# IEEE COMSOC MMTC E-Letter

## IEEE MULTIMEDIA COMMUNICATIONS TECHNICAL COMMITTEE

### E-LETTER

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[IEEE Communications Society](http://www.comsoc.org/~mmc/)
Greetings to our MMTC members,

Hope all of you have a wonderful summer or winter!

You may have noticed that IEEE ICC 2010 is to be held during 23 - 27 May 2010 in Cape Town, South Africa. This will be the first time that ICC has been held in Africa and with Cape Town being voted one of the most beautiful cities in the World. Conference participants will have a wide range of exciting options to add to their conference tour: hikes up the famous Table Mountain; whale watching; tours of the Cape Winelands, Robben Island and even CapePoint where two oceans meet. In this idyllic setting, you will be able to network with friends, colleagues, customers and vendors from around the world. The theme of IEEE ICC 2010, "Communications: Accelerating Growth and Development", is specifically matched to the conference location.

For ICC 2010, Multimedia Services, Communication Software and Services (MCS) Symposium is the ONLY symposium that is fully sponsored by MMTC, hence we encourage all our members to submit your Multimedia related papers to MCS symposium, where a Co-Chair (Prof. Zhu Li) and many TPC members recommended from MMTC would handle all the review process of the multimedia related paper submissions.

Another important issue I would like to draw your attention is the nomination for new EiC of MMTC E-Letter. First, I would like to express my sincere thanks for the great efforts from Haohong (current EiC) and all the editors (Philippe, Chonggang, Guan-Ming, Shigu, and Antonios). Without their great efforts, we cannot continue our E-Letter in such an impressive way. You can check all the E-Letters from http://www.comsoc.org/~mmc/index.asp. 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 term of the current editor-in-chief (EiC) of IEEE MMTC E-Letter is coming to an end by Jan. 2010, and the MMTC has set up a nominating committee to assist in selecting the next EiC. The EiC is responsible for maintaining the highest editorial quality, for setting technical direction of the papers published in E-Letter, and for maintaining a reasonable pipeline of articles for publication.

I would like to take this opportunity to encourage all of you to nominate the capable candidates (including yourself) to nominating committee chair by Nov. 15 2009.

Thank all of you for your continuous great support to MMTC.

Best,
Qian Zhang
Chair of Multimedia Communication TC of IEEE ComSoc
In this Issue, first I would like to thank Dr. Guan-Ming Su (Marvell Semiconductors, USA), who expands our traditional column of Distinguished Position Paper Series into a special issue, namely Special Issue on Future Research Directions, where four position papers contributed from world top-class research teams (of 4 different countries) are presented. Please check out this special issue starting from Dr. Su’s Guest Editorial on page 6.

In the Editor Recommended Paper Column, a paper to appear in ICNP’09 conference on video multicast services is recommended by the Column Editor, Dr. Chonggang Wang (NEC Laboratories America, USA). The website for downloading a pre-print version of the paper is also provided.

After that, a short position paper delivered by Prof. Nabil J. Sarhan (Wayne State University, USA) briefly overviews Mobile TV services and discusses the future research directions in this topic as well as the mobile video-on-demand services.

In the focused technology column, Dr. Ivan V. Bajić and Ms. Xiaonan Ma (Simon Fraser University, Canada) introduce a scalable coding solution for the distributed telepresence performances applications in the new media and performing arts community.

As always, I thank all Editors of the E-Letter, and our authors to make this issue successful.

Thank you very much.

Haohong Wang
Editor-in-Chief, MMTC E-Letter
**IEEE COMSOC MMTC E-Letter**

**HIGHLIGHT NEWS & INFORMATION**

*Did You Submit Your Papers Yet?*

The paper submission deadline (**September 10**) for IEEE ICC 2010 is approaching, we urge all members to submit papers to support this event.

**What:** IEEE ICC 2010  
**When:** May 2010  
**Where:** Cape Town, South Africa  
**Submission Deadline:**  
**September 10, 2009**  

**Action:**  
Submit your papers via EDAS TODAY!!!

**Benefits:**  
- Enjoy beautiful scenic view at Cape Town, South Africa;  
- Meet friends, exchange ideas and build/renew connections in this world largest conference of communications;  
- Your paper would be considered in a Best Paper Award competition, once accepted;  
- Attend MMTC meeting during the conference to find out more opportunities in this TC.

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**How to Submit Papers to ICC 2010**

For ICC 2010, **Multimedia Services, Communication Software and Service Symposium (MCS)** is the **ONLY** symposium that is fully sponsored by MMTC, hence we encourage all our members to submit your Multimedia related papers to MCS 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 ICC 2010:

1. Go to EDAS website: [http://edas.info/index.php](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 “ICC’10” and click the icon on the rightmost column;  
4. In the list of ICC’10 symposia, please select the “ICC’10 MCS” and click the icon on the rightmost column;  
5. Start the normal paper submission process.
Call For Nominations: Editor-In-Chief IEEE MMTC E-Letter

The term of the current editor-in-chief (EIC) of IEEE MMTC E-Letter is coming to an end by Jan. 2010, and the MMTC has set up a nominating committee to assist in selecting the next EIC.

Nominations, including self nominations, are invited for a one-year term as IEEE MMTC EIC, beginning from Feb. 2010. The EIC appointment may be renewed once, after review. This is an entirely voluntary position, but IEEE MMTC will provide appropriate administrative support.

The EIC is responsible for maintaining the highest editorial quality, for setting technical direction of the papers published in E-Letter, and for maintaining a reasonable pipeline of articles for publication. He/she has final say on acceptance of papers, size of the Editorial Board, and appointment of Editors. The EIC is expected to adhere to the commitments expressed in the IEEE Publishing Rights and Responsibilities Policy.

Nominations should include a brief statement of why the nominee should be considered. As MMTC E-Letter is a multidisciplinary field and it covers fast evolving research topics, the nominee should comment on plans to help E-Letter cope with the evolving MMTC research topics. Self-nominations are encouraged. The deadline for submitting nominations is the 15th of November, 2009.

Please send all nominations to the Nominating Committee Chair, Dr. Qian Zhang <qianzh@cse.ust.hk>.

The Search Committee members are:

- Qian Zhang (Chair), Hong Kong University of Science & Technology, China
- Gary Chan, Hong Kong University of Science & Technology, China
- Heather Yu, Huawei Technology, USA
- Haohong Wang, TCL-Thomson Electronics, USA
- Majid Merabti, Liverpool John Moores University, UK
- Rob Fish, Mformation, USA
- Wenjun Zeng, University of Missouri, USA
Multimedia communication has become a fundamental service among the communication network service providers. We all experience these brand new forms of multimedia presentations brought by the emerging multimedia communication technologies. On the other hand, end users always want higher resolution, more fluent playback, stronger security, and real-time interactivity. To satisfy those requirements, there are many technical issues needed to be overcome under the current infrastructure. In this special issue, we invite world class research groups to discuss the new research directions of multimedia communications and contribute their promising solutions. Due to the page limitation, we will publish this special issue in two parts. The first part is published in this issue and the other part is scheduled to publish in Q4 2009.

The first article, “R&D for wireless multimedia communications in next decade,” by Man-On Pun and C. -C. Jay Kuo, discusses the future research issues on the topics of wireless multimedia communications. The authors start to address current R & D in video compression technologies, including H.264 AVC/SVC/HVC and H.265, the communication technologies, including HSPA+, IEEE 802.11n, WiMAX and LTE, and the energy and security issues in multimedia communications. The authors also enumerate the potential future research directions in both theoretical framework and advanced applications.

Real-time interactive multi-user wireless multimedia transmission systems are the challenging and critical services in the resource-limited wireless networks. The second article, “Transmission control games for scalable video coding in wireless multimedia networks” by Jane W. Huang, Hassan Mansour, and Vikram Krishnamurthy, introduces the advantage of adopting scalable video coding in wireless networks and indicates that a stochastic dynamic game theory can be a useful tool. The authors also provide a framework and formulate this system as an optimization problem. The corresponding algorithms and results are also stated to illustrate the effectiveness.

As the growing demands for internet multimedia communications, efficient approaches to utilize the bandwidth to deliver multimedia content to multiple subscribers with satisfactory Quality of Service simultaneously become imperative. The third article, “On application layer multicast” by Nirwan Ansari and Nei Kato, presents the concepts and advantages of application layer multicast (ALM). The authors overview the state-of-the-art techniques for designing efficient application layer multicast trees and examine the pros and cons. At the end of this article, the challenges and the open issues on the topics of ALM are discussed.

Applying forward error correction code to video transmission over unreliable channels is one solution to provide desired video quality. However, using conventional fixed-rate code may not be suitable in a time-varying channel conditions since the multimedia may either be over- or under-protected. The fourth article, “Efficient transmission of video over highly error-prone channels” by Mohammed Ghanbari and Martin Fleury, highlights the advantages of deploying the newly developed rateless channel code (fountain channel code) in the considered scenario. Then, the authors demonstrate how to apply the fountain code on real-time H.264 video transmission system with data partitioning mechanism enabled. The effectiveness of the proposed approach is demonstrated via simulation results.

We would like to express our grateful thanks to all the researchers who contribute to this special issue. We believe the articles in this special issue will provide an insightful overview of the existing literature, and sketch the roadmap toward the next generation of multimedia communication framework. We also hope these papers can inspire readers new research ideas and disruptive applications in the area of multimedia communication.
Guan-Ming Su received the B.S.E. degree in electrical engineering from National Taiwan University, Taipei, Taiwan, in 1996 and the M.S. and Ph.D. degrees in electrical engineering from the University of Maryland, College Park, USA, in 2001 and 2006, respectively.

He was with the Research and Development Department, Qualcomm, Inc., San Diego, CA, during the summer of 2005, and with ESS Technology, Fremont, CA, in 2006. He is currently with video R&D department in Marvell Semiconductor, Inc., Santa Clara, CA. His research interests are multimedia communications and multimedia signal processing.

Dr. Su is an associate editor of Journal of Communications and a guest editor in Journal of Communications special issue on Multimedia Communications, Networking, and Applications. He serves as the Publicity Co-Chair of IEEE GLOBECOM 2010.
The rapid convergence of the voice, data and video networks is driving the wireless communications industry to undergo an unprecedented paradigm shift. Similar to voice services, multimedia services are now an indispensable mobile service component. This is exemplified by Apple’s recent success in selling more than one million units of its next-generation iPhone 3GS on the first debut weekend in June even when the recession is in full swing. In this article, we will review and envisage the R&D development on multimedia technologies and the underlying wireless communication networks, respectively.

We begin with the advancements in multimedia technologies. Clearly, applications and contents will be the key driving factors in the next generation broadband wireless multimedia communications. The H.264/AVC video coding standard [1] has been well received by the multimedia industry. Its scalable extension, namely, H.264/SVC [2], has a great potential since it can meet different quality of services (QoS) requirements easily and offer robust performance in the presence of unreliable wireless channels. On the other hand, the great flexibility of H.264/SVC does impose challenges in system integration for the operators. A substantial amount of industrial R&D is needed to have a good cross-layer design to exploit the advantages of H.264/SVC fully.

High definition (HD) video with a data rate of 10Mbps or higher becomes more popular nowadays. However, H.264/AVC and H.264/SVC are still not effective in the coding of high definition content. New standardization efforts have been initiated along this line, including ITU-T H.265 [3] and MPEG High-performance Video Coding (HVC) [4] activities. A key objective is to reach the same video quality while reducing the bit rate of the state-of-the-art coding standard by one half. Several recent R&D efforts have shown good promises in achieving this ambitious goal. It seems feasible that we will see another quantum jump of the video coding performance after H.264, which will enable wireless HD video within the next decade.

Location-based and network-based services and applications will emerge and become popular. Mobile devices serve as capturing instruments, recording numerous video and audio for local and/or remote storage. Besides, queries are sent to the data center through networks to search information of interest in form of multimedia. Thus, content-based media search, indexing and retrieval will be in great demand. Despite its technical challenges, some major breakthrough is expected along this line in the next decade. Another interesting area to observe is the development of social networks such as Facebook and Twitter, where multimedia data will become more dominant. Clearly, real-time high quality multimedia information sharing among user groups will create a lot of traffic. Network coding [5] offers an excellent framework to efficient video multicast in theory. Its practical implementation [6] is expected in the near future.

To support these advanced multimedia services over mobile broadband networks, the wireless communications industry has to develop innovative technologies capable of providing fast multimedia services over all-IP wireless networks. The existing technologies such as High Speed Packet Access (HSPA) and IEEE 802.11g can typically deliver maximum real-world throughput on the order of 10Mbps in the downlink transmissions, which is increasingly seen as inadequate. To the rescue, HSPA+ is currently being implemented to double the throughput as an affordable and incremental upgrade. In contrast, the emerging IEEE 802.11n improves upon IEEE 802.11g by adopting some latest technologies such as the multiple-input multiple-output (MIMO) technology. By exploiting multiple antennas and bonding wider bandwidth channels, IEEE 802.11n is expected to increase the throughput to the 100Mbps range. However, the mobile data usage is foreseen to strain the current network capacity in a few years time.

This growing demand for even higher throughput, lower latency and longer transmission range has motivated substantial fourth-generation (4G) standardization activities on IEEE 802.16 (also
known as WiMAX) and Long-Term Evolution (LTE). Notwithstanding their different historical origins and subsequently backward compatibility constraints, both WiMAX and LTE are designed to achieve significantly higher data rates compared to any existing standards. More specifically, IEEE 802.16m and LTE-Advanced (LTE-A) are targeting at a peak data rate of 1Gbps [7],[8]. This stringent data-rate requirement demands a wide array of novel technologies to revolutionize the current network design. For instance, both WiMAX and LTE utilize the scalable orthogonal frequency division multiple access (OFDMA) as the air interface technology for its robustness against multipath fading and flexibility in dynamic resources allocation. Furthermore, both capitalize on sophisticated MIMO technologies coupled with advanced scheduling schemes. In addition, to enhance the QoS of cell-edge users, technologies such as relaying and collaborative transmission among multiple base stations are being actively investigated. Finally, one emerging technology developed for LTE-A is carrier aggregation. In carrier aggregation, multiple 20 MHz carrier components in different frequency bands can be flexibly aggregated together into wider bandwidth for very fast data transmission. It is expected that the WiMAX and LTE standardization activities will continue expediting the R&D efforts on these cutting-edge technologies.

However, speed is not the sole feature to characterize future wireless multimedia communications. A few distinguishing features are highlighted in the following. First, due to the broadcasting nature of wireless communications, information security has become an imminent concern. In addition to the conventional cryptography techniques, information-theoretic techniques are being developed to strengthen information security by exploiting wireless channels [9]. Second, “green” design to dramatically improve the system energy efficiency has become increasingly important for high data-rate communications. This has to be achieved by developing novel energy-efficient hardware design and network architecture [10]. Third, to achieve higher spectral efficiency via multiplexing or frequency reuse, the importance of interference management appears even more prominent. More sophisticated signal processing technologies for interference suppression or cross-layer approaches are useful to circumvent the problem [11]. Last but not the least, despite the success in analyzing different theoretical aspects of point-to-point links, little work has been achieved in developing theoretical frameworks for wireless networks comprised of multiple links. Therefore, more fundamental understandings on wireless networks such as their capacity and practical performance will play an essential role in improving holistic network design [12].

Ubiquitous wireless multimedia communications have been envisioned by many pioneers for years. With the arrival of 4G networks, high-quality and robust wireless multimedia communications are no longer science fiction in the near future.

References


Man-On Pun received the Ph.D. degree in Electrical Engineering from the University of Southern California (USC) in 2006. He is a research scientist at Mitsubishi Electric Research Labs (MERL) in Boston, MA. He held research positions at Princeton University, Princeton, NJ from 2006 to 2008 and Sony Corporation, Tokyo, Japan from 1999 to 2001. He received the best paper award - runner-up from the IEEE Conference on Computer Communications (Infocom), Rio de Janeiro, Brazil in 2009, the best paper award from the IEEE International Conference on Communications (ICC), Beijing, China in 2008 and the best student paper award from the IEEE Vehicular Technology Fall Conference (VTC-Fall), Montreal, Canada in 2006.

C.-C. Jay Kuo received the B.S. degree from the National Taiwan University, Taipei, in 1980 and the M.S. and Ph.D. degrees from the Massachusetts Institute of Technology, Cambridge, in 1985 and 1987, respectively, all in Electrical Engineering. He is Director of the Signal and Image Processing Institute and Professor of Electrical Engineering, Computer Science and Mathematics at the University of Southern California (USC). His research interests are in the areas of digital image/video analysis and modeling, multimedia data compression, communication and networking, and biological signal/image processing. Dr. Kuo has guided 94 students to their Ph.D. degrees and supervised 20 postdoctoral research fellows. He is co-author of about 160 journal papers, 780 conference papers and 9 books. He delivered more than 420 invited lectures in conferences, research institutes, universities and companies. He is Editor-in-Chief for the *Journal of Visual Communication and Image Representation* (an Elsevier journal), and has served as Editorial Board member for about 10 international journals. Dr. Kuo received the National Science Foundation Young Investigator Award and Presidential Faculty Fellow Award in 1992 and 1993, respectively. He is a Fellow of IEEE and SPIE.
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Transmission Control Games for Scalable Video Coding in Wireless Multimedia Networks

Jane W. Huang, Hassan Mansour and Vikram Krishnamurthy (IEEE Fellow),
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Video transmission services are generally characterized by bandwidth-intensity, delay-sensitivity, and loss-tolerance. The provision of these services to multiple users over wireless networks is faced with the bandwidth fluctuation and the high rate of packet loss associated with the wireless channel. The time-varying nature of the channel and video content give rise to the need for scalable media which can drop the video bit-rate without significantly sacrificing the decoded video quality. As a result, transmission policies should consider the scheduling, rate adaptation, and buffer management of each user.

Scalable video coding (SVC) provides a quality scalable video bitstream which ensures a graceful degradation in video quality when faced with channel fluctuations [1]. In SVC, quality scalability is achieved using medium-grained or coarse-grained scalability (MGS/CGS) where scalable enhancement packets deliver quality refinements to a preceding layer representation by re-quantizing the residual signal using a smaller quantization step size and encoding only the quantization refinements in the enhancement layer packets [2]. The variation of the video bit rate and distortion over time can be modeled as a stationary first-order Gauss Markov process [3]. Therefore, we quantize the range of that variation to achieve a video variability state space in which the video states constitute a Markov process.

In wireless multimedia networks with reduced functionality base stations (no central authority) and autonomous multimedia users, game theory can be naturally applied to achieve the decentralized scheduling policy. Based on the opportunistic scheduling, each user accesses the spectrum hole in a decentralized manner. The rate adaptation in the wireless multimedia system is then able to be formulated as a constrained general-sum switching control dynamic Markovian game [4, 5]. A switching control game [6, 5, 7] is a special type of dynamic game where the transition probability in any given state depends on only one player. It is known that the Nash equilibrium for such a game can be computed by solving a sequence of Markov decision processes.

Optimization Problem: Given the policies of the other users, the optimal transmission policy of user \( i \), \( p_i^* \), is chosen so as to maximize the overall expected discounted reward subject to its constraint. The computation of the optimal policy \( p_i^* (i=1,2,...,K) \) can be formulated as an optimization problem which can be written as follows [8].

\[
p_i^* = \max_{p_i} \{ p_i \text{ s.t. } R_i(p_i) \leq D_i \}
\]

where \( R_i(p_i) \) denotes the expected discounted reward function of user \( i \) under policy \( p_i \) and \( D_i(p_i) \) denotes its expected discounted delay. \( D_i \) is the latency constraint parameter of user \( i \).

We propose a value iteration optimization algorithm to obtain the Nash equilibrium policy in such a game. Moreover, given the assumptions that the set of policies is non-empty, the transmission reward is supermodular, the holding cost is submodular, and the transition probabilities are stochastically increasing, then the Nash equilibrium transmission policy of each user is a randomization of two pure policies with each policy nondecreasing on the buffer state occupancy [9].

Based on the structural result on the Nash equilibrium policy, we propose a stochastic approximation algorithm to reduce the calculation complexity. The stochastic approximation algorithm is applied to find the
Nash equilibrium policy with simulation results shown in Fig. 1.

![Monotone transmission policy](image)

Figure 1: The Nash equilibrium transmission control policy obtained via stochastic approximation algorithm for user 1. The result is obtained with a 70 ms transmission delay constraint when the states of user 2 are $h_2=1$ and $b_2=1$. The Transmission policy is monotone nondecreasing on its own buffer state.

![Football Y-PSNR](image)

Figure 2: Result of the transmission of the Football sequence comparing the performance in terms of video PSNR and buffer utilization between the proposed switching control game policy and the myopic policy with 80 ms delay constraint. The result shows that the proposed switching control game policy performs better than the myopic policy.

In order to demonstrate the effectiveness of the proposed dynamical switching control game algorithm, we compare its performance to that of a myopic policy that selects at each time slot the user action that maximizes the video PSNR while satisfying the transmission buffer delay constraint. Fig. 2 shows the results for the Football sequence when simulating the transmission of two users Foreman and Football for both the proposed switching control game policy and the myopic policy. The figure shows that under the same channel conditions, the proposed policy has better Y-peak signal-to-noise ratio (Y-PSNR) performance and transmission buffer utilization for both users.

References
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Threshold Policies,” Accepted by IEEE Transactions on Communications, 2009.

Jane Wei Huang (S’07) received her bachelor's degree from Zhejiang University, China in 2005, and M.Phil. from Hong Kong University of Science and Technology, Hong Kong in 2007. She is currently a Ph.D. student in the University of British Columbia. Her research interests include game theory, Markov decision process, cognitive radio, and sensor networks.

Vikram Krishnamurthy (S’90-M’91-SM’99-F’05) was born in 1966. He received his bachelor's degree from the University of Auckland, New Zealand in 1988, and Ph.D. from the Australian National University, Canberra, in 1992. He is currently a professor and holds the Canada Research Chair at the Department of Electrical Engineering, University of British Columbia, Vancouver, Canada. Prior to 2002, he was a chaired professor at the Department of Electrical and Electronic Engineering, University of Melbourne, Australia, where he also served as deputy head of department. His current research interests include computational game theory, stochastic dynamical systems for modeling of biological ion channels and stochastic optimization and scheduling.


Hassan Mansour (S’99) received the B.E. degree in Computer and Communications Engineering from the American University of Beirut, Lebanon, in 2003, and the M.A.Sc. degree in Electrical and Computer Engineering from the University of British Columbia, Vancouver, BC, Canada, in 2005, where is currently working toward the Ph.D. degree and pursuing research in the field of multimedia communication and video coding. He has been a member of the Standards Council of Canada committee on MPEG development (ISO/IEC JTC1/SC29) since 2004.
Abstract
Application Layer Multicast (ALM) is a promising technology for constructing content delivery systems across the Internet with low cost and scalability. Unlike the conventional, well-known IP Multicast paradigm, ALM uses unicast to virtually realize multicast communications. In this article, we present a survey of state-of-the-art techniques for designing efficient ALM trees and the challenges we are facing.

Background
As opposed to IP Multicast (Fig. 1), in which routers play the role of data copying and forwarding at the network layer, ALM composes multicast trees at the application layer (Fig. 2). Basically, there are two types of nodes in ALM, namely, a parent node and a leaf node. Parent nodes are supposed to take the role of routers in IP Multicast. The greatest benefit of ALM is that users can easily design their own content delivery systems without the constraints of other layers. ALM leans on the fact that the bandwidth of networks and the processing speed of end nodes have increased tremendously in recent years, and this trend is sustainable in the foreseeable future.

In ALM, the content server is treated as the root and user nodes are considered as either parent nodes or leaf nodes depending on their locations. By making use of this tree-like structure, ALM allows users to share the contents in a multicast manner. ALM exhibits many attractive features such as scalability and simplicity of using only unicast, yet several issues need to be addressed. The first one is the accumulation of delay at the lower layers in ALM trees. The second is the degradation of QoS caused by frequent joining and seceding of nodes, especially the secession of nodes in the upper layers. The third, which impacts greatly whether quality video streaming can be achieved, is the diversity of networks between the content server and receivers (Fig. 3). These three issues have to be addressed in designing a content delivery system based on ALM.

Existing Proposals
Aiming at achieving high throughput and Quality of Service (QoS), several ideas have been proposed so far. From the view point of a tree structure, they can be classified into two categories, namely, the so called single-tree multicast and multi-tree multicast, respectively.

(1) Single-tree multicast
In the single-tree multicast, the server sends streaming traffic via a singular tree. Yoid [1], SpreadIt [2], ALMI [3], HBM [4], NICE [5], ZIGZAG [6], and Scribe [7] are examples that fall into this category. Yoid [1] and SpreadIt [2] use the Shortest Path Tree to minimize the delay from the server to end nodes, whereas ALMI [3] and HBM [4] use the Minimum Spanning Tree to reduce the overall delay, both without the consideration of bandwidth constraints. As a whole, the objective of these four methods is to minimize the content streaming delivery delay. NICE [5] and ZIGZAG [6] construct the tree by clustering. The purposes of using clustering are to curtail signaling overhead and to speed up tree management. Scribe [7] first uses Pastry [8] to construct the search paths and then makes the delivery tree by tracing back the search paths. Scribe exhibits the merit of limiting the overhead of control messages, but without consideration of the available bandwidth of participating nodes.

(2) Multiple-tree multicast
The inability of single-tree multicast to facilitate node secession is its main drawback. Multiple-tree multicast has thus been proposed to overcome this problem. Multiple-tree multicast adopts Multiple Description Coding (MDC) [9],[10] to first divide the original stream into multiple descriptions, and then construct the single-tree multicast for each description, separately. The size of each description is, in general, much smaller than that of the original stream. Thus, playback is executable upon receiving any description of the original stream. The more the number of descriptions received, the better the stream quality. MDC is now being studied widely for practical deployments such as...
incorporating MDC in the video coding standard H.264/AVC [11]-[13]. Multiple-tree multicast can also prevent nodes from suffering the interruption caused by the secession of nodes. CoopNet [14],[15], SplitStream [16], and THAG [17] are some representative methods that fall into the multiple-tree multicast category. In CoopNet, all trees are managed at the server. Therefore, overloading the server might be a concern. On the contrary, SplitStream [16] makes use of Scribe [7] to construct the trees. Trees in SplitStream are constructed in a distributed manner so that nodes can freely join a tree at any position. These two methods have addressed the available bandwidth issue, but they do not guarantee the node-disjoint property. Consequently, a node secession from the tree may result in simultaneous breakdown in multiple trees. Incidentally, as shown in Fig. 5, the node-disjoint structure allows a parent node in a certain tree to be a leaf node in the other trees. In THAG, the node-disjoint structure is implemented by the hierarchical Arrangement Graph (AG) [18], [19]. As a result, THAG is more robust in terms of node secession as compared to SplitStream or CoopNet. However, the main concern of THAG is that it does not take network heterogeneity into consideration.

Fig. 1. IP multicast

Fig. 2. Application layer multicast
Fig. 3. Heterogeneous network environment where the server and receivers are located.

Server

Fig. 4. Single-tree multicast
On Realizing an Efficient ALM for Heterogeneous Networks
Kobayashi et al. [20] proposed a novel idea referred to as Network-aware Hierarchical Arrangement Graph (NHAG) to construct efficient ALM trees in heterogeneous networks. In NHAG, the size of AG is dynamically changed and adapted to meet various bandwidth requests from nodes. NHAG is scalable, and simulation results have demonstrated that NHAG outperforms other existing methods, especially in heterogeneous networks. We believe that NHAG is a promising approach to further advance the ALM technology.

Summary and Open Issues
We have briefly presented the state of the art of ALM for establishing a robust content delivery system. Although ALM is deemed powerful for constructing the flexible P2P networks over the Internet, it is still at the embryonic stage; besides efficiency, more studies in terms of security and incentive strategies for nodes to participate in the relay are imperative in ultimately commercializing the ALM technology.

References
IEEE COMSOC MMTC E-Letter


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When Transmission of video over highly error-prone channels demands better protection of the compressed bitstream to ensure an acceptable quality of service. Many short distance error-prone channels use Automatic Repeat Request (ARQ), for retransmission of corrupted data [1]. Others suggest multiple description coding of video with spatio-frequency diversity transmission [2]. Some argue that layered coding with unequal error protection can guarantee a minimum picture quality [3]. However, in wireless channels with bursts of errors, in which those burst occur frequently due to slow and fast fading, the above techniques on their own may not be able to provide acceptable video quality. Of course, if the vital elements of layered encoded video can be efficiently protected, then such a technique can be fruitful.

One way to protect the vital video elements is by forward error correction such that they are almost guaranteed to be received safely. However, conventional fixed-rate control may not be suitable, as under varying channel conditions, the data may either be over- or under-protected. The former reduces the transmission efficiency and the latter may still not make the received data useable.

It appears that rateless or Fountain channel coding of video [4] is a simple and efficient method to protect the variable degree of video tolerance to errors that is required. In this coding method, a varying degree of redundancy is incrementally added to a group of symbols, to ensure that the symbols can be decoded under any adverse channel conditions. Thus, unlike fixed-rate Reed-Solomon (RS) codes, the coding rate is not fixed at the time of coding but can be dynamically varied. The degree of redundancy depends on channel error severity and, after a feedback request, symbols are gradually transmitted to the receiver, until the delay limit is exhausted. This not only makes transmission efficient, it also ensures the received data is decodable.

The class of Fountain codes [4] allows a continual stream of additional symbols to be generated in the event that the original symbols could not be decoded. It is the ability to easily generate new symbols that makes Fountain codes rateless. For a group of $k$ symbols, decoding will succeed with small probability of failure if any of $k (1 + \varepsilon)$ symbols are successfully received, where $\varepsilon$ is a small value, typically 0.05 for video applications. The probability of decoder failure is $\varepsilon^k = 2^{-k\varepsilon}$, which for large $k$ approaches the Shannon limit. Luby transform (LT) codes [5] reduce the complexity of decoding a simple Fountain code (which is of order $k^2$) by means of an iterative decoding procedure, provided that the column entries of the generator matrix are selected from a robust Soliton distribution. In the LT generator matrix case, the expected number of degree one combinations (no XORing of symbols) is $S = c \log_s(k\varepsilon)v_k$, for small constant $c$. Setting $\varepsilon = 2 \log_s(S/c)$ $S$ ensures that, by sending $k(1 + \varepsilon)$ symbols, these are decoded with probability $(1 - \varepsilon)$ and with decoding complexity of order $k \log_s k$. Notice that the essential differences between Fountain erasure codes and RS erasure codes are that: Fountain codes in general (not Raptor codes [6]) are not systematic; and that even if there were no channel errors there is a small probability that the decoding will fail. In compensation, they are completely flexible; the Raptor variety has linear decode computational complexity; and generally their overhead is considerably reduced compared to fixed erasure codes. It should be noted that, compressed video unlike alpha-numeric data does not need to be perfectly decoded and such a small probability of failure is acceptable for most video applications. The symbol can be defined as a packet, block, byte or bit. By defining the symbol to be as small as feasible (a block of bits, byte, or bit) $k$ is automatically made larger.

When one considers that the amount of redundancy added to the compressed video is proportional to the length of the compressed bitstream, $k$, then, for efficient rateless coding only the important elements of video may need to be protected. For instance, through layering techniques, the most important parts of the compressed video bit-stream such as motion vectors (MVs), macroblock (MB) addresses, etc., can be protected such that the delivery of a reduced video quality stream at its lowest possible bit rate can be guaranteed. Scalable video coding is a good example, whereby, through a combination of quality, spatial and temporal scalability, a base-layer video at very low bit rates requires a minimal amount of redundancy through rateless channel coding to be effectively decoded. Another example is data-partitioned video, which, unlike scalable video coding, does not impose any overhead and is as
efficient as can be. Data-partitioning (DP) is now an essential part of the Extended Profile of the H.264/Advanced Video Codec (AVC) [7]. In the following, we show how data-partitioned video can efficiently combine with rateless codes and how the same concept can be extended to scalable video.

H.264/AVC conceptually separates the Video Coding Layer (VCL) from the Network Abstraction Layer (NAL). The VCL specifies the core compression features, while the NAL facilitates the delivery of the H.264 VCL data to the underlying protocols such as RTP/UDP/IP, H.32X, or MPEG-2 transport stream. Each NAL unit could be considered as a packet that contains an integer number of bytes including a header and payload. The header specifies the NAL unit type and the payload contains the related data. The standard defines 31 different NAL unit types, when only NAL units 1 to 5 contain different VCL data [7] that will be of interest to this Letter.

The slices of an Instantaneous Decoder Refresh (IDR) or I-picture (i.e. a picture with all intra slices) are located in type 5 NAL units, while those belonging to a non-IDR or non-I-picture (P- or B-pictures) are placed in NAL units of type 1, and in types 2 to 4 when DP mode is active.

In type-1 and type-5 NALs, MB addresses, MVs and the transform coefficients of the blocks, are packed into the packet in the order that they are generated by the encoder. In type-5, all parts of the compressed bitstream are considered to be equally important, while in type-1, the MB addresses and MVs are much more important than the motion compensated DCT coefficients. In the event of errors in this type of packet, the fact that variable length encoded symbols appearing earlier in the bit-stream suffer less from the impact of errors than those which come later means that bringing the more important parts of the video data (such as header data and MVs) ahead of the less important data or separating the more important data altogether for better protection against errors can significantly reduce the impact of channel errors. It is worth noting that in the standard video codecs prior to H.264 [8], partitioning of the DCT coefficients within the same coding unit, which is a type of layered coding, is known as data partitioning. However, in more general terms, DP now refers to partitioning of the coded video stream into its degrees of importance.

However, in H.264, when DP is enabled, every slice is divided into three separate partitions and each partition is located in either a type-2, type-3 or type-4 NAL unit. A NAL unit of type-2, also known as partition A, comprises the most important information of the compressed video bit-stream for P- and B-pictures, including the MB addresses, MVs and essential headers. If any MBs in these pictures are intra-coded, their DCT coefficients are packed into a type-3 NAL unit, also known as partition B. Type-4 NAL, also known as partition C, carries the DCT coefficients of the motion-compensated inter-picture-coded MBs. As in I-slices all MBs are encoded then type-5 NAL units are very long. On the other hand, the A and B partitions of data-partitioned P- and B-slices are much smaller but their C-type partitions can be very long. In this Letter we propose an efficient method for rateless coding of A, B and C type NAL units to make video transmission more efficient.

For I-slices, a type-5 NAL can be Raptor-coded [6] with redundant information D, as shown in Fig. 1a. For data-partitioned P- and B-pictures, partitions A and B can be Raptor-coded with redundant D, as shown in Fig 1b. For partition C of P- and B-slices, a separate Raptor-code can be applied to the A, B and C partitions with redundant data in E, as shown in the Figure. Notice that this strategy is more efficient than were redundant E only to be generated for partition C. The reason is that, as the probability of decoder failure δ=2^{-k}, including safely-detected A+B in C increases the length of data k, reducing the probability of failure. One way to ensure safe delivery of video is as follows:

For every k data symbols within the partition groupings (type-5, A+B, or A+B+C), a Raptor coder generates a rateless redundant data of r symbols. The compressed data can be partitioned into blocks of bits to form a symbol, as in theory r can be infinitely long to ensure all k data symbols can be safely decoded. For transmission purposes, each packet comprises K blocks of data, and the first Y blocks of their redundant data r are sent at the position of Y in Fig 1c. The packet also includes a Cyclic Redundancy Code (CRC) calculated from the K blocks or symbols. Recalculation of the CRC at the receiver and comparison with the sent CRC indicates whether the data decode was successful. In case of error, the transmitted data are stored and in the following packet additional redundant blocks of r, identified by X in Fig. 1c are sent. These new redundant blocks will help to decode the failed decoding and if the decoder still is not able to decode, more redundant blocks in the following packets will be sent. The process is continued, until the block is safely decoded.
Figure 1. (a) I-slices and redundant data (b) Redundant codes for data-partitioned video (c) Packetized rateless coded data with CRC

Of course, for a delay-sensitive service such as video, transmission of additional redundant blocks cannot go on for ever, and there should be a limit. For instance, one could confine the decoding delay to be within a certain number of pictures (e.g. 15 pictures, equal to approximately half a second at 30 Hz.) To limit the number of transmissions of redundant blocks for previous data (X), the length of these blocks in the following packets can be gradually increased. For I-pictures/slices, when the length of type-5 NALs can be very long, the length of redundant code $r$ is much longer than that of P-pictures. Fortunately, in the H.264 standard there are several B pictures/slices after each I-picture, and instead of transmission of B-pictures, one may just send the redundant D blocks of I-pictures. This is because, B-pictures can be easily discarded without significantly impairing video quality. For P- and B-pictures, since A+B is very small, their number of redundant blocks is also small, and can be easily decoded in a few following packets. For partition type C, though such partitions can be long, since their impact on picture quality is negligible and, as already mentioned, their decoder failure probability is small, they can be easily sacrificed in favor of sending the redundant blocks belonging to previous A+B blocks. Where no B-pictures are used, P-picture C type NALs can also be replaced by the redundant data for type-5 I-pictures. This procedure will significantly reduce the decoding delay under extremely adverse channel conditions.

Our simulations indicate that use of rateless coding on data-partitioned video can improve the video quality by 2-3 dB depending on video content. We believe, similar quality gains could be achieved if rateless coding were to be also applied to scalable coded video, as efficient transmission of scalable coded video greatly depends on the safe delivery of base layer pictures.

References

Mohammed Ghanbari is best known for his pioneering work on two-layer video coding for ATM networks (which earned him an IEEE Fellowship in 2001), now known as SNR scalability in the standard video codecs. He has served as an Associate Editor for IEEE Trans. on Multimedia (IEEE-T-MM from 1998-2004) He has registered for eleven international patents on various aspects of video networking and was the co-recipient of A.H. Reeves prize for the best paper published in the 1995 Proc. of IEE on the theme of digital coding. He is the co-author of “Principles of Performance Engineering”, a book published by IET press in 1997, the author of “Video Coding: An Introduction to Standard Codecs”, a book also published by IET press in 1999, which received the year 2000 best book award by the IEE, and the author of “Standard Codecs: Image Compression to Advanced Video Coding” also published by the IET press in 2003. Prof. Ghanbari has authored or co-authored about 450 journal and conference papers, many of which have had a fundamental influence in this field.

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Recent advances in Next generation cellular wireless networks (i.e., 4G) are featured with high data rate and improved coverage, which make it a reality to provide real-time video multicast services, such as IPTV services, live video streaming, and online telecast of sports.

I have the opportunity to read this new paper recently accepted by IEEE ICNP 2009, and to be appearing in October 2009 (The paper currently can be downloaded from http://www.nec-labs.com/~honghai/icnp-2009.pdf). It is coauthored by P. Li, H. Zhang, B. Zhao, and S. Rangarajan from University of Science and Technology of China and NEC Laboratories America, and deals with optimal scalable video multicast in future cellular networks via Adaptive Modulation and Coding (AMC). In fact, AMC and Scalable Video Coding (SVC) are two of the most critical and efficient techniques for wireless video multicast services. Scalable video coding [1] divides a video stream into multiple layers, including one base layer and one or more enhancement layers. The base layer provides a minimum quality, frame rate, and/or resolution of the video and the enhancement layers represent the same video at gradually increasing quality, frame rate, and/or resolution. AMC allows applying different Modulation and Coding Schemes (MCS) on different layers of the scalable video sequence, such that the users in good channel conditions receive more enhancement layers and obtain better video quality, and the users in bad channel conditions receive fewer enhancement layers and obtain basic video quality. Due to the limited bandwidth resources, the MCS for the video layers should be carefully selected, especially in the multiple-video-session scenario. The key issue for this problem is how to allocate the radio resources to multiple multicast video sessions and to multiple SVC layers within a session, and how to assign MCS for each allocated video layer.

In this paper, the authors formulate the problem as maximizing the sum of utilities of all wireless receivers subject to the constraints of available radio resources allocated for multicast services. Several generalizations have been done in their model. First, the utility function is designed to be a general function of the received data rate which only needs to be non-negative and non-decreasing. It can be both user-dependent and session-dependent. Second, the model in this paper does not require that all video layers have equal length as in [2] and therefore allow more flexibility at the video encoding process. Finally, the base layers are not enforced to be received by all wireless clients. An advantage of such a choice is that it can provide an automatic admission control at both the session level and the user level.

The authors show that the formulated problem is NP-hard. To solve the problem, they develop an optimal MCS assignment algorithm for single session and an optimal resource allocation algorithm for multiple sessions. Both algorithms are based on dynamic programming and can run in pseudo-polynomial time. Detailed simulations have been conducted to verify the proposed algorithms. PSNR function is employed to measure the video quality based on the results in their previous paper [3]. Numerical results show that both algorithms obtain significant improvement on the video quality compared to the existing algorithms.

References

* The Column Editor recommending this paper is Dr. Chonggang Wang.
Multimedia services have become an integral part of mobile networks. Mobile Television is one of these services that have attracted a strong interest worldwide. A new report by Juniper Research predicts that the revenues of Mobile TV services will rise from under $1.4bn in 2007 to nearly $12bn in 2012 [1].

This paper provides an overview of Mobile TV and discusses new and future research directions in Mobile TV and upcoming full-fledged Mobile Video-on-Demand (VOD) services.

Mobile TV Systems/Standards
With Mobile TV, users enjoy live and/or on-demand TV using their own mobile devices, such as TV-capable wireless phones and PDAs. The delivery of Mobile TV can be achieved through terrestrial broadcast, satellite broadcast, a combination of terrestrial and satellite broadcasts, and cellular networks. Table 1 lists popular systems/standards for each of one these categories. In contrast with the pure broadcast systems, MBMS and BCMCS are standardized by the 3rd Generation Partnership Project (3GPP) and 3GPP2 for providing resource-efficient, Mobile TV using the GSM/WCDMA and CDMA2000 cellular networks, respectively. These two standards allow the coexistence of unicast, multicast, and broadcast services [2]. They can be implemented by introducing only minor changes to existing radio and control network protocols in order to reduce the implementation costs in the mobile terminals and network [3].

Various Mobile TV systems/standards differ in many aspects, including robustness of transmission and quality of service expected in indoor and outdoor environments, power-saving features, channel switching times, handset requirements, spectrum utilization, operating costs, charges, countrywide availability, roaming, and provided services [4].

New and Future Research Directions in Mobile TV
A successful implementation of Mobile TV requires accounting for the unique characteristics of the wireless environment (such as noise and multi-path interference), the limited capabilities of mobile devices (such as computing and energy resources), and a distinctive use-case context [2]. In particular, minimizing the energy consumption in Mobile TV systems is a critical problem because of the limited energy supply in the battery-powered mobile devices. Study [5] considered the power optimization problem in broadcast TV systems (DVB-H in particular) through burst scheduling of TV channels, which may be encoded with different bit rates. This scheduling problem was shown to be NP complete.

<table>
<thead>
<tr>
<th>Mobile TV Delivery Type</th>
<th>Examples</th>
</tr>
</thead>
<tbody>
<tr>
<td>Terrestrial Broadcast</td>
<td>Digital Video Broadcast to Handhelds (DVB-H), Terrestrial Digital Multimedia Broadcasting (T-DMB), MediaFLO</td>
</tr>
<tr>
<td>Satellite Broadcast</td>
<td>China Multimedia Mobile Broadcasting (CMMB), Satellite Digital Multimedia Broadcasting (SDMB)</td>
</tr>
<tr>
<td>Terrestrial and Satellite</td>
<td>Digital Video Broadcasting-Satellite Services to Handhelds (DVB-SH)</td>
</tr>
<tr>
<td>Cellular Networks</td>
<td>Multimedia Broadcast Multicast Service (MBMS), Broadcast and Multicast Service (BCMCS)</td>
</tr>
</tbody>
</table>

Table 1: Systems/Standards for Delivering Mobile TV

Interactivity is another major area of research. Mobile phones support user-service interaction by using a back-channel. These interactions include voting, quiz taking, as well and browsing of side information while watching the program. Social interaction enhances interactivity and provides a more enjoyable TV watching experience by allowing interactions with other peers while watching the TV programs. Study [6] explored peer interaction enablers, including text and audio chat, synchronized zapping and “See-what-I-see” message.
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Finally, there has also been a rising interest in providing interoperability among Mobile TV standards. For example, there has been a need for providing a bearer-independent quadruple service (TV, Telephony, Internet, and Wireless) [2]. The objective here is to provide a common system such that the same service-layer functionalities can be used for mobile TV over different broadcast and access networks.

Future Directions in Mobile VOD Services
The main challenge with providing scalable Mobile VOD services is that not all viewers of a video will be at the same playback point. Therefore, the mere use of multicast will not lead to significant reductions in the required load and bandwidth of the server and network. Fortunately, stream merging [7] (and references within) and period broadcasting [8] (and references within) techniques can be used to address this problem. These techniques, however, were not developed for wireless mobile networks, and thus they must be adapted to account for the unique characteristics of these networks. Providing efficient support for VCR-like operations while using stream merging techniques is another major research challenge.

References


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Introduction

With the development of broadband networking technology, the focus in the new media and performing arts community is shifting towards distributed telepresence performance (also known as telematic performance) – a scenario where two or more groups of performers, located in geographically distant sites, are able to perform interactively for the audience located at either site, or in an independent venue. For a number of years, universities and research centers in North America have been linked with gigabit optical networks whose high bandwidth makes such performances feasible. Yet, we are not seeing many of such events available to the wider audience. Part of the reason is that the traditional performance venues, theaters and music halls, are not linked to the high-bandwidth research networks and therefore cannot accommodate the high bandwidth requirement associated with these events.

Compression, especially video compression, appears to be the answer to some of these challenges. However, careful examination of the requirements of a telepresence performance reveals that there is more to it than simply using an off-the-shelf video codec. First, video coding is found to be detrimental to interactive communication since it introduces additional delay and makes the resulting compressed bit stream more sensitive to packet losses in the network [1]. In fact, high-end videoconferencing solutions try to stay away from video compression in order to improve user experience [2]. Commercial products, such as LifeSize, opt for hardware compression which may be suitable for corporate videoconferencing, but lacks the flexibility and accessibility needed for artistic performance.

In order to answer some of these challenges, we have developed a real-time software-only video codec for the most popular software environment used by media artists – Max/MSP/Jitter [3]. The codec is based on the well-known SPIHT algorithm [4], and feeds the video stream directly into the Max/MSP/Jitter environment, enabling easy on-stage video manipulation. Both the codec and its software environment are briefly described below.

Max/MSP/Jitter

Max/MSP/Jitter (or Max, for short) is a graphical programming environment for multimedia [3], widely used by media artists in new media productions. This environment supports real-time manipulation of video, 3-D graphics, audio, and other data in a unified processing architecture, enabling seamless on-stage performance. It is available on Mac OS and Windows platforms.

In Max, programs (called “patches”) are created by connecting objects which perform specific functions. These objects can be written in C++, Open GL, Java, or simply be composed of other sub-patches. Data flow is controlled by the chords connecting these objects. All data are abstracted as multidimensional arrays (matrices). Data exchange between objects is synchronized by the internal clock which can be set at millisecond precision. If a particular chord carries a matrix or higher-dimensional array, it will appear as thick and green, otherwise it will be thin and black. Detailed information on Max can be found at [3].

Scalable Video Codec for Max/MSP/Jitter

We have developed a scalable video codec for Max based on SPIHT [4]. In the current version, only intra-frame coding is supported. While this means that compression efficiency is not as high as it could be with inter-frame coding, the codec has several advantages which are very important in a live performance. First, its CPU usage is low, allowing other video and audio manipulation tasks to run on the same machine. Second, it provides the level of robustness against dropped frames which cannot easily be achieved with inter-frame coding.

The codec has been used in "t2", a telematic dance performance premiered in Vancouver in July 2009. The trailer for the performance can be found at [3].
found at [5]. The performance involved live video transmission between two sites in downtown Vancouver via a conventional residential Internet connection at roughly 500 kbps. In addition, live video from a moving car, transmitted via a 3G mobile Internet connection at roughly 100 kbps, was a part of the performance. This appears to be one of the first (if not the first) use of mobile wireless video in a live telematic dance performance.

Figs. 1 and 2 below show simplified sending and receiving Max patches, respectively, utilizing our codec. In this example, the machine with IP address 142.58.88.162 is sending live video to another machine with IP address 199.60.10.135 on port 8000. The encoder appears as the object `mcl.jit.spihtaritenc` in the sending patch (Fig. 1), while the decoder is `mcl.jit.spihtaritdec` in the receiving patch (Fig. 2). Captured and decoded frames are also shown for reference in their respective patches.

In addition to the simple live video unicast, the codec has also been demonstrated in the live scalable video multicast. In this scenario, live video was encoded and sent to the transcoding peer, whose task was to adjust the bitrate for several heterogeneous receivers. The transcoding was easy due to the progressive nature of SPIHT bitstreams - it amounted to simply truncating the bitstream of each frame to the points appropriate for receivers’ bandwidths. Further illustration and several demo clips about our collaborative work on telepresence in the performing arts are available at [6]. External codec objects for Windows and Mac OS versions of Max, as well as sample Max patches that show how to use them, can be obtained from the authors.

Acknowledgement

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References

[5] T2 trailer: barry.is2.net/T2/t2trailer.mov

http://www.comsoc.org/~mmc/
Ivan V. Bajić is an Assistant Professor in the School of Engineering Science at Simon Fraser University, Burnaby, BC. His research interests revolve around signal processing and its applications in image and video coding, multimedia communications, and computational biology. He has authored about a dozen, and co-authored another three dozen publications in these fields. He was recently the Telepresence Architect for t2, a telematic dance performance premiered in Vancouver in July 2009.

Xiaonan Ma received the B.A.Sc. degree in Electrical Engineering from the University of Ottawa in 2007. She is currently pursuing the M.A.Sc. degree in the School of Engineering Science at Simon Fraser University, Burnaby, BC. Her research interests include video coding and multimedia networking for new media and performing arts. She was recently the Video Coding Engineer for t2, a telematic dance performance premiered in Vancouver in July 2009.
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