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

R-LETTER

Vol. 2, No. 1, February 2011

IEEE COMMUNICATIONS SOCIETY

CONTENTS

Message from R-Letter Director

Thank you for your great support to MMTC R-letter and welcome to the first issue of IEEE MMTC Review-Letter (R-Letter) in 2011. Happy New Year! In this issue, we are pleased to introduce seven high quality papers. Three papers are from Comsoc-sponsored journals, two from MMTC-sponsored conferences, and two from multimedia communication related journals.

The selected papers in this issue cover a wide spectrum on multimedia communication: rateless coding application, video streaming, cross-layer security, and game theory application. The first paper published in *IEEE Transactions on Communications* studies the broadcasting problem in wireless multi-hop networks via rateless coding method. The second paper investigates the problem of scheduling delay-constrained multimedia traffic to multiple users on the shared wireless system and was published in *IEEE Transactions on Wireless Communications*. The third paper, which was published in *IEEE Transactions on Multimedia*, proposes a novel approach via fountain codes for the scenario of scalable video multicasting.

Two papers from *IEEE Global Communications Conference* 2010 are introduced in this issue: one paper proposes the efficient satisfaction equilibrium concept for the self-configuring network scenario, and the other paper addresses the cross-layer attack and defense issues in cognitive radio networks.

The sixth paper, which was published in *IEEE Transactions on Mobile Computing*, studies the energy efficiency issue in mobile ad hoc networks routing. The last paper examines the video traffic characteristics of H.264 SVC and provides suggestions on the design the SVC transmission system.

I would like to thank all editors for their great efforts: Vladimir Stanković, Cheng-Hsin Hsu, Kristian Nybom, Walid Saad, Man-On Pun, Ai-Chun Pang, and Kalpana Seshadrinathan. We hope you will find this issue informative.

Guan-Ming Su

Director, IEEE ComSoc MMTC R-Letter
E-mail: guanmingsu at ieee.org

Energy-Efficient Multihop Networks Broadcasting: A Rateless Coding Approach

A short review for "FTS: A distributed energy-efficient broadcasting scheme using fountain codes for multihop wireless networks"

Edited by Vladimir Stanković

B.N. Vellambi, N. Rahnavard, and F. Fekri, "FTS: A Distributed Energy-efficient Broadcasting Scheme using Fountain Codes for Multihop Wireless Networks," IEEE Transactions on Communications, vol. 58, no. 12, pp. 3561-3572, December 2010.

Wireless multihop networks are transforming the traditional way of networking by introducing the concept of cooperation via relaying, pushing the bounds on the amount of information that can be delivered and opening fantastic application possibilities. Wireless multihop networks are natural solutions to monitoring and control with sensor networks, communications in smart grids, smart homes/buildings, video surveillance, rescue and exploratory missions, etc. Real-time multicasting/broadcasting of multimedia content over such networks is especially challenging due to high bandwidth demands, sensitivity to losses, and small delay tolerance of real-time media.

Multihopping brings the possibility of maximising energy efficiency and hence the life time of the network, distributing the communications load, and increasing the overall network throughput. However, multihopping can easily lead to a network overflow with many redundant transmissions. Indeed, by simply forwarding all received packets, the network nodes will usually end up with multiple copies of the same information packets wasting the available resources and causing network congestion. To cope with this problem, it is essential to provide transmission strategies that will minimise the number of packets injected into the network while still keeping high reliability and energy efficiency with low latency.

Network coding [1] has been widely recognised as a promising solution for wireless multihop communications since it ensures (with high probability) that each injected packet is innovative, in the sense of bringing novel information. Thus, instead of duplicating and re-sending a received packet, with network coding each packet leaving an intermediate node provides novel and useful information. Network coding via random linear coding (RLC) over a large enough field is a practical method that can be used to provide data gathering and efficient broadcasting [2, 3]. However, a key drawback of using RLC is high polynomial decoding complexity, via Gaussian elimination, that is often an obstacle in multihop networks, where nodes are energy constrained. Thus, instead of RLC, rateless erasure codes that use sparse binary combinations of incoming packets have become popular [4, 5]. That is, digital fountain codes such as LT or Raptor codes can be used with low encoding and decoding complexity.

The two key problems associated with application of rateless codes to broadcasting are: (i) routing, i.e., determination of sub-graphs over which packets should be sent and coding performed; (ii) coding, i.e., which codes to use. This paper provides a solution to these problems bringing a broadcasting scheme with rateless codes, called fractional transmission scheme (FTS). It is assumed that all transmissions are subject to distance attenuation and Rayleigh fading, and if the received SNR drops below a certain level, the packet is assumed to be lost. Thus, all communication channels are modeled as packet erasure channels, and a slotted CSMA is used to avoid collisions.

The FTS employs the decode-and-forward strategy, that is, each node waits until it receives enough packets to decode the message, re-encodes it using the same rateless code, and then forwards novel encoded packets. This way, powerful, conventional digital fountain codes can be used independently in each hop. A neighbourhood of a node n is defined as a set of nodes which node n can communicate to. All neighbourhood nodes are classified as parent nodes, which should send packets to n, or children nodes, which expect packets from n.

A goal of the FTS is to provide an effective load balancing between all the nodes in the network by allowing each node to locally determine the fraction of data it needs to transmit to its children based on the number of its neighbours. This way, each node knows how many packets it should receive from each of its parents. A beauty of using rateless codes is that packets from different parents can be combined when decoding. Moreover, it is not important from which parent packets arrive, as long as the total number of received packets is enough for decoding.

The FTS introduces two acknowledges (ACKs): one, partial, ACK to signal to each parent that all packet from him/her are received, and another ACK to signal to all parents that successful decoding is done, after which the node transit to the forward mode and prepares transmission to its children. It is shown that the partial ACK provides energy savings by reducing the number of redundant transmissions.

Analytical and simulation study is performed for grid and random deployment networks, showing that the proposed FTS provides comparable performance to the more complex network coding [1, 2, 3] and broadcast incremental power (BIP) [6] schemes.

The proposed approach uses conventional centralized rateless coding for each hop. An interesting alternative is employing decentralized rateless codes. Indeed, in [7], a distributed Raptor code design is proposed for data gathering in wireless sensor networks. The main idea of the approach is to generate, encode in a *distributed fashion*, and disperse uniformly across the network a sufficient number of encoded packets, where each packet shares the same properties as if it were generated by the centralized Raptor encoder. Collecting any subset of the packets slightly larger than the size of the distributed network data content is sufficient for successful data recovery with high probability. The approach is packet-centric, as it is controlled by the packets that travel within the network. The advantage of the distributed rateless coding approach is that it does not need any topology information and it avoids decoding at the network nodes. A disadvantage is the slightly reduced performance.

In summary, it is well known that rateless codes are very efficient for energy-constrained broadcasting. This paper's main contribution lies in decentralized protocol designs that reduce the total number of transmissions in each hop. This is one of many papers that have established potential of rateless codes via analytical studies. It is the right time now to provide practical deployment solutions and see rateless codes in action for data gathering and energy-efficient broadcasting

References:

[1] R. Ahlswede, N. Cai, S.Y.R. Li, and R.W. Yeung, "Network information flow," *IEEE Trans. Inform. Theory*, vol. 46, pp. 1024-1216, July 2000.

[2] D.S. Lun, M. Medard, T. Ho, and R. Koetter, "Network coding with a cost criterion," in *Proc. Int. Symp. Inform. Theory Applications*, pp. 1232-1237, Oct. 2004.

[3] D.S. Lun, M. Medard, and R. Koetter, "Efficient operation of wireless packets networks using network coding," in *Proc. Int. Workshop Convergent Technol.*, June 2005.

[4] M. Luby, "LT Codes," in *Proc. of the 43rd Annual IEEE Symp. Foundations of Computer Science (FOCS)*, Vancouver, Canada, Nov. 2002.

[5] A. Shokrollahi, "Raptor codes," *IEEE Trans. Inform. Theory*, vol. 52, no. 6, pp. 2251–2567, 2006.

[6] J.E. Wieselthier, G.D. Nguyen, and A. Ephremides, "Energy-efficient broadcast and multicast trees in ol. 7, no. 6, pp. 481-492, Dec. 2002. wireless networks," *Mobile Networks and Applications*, v

[7] D. Vukobratovic, C. Stefanovic, M. Stojakovic, V. Stankovic, "Raptor packets: A packet-centric approach to distributed Raptor code design," in *Proc. IEEE Int. Symp. Inform. Theory*, 2009.

Vladimir Stankovic (M'2004, SM'2010) received his Dipl-Ing degree in Electrical Engineering from the University of Belgrade, Serbia, in 2000 and Dr-Ing degree from the University of Leipzig, Germany in 2003. From 2003-2006 he was with Texas A&M University, College Station, as a Postdoctoral Research Associate and a Research Assist Prof, and Lancaster University as a Lecturer. From 2007, he has been with the Dept Electronic and Electrical Engineering, University of Strathclyde, Glasgow, where he currently holds a Senior Lecturer post (equivalent to Associate Prof).

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

Scheduling Video Packets in Next Generation Cellular Networks

A short review for "Channel, deadline, and distortion aware scheduling for video streams over wireless"

Edited by Cheng-Hsin Hsu

A. Dua, C.W. Chan, N. Bambos, and J. Apostolopoulos, "Channel, Deadline, and Distortion (CD^2) Aware Scheduling for Video Streams over Wireless," IEEE Transactions on Wireless Communications, vol.9, no.3, pp.1001-1011, March 2010.

The explosive growth of third and fourth generation cellular networks like HSPA+ and LTE has sparked an ever increasing interest in mobile wireless multimedia applications such as video streaming. Transporting multimedia traffic over wireless links poses significant theoretical as well as practical challenges. These challenges stem from the temporal and spatial variations in wireless channel quality, contention amongst competing users for shared network resources such as bandwidth, and the unique characteristics of multimedia traffic such as packet interdependencies and deadline constraints.

This paper studies the problem of scheduling delay-constrained multimedia traffic to multiple users on the shared downlink of a wireless cellular system. The authors formulate the multimedia scheduling problem in a dynamic programming framework. In the scenario examined in the paper, multimedia packets are associated with strict deadlines and are equivalent to lost packets if they arrive at the receiver past their deadlines. Lost packets result in degradation of playout quality at the receiver, which is quantified in terms of the distortion cost associated with each packet. The objective of the scheduler is to minimize the aggregate distortion cost across all users over a finite time horizon.

The authors apply well justified modeling reductions and extensively characterize key structural properties of the optimal control policy associated with this dynamic programming problem. More specifically, the authors first show that, for a system with two users, there is an optimal yet time-invariant scheduling policy. Next, the authors employ a pair-wise comparison approach to derive an optimal scheduling policy for a system with more than two users. These provable structural properties of the optimal control lead to the low-complexity **C**hannel, **D**eadline, and **D**istortion (CD^2) aware scheduling policy. The CD^2 scheduling policy determines the packet transmission schedule based on channel characteristics, packet delay deadlines, and application specific information (per packet distortion costs), and prioritizes transmission of packets within a stream as well as across multiple streams, in order to minimize the expected aggregate distortion across all the receivers. The CD^2 policy has a very low computational complexity of $O(N)$, where N is the number of users. Therefore, it is amenable to implementation in practical wireless systems.

The performance of CD^2 is evaluated via trace driven simulations (using H.264/MPEG-4 AVC coded videos) under a variety of wireless channel conditions and contrasted to benchmark scheduling policies such as Round Robin, Best Channel First, and Earliest Deadline First. Peak Signal to Noise Ratio (PSNR) is used as the video quality metric for comparing different scheduling policies. Simulation results demonstrate that CD^2 comfortably outperforms all benchmark scheduling policies considered in the paper. CD^2 delivers 2-12 dB enhancements in average PSNR over benchmark schedulers and is also shown to have better fairness properties. Moreover, CD^2 is able to deliver this enhanced performance at a computational complexity which only grows linearly with the number of users in the system.

Scheduling delay-tolerant wireless traffic to maximize throughput has been thoroughly studied, e.g., see [1] and the references therein. Scheduling delay-constrained traffics over dynamic wireless channels has also been considered, e.g., Shakkottai and Srikant [2], show that a variation of the Earliest Deadline First policy minimizes the number of packets missing their deadlines. While these early studies account for both channel conditions and packet deadlines, they do not take video characteristics into consideration.

In contrast, this paper explicitly takes account for video characteristics under the Rate-Distortion (R-D) optimization framework [3], and studies channel, deadline, and distortion aware scheduling of multiple multimedia streams over a shared wireless link in a systematic and unified manner. Despite the inherent complexity of the considered multimedia system, this work leverages provable structural properties of the optimal controller to develop a high-performance scheduling policy. I believe the simplicity of the proposed CD^2 scheduling policy makes it an excellent candidate to be implemented and deployed in the wild. Along this direction, I look forward to see future systems research on integrating CD^2 scheduling policy with third and fourth generation networks, including the emerging HSPA+ and LTE.

References:

[1] H. Fattah and C. Leung, "An overview of scheduling algorithms in wireless multimedia networks," *IEEE Wireless Commun. Mag*., vol. 9, no. 5, pp. 76–83, Oct. 2002.

[2] S. Shakkottai and R. Srikant, "Scheduling real-time traffic with deadlines over a wireless channel," *ACM/Baltzer Wireless Networks*, vol. 8, no. 1, pp. 13–26, Jan. 2002.

[3] P. Chou and Z. Miao, "Rate-Distortion optimized streaming of packetized media," *IEEE Trans. Multimedia*, vol. 8, no. 2, pp. 390–404, Apr. 2006

Cheng-Hsin Hsu received the Ph.D. degree from Simon Fraser University, Canada in 2009, the M.Eng. degree from University of Maryland, College Park in 2003, and the M.Sc. and B.Sc. degrees from National Chung-Cheng University, Taiwan in 2000 and 1996, respectively. He is a senior research scientist at Deutsche Telekom R&D Lab USA, Los Altos, CA, where he leads the New Media research group. His research interests are in the area of multimedia networking and distributed systems. He has published more than 30 papers in leading journals, conference, and workshops. He and his colleagues in Simon Fraser University developed a mobile TV testbed, which won the Best Technical Demo Award in the ACM multimedia 2008 Conference. He is on the Preservation Committee of ACM Special Interest Group on Multimedia (SIGMM), and on the Review Board Committee of IEEE Technical Committee on Multimedia Communications (MMTC). He served as the Proceeding and Web Chair of the ACM International Workshop on Network and Operating Systems Support for Digital Audio and Video (NOSSDAV 2010). He was on several technical program committees, including the ACM International Workshop on Network and Operating Systems Support for Digital Audio and Video (NOSSDAV 2010).

Unequal Error Protection Fountain Codes for SVC Broadcasting

A short review for "Scalable video multicast using expanding window fountain codes"

Edited by Kristian Nybom

D. Vukobratović, V. Stanković, D. Sejdinović, L. Stanković, and Z. Xiong,. "Scalable Video Multicast Using Expanding Window Fountain Codes", IEEE Trans. Multimedia, vol. 11, no.6, Oct. 2009. pp-1094-1104.

Fountain (or rateless) codes are an exciting recent development in coding theory proposed to simplify and improve multicasting of large files over lossy packet networks. In a multicast system employing fountain codes, the multicast server produces a potentially infinite amount of encoded packets by a carefully designed random encoding procedure applied over the source file packets. A user collects any set of encoded packets, containing slightly more packets than there are packets in the source file, after which it is able to recover the file. The feedback is required only at the end of the process when the user reports to the server that the file recovery is complete.

The fountain coded multicast system, as described above, was conceptually introduced in [1], and the first practical fountain codes called LT codes were introduced in [2]. Soon after the introduction of LT codes, Raptor codes [3] were proposed, which reduce the encoding and decoding complexity to linear in the file length. In parallel, commercial Raptor code solutions designed by the Digital Fountain company [4] are standardized as part of the application layer forward error correction for file multicast within recent wireless mobile broadcasting systems, such as DVB-H and 3GPP MBMS [5][6].

Apart from file multicasting, fountain coding is becoming increasingly popular for multimedia streaming over wireless networks. There are two key problems in using fountain codes for scalable video multicasting. Firstly, scalable video coders such as H.264/SVC [7] produce a layered source content which can be efficiently protected by unequal error protection (UEP) codes for wireless broadcast to heterogeneous receivers. However, as originally studied, fountain codes are equal error protection (EEP) codes and cannot adapt to the varying importance of scalable coded video layers. Secondly, applied fountain codes are limited to produce "only" a finite number of encoded symbols due to real-time constraints and lack of retransmission protocols

The recently proposed expanding window fountain (EWF) codes are a class of UEP fountain codes that naturally extend the fountain coding concepts to provide UEP [8]. EWF codes introduce a set of nested expanding windows aligned with the importance structure of the source message. The fountain encoding is slightly modified to include probabilistic window selection which may be designed to provide a desired UEP performance. An asymptotic erasure recovery performance of EWF codes for source packets belonging to different importance classes is derived in [8] and serves as a fundamental tool for the EWF code design.

Current broadcast systems are designed with bandwidth efficiency in mind. This is due to the frequency spectrum being crowded with several different services. Simultaneously, new broadcast systems are required to deliver services with higher and higher bit-rates, requiring the use of state-of-the-art technologies. For example, the second generation DVB standards aim at transmitting high definition TV (HDTV) to receivers and to achieve this, not only are the standards designed with state-of-the-art building blocks but also, the content transmitted need to have high compression rates.

Traditional solutions encode each video layer with a separate digital fountain code, and UEP is achieved by assigning different coding rates to different layers based on their importance. However, using multiple fountain codes requires complex rate allocation optimization and control and increases coding complexity. Also, separate codes are shorter and thus usually less efficient.
Targeting wireless video multicast applications, this paper develops EWF code designs to provide optimal UEP of H.264/SVC coded video. The key advantage of the proposed designs is that a *single code* can be used to provide optimal

UEP performance, very close to the analytically derived asymptotic limits.

The real beauty of the developed solution lies in the design flexibility which is enabled by many parameters that can be tuned to simultaneously satisfy different clients' and network demands. The main design challenge is to optimally adapt unequal data importance to the reception conditions of heterogeneous receiver classes. In particular, by setting clients' and network demands as constraints on the probabilities of decoding of different layers, an optimization problem was cast and an analytical solution found as a region of coding parameters that satisfy the constraints. Finally, using rate-distortion optimization, the optimal point within the found region can be selected. This way, the optimal designs satisfy the constraints imposed by heterogeneity of the clients, and the coding scheme is optimally adapted to the different levels of importance of different layers of the video stream. Results with H.264/SVC coded video and EWF with and without Raptor-like precoding show very good match between theoretical asymptotic analysis and simulations.

In summary, SVC is a promising video compression technique for improving bandwidth efficiency in multimedia broadcast networks with heterogeneous receiver capabilities. However, due to the nature of SVC, UEP techniques may be required in future broadcast networks to ensure high quality reception of the content. To this end, EWF codes are good candidates, allowing receivers to decode only a fraction of the data stream according to their needs and capabilities.

The performance of EWF codes for scalable video broadcasting has been investigated recently through field measurements in a DVB-H network [9], where it was shown that EWF codes provide competitive end-user experiences. As future work, it is important to compare the EWF solution with other UEP solutions, in particular with respect to bandwidth efficiency. Another research direction is to design transmission schemes, such that receivers experience close to optimal performances in terms of required number of received packets when the full SVC stream is not of interest, e.g., when receivers are interested only in the base layer. Finally, designing systematic EWF codes for allowing fast access to correctly received data would be of interest for real-time applications.

References:

[1] J.W. Byers, M. Luby, and M. Mitzenmacher, "A Digital Fountain Approach for Asynchronous Reliable Multicast," *IEEE J. Sel. Areas Comms*, vol. 20, no. 8, pp. 1528-1540, Oct. 2002.

[2] M. Luby, "LT Codes," in *Proc. of the 43rd Annual IEEE Symp. Foundations of Computer Science (FOCS)*, Vancouver, Canada, Nov. 2002.

[3] A. Shokrollahi, "Raptor codes," *IEEE Trans. Inform. Theory*, vol.52, no. 6, pp. 2251–2567, 2006.

[4] T. Stockhammer, A. Shokrollahi, M. Watson, M. Luby, and T. Gasiba, "Application Layer Forward Error Correction for Mobile Multimedia Broadcasting," *Handbook of Mobile Broadcasting*, CRC Press, 2008.

[5] ETSI TS 102 472 v1.2.1, IP Datacast over DVB-H: Content Delivery Protocols, 2006.

[6] 3GPP TS 26.346 Multimedia Broadcast/Multicast Service: protocols and codecs. V6.1.0 2005.

[7] H. Schwarz, D. Marpe, and T. Wiegand, "Overview of the Scalable Video Coding Extension of the H.264/AVC Standard," *IEEE Trans. Circuits and Systems for Video Techn.*, vol. 17, pp. 1103–1120, Sept. 2007.

[8] D. Sejdinovic, D. Vukobratovic, A. Doufexi, V. Senk, and R. Piechocki, "Expanding Window Fountain Codes for Unequal Error Protection," *IEEE Trans. Commun.*, pp. 2510-2516, Sept. 2009.

[9] K. Nybom, S. Grönroos, and J. Björkqvist, "Expanding Window Fountain Coded Scalable Video in Broadcasting," in *Proc. IEEE Int. Conf. Multimedia & Expo (ICME)*, Singapore, July 2010.

Kristian Nybom received his M.Sc. Eng. degree in Embedded Systems and D.Sc. (Technology) in 2005 and 2009, respectively, from Åbo Akademi University, Turku, Finland. During his post graduate studies he has been working within the European EUREKA CELTIC Wing-TV and B21C research projects, focusing on the evaluation of the DVB-H and the DVB-T2 standards. From 2009 he has been with the Department of Information Technologies at Åbo Akademi University, where he is a Research Fellow. He is currently working within the European EUREKA CELTIC ENGINES project, focusing on the evaluation of the DVB-NGH system. His research interests include DVB systems, forward error correction codes, and multimedia streaming.

QoS Provisioning in Self-Configuring Networks: Beyond Nash Equilibrium

A short review for "Satisfaction equilibrium: A general framework for QoS provisioning in self configuring networks"

Edited by Walid Saad

S. M. Perlaza1, H. Tembine, S. Lasaulce, and M´. Debbah, "Satisfaction Equilibrium: A General Framework for QoS Provisioning in Self Configuring Networks", IEEE Global Communications Conference, Dec. 2010.

In recent years, it has become imperative to enable wireless devices to communicate with on another, in a distributed and autonomous manner, in order to perform different tasks such as updating operation settings, exchanging user data, or collecting information so as to enhance the users' overall wireless experience and ensure service satisfaction.

In this respect, future wireless networks are expected to be composed of a dense number of radio devices operating in the same frequency band (e.g., the ISM band in 2.4Ghz) and, thus, subject to significant mutual interference. Unlike in classical networks (e.g., cellular networks), future wireless networks are envisioned to be highly decentralized with little or no reliance on any central entity such as a base station that can gather complete information of the network and determine the optimal transmission configuration for each device given their individual quality of service (QoS) requirements. Moreover, the large diversity of physical layer technologies, e.g., Wi-Fi, Bluetooth, ZigBee, *etc*, naturally constrains any kind of signaling or message exchange between all devices which might be simultaneously sharing the same spectrum. Therefore, one key challenge of future networks is to design a QoS-aware scheme that enables the wireless devices to coexist and operate within the frequency band while maintaining a certain QoS level. To overcome this problem, radio device manufacturers, network designers and service providers are seeking novel design methodologies that allow the radio devices to, autonomously, determine their own optimal transmission parameters given their individual QoS requirements. The transmission parameters can pertain to different metrics such as power allocation polices, coding-modulation schemes, timing to transmit policies, among others.

By enabling such a decentralized operation, the wireless nodes would need to self-configure based only on their local information while relying on a sequence of observations of their environment, e.g., feedback of the signal to interference plus noise ratio (SINR) at the corresponding receiver or messages such as ACK/NACK (acknowledgment/ non-acknowledgment). This need for distributed self-configuring devices motivates the use of game theory.

For instance, game theory is a branch of applied mathematics which studies the interaction of several decisions makers that are seeking to optimize a common or individual metric (e.g., maximize their profit, minimize their cost, etc.). In this work, to model the interactions between the distributed radio devices, we are particularly interested in the so-called non-cooperative games in normal form [1]. In essence, a *non-cooperative game in normal form* consists three elements: (*i*) a number of players, e.g., the wireless devices, that are competing over a certain resource (e.g., the radio resources), (*ii*) the set of actions or strategies which correspond to the feasible transmission configurations that each radio device can adopt or select, and (*iii*) a utility function per device which measures the benefit obtained with the current transmit configuration given the transmit configurations adopted by the others. In a wireless network, a utility function often measures key QoS parameters such as spectral efficiency in bits per second per Hertz, bit error rates, energy efficiency in bits per second per Joule, and so on. The use of non-cooperative games in normal form has recently attracted a lot of attention in the wireless community, notably as a suitable model for distributed radio resource allocation problems (e.g., see [2][3] and references therein). The most renowned solution for a non-cooperative game is the *Nash equilibrium* (NE) introduced by John Nash in [4]. Within a wireless setting, an NE can be interpreted as the network state at which radio devices cannot improve their individual performance (as captured by a utility function) through a unilateral change of their strategy, i.e., of their adopted transmission scheme such as their power level. When operating at the NE, each radio device attains its highest possible performance given the transmission schemes of its counterparts.

Although the NE is an attractive solution concept, from a practical point of view, a wireless operator, a service provider, or even an end-user, are often more interested in guaranteeing a certain minimum QoS level (e.g., for voice or multimedia services) rather than in reaching the highest achievable performance. For example, consider a radio device that needs to operate voice services with telephone voice quality over the radio resources of a self-configuring decentralized wireless network. In such a setting, a typical utility function (assuming the network latency is shorter than 200 ms) for such a device is the transmission rate. For such a voice service, the radio device mainly seeks to ensure a minimum transmission rate of 8000 samples/sec [5]. Beyond this minimum QoS level, there is no particular incentive for the device to change its own transmission configuration since the required quality of service is already guaranteed. In fact, beyond certain thresholds, the human ears cannot even detect any difference in the QoS.

In such scenarios, using the NE as a solution concept might fail to model the real behavior of self-configuring wireless communication networks. As a result, in the presence of minimum QoS requirements, a more suitable solution concept for non-cooperative games is the generalized NE (GNE) that was proposed by Debreu in [6] and later by Rosen in [7]. Within a wireless network, a GNE is a game outcome in which all transmitters select their action in such a way that their performance cannot be improved by unilateral deviations and, at the same time, certain QoS levels can be guaranteed. This concept is of particular interest for scenarios in which one needs, not only a stable state of the game, but also certain guaranteed QoS levels. However, depending on the QoS metrics and the network topology, the GNE might not exist [2]. Even when it exists, at the GNE, a transmitter still end up achieving the highest achievable performance, which, in several scenarios, turns out to be costly in terms of energy consumption, spectral efficiency, *etc*.

Motivated by the need for an equilibrium state with QoS guarantees, this work introduces a new concept of equilibrium, named the *satisfaction equilibrium* (SE), in which players aim to exclusively guarantee certain QoS level independently of the actual achieved performance. At the SE, wireless radio devices have no incentive to modify their transmit configuration as long as their QoS conditions are satisfied. Following this definition, it can be immediately implied that the SE is not often unique. In fact, many transmission configurations might ensure the respective QoS requirements and each of them constitutes an SE. Note that unilateral deviations from a given SE might bring important performance improvements, however, given services that require only QoS guarantees, performance maximization is no longer the goal of the players, and does not provide any incentive for the deviation.

Interestingly enough, the only condition for the existence of the SE is the feasibility of all the QoS requests, which makes the SE concept less restrictive than other existing equilibrium concepts. For instance, there exists many wireless network games in which no NE or GNE exist (in pure strategies) but an SE exists. In such a setting, the SE is an outcome that the wireless users can reach, in a decentralized manner, while ensuring certain QoS guarantees. Additionally, the non-uniqueness of the SE can be seen as another advantage with respect to NE and GNE. In general, the set of GNE is a subset of the set of SE of a particular game. Thus, since the set of SE is larger, the time required to achieve it as a result of a trial and error learning process is likely to be shorter than the time required achieving any NE or GNE.

Beyond the SE concept, this work also presents an interesting refinement which is the concept of *efficient satisfaction equilibrium* (ESE). To better understand the ESE consider the following. Let each radio device arbitrarily define a mapping from its set of actions to the interval [0, 1] and let it quantify the effort associated with each one of its actions. Then, the ESE is a network state where all players satisfy their QoS requirements by using the feasible action which requires the smallest effort. It is important to note that, unlike the NE and GNE, for a given game, at least one ESE always exists as long as the QoS requirements are feasible and the set of actions is finite. Interestingly, this result is independent of the explicit form of the QoS metrics. In this case, the set of ESE is smaller than the set of SE of a given game. Thus, learning an ESE might take longer than learning a simple SE.

The concepts of an SE and an ESE are of both theoretical and practical importance for the design of future decentralized wireless networks. First, this work establishes and discusses the existence and uniqueness of these concepts within non-cooperative games in normal form. However, many theoretical aspects remain open. For instance, in this work, the concepts are restricted to *static* games, i.e., games in which the players act only once. An important future direction is to provide a generalization of the SE and ESE concepts suitable to dynamic games (e.g., stochastic games) in which the wireless devices act over time. The need for a dynamic version of the SE and ESE stems from the inherent time-varying nature of wireless communications networks. Moreover, while the SE and ESE have been well characterized in this work, a key next step is to design general algorithms that enable the wireless devices to reach an SE or an ESE in a fully decentralized fashion.

From a practical point of view, many of the emerging services, notably voice-based or multimedia services, require the mobile devices to guarantee certain QoS levels. While concepts such as NE or GNE have been abundant in the literature, these concepts do not enable any QoS guarantees for future services. Moreover, the recent interest in energy efficient and green communication motivates the adoptions of solution concepts such as the ESE, which allow the wireless devices, not only to maintain their minimum desired QoS levels, but also to do so with the lowest effort, e.g., lowest energy consumption. Hence, in essence, the SE and ESE concepts that were introduced in this work are bound to constitute an important operating point for future decentralized wireless systems, notably for voice-based or multimedia services.

References:

[1] D. Fudenberg and J. Tirole, “Game theory,” MIT Press, 1991.

[2] S. Lasaulce, M. Debbah, and E. Altman, “Methodologies for analyzing equilibria in wireless games,” *IEEE Signal Processing Magazine*, *Special issue on Game Theory for Signal Processing*, vol. 26, no. 5, pp. 41–52, Sep. 2009.

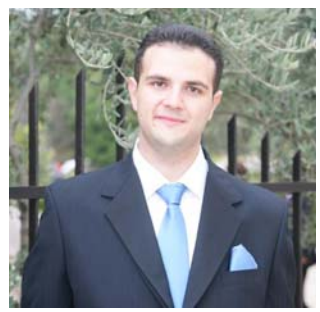
[3] V. Srivastava, J. Neel, A. Mackenzie, R. Menon, L. Dasilva, J. Hicks, J. Reed, and R. Gilles, “Using game theory to analyze wireless ad hoc networks,” *IEEE Communications Surveys & Tutorials*, vol. 7, no. 4, pp. 46–56, Quarter 2005.

[4] J. F. Nash, “Equilibrium points in n-person games,” *Proceedings of the National Academy of Sciences of the United States of America*, vol. 36,no. 1, pp. 48–49, 1950.

[5] D. Minoli and E. Minoli, Delivering voice over IP networks. New York, NY, USA: John Wiley & Sons, Inc., 1998.

[6] G. Debreu, “A social equilibrium existence theorem,” *Proceedings of the National Academy of Sciences of the United States of America*, vol. 38, no. 10, pp. 886–893, October 1952.

[7] J. B. Rosen, “Existence and uniqueness of equilibrium points for concave n-person games,” *Econometrica*, vol. 33, no. 3, pp. 520–534, 1965.



Walid Saad received his B.E. degree in Computer and Communications Engineering from the Lebanese University, Faculty of Engineering II, in 2004, his M.E. in Computer and Communications Engineering from the American University of Beirut (AUB) in 2007, and his Ph.D from the University of Oslo in 2010. From August 2008 till July 2009 he was a visiting scholar in the Coordinated Science Laboratory at the University of Illinois at Urbana Champaign where he pursued research activities during his 2nd year of the PhD.

Currently, he is a postdoctoral research associate at the Electrical Engineering Department at Princeton University. His research interests include applications of game theory in wireless networks, coalitional game theory, cognitive radio, autonomous communication and wireless communication systems (UMTS, WiMAX, LTE, etc.). He was the first author of the paper that received the Best Paper Award at the 7th International Symposium on Modeling and Optimization in Mobile, Ad Hoc and Wireless Networks (WiOpt), in June 2009 and the paper that received a Best Paper Award at the 5th International Conference on Internet Monitoring and Protection (ICIMP 2010) in May 2010.

Paradigm Shift in Network Security: Cross-Layer Attack and Defense

A short review for "Cross-layer attack and defense in cognitive radio networks"

Edited by Man-On Pun

W. Wang, Y. Sun, H. Li, and Z. Han, "Cross-Layer Attack and Defense in Cognitive Radio Networks", IEEE Global Communications Conference, Dec. 2010.

Cognitive radio networks (CRNs) have been envisaged as a promising solution to the radio spectrum scarcity problem. By allowing secondary users (SU) to "opportunistically" operate over under-utilized frequency bands licensed to primary users (PU), CRNs can dramatically improve spectral utilization [1-2]. However, the ubiquitous and distributed nature of CRNs makes them vulnerable to challenging security threats. In the literature, extensive studies have been devoted to different aspects of security issues in CRNs. In particular, [3] investigated the primary user emulation (PUE) attack in the physical (PHY) layer whereas [4] evaluated the small-backoff-window attack in the multiple access (MAC) layer. Despite their good performance in mitigating malicious attacks, most existing works assumed that the adversaries launched attacks only in one network layer. As a result, their performance remains unclear if an adversary can intelligently exploit the security vulnerabilities across multiple network layers coordinately.

Motivated by this drawback, the authors investigated cross-layer attacks in CRNs comprised of multiple SUs who jointly perform collaborative spectrum sensing. This work has provided interesting insights into the problem by investigating the case that an adversary intends to reduce spectral utilization by *simultaneously* launching the reporting false sensing data (RFSD) attack in the PHY layer and the small-backoff-window (SBW) attack in the MAC layer.

The major contributions of this work are twofold. First, improved defense schemes have been developed to derive trust values for PHY and MAC layers, respectively. Second, a unified defense architecture has been proposed to perform joint anomaly detection by exploiting the trust values reported from different layers. Through extensive simulation results, the authors have successfully demonstrated that (1) cross-layer attacks can potentially inflict significant damages to CRNs and (2) their proposed defense architecture can substantially reduce the maximal damages. In the rest of this review, each contribution will be briefly discussed.

In the RFSD attack, a malicious attacker intends to deceive SUs into thinking the presence of PU, even though the PU is actually absent. The attacker achieves this attack by reporting an intentionally inflated sensed energy level to the common fusion receiver. If the fusion receiver fails in detecting the false report, it will simply summarize all sensed energy reports from SUs as well as the attacker. As a result, it is more likely that the fusion receiver is misled to announce the presence of PU. Subsequently, all SUs will keep silence even in the absence of PU, which leads to inefficient spectrum utilization. To circumvent the RFSD attack, the authors have proposed a three-step defense scheme. The scheme first performs Neyman-Pearson hypothesis testing to detect the presence of PU. If it is determined that PU is absent, the scheme then proceeds to its second step and check if all SUs faithfully report their sensed energy levels. Finally, the scheme calculates the PHY-layer trust value by exploiting the results in the second step.

On the other hand, the SBW attack in the MAC layer maliciously provokes more frequent channel collisions by using a much smaller backoff window size than that of honest SUs. As a result, honest SUs are deprived of the transmission priority. However, it is technically challenging to detect the SBW attack as both regular and malicious backoff schemes are random processes. To cope with the SBW attack, the authors have proposed a modified Cramer-von Mises (C-M) test to detect malicious users of a reduced backoff window size. By exploiting a limited number of observations on each SU's channel contention behaviors, the modified C-M test measures each SU's backoff window size as compared to the regular one. The modified C-M test result is then utilized to calculate the trust value of the MAC layer. It is worth mentioning that the modified C-M test proposed by the authors is believed to be more advantageous compared to the conventional Kolmogoriv-

Smirnov (K-S) test [4] that only compares two different distributions by evaluating the difference of their maximum values.

Upon obtaining the trust values from the PHY and MAC layers, a fusion algorithm has been proposed to combine these trust values to generate an overall trust value. More specifically, the fusion algorithm produces a weighted sum of all trust values. Clearly, the weighting coefficients should describe the effectiveness of the trust values. In this work, the authors have proposed to determine the weighting coefficients by using the variances of the trust values. Finally, the overall trust value provided by the fusion algorithm is passed to an anomaly detector. Since this overall trust value summarizes information from different layers, it represents a more reliable and effective metric for anomaly detection.

In the last part of the paper, the authors have confirmed the performance of their proposed defense architecture via extensive computer experiments. The authors first investigated the damages caused by single-layer attacks, i.e. either RFSD or SBW attacks. Interestingly, it was found that the RFSD attack presents a much bigger threat to CRNs than SBW. Furthermore, the simulation results have also suggested that the conventional single-layer defense schemes are ineffective in capturing cross-layer attacks that caused substantially more damages than conventional single-layer attacks. Finally, it has been demonstrated that the proposed cross-layer defense architecture can significantly mitigate the maximal damages caused by cross-layer attacks.

To the reviewer's best knowledge, this work stands for the first report on defending cross-layer attacks by exploiting the correlation among attacks in different layers. It has clearly demonstrated the potentially severe damages caused to CRNs by non-conventional yet more intelligent cross-layer attacks. As the cross-layer attacks as well as the proposed cross-layer defense architecture can also apply to general communication networks, it is expected that this work will stimulate more research interests and awareness to safeguard communication networks (including CRNs) from cross-layer attacks.

References:

[1] I.F. Akyildiza, W.Y. Leea, M.C. Vuran and S. Mohanty, "NeXt generation/dynamic spectrum access/cognitive radio wireless networks: A survey," *Computer Networks,* vol. 50, no. 13, pp. 2127-2159, September 2006.

[2] S. Haykin, "Cognitive radio: brain- empowered wireless communications," *IEEE Journal on Selected Areas in Communications,* vol. 23, no. 2, pp. 201 - 220, Feb. 2005.

[3] R. Chen, J.M. Park and J.H. Reed, "Defense against Primary User Emulation Attacks in Cognitive Radio Networks," *IEEE Journal on Selected Areas in Communications,* vol. 26, no. 1, pp. 25 - 37, Jan. 2008.

[4] A.L. Toledo and X. Wang, "Robust Detection of Selfish Misbehavior in Wireless Networks," *IEEE Journal on Selected Areas in Communications,* vol. 25, no. 6, pp. 1124 - 1134, Aug. 2007.

Man-On Pun received the BEng (Hon.) in Electronic Engineering from the Chinese University of Hong Kong in 1996, the MEng. degree in Computer Science from University of Tsukuba, Japan in 1999 and the Ph.D. degree in Electrical Engineering from the University of Southern California (USC) in 2006, respectively. He joined the Mitsubishi Electric Research Labs (MERL), Cambridge, MA as Research Scientist in 2008. Prior to MERL, he held research positions at Princeton University from 2006 to 2008 and Sony Corporation, Tokyo from 1999 to 2001. Dr. Pun received the MERL president's award in 2009 and three best paper awards from Infocom 2009, ICC 2008 and VTC-Fall 2006. He serves as Associate Editor of the IEEE Transactions on Wireless Communications.

Green Multimedia Multicasting in Mobile Ad Hoc Networks

A short review for "A maximum-residual multicast protocol for large-scale mobile ad hoc networks "

Edited by Ai-Chun Pang

P.-C. Hsiu and T.-W. Kuo, "A Maximum-Residual Multicast Protocol for Large-Scale Mobile Ad Hoc Networks," IEEE Trans. on Mobile Computing, vol. 8, no. 11, pp. 1441-1453, Nov. 2009.

Advances in wireless communications have increased the popularity of mobile devices, thereby motivating a large variety of mobile applications and services to meet the various needs of users. In recent years, mobile users have become increasingly addicted to multimedia streaming and multimedia social network communities. However, communications are usually energy-intensive, and mobile devices are battery-operated. Therefore, how to reduce the energy consumption and prolong the usage time of mobile devices for communications is a major concern.

Mobile ad hoc networks are autonomous systems comprised of mobile devices, where packets are transmitted via intermediate devices, instead of an established infrastructure. Routing in such networks is very challenging because of the dynamic nature of network topologies and critical energy efficiency considerations. The problem is compounded by the existence of a huge population of devices and the need for efficient utilization of communication bandwidth, such as the replacement of multiple unicasts with a multicast. To address the above issues, this paper presents the Maximum-Residual Multicast Protocol (MRMP), a power-aware multicast protocol specially designed for multimedia multicasting in large-scale mobile ad hoc networks.

In the past decades, many excellent routing protocols have been proposed for mobile ad hoc networks. Each protocol attempts to optimize certain routing and performance metrics for different application scenarios, e.g., the propagation delay and the delivery ratio. MAODV [1], ODMRP [2], and DDM [3], which exemplify the best solutions, were submitted to the IETF MANET Working Group [4] as candidates for standardization. Routing over mobile ad hoc networks is complicated by considerations of energy efficiency, while the shortest paths are not favored in routing. In recent years, power-aware routing has received a great deal of attention and has yielded a class of fundamental optimization problems based on various routing metrics, such as minimum-energy routing [5] and maximum-lifetime routing [6]. However, most existing solutions rely on knowledge of certain global information, such as the remaining energy of all mobile devices and/or the minimum transmission power between every pair of devices. The maintenance problem of similar global information is highly challenging in protocol design because of the difficulty and cost of maintaining up-to-date information. As a result, various assumptions, such as static network topologies and/or fixed traffic patterns, are made to reduce the complexity of power-aware routing.

Unlike the past work, this paper considers applications that have a huge population of mobile devices, such that no global information can be efficiently maintained at any device. Depending on the duration of a multimedia multicast and the degree of network mobility, the multicast is partitioned into multiple sessions. The problem, referred to as maximum-residual routing, is formulated as the maximization of the minimum remaining energy of all devices in the network after each multicast session. The objective is to prolong the time before the first device fails, especially when network topologies and data traffic may change frequently in an unpredictable way.

To achieve the above objective, the authors propose a distributed algorithm for maximum-residual multicasting and prove its optimality without considering device movements and control overheads. When mobility and control message collisions are taken into account, it is shown that every derived route remains loop-free and converges toward an optimal solution in terms of maximizing the minimum residual energy. In addition, based on the proposed algorithm, the authors develop a source-initiated on-demand routing protocol, called MRMP, which is adaptable to network topologies and resources that may change over time. In MRMP, no periodic control message is employed to collect routing information or repair link breakages. Neither group membership nor neighbor relationship is maintained at any device by explicit control messages. When a source requires a route, it invokes a route discovery procedure over the network, and the individual decisions of intermediate devices form a loop-free multicast tree naturally.

For the performance evaluation, the protocol was implemented over NS2 [7]. The simulation results demonstrate that MRMP is effective and efficient in power-aware routing based on the parameter settings of Intel wireless devices [8] and the performance metrics concerned by the IETF MANET Working Group [4]. The proposed distributed methodology is also generalizable to various related optimization problems, such as the minimization of the total energy consumption of any path from a source to a destination, and provides useful insights into protocol design when network resources, such as bandwidth, may change over time.

References:

[1] E.M. Royer and C.E. Perkins, "Multicast Operation of the Ad Hoc On-Demand Distance Vector Routing Protocol," Proc. *ACM MobiCom*, pp. 207-218, 1999.

[2] S.-J. Lee, M. Gerla, and C.-C. Chiang, "On-Demand Multicast Routing Protocol," Proc. *IEEE WCNC*, pp. 1298-1304, 1999.

[3] L. Ji and M.S. Corson, "Differential Destination Multicast—A MANET Multicast Routing Protocol for Small Groups," Proc. *IEEE INFOCOM*, pp. 1192-1201, 2001.

[4] S. Corson and J. Macker, "Mobile Ad Hoc Networking (MANET): Routing Protocol Performance Issues and Evaluation Considerations," IETF RFC 2501, 1999.

[5] P.-J. Wan, G. C_alinescu, X.-Y. Li, and O. Frieder, "Minimum-Energy Broadcast Routing in Static Ad Hoc Wireless Networks," Proc. *IEEE INFOCOM*, pp. 1162-1171, 2001.

[6] A. Sankar and Z. Liu, "Maximum Lifetime Routing in Wireless Ad-Hoc Networks," Proc. *IEEE INFOCOM*, pp. 1089-1097, 2004.

[7] K. Fall and K. Varadhan, The ns Manual. UC Berkeley and LBL and USC/ISI and Xerox PARC, 2007.

[8] Intel PRO/Wireless 2011B LAN PC Card, ftp://download.intel.com/support/wireless/wlan/pro2011/wireless.pdf, 2008.

Ai-Chun Pang received the B.S., M.S. and Ph.D. degrees in Computer Science and Information Engineering from National Chiao Tung University, Taiwan, in 1996, 1998 and 2002, respectively. She joined the Department of Computer Science and Information Engineering (CSIE), National Taiwan University (NTU), Taipei, Taiwan, in 2002. Currently, she is a Professor in CSIE and Graduate Institute of Networking and Multimedia of NTU, Taipei, Taiwan. Her research interests include wireless networking, mobile computing, and performance modeling. She is a Senior Member of IEEE.

Meeting the Challenges of Internet Video Traffic via Scalable Video Coding

A short review for "Traffic and quality characterization of the H.264/AVC scalable video coding extension"

Edited by Kalpana Seshadrinathan

G. van der Auwera, P. T. David, M. Reisslein, and L. J. Karam, "Traffic and Quality Characterization of the H.264/AVC Scalable Video Coding Extension", Advances in Multimedia, Article ID 164027, Jan. 2008.

One of the challenges of multimedia communication today is to ensure consistent video quality and playback to a client that often moves through heterogeneous networks and/or sees widely varying network bandwidth. The increasing amount of video traffic on today's networks has seen the rapid deployment of adaptive streaming strategies that have been adopted in Apple's HTTP adaptive bitrate streaming, Adobe Flash's dynamic streaming, Microsoft Silverlight's smooth streaming *etc*. Many of these streaming strategies utilize a simple switching policy, where the server contains multiply encoded versions of a video and switches between these to match the current bitrate seen by the client. Scalable Video Coding (SVC) overcomes the inefficiencies in multiply encoding the same video content and is currently used in Google video chat.

The scalable video coding (SVC) extension (Annex G) to the H.264/AVC video coding standard [1] has unprecedented compression efficiency while supporting a wide range of scalability modes. SVC provides temporal scalability, spatial scalability, coarse (CGS) and medium (MGS) granularity scalability, as well as combined spatiotemporal SNR scalability, which allows a restricted set of spatiotemporal-SNR points to be extracted from a global scalable bit stream. While earlier scalable video encoders and receivers, such as MPEG-4 Part 2, did not gain wide market deployment, SVC is starting to be adopted by the videoconferencing industry and is expected to play a major role in providing video services over heterogeneous networks due to the significantly improved rate-distortion efficiency with respect to MPEG-4 Part 2.

Earlier work has studied the network traffic statistics of single-layer H.264/AVC streams [2] and single-layer SVC streams [3]. In this paper, the SVC traffic statistics of (i) temporal scalability with three temporal layers, (ii) spatial scalability with a QCIF base layer and a CIF enhancement layer, as well as (iii) quality scalability mode MGS are considered. The bit rate distortion and bit rate variability distortion are studied with long CIF resolution video sequences and compared against the corresponding MPEG-4 Part 2 traffic statistics. The traffic characteristics, especially the bit rate variabilities, of the individual layer streams critically affect their network transport.

Overall, it is found that SVC achieves significantly higher compression ratios than MPEG-4 Part 2, but produces unprecedented levels of traffic variability, thus presenting new challenges for the network transport of scalable video. It is also found that separately analyzing the traffic of temporal-scalability only encodings gives reasonable estimates of the traffic statistics of the temporal layers embedded in combined spatiotemporal encodings and in the base layer of combined SNR-temporal encodings.

All encodings of this study are publicly available as video traces at http://trace.eas.asu.edu/. Video traces are files mainly containing video frame time stamps, frame types (e.g., I, P, or B), encoded frame sizes (in bits), and frame qualities (PSNR). Video traces are employed in simulation studies of transport of scalable video over communication networks. Key advantages of simulating with video traces over experiments with actual video are that only very basic knowledge of video encoding is required for simulations with video traces and that video traces are freely available without copyright protection. Also, network simulations with video traces can be conducted with standard network simulation programs and integrated in network simulation modules (see, e.g., [4]), whereas experiments with actual video require in-depth video coding expertise and large computational resources for the encoding of many long video sequences.

The study of H.264 SVC traffic statistics in this paper and the video traces provided by the authors are valuable tools for researchers in designing the next generation scalable streaming technologies.

References:

[1] *Advanced video coding and generic audiovisual services*, ITU-T Rec. H.264 & ISO/IEC 14496-10, Edition 5.0, Mar. 2010. Available: http://www.itu.int/rec/T-REC-H.264

[2] G. Van der Auwera, P. T. David, and M. Reisslein, "Traffic Characteristics of H.264/AVC Variable Bit Rate Video," *IEEE Communications Magazine*, vol. 46, no. 11, pp. 164-174, Nov. 2008.

[3] G. Van der Auwera, P. T. David, and M. Reisslein, "Traffic and Quality Characterization of Single-Layer Video Streams Encoded with H.264/MPEG-4 Advanced Video Coding Standard and Scalable Video Coding Extension," *IEEE Trans. Broadcasting*, vol. 54, no. 3, pp. 698-718, Sept. 2008.

[4] O. Marques, P. Auger, and L. M. Mayron, "SimViKi: a tool for the simulation of secure video communication systems," in Proc. of 4th IASTED *Int. Conf. on Commun., Internet, and Info. Techn*. (CIIT '06), St. Thomas, Virgin Islands, USA, Dec. 2006

Kalpana Seshadrinathan received the B.Tech. degree from the University of Kerala, India in 2002 and the M.S. and Ph.D. degrees in Electrical Engineering from the University of Texas at Austin, in 2004 and 2008, respectively. She is currently a Research Scientist with Intel Corporation in Santa Clara, CA. Her research interests include image and video quality assessment, computational aspects of human vision, motion estimation and applications, computational photography and statistical modeling of images and video. She is a recipient of the 2003 Texas Telecommunications Engineering Consortium Graduate Fellowship and the 2007 Graduate Student Professional Development Award from the University of Texas at Austin. She was Assistant Director of the Laboratory for Image and Video Engineering (LIVE) at the University of Texas at Austin from 2005-2008. She is a member of the IEEE.

Paper Nomination Policy

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

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

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

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