanding services such as multimedia or gaming implies that future wireless network infrastructures must be able to support multiple simultaneous real-time video streaming applications. One of the key challenges associated with enabling multi-user video streaming over wireless networks is the dynamic allocation of the scarce network resources among heterogeneous users experiencing different time-varying network conditions and traffic characteristics.

Existing literature on video streaming over wireless networks often ignores these challenges, and many multi-user video transmission solutions are based on the network utility maximization (NUM) framework [1] with the key assumption that each user has a static utility that is a function of the (average) allocated transmission rate. However, in practice, due to the dynamic wireless channel conditions and the heterogeneous traffic characteristics (e.g. data arrival rates, data dependencies, and data importance, etc.), the NUM framework becomes unsuitable for multi-user video transmission over wireless networks. Moreover, most of the work that addressed single-user video transmission such as in [2] is often based on heuristic schemes that myopically adapt the transmission strategies (e.g. transmission power allocation, retransmission, etc.) based on the experienced wireless channel conditions and observed traffic characteristics. As a result, such schemes often yield a suboptimal performance because they do not account for the future channel conditions and the video traffic.

In this respect, the paper by Fu and van der Schaar introduces a systematic framework for coordinating a network of wireless users seeking to stream video data so as to share the network resources in a dynamically changing environment. To do so, the first step is to formulate a foresighted framework that can dynamically adapt the cross-layer transmission strategies (e.g. packet scheduling and resource acquisition, etc) of multiple users using a weakly-coupled multi-user MDP (MUMDP) problem [3]. The MUMDP formulation allows each wireless user to make foresighted transmission decisions by taking into account the impact of its current decisions on the long-term utilities of all the wireless users.

To solve the problem, a distributed solution based on Lagrangian relaxation is used so as to decompose the weakly-coupled MUMDP problem into multiple local MDPs, each of which can be separately solved by the individual wireless users. Hence, from a practical perspective, the proposed solution improves significantly from existing centralized solutions [4] to the MDP, which often lead to increasing delays due to their high computational complexity and communication overheads. The proposed decomposition distinguishes itself from conventional dual solutions [1][5] to the multi-user NUM-based video transmission problem in two ways. First, instead of maximizing the static utility at each transmission time, the proposed approach allows each wireless user to solve the dynamic optimization problem (formulated as a local MDP), which is vital for the delay-sensitive applications. Second, instead of updating the Lagrangian multipliers based only on the current resource requirements of the wireless users, the proposed approach updates the multipliers depending on the their future predicted resource needs, such that the long-term utility of all the wireless users is maximized (thereby maximizing the long-term global utility of the wireless network).

The paper also introduces a novel traffic state representation that can capture the dynamic features (e.g. delay deadlines, distortion impacts and dependencies etc.) of the media data at each time slot. For instance, this traffic state concept makes it possible to allow a dynamic scheduling for the media transmissions using a Markov decision process that explicitly considers the users' heterogeneous multimedia data characteristics and time-varying network conditions.

Most existing literature in networking assumes that the environment (traffic and network conditions) and the resulting utilities for different transmission decisions are known. However, this assumption does not hold in practice and therefore one needs to devise schemes that can cope with the unknown and time-varying nature of wireless environments. To do so, this paper proposes low-complexity online learning algorithms which take advantage of the structural properties of the post-decision state-value functions. Instead of learning the value function in every possible state, the proposed solution needs only to learn it in a limited number of states. This can significantly accelerate the convergence rate compared to existing state-of-the-art learning solutions, which is essential for the performance of delay-sensitive video streaming applications.

The presented MUMDP framework can be applied to other classes of applications in which the decisions need to be adapted dynamically, over time, and which operate in unknown environments. Examples of such applications include adaptive media encoding/decoding [6], dynamic resource allocation for large-scale data centers [7], stream mining systems [8], etc. Hence, the presented framework is envisioned to be a key enabler for the transmission of a variety of delay sensitive and resource demanding applications over practical wireless networks in the future.

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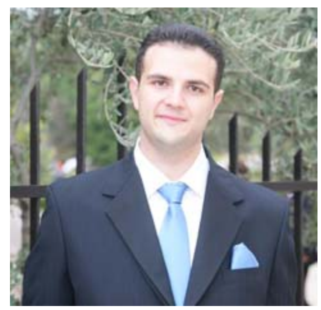

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Next Generation P2P Live Streaming Systems

A short review for "LayerP2P: using layered video chunks in P2P live streaming"

Edited by Cheng-Hsin Hsu

Z. Liu, Y. Shen, K. W. Ross, S. Panwar, and Y. Wang, "LayerP2P: Using Layered Video Chunks in P2P Live Streaming," IEEE Transactions on Multimedia, vol. 11, no. 7, pages 1340—1352, November 2009.

P2P live video streaming has become a popular service in the Internet owing to the widespread adoption of broadband access, advanced video codec technology, and faster computation ability equipped hardware. P2P live streaming systems can be categorized into two classes: (i) tree-based systems in which one or multiple trees are constructed among peers for transferring video content and (ii) mesh-based systems in which each peer connects to a few other peers without an explicit overlay topology. Studies indicate that mesh-based live streaming systems outperforms tree-based ones in several aspects, such as better perceived quality [1], lower maintenance overhead, higher resilience to network dynamics, and lower implementation complexity [2].

Most live streaming systems in operation today are mesh-based. In these systems, each live channel is encoded into a non-scalable, single-layer video, which is then partitioned into multiple pieces. The pieces are distributed in the overlay network formed by peers, and each peer reassembles the received pieces back to the single-layer video before rendering it to the user.

These existing P2P live streaming systems are not scalable to bandwidth fluctuation, because peers must receive all pieces in time to decode the single-layer video. Bandwidth fluctuations are common, and can be due to congestion in underlying networks as well as peer churn in overlay networks. For peers that do not receive all pieces in time, they simply cannot render the video and have to pause the playback for re-buffering. Re-buffering instances significantly degrade the viewing experience, and excessive interruptions drive the users away from the service. Hence, P2P live streaming systems need an efficiency way to adapt to bandwidth fluctuations.

Layered video coding has been proposed to cope with bandwidth fluctuations. With layered videos, each channel is encoded into a multi-layer video, and peers that do not have enough bandwidth to receive the complete video stream can still render a subset of layers for a slightly lower perceived video quality rather than suffering from annoying playback pauses. Layered videos, however, impose complicated interdependencies among layers, which must be carefully accounted for. Using layered videos in P2P live streaming is even more challenging than doing so in traditional client-server streaming systems, because the high churn rate in P2P systems. To my best knowledge, the authors' prototype, deploy, and evaluate the first layered P2P live streaming system [3, 4], called *LayerP2P*. Their work stimulates the systems research on streaming layered videos in P2P networks.

LayerP2P leverages on layered videos in mesh-based P2P live streaming systems. In LayerP2P, when the system has abundant bandwidth, where the average upload bandwidth supply is higher than the full video rate, every peer receives all layers of the video and enjoys the full video quality. However, when the system is in a bandwidth deficient state, a peer's perceived video quality is proportional to its upload contribution to the system. A P2P live streaming system enters the bandwidth deficient state when most peers have low uplink bandwidth or do not want to contribute. LayerP2P encourages peers to contribute more by providing higher (lower) quality videos to the peers that contribute more (less). Hence, layerP2P not only better adapts to bandwidth fluctuation, more resilient to packet loss, but also provides a strong incentive to peers: free-riders can only receive at rather poor video quality. Such incentive is crucial for scalable and robust P2P live streaming, and is not possible with single-layer videos.

The authors' prototype a complete LayereP2P system, which includes seed, peer, and tracker. They deploy the prototype in PlanetLab and conduct extensive experiments using real layered videos. To account for peer churn, they also conduct trace-drive simulations using the peer dynamics captured from a commercial P2P live streaming system.

The experimental and simulation results show that LayerP2P achieves high perceived video quality at all peers when the overall upload bandwidth is high enough. LayerP2P provides differentiated services for different peers when the overall upload bandwidth is insufficient. The authors also consider two single-layer P2P live streaming systems as the benchmarks, and show that LayerP2P delivers better video quality to cooperative peers.

The authors clearly demonstrate the great potential of using layered videos in P2P live streaming systems. With advances in scalable video codecs in both International standards [5, 6] and open-source communities [7], we are not far away from large deployments of next generation, layered P2P live streaming systems.

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Layered IPTV Multicasting: Toward Maximizing Group Visual Experience

A short review for "Adaptive resource allocation for layer-encoded IPTV multicasting in IEEE 802.16 WiMAX wireless networks"

Edited by Ai-Chun Pang

W.-H. Kuo, W. Liao, and T. Liu, "Adaptive Resource Allocation for Layer-Encoded IPTV Multicasting in IEEE 802.16 WiMAX Wireless Networks," IEEE Transactions on Multimedia, vol. 13, no. 1, pages 116—124, Feb. 2011.

IEEE 802.16 (WiMAX) [1], an emerging broadband wireless access technology, is an excellent platform for providing IPTV multicast service to residential users. In IPTV multicast service, each video stream is multicast to a set of users tuned to the same program channel. Due to location-dependent and/or time-varying factors in the wireless environment, the link condition of users in the same multicast group may not be identical. In WiMAX networks and other next generation wireless networks, adaptive modulation and coding (AMC) is supported for better spectrum usage, and each Subscriber Station (SS) located at the client side must negotiate its burst profile with the Base Station (BS) before the connection starts. To ensure that users with different burst profiles in the same group can receive the program with the same video quality, the IEEE 802.16 standard requires that the video be encoded by the BS with the most robust burst profile (corresponding to the user(s) of the worst link channel quality) in the group. As a result, having more users in a group implies that the video stream multicast to the group of users is more likely to be encoded with a more robust profile, and therefore, more resource is consumed to transmit the same amount of data. Since the wireless resource is relatively scarce, it is impossible to always serve all subscribers with all layers of multicast streams. This calls for an efficient and effective resource allocation scheme.

In this paper, the authors tackle this problem for layer-encoded IPTV multicasting over WiMAX networks. Each video stream is encoded at the highest resolution and divided into layers such that each receiver can decode the stream at the preferred rate and resolution with a set of layers [2-3]. For each layer-encoded stream multicast through the network, different users may receive different numbers of layers depending on their channel qualities. In a wireless environment where users may have different channel conditions, this layering approach ensures more flexible in resource utilization and also allows each user to experience different video quality according to its channel quality.

To efficiently assign different layers of the stream to different SSs, each layer of a video stream is assigned a utility value and the number of layers in each program each user can receive is adjustable. The objective is to maximize the total utility (i.e., all users' satisfaction) and the system resource utilization, subject to users' channel conditions, the popularity of video programs, and the total available radio resource. The authors prove that this problem is NP-hard by showing that it is more general than the famous NP-complete Knapsack problem [4]. Based on the concept of an "envelop function," they design a polynomial-time utility-based resource allocation heuristic for layered-encoded multicasting, called UE-LEM. The allocation utility of each layer is first transferred into an approximate envelop function. Then, UE-LEM allocates resources to subscribers accordingly.

Through analysis, they show that the difference in the performance of UE-LEM and the optimal solution is tightly bounded, since the "envelop function" on which UE-LEM is based approximates the actual utility. The authors have also demonstrated that UE-LEM has an acceptable polynomial time complexity. The performance of UE-LEM is also evaluated by simulations. The results indicate that this scheme can make flexible and reasonable resource allocation according to the utility function of each program, the popularity of each program, the channel condition of subscribers, and the amount of total resource available in the network.

UE-LEM is a promising scheme for allocating layer-encoded video streams over adaptive modulation and coding supported broadband wireless access networks such as IEEE 802.16 WiMAX and 3GPP LTE-Advanced. Due to its low computation overhead, it can be executed periodically or on demand to better reflect time-varying channel conditions. Also, it supports single layer media streams, and can also be applied to unicast media streams by treating each unicast stream as a multicast stream with only one receiver. More importantly, it can be integrated with the multicast and broadcast service (MBS) mechanism defined in the WiMAX standards, and can also be applied to wireless networks which support adaptive modulation and coding schemes. In summary, UE-LEM can be implemented to improve the resource utilization of both unicast and multicast transmissions, in both single layer and multi-layer environments for next generation wireless networks.

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Scalable Video Coding over DVB-SH: Unequal Error Protection Approach

A short review for "Study of the performance of scalable video coding over a DVB-SH satellite link"

Edited by Christian Timmerer

G. Liebl, K. Tappayuthpijarn, K. Grüneberg, T. M. de Moraes, C. Hellge, T. Schierl, C. Keip, H. Stadal, and N. Pham, "Study of the Performance of Scalable Video Coding over a DVB-SH satellite link," Proc. of International Conf. on Multimedia and Expo, pages 522—527, 2010.

Digital Video Broadcasting - Satellite services to Handhelds (DVB-SH) is a standard for broadcasting IP-based media content and data to handheld terminals [1]. Transmission is done in parallel via (quasi) geostationary satellites and a network of terrestrial gap-fillers, called Complementary Ground Component (CGC), which reuse the same frequency allocation.

The DVB-SH standard in its current version offers a lot of different features for tuning the system performance: On the satellite link, one has the choice between an optimized single-carrier time-division multiplexing (TDM) waveform and an orthogonal frequency-division multiplexing (OFDM) waveform with optional hierarchical modulation (HMOD). On the terrestrial link, only OFDM is possible, since it is more resilient to multipath propagation. The usage of Turbo codes with a variable length time interleaver at the physical layer allows correcting most of the transmission errors. Furthermore, additional protection at session level via multi-protocol encapsulation–inter-burst forward error correction (MPE-IFEC) is possible. Additionally, unequal error protection (UEP) can be realized in DVB-SH either via HMOD or MPE-IFEC.

In order to efficiently exploit this large variety of transmission options, future broadcast applications may require further optimization also. One frequently debated solution is to introduce advanced media encoding techniques that may or may not "scale" with varying terminal capabilities and/or channel conditions. This can be achieved, for example, by replacing Advanced Video Coding (AVC) specified for DVB-SH with its scalable extension referred to as Scalable Video Coding (SVC) [2].

An SVC bit stream consists of two or more layers which correspond to different levels of spatial or temporal resolution or fidelity in terms of peak signal-to-noise ratio (PSNR). Upper layers depend on lower ones due to the prediction of texture or motion data from them. The lowest (or base) layer is an AVC compatible bit stream which can be processed by legacy decoders. The term scalability implies that part of the bit stream can be discarded, leaving a remaining stream which can still be decoded, albeit at lower quality.

Since until recently it had not been clear whether this type of scalability is efficiently supported by the DVB-SH standard, a study funded by the European Space Agency (ESA) was conducted from 2008-2010 [3]. Within the reviewed paper, one specific reference use case of the ESA study is presented and discussed. It concerns the exploitation of heterogeneous reception conditions on the satellite link and can be described as follows. Consider an area where signal reception is only possible via the satellite, i.e., where there are no nearby CGCs. Furthermore, within this area, two different environmental conditions, rural and suburban, exist, of which the latter one is less critical for transmission. Provided that the average reception level, which depends on transmit power, satellite elevation, distance-dependent path loss, antenna configuration, and system-internal loss, is the same, better reception quality can be achieved in the suburban environment.

However, without scalability in the video stream, the system operator must choose the physical and/or link layer code rate, as well as the modulation scheme such that all terminals achieve a decent quality regardless of their environment. Hence, the system has to be designed for the worst case, i.e., for rural reception. In case most terminals are in a suburban environment, a large portion of the bandwidth is thus wasted for unnecessarily high protection, which could otherwise be used to enhance the quality and/or number of services.

With scalability in the video stream, the system operator may adapt the design as follows. The encoded stream contains two fidelity-scalable layers, a base layer (BL), which achieves a minimum-acceptable quality, if received alone at a terminal, and an enhancement layer (EL), which when added on top of the BL almost achieves the same quality as with non-scalable encoding. The two layers are transmitted as two separate sessions per service, such that UEP can be applied either at the physical layer via HMOD or at the link layer via MPE-IFEC. The design of the protection levels is as follows. The BL is required in both environments and is thus protected with the "High Priority" (HP) parameter set, which must match the more critical rural environment. The EL is only required in the favorable suburban environment and is thus protected with the "Low Priority" (LP) parameter set. Hence, the objective is to increase the number of offered services by trading in some small quality loss in rural reception.

Simulation and performance evaluation of these two system designs has been performed with the help of a simulation platform described in [4]. Two sets of experiments have been conducted which target different receiver classes in DVB-SH [5]. In the first set, a class-2 receiver capable of handling a long interleaver at the physical layer has been assumed. Hence, only physical forward error correction (FEC) is applied and UEP must be realized via HMOD. In the second set, a class-1 receiver with a short interleaver is considered. Hence, both physical FEC and MPE-IFEC are applied and UEP must be realized via different IFEC code rates.

The outcome of this analysis can be summarized as follows. If HMOD is used for achieving UEP, the necessary change in waveform from TDM (best non-scalable reference system) to OFDM (scalable system) requires relatively low HP and LP code rates, otherwise the reception quality is below target. This directly affects the number of offered services, which actually starts to decrease. Considering also the inherent degradation in video quality in rural reception, it must be concluded that SVC+HMOD does not satisfy the objective of the use case and results in both less services and worse quality at the same time.

If the same comparison is made with MPE-IFEC enabled at the link layer for both reference and scalable system, a slight gain in services for the latter one seems achievable, if one trades in some quality in rural reception. Hence, the results indicate a strong dependency on the applied UEP mechanism which does not render this use case as a strong advocate for scalability in DVB-SH.

Further analysis of other, specifically terrestrial use cases within the ESA project show that SVC can be of definitely of interest for certain deployments [3].

These kinds of studies – either simulation or (real-world) test-beds – are important for the deployment of novel coding and transmission techniques and often provide the basis for future research as they point out issues not foreseen when drafting related standards. Practitioners and researchers may use this paper as a basis in order to setup their own simulation environments or test-beds and future research in the area of SVC deployments in heterogeneous environments is hereby solicited.

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Wyner-Ziv Video Coding: Pushing Limits via Effective Correlation Estimation

A short review for "Transform-domain adaptive correlation estimation for Wyner-Ziv video coding"

Edited by Vladimir Stanković

X. Fan, O.C. Au, N.M. Cheung, "Transform-domain Adaptive Correlation Estimation (TRACE) for Wyner-Ziv Video Coding, IEEE Transactions on Circuits and Systems for Video Technology, vol. 20, no. 11, pages 1423-1436, November 2010.

Since its appearance in 2002 [1, 2], Wyner-Ziv video coding (WZVC) or distributed video coding (DVC) has been the subject of intense debate and criticisms in the image/video coding community. Having its roots in the 1970's information-theoretical results [3, 4] on source compression with decoder side information, WZVC brought a very neat possibility of avoiding motion estimation at the encoder side within a probabilistic coding framework. Being surprised and thrilled with this revolutionary novel video coding thinking, many saw in WZVC a remarkable possibility of distributing computation load between encoders and decoders which suited very well at that time emerging distributed networks, such as sensor/surveillance networks. That was enough for research on WZC to ignite with many contributions appearing from leading video coding centers in the years to come. However, the initial excitement has been replaced by some level of disappointment as we see a decline in WZVC contributions. We have yet to see any commercial product using WZ coding principles in any way or form. Is that a sign that WZVC will remain as an elegant theoretical attempt only?

The key feature of WZVC is avoiding computationally expensive temporal prediction loop in video encoding, which makes it desirable to capturing devices of limited computing power, such as smart phones, disposable cameras, and wireless surveillance cameras. Additionally, WZVC possesses built-in error resilience since it does not use a prediction loop at the encoder side. Moreover, its statistical coding nature makes it a more suitable fit to statistically varying channels, than traditional coders. WZVC is also a natural solution for exploiting inter-view correlation in multi-view setups where many closely located cameras are used to capture the scene. These promising potential applications, assessed and analyzed in [5], have been motivation drivers for WZVC research.

However, the main drawbacks are high decoding complexity and, more importantly, significantly lower compression efficiency than state-of-the-art encoder motion-compensation based video coding. This comes as disappointment, especially because information theory predicts that it should be possible to come much closer to the encoder-prediction performance. However, the main reason for this "disagreement" between theory and practice lies mainly in the non-stationary nature of video, which cannot be captured. That is, video signals change unpredictably causing source statistics to vary significantly, and the exact knowledge of statistics at the encoder and decoder is the main assumption used to derive theoretical bounds. The WZVC decoder, which drives the whole process, relies purely on statistics to recover video, and if these statistics are unreliable or varying significantly from their true value, significant rate loss will be incurred.

The main prerequisite in designing WZVC solutions is providing a proper estimation of the correlation between the frame to be encoded and the generated decoder side information [1, 2], and this paper is an important step forward. Being recognized as a key performance contributor, correlation estimation has been the focus of the WZVC research community for a while. Usually the difference between the side information and the source in WZVC is modeled as Laplacian, which is used to initialize decoding algorithms, provide optimal MMSE reconstruction, and estimate the required rate. In early works, this "correlation noise" is assumed to be stationary within the sequence, but later it has been noticed that improved performance is obtained by estimating the level of noise per frame, block, pixel, bitplane band and transform coefficient (for transform-based bitplane codecs), see, e.g., [6, 7]. It was shown in [6] that coefficient-level correlation estimation, where correlation is estimated for each DCT coefficient, can lead to significant performance gains as it can potentially capture the finest correlation information. However, the danger is that not enough statistics might be available for a precise-enough estimation.

The main difficulty in the correlation estimation lies in the fact that only the encoder possesses the source data, and only the decoder has access to the side information data. This paper proposes a transform-domain adaptive correlation estimation method, called TRACE for the coefficient-level estimation independent of the way the side information is generated. It is assumed that correlation at any level can be approximated as a zero-mean Laplacian random variable, with unknown variance that needs to be estimated. A key contribution is the observation of a linear relationship between the variance of a transform coefficient to the pixel-domain variance of the block that this coefficient belongs to, something that previously was assumed to be uncorrelated. The paper shows that variances of two neighboring coefficients are proportional to each other and thus a bitplane-based decoder can perform coefficient-level correlation estimation using the band-level correlation and previously decoded bitplanes. The paper further shows that the bitplane band-level variance [6] can be optimally estimated using convex optimization. Simulation results show considerable gains over previous methods, and applicability of the solution to many different WZVC schemes.

As future work, the authors suggest extension to scalable WZVC. It is interesting to combine the proposed method with a side information dependent correlation noise model proposed in [8], where the standard deviation of the Lapacian model is a function of a particular realization of the side information at each pixel position. Another interesting approach taken recently is to *track* correlation changes over time at the decoder side by integrating particle filtering into the Slepian-Wolf decoding process [9]. Combining fine-level estimation for initialization and effectively tracking correlation at bit-level seems an exciting research direction.

In summary, for statistically-based WZVC, good correlation estimation is essential for reaching high performance. However, how much can be achieved with estimation/tracking towards closing the performance gap to the conventional video coders is questionable, since there are other problems such as rate control/adaptation and low code length performance. Additionally, deploying experimentations in practical networks is essential to demonstrate effectiveness.

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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

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- 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**
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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.

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Second review complete/acceptance:	March 10, 2012
Preliminary review results:	December 15, 2011
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Call for Papers: IEEE Communication Surveys and Tutorials

Special Issue on Energy-Efficient Multimedia Communication

(pending final approval)

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
- Use of cloud computing and other grid-based technologies for energy-efficient multimedia processing, hosting, and communication
- Metrics, benchmarks, and data-sets to evaluate the trade-offs between multimedia processing energy and the perceived multimedia quality
- Tools for measuring and analyzing energy consumed during multimedia communication

Manuscript Submission

IEEE Communications Surveys & Tutorials is a ComSoc publication. It provides researchers and other communications professionals with the ideal venue for publishing on-line tutorials and surveys which are exposed to an unlimited global audience. Articles should be written in a style comprehensible and appealing to readers outside the specialty of the article. For details, please refer to "Call for Papers" and "Information for Authors" at the IEEE Surveys and Tutorials web site at http://dl.comsoc.org/surveys/.

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