IEEE MULTIMEDIA COMMUNICATIONS TECHNICAL COMMITTEE

E-LETTER

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IEEE COMMUNICATIONS SOCIETY

Message from MMTC Chair

First of all, as the new chair of IEEE Communication Society Multimedia Communication Technical Committee (MMTC) during 2008-2010, I would like to take this opportunity to extend my best wishes for a happy and prosperous New Year 2009 to all our members. Also, I would like to express my gratitude to all the past MMTC officers as well as many earnest volunteers involved in the MMTC related activities.

The scope of MMTC includes conversational, presentational, and transactional applications and the underlying networking systems to support them. It further includes media servers, gateways, user devices and interfaces, software for distributed applications, and personal communication/ software agents to facilitate applications. MMTC's main activity is the sponsoring of symposia and workshops at IEEE ComSoc conferences (specifically IEEE ICC and Globecom). Our MMTC is now co-sponsoring Communications Software and Services Symposium in IEEE ICC and Globecom. We have seen degradation in terms of number of submissions from our TC members in the recent several years. In this respect, we are now in the stage to further expand our technical activities so as to generate higher impact to our society. No doubt, this will bring us a lot of challenges. But, on the other hand, we are surrounded by many opportunities too. We expect your strong supports and contributions. We will do our best for sure and let us work together in a more effective way to create an even brilliant future of our MMTC. Thank you very much in advance for your strong supports and collaboration.

Sincerely yours,

Qian Zhang

Message from Editor-in-Chief

After two and half years break, the E-Letter continues to serve for MMTC members. I would like first thank for Qian, the MMTC Chair, who has the wisdom and long-term vision to push for this effort. The primary goal of the E-Letter is to disseminate issues that share ideas, opinions, and perspectives in various areas of multimedia communications related technologies. The topic can be anything along the multimedia content lifecycle, from content acquisition, analysis, processing, coding, delivery, to rendering. It can also be any communication and networking system, algorithm, protocol, platform or application where the multimedia data are transmitted.

The E-Letter would be published monthly and delivered to **1000+** scholars who are working in the multimedia communications related areas. I hope it would become an effective means for promoting future collaborations and idea exchanges among our fellow members.

In this issue, we have 6 scientific articles published in the technology session. The session begins with a Distinguished Position paper on IPTV over WiMAX, delivered by Dr. Jeng-Neng Huang (University of Washington). The aim of the series of Distinguished Position papers is to share the visions and perspectives of top researchers in specific areas to our fellow members.

The second article is a short review on a selected paper by our Column Editor, Dr. Guan-Ming Su (Marvell Semiconductors). The selected paper is on the topic of cooperative multimedia communications. The aim of this column is to recommend papers with great potential to our members, and thus trigger more collaboration, discussion, or debating in the community to promote the technology development.

The third paper is a short overview paper on network coding and its applications on multimedia communications, delivered by Dr. Athina Markopoulou et al. (University of California). In the fourth paper written by Dr. Shujun Li (Universität Konstanz), the multimedia encryption for communications is overviewed and a few open topics and challenges are pointed out for further investigations.

The fifth article by Dr. Wenjun Zeng (University of Missouri) highlights a few latest advances in the wireless multimedia communications and networking area. The column of "Focused Technology Advances" is featured for introducing new inventions, latest products or new concepts and applications.

The last article by Dr. Song Ci (University of Nebraska-Lincon) discusses the latest advances and activities of the Cross-Layer Multimedia Communication Interest Group. The "IG Corner" is a reserved place for MMTC Interest Group to introduce recent development, exchange ideas, and announce new activities.

In this E-Letter, we look forward to seeing new features or columns proposed by our TC members, and we encourage all of you to contribute short articles to share with us your research achievements and your perspectives on specific research topics.

At the end, I would like to thank all Editors of the E-Letter, and our authors to make this issue successful.

Thank you very much.

Haohong Wang

HIGHLIGHT NEWS & INFORMATION
TC Promotion & Improvement Sub-Committee Appointed

A new sub-committee of MMTC, namely, TC Promotion & Improvement (TCPI) sub-committee, has been appointed by the MMTC Chair recently. TCPI is expected to take immediate actions to strength the communications and interactions among MMTC members, and improve the member involvements in MMTC future events. Specifically, TCPI's tasks include but not limited to the following items:

- Bring in new members that are active in the field to enhance the influence of MMTC in the research community;
- Come up creative strategies and incentive plans to trigger MMTC members' enthusiasm to participate in MMTC activities;
- Promote the paper submission volume of MMTC community to ICC/GLOBECOM to improve the visibility and reputation of MMTC in IEEE society;
- Reactivate the MMTC E-letter and use it as an effective means to activate the interactions and communications among MMTC members;
- Setup a workshop to be held regularly in conjunction with the ICC/GLOBECOM to help gather the community as well as a complement event for ICC/GLOBECOM.

TCPI effort is led by Dr. Haohong Wang (Marvell Semiconductor, USA), with the supports of the following capable sub-committee members:

Dr. Antonios Argyriou
Phillips Research
Netherlands

Prof. Zhihai He
Univ. of Missouri
USA

Dr. Christian Hoene
University of Tübingen
Germany

Prof. Zhu Li
Hong Kong Polytechnic University
China

Dr. Shiguo Lian
France Telecom R&D Beijing
China

Prof. Philippe Roose
IUT of Bayonne
France

Dr. Guan-Ming Su
Marvell Semiconductors
USA

Dr. Chonggang Wang
NEC Laboratories America
USA

Prof. Jiang (Linda) Xie
Univ. of North Carolina at Charlotte,
USA

If you have any comments, suggestions or volunteering to help, please contact directly with Dr. Haohong Wang at haohong@ieee.org.

IEEE GLOBECOM 2009 @ Honolulu, Hawaii

IEEE GLOBECOM 2009 is to be held during 30 November - 4 December 2009 in Honolulu, Hawaii. The Honolulu venue will also offer participants - and those accompanying them - an opportunity to explore the paradise of the Hawaiian Islands. In this idyllic setting, you will be able to network with friends, colleagues, customers and vendors from around the world.

The theme of IEEE GLOBECOM 2009, "Riding the Wave to Global Connectivity," is specifically matched to the conference location, setting the conference's focus on leading edge coverage of communications with and across all parts of our planet (and beyond).

As shown in the map above, Honolulu is the capital and largest city of Hawaii, on the southeast coast of the island of Oahu. It is situated in the central Pacific Ocean 2,397 mi west-southwest of San Francisco. Honolulu's name derives from the native words hono, meaning “a bay,” and lulu, meaning “sheltered.” Millions of visitors are drawn annually to Honolulu's mild, semitropical climate and to the beautiful scenic views.

Honolulu is home to some of America's best-looking beaches and infamous for adjacent Pearl Harbor. Green mountains, the Koloau, flank Honolulu to the east, while the white sands of Waikiki and luminescent blue water running along two-miles to the southeast have long lured visitors to the area. With a skyline that properly distinguishes the downtown as a bustling metropolis, there are plenty of cultural offerings to be found in Honolulu between forays into the jungle and hard days on the beach.

With dozens of coves and bays, Oahu has almost endless snorkeling opportunities for people of all ages and skill levels. There are also many boats that offer at-sea snorkeling. Snorkeling is a very popular and common activity here, as, on a typical Oahu day, the water is warm and crystal clear, offering excellent opportunities to observe abundant and colorful sea life up close. In many instances, sea creatures will swim right up to you. There is no shortage of places from which to rent or buy the necessary equipment. To be specific, Hanauma Bay, Shark's Cove and Turtle Bay are all popular locations for snorkeling and other water activities.

If you have not started to plan for your family's winter vacation in 2009 yet, please do consider Honolulu and Hawaii. Remember to have your papers (or proposal) ready as soon as possible for GLOBECOM 2009 following the deadlines below:

Paper Submission:	**March 15**, 2009
Tutorial Proposal:	**March 15**, 2009
Workshop Proposal:	**March 15**, 2009

See you in Hawaii this winter! Aloha!

How to Submit Papers to GLOBECOM 2009

For GLOBECOM 2009, **Communication Software Symposium (CSS)** is the **ONLY** symposium that is fully sponsored by MMTC, hence we encourage all our members to submit your Multimedia related papers to CSS symposium, where a Co-Chair and many TPC members recommended from MMTC would handle all the review process of the multimedia related paper submissions.

Here are the few steps to submit your papers to GLOBECOM 2009:

(1) Go to EDAS website: http://edas.info/index.php and sign into your account;
(2) Click the "**Submit paper**" Tab on the top of the webpage;
(3) In the list of the conferences, find "**Globecom'09 CSS**" (or "GC'09 CSS" as the) and click the icon on the rightmost column;
(4) Start the normal paper submission process.

TECHNOLOGY ADVANCES

Distinguished Position Paper Series

Scalable IPTV over Mobile WiMAX

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The emerging deployment of internet protocol TV (IPTV), which will provide a more function-rich and user-interaction form of TV to the consumers over the existing IP infrastructure. calls for sufficient bandwidth and quality of service (QoS). To further make IPTV streaming more ubiquitous and seamless no matter where the users are located, a new frontier of wireless technology that can deliver the content to mobile users is needed. Worldwide Interoperability for Microwave Access (WiMAX) technology, based on current IEEE 802.16e and IEEE 802.16m in the future for mobile wireless access in metropolitan area networks (MAN), is thus introduced as one of the most competitive solutions for the all-IP based 4th generation wireless technology. There are several important characteristics of WiMAX that can be taken advantage of to support the dissemination of IPTV. More specifically, WiMAX supports multicast and broadcast service (MBS) while utilizing orthogonal frequency division multiple access (OFDMA) with adaptive modulation and coding scheme (MCS) capability, which enables a better throughput-delay tradeoff by changing the targeted/scheduled users in every transmission. Furthermore, it also provides real-time polling service (rtPS) priorities to ensure the required bandwidth and minimum latencies for video streaming so that quality of service (QoS) can be assured. To ensure efficient use of the WiMAX resources and for QoS assurance, effective subscribers' scheduling and radio resource allocation mechanisms are critically needed, however, these are not specified in the WiMAX standard, therefore service providers should have their scheduling and allocation designs. A better design would jointly integrate the scheduling and resource allocation tasks to achieve superior system performance.

There are two kinds of IPTV streaming, one is the real-time live streaming of broadcasting video content and the other is on-demand streaming of specifically requested video content. The real-time live IPTV streaming, characterized by groups of users acquiring the same popular video programs over a fading channel, can be effectively disseminated by the multicast support of WiMAX with stringent QoS constraints in terms of delay, jitter and packet loss. Therefore, efficient resource allocation for multicast is the key to ensure effective dissemination to users. More specifically, a base station (BS) attempts to multicast a live IPTV stream to multiple mobile stations (MSs), which have varying and fluctuating channel qualities. Obviously, if we schedule all MSs on every WiMAX frame transmission to ensure continuous reception, very low efficiency MCS would be used to satisfy such a diversified set of MSs. However, it is proven better to schedule only some MSs with better relative channel qualities on each transmission with higher efficiency MCS, and schedule the rest of MSs later when their relative channel qualities are improved. This concept is called opportunistic multicasting. In contrast to opportunistic unicasting, opportunistic multicasting searches for the best subset of users in each transmission to achieve higher throughput in multicasting.

To further accommodate various bandwidth requirements and adaptive perceptual quality to different users, video streams are coded into base and enhancement layers using scalable video coding technology, such as the recently standardized scalable extension of the H.264/AVC which provides not only the desired scalability but good coding efficiency. The concept of receiver-driven layered multicast can thus be applied in WiMAX IPTV live streaming, where all layers (base layer and enhancement layers) of the IPTV video are sent to the WiMAX base station, with each layer being sent as a multicast group. Based on the measured fluctuating channel quality of each receiving MS, the BS can optimally determine the set of users to schedule opportunistically as well as the MCS used in each layer in the MBS, i.e., to assign the most appropriate MCS so as to provide best video quality to as many users. More specifically, the minimum effective throughput can be maximized across all users for mandatory (base) layer delivery through adapting MCS. At the same time, optional (enhancement) layers are allocated to maximize total utility, such as the cumulative perceived quality by all users. Thus the basic video quality can be efficiently guaranteed to all subscribers while making the most out of limited resources on enhancement information.

With the use of opportunistic layered multicasting of scalable IPTV content, many MSs can only receive the mandatory base layer without the optional enhancement layers, especially for those MSs who are on the verge of the WiMAX hot zone. To further improve the service satisfaction among all users, the WiMAX IPTV live streaming can be combined with different wireless networking infrastructures, e.g., the upcoming 802.11p to be standardized for vehicle-to-vehicle and vehicle-to-infrastructure communication. This combination will allow those optional enhancement layers of IPTV content to be relayed to some vehicles which cannot receive those layers directly from WiMAX layered multicasting.

For on-demand streaming of IPTV content, the requested unicast connection needs to compete with other available services for resource in a downlink frame. A joint scheduling and resource allocation scheme, which can achieve better resource utilization efficiency, can be used to handle traffic flows of all three competing services, i.e., real-time polling service (rtPS) for IPTV, non-real-time polling service (nrtPS) for file transfer, and best effort (BE) for web browsing. By taking into account the service priority, QoS satisfaction factor, channel quality as well as available queue length, all packets in each service flow will be assigned an appropriate scheduling priority index to be used in the scheduler. The resource allocator can then iteratively decide the allocation order by jointly considering the average channel quality of each flow based on some heuristics, e.g., the flow with less mean channel quality has a higher ranking order to be assigned so that the flow has a higher probability to be allocated in its relatively good channel. The allocator will adaptively request the scheduler for more packets to transmit until the resource runs out or the unallocated resource is not enough for a packet.

Jenq-Neng Hwang received the BS and MS degrees, both in electrical engineering from the National Taiwan University, Taipei, Taiwan, in 1981 and 1983 separately. After two years of obligatory military services, he enrolled as a research assistant in 1985 at the Signal and Image Processing Institute, Dept. of Electrical Engineering, University of Southern California, where he received his Ph.D. degree in December 1988. He was also a visiting student at Princeton University, New Jersey, from 1987 to 1989.

In the summer of 1989, Dr. Hwang joined the Department of Electrical Engineering of the University of Washington in Seattle, where he has been promoted to Full Professor since 1999. He also served as the Associate Chair for Research & Development in the EE Department from 2003 to 2005. He has published more than 250 journal, conference papers and book chapters in the areas of image/video signal processing, computational neural networks, multimedia system integration and networking (including the recent authored textbook, entitlted "Multimedia Networking: from Theory to Practice," published by Cambridge University Press in 2009). Dr. Hwang received the 1995 IEEE Signal Processing Society's Annual Best Paper Award (with Shyh-Rong Lay and Alan Lippman) in the area of Neural Networks for Signal Processing.

Dr. Hwang served as the Secretary of the Neural Systems and Applications Committee of the IEEE Circuits and Systems Society from 1989 to 1991, and was a member of Design and Implementation of Signal Processing Systems Technical Committee in IEEE Signal Processing Society. He is also a founding member of Multimedia Signal Processing Technical Committee of IEEE Signal Processing Society. He served as the Chairman of the Neural Networks Signal Processing Technical

Committee in IEEE Signal Processing Society from 1996 to 1998, and was the Society's representative to IEEE Neural Network Council from 1996 to 2000. He served as an associate editor for IEEE Transactions on Signal Processing from 1992 to 1994, an Associate Editors for IEEE Transactions on Neural Networks from 1992 to 2000, and an Associate Editor for IEEE Transactions on Circuits and Systems for Video Technology from 1998 to 2006. He is now an Associate Editor for IEEE Trans. on Image Processing and an Editor for Journal of Information Science and Engineering. He is also on the editorial board of Journal of VLSI Signal Processing Systems for Signal, Image, and Video Technology. Dr. Hwang chaired the tutorial committee for IEEE International Conference on Neural Networks (ICNN'96) held in Washington DC, June 1996. He was the Program Co-Chair of International Conference on Acoustics, Speech, and Signal Processing (ICASSP), Seattle 1998. He was the Special Session Co-Chair of ISCAS 2008 and is the Program Co-Chair of International Symposium on Circuits and Systems (ISCAS) 2009. Dr. Hwang is a fellow of IEEE since 2001.

Editor's Selected Paper Recommendation

H. Y. Shutoy, D. Gündüz, E. Erkip, and Y. Wang, "Cooperative Source and Channel Coding for Wireless Multimedia Communications", *IEEE Journal on Selected Topics on Signal Processing,* vol. 1, no. 2, pp. 295-307, August 2007.

Multimedia transmission over wireless networks has seen an increasing demand in the recent years. Many emerging applications require transfer of high data rate sources over unreliable fading wireless links. The key to satisfy the increasing demand for high-quality end-user experience for these applications is to increase the channel reliability and error resilience without sacrificing bandwidth efficiency.

To improve the end-to-end quality for video transmission over wireless networks, there have been many strategies proposed in the literature in both application and physical layer. A recent promising approach in the physical layer is to adopt cooperative communications, in which wireless transmissions overheard by nearby terminals (called relays) are processed and forwarded to the intended destination. The destination combines the different copies of the underlying signal for improved reliability. This spatial diversity scheme provides adaptation to the time-varying channel state by exploiting the network resources rather than limiting itself to the bottlenecks of the point-to-point channel. Several multimedia communication works based on the concept of cooperative communication have been proposed [1]-[3].

Over the past decade, unequal error protection (UEP) for video streaming has been demonstrated as an effective approach to improve the received quality because of unequal importance in different syntax elements and corresponding different level of error protection. The contribution of this work is to take the cross-layer design approach to bring together the UEP technique and the cooperative communications.

The main transmission protocol for scalable coding proposed in this paper is called "layered cooperation", in which the base layer is better protected against channel errors through cooperation, while the enhancement layer is transmitted directly. This paper considers a total bit budget, and optimizes the bit allocation among source coding (for both compression and error-resilience), channel coding, and cooperation.

A theoretical comparison of four modes of operation (direct transmission, layered transmission, cooperative transmission, and layered cooperation) for a Gaussian source is presented in this paper. It is concluded that fixing the modulation scheme limits the potential improvements with layered cooperation. Therefore, adaptive modulation coupled with layered cooperation would be essential for better reconstructed signal quality. A simulation based on H.263+ codec with SNR scalability is also conducted in this work. Results show that cooperation brings significant improvement at low to medium SNR range, and layered cooperation offers distinct improvement over all other modes in the intermediate SNR range.

This paper highlights the benefits of layered video compression together with cooperative coding. The idea has been extended to other applications, such as wireless video multicast scenario [4].

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** The Column Editor recommending this paper is Dr. Guan-Ming Su.*

Network Coding meets Multimedia: Opportunities and Challenges

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Abstract: *Network coding is a new paradigm that extends the traditional routing by allowing intermediate nodes in a network to combine packets, thus increasing the information per packet. Network coding research has –for the most part- considered flows from an information theoretical perspective, or data flows and content distribution scenarios. In this article, we focus on the issues that arise when network coding is applied specifically to multimedia streams. We identify some unique challenges and opportunities that arise in this case, and we give an overview of existing work as well as of open research questions in this area.*

Network Coding

Today's networks are based on the fundamental principle that the network forwards data but the information itself is only processed at the end-systems. Network coding is a research field that emerged over the past decade and breaks this fundamental assumption: it advocates that, in addition to forwarding packets, intermediate nodes should be allowed to also process and re-combine several incoming packets into one or more outgoing packets [1,2]. This breakthrough idea has spawned a significant effort [3,4], initially in the information theory and computer science communities and more recently in the networking and multimedia communities.

From a theoretical point of view, researchers have been studying the design of coding schemes and quantifying the benefits (in terms of throughput, delay and robustness) as well as the cost of network coding, for various traffic scenarios and network topologies. From a practical point of view, researchers are exploring potential applications to practical networking problems at various layers of the protocol stack. For example, in the context of wireless mesh networks, one-hop opportunistic network coding has been shown to increase throughput by mixing packets from different flows into a single packet and by broadcasting over the wireless medium [5,6]. In the context of peer-to-peer content distribution, random network coding has been shown to facilitate distributed scheduling, reduce the download times and increase robustness to node failures [7,8]. In the context of TCP, window-based intra-flow network coding combined with retransmissions has been shown to successfully mask wireless losses from TCP congestion control [9]. There is also a growing body of work within the multimedia community that studies network coding techniques for multimedia and delay-sensitive traffic.

In this short article, we are interested in this last question: what are the novel challenges and opportunities that arise when network coding is applied to multimedia streams in particular? We review some of the existing work and identify some open research questions. *Disclaimer:* this is by no means a complete survey of the subject but is rather a view from our own research perspective.

Network Coding for Multimedia

Below we discuss some fundamental properties of multimedia traffic and their implications for network coding.

Unequal Packet Importance. The fact that different packets, within the same media stream, have different contributions to distortion (due to video content, encoding, or playout deadlines) is well understood in the multimedia community. This fact lies at the heart of multimedia streaming: the unequal importance of packets is used to guide prioritized transmission over a network. Depending on the transmission scenario, available differentiation mechanisms are used to ensure that the most important packets of a particular stream are given priority, thus providing a graceful degradation in the presence of adverse network conditions. One challenge that arises from this fundamental property of multimedia, with respect to network coding, is that network coding, so far, has been agnostic to the content of the packets that are coded together.

In inter-session network coding, the goal is to mix together several packets from different flows, thus increasing the information per packet and eventually the throughput. However, for media streaming it is not only the quantity of delivered packets that matters but also their quality. In [10], we considered video traffic transmitted over wireless networks with opportunistic one-hop network coding [6]. We designed video-aware network coding schemes that take into account both the decodability of network codes by several receivers and the unequal importance of video packets, namely, distortion values and playout deadlines. We demonstrated that these schemes improve the video quality without compromising the MAC throughput. In a sense, this work combined two orthogonal aspects of packet scheduling: (i) network coding to mix packets from different flows and increase throughput and (ii) radio-distortion optimized streaming of packets within the same stream to maximize video quality.

In intra-session network coding, the importance of packets coded together can be taken into account in the composition of generations, *i.e.,* in selecting which packets should be coded together [11]. In [11] Chou et al. also mentioned the idea of incorporating error protection in a generation, either in terms of simple redundancy or in more sophisticated ways, such as priority encoding transmission (PET), which is particularly well-suited for unequal error pretection of layered media.

Different Flow Characteristics and Requirements. Moving from the granularity of packets to the granularity of flows, we observe that entire flows may also have different importance, *e.g.,* due to their traffic characteristics, sensitivity or pricing. When there are multiple media and/or data flows in a system, the question is which flows should be coded together? The rate and delay requirements of media streams should be taken into account when deciding which of them to code together and/or with data flows. For example, we might decide to not code audio traffic, due to its importance and tight delay requirement, but code together similar video streams. Some existing standards (e.g., 802.11e or DiffServ) explicitly specify traffic priorities and allow for separation of media from data traffic, *e.g.,* via priority queuing or preferential dropping mechanisms. However, there is currently no explicit network coding differentiation mechanism. One approach would be to extend the framework in [10] and assign importance to packets based, not only on their distortion value and playout deadline, but also on their traffic type and priority.

Delay Requirements. Another inherent characteristic of media streaming and real-time communications is that they have strict delay requirements, which poses both a challenge and an opportunity when network coding is used. On one hand, network coding increases delay due to additional encoding/decoding and possibly due to waiting at intermediate nodes for enough packets to arrive and be coded together. On the other hand, the increase in throughput can decrease the end-to-end delay. The design of scheduling and coding algorithms can trade-off throughput for delay so as to meet media requirements. In [10], we implicitly took the delay requirement into account, by incorporating it into the importance of a packet. In [12], Nguyen et al. looked a downlink scenario and formulated the scheduling and coding problem within a markov decision process framework, which can also incorporate delay through its contribution to distortion. In the context of generation-based network coding [11], the throughput vs. delay tradeoff can be explicitly controlled by tuning the generation size.

Rate Requirements. One of the main advantages of network coding is that it extends the achievable rate region compared to traditional routing. Interestingly, this has a non-trivial interaction with the rate requirements and adaptation at higher layers. For example, video streams have their own, typically time-varying, rates that need to be adapted to match the rate region offered by the underlying network. Conversely, the rates at the video/application layer affect the availability of network coding opportunities at the underlying network coding layer and thus the achievable region. In our recent and ongoing work [13], we investigate video rate adaptation jointly optimized with network coding.

Network Coding used for Error Correction. Error resilience through redundancy and retransmissions is extensively studied in multimedia streaming. On one hand, network coding can be combined with well-known techniques, such as ARQ [9] and FEC [11,14]. On the other hand, network coding can be seen as an extension of FEC, applied not only at the source but also at intermediate nodes. In the context of peer-to-peer content storage and distribution, random network coding has been shown to be more robust than traditional FEC against failures or departures of nodes [7,8,15]. The intuition is that, in case of a block being lost, network coding produces unique innovative blocks, while FEC-based schemes can replicate the same block (original or redundant). Issues specific to peer-to-peer systems that use network coding and support video include: the need for low delay and continuous playout, the need for unequal error protection [16] and the interaction between video/FEC rates and the underlying rate region achieved by network coding.

Summary and Directions

Media traffic has some characteristics and requirements (such as the difference importance of packets, the strict delay requirements and the time-varying video rate) that introduce unique challenges and opportunities for network coding. We advocate the need for cross-layer design of video-aware network coding schemes that specifically take these features into account. Combining techniques from network coding and media streaming can make the best of both worlds.

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Athina Markopoulou is an assistant professor in the EECS Dept. at the University of California, Irvine. She received the Diploma degree in Electrical and Computer Engineering from the National Technical University of Athens, Greece, in 1996, and the M.S. and Ph.D. degrees, both in Electrical Engineering, from Stanford University in 1998 and 2002, respectively. She has been a postdoctoral fellow at Sprint Labs (2003) and at Stanford University (2004-2005), and a member of the technical staff at Arastra Inc. (2005). Her research interests include media transmission over IP networks, network measurement, network security, and applications of network coding. She received the NSF CAREER award in 2008.

Hulya Seferoglu received the B.S. degree in Electrical Engineering from Istanbul University, Turkey, in 2003 and M.S degree in Electrical Engineering and Computer Science from Sabanci University, Turkey in 2005. She is currently a Ph.D. student in the EECS Dept. at UC Irvine. Her research interests are in the areas of multimedia streaming and network coding.

Encryption-Friendly Multimedia Coding and Communications: Is it Necessary and Possible?

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Abstract— This letter discusses some key issues and problems about multimedia encryption, leading to a call for research in an interesting new direction of encryption-friendly multimedia coding and communications.

Keywords—Multimedia security; multimedia coding and communications; encryption-friendliness; selective encryption

I. Introduction

Security is a very important issue in multimedia communication applications. One of the most frequently demanded security functions in multimedia communications is multimedia content protection, i.e., encryption of plain multimedia data at the sender side and decryption of the encrypted data at the receiver side. Typical applications of multimedia encryption include secure video streaming and secure multimedia file sharing over open networks like Internet.

Since the 1990s, a lot of research efforts have been devoted to find good solutions to multimedia encryption [1-5]. While a large number of multimedia encryption schemes have been proposed in the literature and some have been used in real products, cryptanalytic work has shown the existence of security problems and other weaknesses in most of the proposed multimedia encryption schemes [6].

Some of the problems and weaknesses are related to the essential conflicts between available multimedia encryption techniques and the underlying multimedia coding standards/specifications. As a consequence, it becomes a really challenging task how to design a multimedia encryption algorithm with all the desired features. In most cases, requirements on some features have to be relaxed to make it possible to find a feasible solution to multimedia encryption. It is actually not a surprise since security was not considered as an essential concern at the design stage of any multimedia coding standard/specification. Security-related multimedia standards like JPSEC and MPEG4-IPMPX do exist, but their main goal is to provide a general framework of adding various security tools into the multimedia coding systems, thus enhancing the interoperability of different components and the renewability of broken security tools.

Then, it becomes a question if multimedia coding and communication systems can be made more encryption-friendly. While the necessity of developing such systems has been justified, it is not clear yet if we will finally be able to find a feasible solution. This calls for more research in this interesting new direction.

In the following sections, I first give a brief survey of research on multimedia encryption and then suggest some open research topics about encryption-friendly multimedia coding and communications.

II. Multimedia Encryption: A Brief Survey

There are a number of features we want to get from a multimedia encryption scheme. The most important ones are listed as follows.

- *A reasonable level of security*: the definition of “reasonable” depends on the target applications.
- *Minimized influence on compression efficiency*: in the ideal case, each syntax element should not be influenced and keep its original size after encryption. This feature is called “size preservation”.
- *Format compliance [7]*: encrypted multimedia data can still be decoded by any compatible decoder without the knowledge of the decryption key. This feature serves as a basis of many multimedia communication applications such as post-processing of encrypted multimedia data performed between the sender and the receiver. Some special forms of multimedia encryption, such as *perceptual encryption* [8] and *scalable encryption* [9], also require format compliance.
- *Low computational complexity*: this feature is of particular importance in applications where the computational resource is limited and/or fast encryption speed is demanded. Examples include video encryption on mobile phones

and real-time video encryption occurring at a busy VoD server.

Note that size preservation is much more useful than it looks like. Since the encryption process does not change the size of any syntax element, only a small buffer will be enough to support encryption, which help makes the encryption suitable for those portable devices with very limited memory. Furthermore, size preservation also implies that the location of each syntax element in the multimedia bitstream will not change, either, so on-the-fly encryption and simultaneous encryption at different positions will be possible. One interesting application of this feature is a wiki-like distributed video editing service, where multiple editors can access an online video simultaneously, edit selected parts of the video and lock these parts temporarily or permanently via encryption.

According to at which point encryption can be added into the normal multimedia encoding process, there are three possible approaches to realize multimedia content protection: encryption after encoding, joint encryption-encoding, and encryption before encoding.

The first approach is the most natural and simplest way to realize multimedia encryption, and is often called naïve encryption in the literature. The main problem of this approach is its incapability to maintain format compliance.

The third approach was mainly used for encrypting uncompressed multimedia data like BMP images. Since encryption generally leads to a random-like output, unfortunately, it will be very difficult (if not impossible) to further compress the encrypted multimedia data. Recently an innovative idea was proposed to realize compression after encryption at the encoder side by making the encryption key as useful side information available at the decoder side [10]. This proposal can achieve compression after encryption at the encoder side, but decompression and decryption at the decoder side are still inseparable. As a consequence, format compliance cannot be easily fulfilled, either.

Now it becomes clear that joint encryption-encoding is the most appropriate approach to multimedia encryption. To ensure format compliance, some syntax elements should be left unencrypted. That is, the idea of *selective encryption* (also called *partial encryption*) [11-13] should be used. Note that selective encryption can help get a better balance between requirements on security and complexity. According to which syntax elements are selected for encryption with what encryption technique(s), the following basic encryption methods have been reported in the literature: secret permutations of various syntax elements, FLC (fixed-length codeword) encryption, VLC (variable-length codeword) index encryption, secure entropy coding, "virtual full encryption" working with arithmetic coding or adaptive entropy coding, header encryption, and so forth. Although most of the basic encryption methods can provide acceptable solutions to some applications, there always exist tradeoffs between different aspects of the overall performance of the joint encryption-encoding systems. Some known tradeoffs are shown in the following list.

- Secret permutations can easily ensure format compliance and size preservation, but it is not secure against plaintext attack when being used alone.
- FLC encryption can absolutely maintain format compliance and size preservation, but it is unable to provide a very high security level.
- VLC index encryption can be configured to ensure format compliance and security, but it always influences the compression efficiency (though slightly in most cases) and cannot maintain size preservation.
- Most secure entropy coders are either insecure against chosen plaintext attack or unable to offer a better performance than naïve encryption (see Sec. IV.B of [19] for an example of the second case). Secure entropy coding is often unable to keep format compliance.
- "Virtual full encryption" and header encryption cannot ensure format compliance sometimes.

It has been found that most of the tradeoffs are related to essential conflicts between the encryption techniques involved and the underlying multimedia coding standards/specifications. In fact, plenty of cryptanalysis work on error concealment attacks [8, 14-17] has clearly shown that selective encryption cannot conceal all perceptual information effectively, because some perceptual information is coded in such a way that it cannot be encrypted effectively as long as format compliance and/or low computational complexity has to be maintained. This justifies the necessity of the quest for encryption-friendly multimedia coding and communication systems.

Another interesting observation about selective encryption is the impossibility to offer 100% security. Assuming we choose to encrypt ALL sign bits of ALL DCT coefficients in an MPEG video, then only 50% sign bits will be flipped in an average sense. A similar phenomenon was demonstrated with a uniform quantizer in [12]. This problem is obviously related to the relatively short sizes of some independently coded syntax elements (such as sign bits) in coded multimedia data.

III. Open Research Topics

According to our discussion given in the last section, I feel the following research topics will be of particular importance for the research on encryption-friendly multimedia coding and communications, and for multimedia encryption as a whole.

A. New objective performance metrics tailored for the specific needs of multimedia encryption like ESS proposed in [18].

A number of metrics are needed to answer at least the following four research questions:

- How to better measure the visual quality degradation of selectively encrypted multimedia data?
- How to better evaluate the performance of error-concealment attack?
- How to define multimedia-friendliness for different multimedia coding and communications systems?
- How to better evaluate the overall performance of a multimedia encryption solution to a specific application?

B. Measurable technical limits of current multimedia encryption techniques.

Based on the objective metrics developed for the last suggested research topic, we can quantitatively study how far different multimedia encryption techniques can go on a given multimedia coding and communication system.

C. Development of a general-purpose benchmarking system for evaluating the overall performance of different multimedia encryption algorithms.

An interface between various multimedia encryption algorithms and multimedia coding/communication platforms should be designed to ease the usability of such a benchmarking system.

D. Possible amendments to existing multimedia coding/communications systems or new paradigms of multimedia coding/communications, for the benefit of enhancing encryption-friendliness.

One possible direction is to further improve the compression-after-encryption scheme proposed in [10].

E. Joint watermarking-encryption and joint fingerprinting-decryption.

By combining watermarking with encryption, we will be able to not only enhance encryption-friendliness of a multimedia coding/communication system, but also its friendliness to other security factors as a whole. In addition, it is also possible to use the embedded side information to enhance the security of the multimedia encryption algorithms.

F. Information retrieval and other signal processing operations performed on encrypted multimedia data.

It certainly makes sense to do information retrieval and signal processing operations on selectively encrypted multimedia data. Furthermore, it is also possible to do the same task on fully encrypted multimedia data, according to some recent research in cryptography. Embedding watermarks in encrypted data might also be yet another way to achieve the same goal.

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Focused Technology Advances Series

Latest Advances in Wireless Multimedia Communications and Networking

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In the technology world, 2009 started with CES, the world's largest consumer technology tradeshow, which hopefully will help lead the way to economic recovery. Multimedia technology is again at the center of the new trends unveiled at the CES such as 3D HDTV, Internet TV, advances in OLED, new smart phones, and digital entertainment.

There is no doubt that multimedia has become an integral part of our daily life, and the technology is still evolving rapidly. This column will focus on the communication aspect of multimedia technologies. It will highlight some of the recent advances in wireless multimedia communications and networking.

With the wide deployment of wireless access networks (e.g., WLAN and 3G networks) and various wireless backhaul technologies (e.g., wireless mesh networks (WMN) and WiMax), much research and development effort has been devoted to the problem of streaming multimedia data over wireless networks. The unique characteristics of wireless networks such as limited and dynamic bandwidth, unreliability, interferences, and user mobility, however, pose a number of significant challenges for high quality video streaming. Advanced technologies are deemed necessary to address these challenges to enable ubiquitous multimedia experience. We highlight a few hot topics in recent years.

Multi-hop wireless mesh networks are emerging as a promising technology that has applications in metro-area Internet access (e.g., Google's WiFi muni mesh network), public safety, disaster response, battlefield, etc. Delivering high quality video over WMN however remains a challenge. Video streaming over mobile ad hoc networks and wireless mesh networks have been studied recently, with various proposed server client techniques such as multiple description coding and path diversity [1][2]. Cross-layer approaches have also been explored to improve the transport efficiency [3]. Traditional client-server model based approaches however do not scale very well. Content delivery technologies such as content distribution network (CDN) and emerging peer-to-peer (P2P) technologies can be leveraged to address some of the challenges. CDN can reduce the traffic within the network, shorten the users' startup delay, and improve the user viewing quality. There is a significant difference in WMN though. In the Internet, the bottleneck is either at the server or at the client. In wireless mesh networks, the bottleneck may be within the network. So it is necessary to consider caching within the backhaul network too [5]. P2P video streaming has recently emerged as an alternative with low server infrastructure cost and good scalability. P2P streaming in WMNs environment is studied in [4], where a central server is used to find the best route for each client to other peers that minimizes the received video distortion. A unified P2P and cache framework for high quality video-on-demand services over infrastructure WMNs is proposed in [5]. To increase the capacity of the streaming services and ensure high video quality, the video content is cached at selected wireless mesh access points in the mesh network. Peer devices also help each other on video downloading in a best effort manner to further balance the network traffic load and reduce the network resource consumption along the path from the source to the client. Cache server selection, wireless multi-hop path selection, peer selection, and admission control are among the important issues to be addressed, typically using a cross-layer approach.

Another important recent trend is the application of the emerging network coding paradigm [6] to wireless media streaming. By providing coding capability at intermediate nodes, network coding has been recently shown to improve network throughput in broadcast/multicast wireless networks [7][8]. For example, [7] implemented opportunistic listening and a coding layer between IP and MAC that is used to detect coding opportunities and pack packets from different flows into a single transmission, thus increasing the throughput of wireless mesh networks. Rate-distortion optimization on a packet-by-packet basis for a wireless router has also been considered in network-coding schemes for video streaming [8][9]. Study of practical network coding techniques for streaming media applications has shown that randomized network coding can be designed to be robust to random packet loss, delay, as well as any changes in network topology and capacity [10], and that randomized coding can be used for both elastic content distribution [11] and P2P live content streaming [12] to significantly improve the throughput.

Exciting development in wireless multimedia communication has emerged in recent years. Looking forward, more sophisticated and powerful techniques that leverage new ideas such as network coding, caching, P2P and cross layer optimization are expected to be developed.

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IG Corner: Cross-Layer Multimedia Communications

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In recent years, pervasive computing devices such as laptop computers, PDAs, smart phones, and wearable computers have been ever growing in popularity and capability. Therefore, there has been a strong user demand for providing multimedia services to those devices such as i-Tunes, PPlive, MSN, and YouTube. However, bringing delay-sensitive loss-tolerant multimedia services to the current Internet is a very challenging task due to the fact that the original design goal of the Internet is to offer simple delay-insensitive loss-sensitive data services with little QoS consideration. Therefore, this shift of design goal urges us to rethink the current Internet architecture and develop a new design methodology for multimedia communications over the current and future wired and wireless Internet.

So far, cross-layer design has been thought as one of the most effective and efficient ways to provide Quality of Service (QoS) over the Internet, especially the wireless Internet, and has been receiving many research efforts. The basic idea of cross-layer design is to fully utilize the interactions among design variables (system parameters) residing in different network functional entities (network layers and nodes) to achieve the optimal design performance of time-varying networks. However, most existing cross-layer designs are mainly focusing on improving network QoS in terms of throughput, delay, and jitter; user perceived quality has long been ignored in current multimedia communications. Furthermore, most existing cross-layer designs take the piecemeal approach and lack of a systematic framework of modeling and optimization, leading to suboptimal solutions and/or proprietary designs. Since for most multimedia communications, user perceived quality is the ultimate goal of communications. Thus, quality-driven cross-layer design is one of the enable technologies to the next-generation quality-aware service-oriented multimedia networks.

In order to achieve the global optimality of cross-layer design, we also need to consider design variables and the interactions among them as much as possible. However, more does not necessarily mean better. The more design variables we considered, the more difficult is orchestrating them to work harmonically and synergetically. From the point of view of system optimization, the number of design variables increases, the size of state space of the objective function will increase exponentially, making the optimization problem intractable. To overcome this problem, an often-used approach is to reduce the size of the problem at the system modeling phase and then solve the simplified problem by using various optimization algorithms such as gradient-based local search, linear/nonlinear programming, genetic algorithm, exhaustive search, and artificial neural networks. However, reducing a high-dimensional cross-layer optimization problem to a low-dimensional problem in the system modeling phase raises a series of questions:
1) how to evaluate the fidelity of the simplified problem compared with the problem as what it should be; 2) how to evaluate the quality of the suboptimal solution to the global optimum; and 3) how to evaluate the robustness of a solution, that is, whether the solution can guarantee the predictable sound results at all possible circumstances. Unfortunately, at the time of this writing, we have no clear answers to all these three questions.

Moreover, reducing the size of the problem in the problem formulation means that only part of the current Internet architecture can be considered, causing a shift of the design goal of multimedia services from the best user experience to some layer-specific performance metrics such as distortion at the application layer, delay at the network layer, and goodput at the MAC/PHY layer. This shift of the design goal may cause Ellsberg Paradox, where each individual design variable makes its good decision for maximizing the objective function, respectively. But the overall outcome violates the expected utility function. In other words, breaking a big problem into several smaller problems in the system modeling phase can only increase the solvability of the original problem but cannot guarantee it is a good solution. The Ellsberg Paradox also tells us that the traditional additive measure such as probability measure may no longer hold in the context of cross-layer design due to the possible strong coupling (interdependency) among design variables.

Essentially, all aforementioned difficulties in the area of cross-layer design in multimedia communications are caused by lacking of a methodological foundation and an in-depth understanding of cross-layer behaviors. Therefore, a flexible yet scalable theoretical cross-layer framework has to be developed to accommodate all major design variables of interest, spanning from application layer to physical layer, for delay-bounded multimedia communications over the next generation Internet. Moreover, the complexity of cross-layer optimization should be examined, and complexity-scalable optimization needs to be further investigated.

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MMTC COMMUNICATIONS & EVENTS
10 Quick Ways to Contribute to MMTC

One of the most popular questions we have received from our members is “how can I contribute to MMTC?” To address this common concern, we put below 10 quick ways to contribute to MMTC for your reference, please do not hesitate to send your questions or concerns to either MMTC Chair (qianzh@cse.ust.hk) or TCPI team (haohong@ieee.org).

- Participate in the TCPI efforts, please send an email to TCPI team, you will be contacted shortly;
- Propose constructive proposals or suggestions to MMTC Chair or TCPI team;
- Nominate new active members to MMTC, please send his/her CV and your reasons of nomination to either MMTC Chair or TCPI team;
- Propose new Interest Group in MMTC, or participate in the existing IG events, please contact the MMTC Chair or IG Chairs;
- Volunteer to help with the MMTC E-Letter or MMTC workshops, please contact TPCI team for such opportunities;
- Volunteer to help with the future GLOBECOM/ICC (as TPC members or reviewers), please send email to the MMTC Chair;
- Contribute short paper, letter, or paper/book review to MMTC E-Letter;
- Submit high-quality papers to the MMTC sponsored symposium in future GLOBECOM/ICC conferences;
- Attend MMTC regular meetings in GLOBECOM/ICC and provide suggestions and feedbacks;
- Disseminate the MMTC E-Letter and other events information to your friends and colleagues to let more people know of this organization and participate.

Call for Best Paper Award Nominations

Call for Nominations (2009 MMTC Best Journal Paper Award and Best Conference Paper Award) The Multimedia Communications Technical Committee, IEEE Communications Society, calls for nominations for 2009 MMTC Best Journal Paper Award and Best Conference Paper Award.

Eligibility for 2009 MMTC Best Journal Paper Award:

Any paper that is published in the multimedia communication area and appeared in an IEEE ComSoc journal/magazine in 2007 and 2008, is eligible. The list of eligible journals includes IEEE Journal Selected Areas of Communications, IEEE Transactions of Multimedia, IEEE Multimedia.

Eligibility for 2009 MMTC Best Conference Paper Award:

Any paper that is published in the multimedia communication area and appeared in the proceedings of an IEEE ComSoc-sponsored conference/workshop/symposium, in 2007 and 2008, is eligible. The list of eligible conferences includes IEEE CCNC, IEEE ICME, IEEE Globecom, IEEE ICC and IEEE Infocom.

Nominations have to be sent by email to **wu@ece.ufl.edu**, with subject line **'MMTC-BPA Nomination'**. The nomination should include the complete reference of the paper, author contact information, a brief supporting statement (maximum one page), the name of the nominator, and an electronic copy of the paper if possible.

The hard deadline for paper nomination is set to **Feb. 15th, 2009**.

An independent subcommittee has been created to evaluate nominated papers, and the Best Journal Paper Award and Best Conference Paper Award 2009 will be presented at ICC 2009, by the MMTC chair, to one of the authors of the best papers. The authors will be notified at least 6 weeks prior to ICC 2009.

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Workshop on IPTV Technologies and Multidisciplinary Applications
(Co-located with ConTEL 2009 - The 10th Int'l Conference on Telecommunications)

Website:	http://www.contel.hr/2009/workshops
Dates:	June 8 - 10, 2009
Location:	Zagreb, Croatia
Submission Due:	**Feb. 19, 2009**

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