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

R-LETTER

Vol. 1, No. 2, December 2010

IEEE COMMUNICATIONS SOCIETY

CONTENTS

Message from R-Letter Co-Director

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

Building on the success of the first issue, we are pleased to introduce four high quality papers in this issue. Two papers are from MMTC-sponsored conferences and two from IEEE journals.

The introduced papers span a variety of topics in multimedia communication: cognitive radio networks, video rate adaptation, video error correction, and multimedia security. The first paper analyzes voice capacity in cognitive radio networks and was published in the IEEE International Conference on Communications 2010. The second paper, also from the IEEE International Conference on Communications 2010, deals with rate adaption for streaming scalable video to wireless devices. The third paper, published in the IEEE Transactions on Image Processing, considers application-layer forward error protection for wireless video broadcasting. The fourth paper, published in the IEEE Transactions on Information Forensics and Security, considers behavior forensics in multimedia social networks.

We hope that this issue will be both informative and a pleasure to read.

Finally, I would like thank all the editors of this issue for their great work: Ai-Chun Pang, Cheng-Hsin Hsu, Vladimir Stankovic, and Hafiz Malik. I also would like to thank the R-Letter Director Guan-Ming Su for his vision and efforts that made the MMTC R-Letter possible.

Nabil J. Sarhan

Co-Director, IEEE ComSoc MMTC R-Letter
E-mail: nabil at ece.eng.wayne.edu

Voice over Cognitive Radio Networks - Capacity Analysis

A short review for "Voice capacity of cognitive radio networks"

Edited by Ai-Chun Pang

S. Gunawardena and W. Zhuang, "Voice capacity of cognitive radio networks,", Proc. of IEEE International Conf. on Communications 2010, May 2010.

The scarcity of the radio spectrum together with inefficient spectrum management has become a limiting factor for the development of wireless networks. The concept of cognitive radio networks (CRNs) has emerged as an efficient technique for improving radio spectrum utilization to support an increasing demand for wireless communication services. As an emerging networking technique, it is important to support popular multimedia applications such as voice and video conferencing, which have been successfully implemented over the existing legacy networks. However, supporting voice communication in cognitive radio networks is technically challenging, due to the randomness of spectrum availability and the real-time nature of voice communication. This piece of work provides some insights to the voice service support over CRNs.

The key service quality requirement of a voice call is based on the delay of received speech from the other party. In order to have better call quality, it is required to keep the mouth-to-ear delay of received speech within a desirable limit (i.e., to keep the end-to-end delay of a voice packet within a particular delay bound). The delay performance of a voice call is highly dependent on the spectrum resource availability for the particular call. In the GSM/CDMA type of cellular networks, each call is provided with a dedicated channel and a specific data rate, such that the delay requirement is always satisfied. However, in a wireless networking environment such as IEEE 802.11 based networks in which the users share the spectrum, meeting a deterministic delay requirement consumes more resources due to a less extent of statistical multiplexing. Therefore, instead of a deterministic delay requirement, a stochastic delay requirement is considered [1]. As secondary users of a CRN shares the spectrum opportunistically while giving priority to the licensed users, the spectrum availability of the CRN is random, and highly depends on the spectrum usage behavior of the licensed users.

The timely channel access of the secondary users plays an important role in providing timely packet delivery, which is the key aspect of providing delay guarantees. The channel access of the secondary users are governed by the scheduling scheme of a central controller or channel access schemes used by the secondary users for centralized and distributed network coordination scenarios, respectively. Some quality-of-service (QoS) based channel access schemes are studied in the literature [2]-[5], in terms of delay requirements. In [4][5], performance of the channel access schemes is analyzed in terms of the maximum number of simultaneous voice calls that can be supported by the system without violating the service quality requirement. In order to compare the efficiency of channel access schemes and packet scheduling schemes, some performance benchmarks of legacy schemes in CRNs are required.

Human speech in a voice call is digitized into a voice packet stream by the voice coder of a source node. A constant-rate voice encoder always generates voice packets at a constant rate irrespective of the silent periods or talk spurts of the speech. However, some voice encoders do not generate any packets during the silent periods of speech, which is equivalent to modulate a constant-rate voice packet flow by an on-off signal. It is called a silent suppressed (on-off) voice traffic flow. It is shown in the literature that the number of on-off voice traffic flows that can be accommodated in a system is higher than that of the constant-rate voice traffic flows, due to the reduced bandwidth requirement [6]. Therefore, it is important to study the system performance based on the on-off voice traffic flows.

This work focuses on providing a benchmark performance analysis for the first-in first-out (FIFO) scheduling scheme in a centrally coordinated cognitive radio voice network. In particular, it analyzes the maximum number of simultaneous on-off voice traffic flows that can be carried out by the CRN, satisfying their delay requirement. The voice traffic flows under consideration are independent single-hop traffic flows. The primary network under consideration is a time-slotted multichannel network, and the transmissions of licensed users are synchronized with time-slots. Different spectrum utilization statistics of licensed users are considered in the analysis.

The end-to-end delay of a voice packet depends on the processing delay, queuing delay, transmission delay, and the propagation delay. As the propagation, processing, and transmission delays of voice packets are very small when compared with the queuing delay, the end-to-end delay is governed by the waiting time of a voice packet in the queue. Therefore, the end-to-end delay requirement of a voice call is reduced to a queuing delay requirement, such that the waiting time of a voice packet in the source buffer should be kept below a certain delay bound with a certain probability. The analysis in [4][5] takes the queuing delay of voice packets as the QoS requirement for constant-rate voice traffic flows. Different from [4][5], in this work, since the source calls are modeled as on-off traffic flows, voice packet arrivals are random, and the Markov chain modeling used in [4][5] cannot be directly applied.

When the arrival rate and the service rate of a system are time varying, the theory of effective bandwidth (EB) [7] and effective capacity (EC) [8] can be used to analyze the maximum number of voice traffic flows that can be supported by the system, satisfying a certain delay requirement [1]. The theory of EB characterizes the required constant bandwidth to support a time-varying arrival process as a function of the QoS requirement of the service. The theory of EC is considered as the dual of EB. It is important to note that the validity of the theory of EC is conditioned on having a FIFO service process and voice traffic flows with identical QoS requirement. The system under consideration as a whole acts in a FIFO manner, and the delay requirements are identical in all voice traffic flows. Therefore, the theories of EB and EC are used to analyze the system capacity in terms of the number of simultaneous voice traffic flows.

It is shown that the analytical results match well with simulation results and stays slightly lower than the simulation results due to the conservative nature of the EB and EC theories, and the mean duration of channel being unavailable to the secondary users has a significant impact on the system capacity. It is assumed that the base station has packet timing information of all voice nodes in order to accomplish the FIFO scheduling. Even though it is very difficult to make this assumption a reality, it is an ideal situation for the base station to schedule voice packets when the delay requirements and packet sizes are the same for all voice traffic flows in the network. Therefore, this analysis can be considered as a benchmark for the performance analysis of more realistic scheduling schemes. With proper medium access control, the capacity analysis can help to develop a call admission control policy for QoS provisioning in cognitive radio networks

References:

[1] A. Abdrabou and W. Zhuang, “Stochastic delay guarantees and statistical call admission control for IEEE 802.11 single-hop ad hoc networks,” *IEEE Trans. Wireless Comm.* , pp. 3972-3981, Oct. 2008.

[2] X. Zhu, L. Shen, and P. Yum, “Analysis of cognitive radio spectrum access with optimal channel reservation,” *IEEE Communications Letters*, vol. 11, no. 4, pp. 1-3, Apr. 2007.

[3] L. Wang, A. Chen, and D. Wei, “A cognitive MAC protocol for QoS provisioning in overlaying ad hoc networks,” in Proc *IEEE CCNC ’07*, pp. 11391143, Jan. 2007.

[4] P. Wang, D. Niyato, and H. Jiang, “Voice service support over cognitive radio networks,” in Proc. *IEEE ICC’09*, June 2009.

[5] P. Wang, D. Niyato, and H. Jiang, “Voice service capacity analysis for cognitive radio networks,” *IEEE Trans. Vehicular Technology*, vol. 59, no. 4, pp. 1779-1790, May 2010.

[6] L. Cai, Y. Xiao, X. Shen, L. Cai, and J. Mark, “VoIP over WLAN: voice capacity, admission control, QoS, and MAC,” *Int. J. Communication Systems*, pp. 491-508, 2006.

[7] D. Wu and R. Negi, “Effective capacity: a wireless link model for support of quality of service,” *IEEE Trans. Wireless Communications*, vol. 2, no. 4, pp. 630-643, July 2003.

[8] D. Wu and R. Negi, “Effective capacity-based quality of service measures for wireless networks,” in Proc. *IEEE BroadNets’04*, pp. 527-536, Oct. 2004.

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.

Practical Rate Control of Scalable Video Streamed to Your Smartphones

A short review for "Advanced rate adaptation for unicast streaming of scalable video"

Edited by Cheng-Hsin Hsu

C. Liu, I. Bouazizi, and M. Gabbouj, "Advanced Rate Adaption for Unicast Streaming of Scalable Video", Proc. of IEEE International Conf. on Communications 2010, May 2010.

With the accelerated developments in the wireless communication technologies, mobile multimedia streaming rises as one of the most popular applications. In fact, a recent Cisco report [1] indicates that the multimedia traffic will represent more than 66% of the total mobile data traffic by 2014. The most critical feature of streaming services is the in-time delivery of media data for timely playback at the receiver. However, wireless networks are characterized by varying bandwidth (usually dependant on the number of users sharing the same channel) and by link layer loss. To cope with the dynamic characteristics of wireless networks, a plurality of rate adaptation approaches have been proposed and deployed. Among others, a rate adaptation method [2] based on the client buffer feedback mechanisms in 3GPP Packet-Switched Streaming Service (PSS) [3] was proposed. The 3GPP client buffer feedback-based rate adaptation method detects network bandwidth changes by continuously monitoring the amount of buffered media time, which ultimately results in a superior behavior compared to packet loss-based rate adaptation [4]. The reason is that reacting to changes in the buffered media time enables to smooth out short-term bandwidth variations, detect congestion early enough, and differentiate between packet loss due to congestion and that caused by the wireless transmission medium.

Recent advances in scalable video coding such as SVC/H.264 [5] have paved the way for further development of flexible and adaptive media streaming applications. One of the key advantages of SVC/H.264 is its support for coarse-granularity scalability (CGS) and medium-granularity quality scalability (MGS). MGS provides a larger number of quality layers with reasonable stepwise bitrate gaps between the different quality layers and with small decoding complexity compared to CGS. Furthermore MGS supports more flexible switching between the different quality layers than CGS by providing packet-based quality scalability [6]. Hence, deploying MGS in combination with rate adaptation was sufficiently promising to trigger several research efforts as well as some initial commercial implementations. To fully utilize the switching flexibility in SVC, knowledge of the reception status for each MGS sub-stream is required at the server. However, 3GPP PSS lacks the capability to signal client buffer feedback at the sub-stream level, particularly when the different layers are transmitted over the same RTP session to avoid continuous setting up and tearing down of RTP sessions.

Without the sub-stream level feedback, the rate adaptation method specified in 3GPP PSS fails to correctly estimate the buffered media time, since the server bases its estimation on the stream level feedback, thus assuming the same amount of buffered media for all sub-streams. Furthermore, the authors assert that it is worthwhile to assign different buffering targets to the different layers, as this will improve playback continuity through graceful degradation in case of drastic bandwidth drops. Consequently, the authors propose a novel fine-granular estimation of the client buffer status in conjunction with MGS for rate adaptation in unicast streaming sessions.

The MGS switching flexibility is exploited when adapting the transmission rate to the varying wireless link conditions. The feedback mechanism is defined as an extension to the PSS feedback tools. Based on the multi-buffer feedback mechanism, the server continuously monitors the reception status of each sub-stream and adapts its transmission rate accordingly to provide accurate and fast reaction to congestion signals. Furthermore, an inverse pyramid-like threshold selection for the buffer level of the different sub-streams is used to improve playback continuity in case of a sudden drop in bandwidth.

Simulation results show that the proposed multi-buffer feedback-based rate adaptation algorithm outperforms the traditional 3GPP PSS compliant rate adaptation method by quickly adapting the temporal/spatial resolution to the varying network resources without unnecessary bouncing among different temporal/spatial resolutions. As graceful degradation, interruption-free playback, and video quality stability are essential for a good user experience, the proposed algorithm shows the potential of achieving the highest possible video quality and the best user experience for unicast streaming applications in wireless environments.

Modifications to the 3GPP PSS standard to support the proposed rate control algorithm are being considered in 3GPP in conjunction with the adoption of SVC for high-quality mobile streaming services. Other future research directions include: (i) quantitative comparison between an SVC-based streaming system with the proposed rate adaptation algorithm against a nonscalable streaming system with 3GPP PSS compliant rate adaptation method, and (ii) extension of the proposed rate adaptation algorithm to video multicast services which may become more popular in next generation multicast-enabled wireless networks, such as WiMAX and LTE.

References:

[1] Cisco visual networking index: Forecast and methodology, 2009—2014, http://www.cisco.com/en/US/solutions/collateral/ns341/ns525/ns537/ns705/ns827/white_paper_c11-481360.pdf.

[2] T. Schierl, M. Kampmann, and T. Wiegand, “3GPP Compliant Adaptive Wireless Video Streaming Using H.264/AVC,” in Proc. of *IEEE Int. Conf. Image (ICIP’05)*, Genova, Italy, Sep. 2005.

[3] 3GPP TS 26.234, “Transparent End-To-End Packet-Switched Streaming Service (PSS): Protocols and Codecs (Release 9)”, Sep. 2009.

[4] S. McCanne, V. Jacobson, and M. Vetterli, "Receiver-driven layered multicast," in Proc. of *ACM SIGCOMM'96*, Stanford, CA, Aug. 1996, pp. 117–130.

[5] ITU-T Rec. H.264 and ISO/IEC 14496-10 (MPEG-4 AVC), ITU-T and ISO/IEC JTC 1, Advanced Video Coding for Generic Audiovisual Services, Version 8 (including SVC extension): Nov. 2007.

[6] H. Schwarz, D. Marpe, T. Wiegand, “Overview of the Scalable Video Coding Extension of the H.264/AVC Standard,” *IEEE Trans. Circuits Syst. Video Technol*., vol. 17, no. 9, pp. 1103 – 1120, Sep. 2007

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).

Application Layer Forward Error Protection for Wireless Video Broadcasting

A short review for "Sliding-window Raptor codes for efficient scalable wireless video broadcasting"

Edited by Vladimir Stankovic

P.Cataldi, M. Grangetto, T. Tillo, E. Magli, and G. Olmo, "Sliding-window Raptor codes for efficient scalable wireless video broadcasting with unequal loss protection," IEEE Trans. on Image Processing, vol. 19, no. 6, pp. 1491-1503, June 2010.

Multimedia streaming applications over multiple content delivery networks to users with different receiver capabilities (e.g., display resolution, power resources, and bandwidth) are emerging. Network and user heterogeneities impose many technological challenges to multimedia processing calling for efficient compression tools, robust and error resilient coding, and scalability.
A promising development in scalable video coding is recent Annex G of H.264/AVC standard known as H.264 Scalable Video Coding (SVC) [1], which provides efficient scalability functionalities on top of the high coding efficiency of H.264/AVC. SVC encodes the video source into a base layer and several enhancement layers providing temporal, spatial, and quality scalability and any combination thereof, bringing huge performance gains compared to transcoding or simulcasting.

Source scalability introduces significant disproportion in importance of the video output. Indeed, the base layer is the most important and its loss makes all subsequent layers useless. To achieve graceful degradation, maximize performance and keep the overall redundancy low, unequal error protection (UEP) strategies are necessary. UEP can be achieved at different layers. Physical-layer coding and processing can provide UEP via adaptive modulation and coding and hierarchical modulation. Some wireless systems, such as DVB-H [2], adopted optional systematic Reed-Solomon (RS) coding for packet loss protection at the link layer where by means of puncturing, coding rate may be adapted to network and source characteristics. Though being capacity-achieving, RS decoding complexity requires hardware implementation at the lower stack layers, which limits their adaptability to the source content.

As an alternative, recently, application-layer forward error protection (AL-FEC) coding [3] has been considered as a solution for packet loss protection in wireless systems. The main advantage of AL-FEC is its flexibility and hence increased adaptability to the source content since it is implemented in software at the AL. Digital Fountain Raptor codes [4] are adopted as an AL-FEC solution in the DVB-H standard [2] for IP datacasting services, such as file transfer, and standardized by the 3GPP in the context of multimedia broadcast multicast services (MBMS) [5]. Raptor codes are being considered for real-time multimedia services due to their near-optimal performance, linear decoding complexity, and the rateless property which enables simple adaptation to heterogeneous network conditions (see [3, 6] and references therein).

Raptor codes are initially designed for large source messages [4], and for short messages their performance suffers small rate penalty. Thus Raptor codes originally proposed in [4] are not an ideal solution for real-time delay-sensitive applications which require short data blocks.

One possible solution to this problem is the sliding-window approach applied to Raptor codes. The main idea behind the sliding-window approach is to apply Raptor coding not on the disjoint source blocks, but instead on a sliding window of source symbols.

In traditional (Raptor) coding, each block of length K symbols is encoded producing a stream of output symbols. After encoding of a data block, the encoding window is shifted for $S=K$ symbols, enabling encoding of the next K symbols. In the sliding window approach, the shift S is kept smaller than K, implying that each symbol enters more encoding windows. This process virtually increases the length of the source block from K symbols to $2K-S$ symbols.

The code design challenge is solved by imposing the pre-code block length to be equal to S. That is, the source block is first encoded by a binary block code (LDPC code) that spans S symbols, and then the encoded bits are grouped into symbols and fed to the conventional LT code [4].

Sliding-window (SW) Raptor codes are suitable for UEP as shown by simulation results. Each H.264/SVC group of pictures (GOP) of each layer is separately encoded using an SW-Raptor code, and full search is conducted to find the optimal source-channel bit allocation. Performance improvement of roughly 0.25 dB compared to the traditional Raptor code [4] is observed; additionally, the SW Raptor code is robust to channel parameters mismatch.

The price to pay is a small increase in decoding complexity, which decreases with the increase of the code overhead ε. Unfortunately, the main performance gains are for low overheads.

The SW approach differs from the expanding window (EW) approach [7], recently developed for UEP with digital fountain and random linear codes (see [8] and references therein). The SW approach improves error protection capability of the Raptor code [4] by increasing decoding complexity, and it applies UEP in the same way as the traditional Raptor codes (each layer is protected with one code). EW [7], on the other hand, is a flexible solution for UEP, sometimes at the expense of a slight decrease in error protection capability.

It would be important to compare the SW Raptor coding approach to the standardized 3GPP systematic Raptor code, which performs almost equally well for large and short block sizes and comes very close to optimal performance. The SW Raptor code seem to be a natural solution for online applications, where the importance of the content decreases with time. Another research direction is providing fast and efficient optimization methods for source-channel symbol allocation when multiple layers are used.

References:

[1]H. Schwarz, D. Marpe, and T. Wiegand, "Overview of the Scalable Video Coding Extension of the H.264/AVC Standard," *IEEE Trans. Circuits & Sys. Video Techn*, vol. 19, pp. 1103-1120, Sept. 2007.

[2] DVB-H [online] http://www.dvb-h.org.

[3] T. Stockhammer, A. Shokrollahi, M. Watson, M. Luby, and T. Gasiba, Application Layer FEC for Mobile Multimedia Broadcasting (in Handbook of Mobile Broadcasting), CRC Press, 2008.

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

[5] 3GPP [online] http://www.3gpp.org.

[6] D.G. Barquero, K. Nybom, D. Vukobratovic, and V. Stankovic, "Scalable video coding for mobile broadcasting DVB systems," Proc. *ICME-2010 IEEE Intl. Conf. Multimedia & Expo*, Singapore, July 2010.

[7] D. Vukobratovic, D., V. Stankovic, D. Sejdinovic, L. Stankovic, and Z. Xiong, "Scalable video multicast using Expanding window fountain codes," *IEEE Trans. Multimedia*, vol. 11, pp. 1094-1104, Oct. 2009.

[8] D. Vukobratovic and V. Stankovic, "On unequal error protection random linear coding for scalable video broadcasting," Proc. *PV-2010 IEEE Packet Video Workshop*, Hong Kong, China, Dec. 2010.

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 Assistant Professor, and Lancaster University as a Lecturer. From 2007, he has been with the Department of Electronic and Electrical Engineering, University of Strathclyde, Glasgow, where he currently holds a Senior Lecture post (equivalent to Associate Professor).

He has co-authored over 100 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.

Multimedia Social Networks: Behavior Forensics

A short review for "Behavior forensics with side information for multimedia fingerprinting social networks"

Edited by Hafiz Malik

W.S. Lin, H.V. Zhao, and K.J.R. Liu, "Behavior Forensics with Side Information for Multimedia Fingerprinting Social Networks", IEEE Trans. on Information Forensics and Security, vol.4, no. 4, pp.911 -927, Dec 2009.

Recent advances in communication, networking and multimedia production and manipulation have sparked proliferation of multimedia applications. The emergence of large-scale multimedia social network communities such as Napster, YouTube, Dailymotion, CoolStreaming, PPLive, etc. where millions of users form a distributed infrastructure to share multimedia content [1]–[3], has also prompted research activities in the areas ranging from secure media distribution to infringement, modeling of user dynamics that influence human behavior and impact of human factors on multimedia systems [4], [5], and influence of user cooperation on the performance of "*traitor tracing*" systems. User cooperation while sharing multimedia on social networks enables users to access extra resources from their peers and thus receiving higher payoffs, while each user also needs to contribute his/her own resource to help others. Recent studies have shown that it is easy for users to cheat and manipulate such cooperation based systems to further increase their payoffs, for example, peer-to-peer file sharing [6], peer-to-peer gaming [7], visual cryptography for secret sharing [8], etc. Cheat detection and prevention, therefore, is one of the fundamental requirements to ensure *fair* user cooperation in multimedia social networks.

To address aforementioned problems, the authors investigate two important issues in multimedia social networks, that is, 1) impact of the dynamics between two groups of users (e.g. colluders and the fingerprint detector) in the social network when side information is available, and 2) modeling of user dynamics using a game-theoretic framework and finding the optimal strategies for all users.

In multimedia fingerprinting, colluders and the fingerprint detector influence each other's decision and performance. To maximize their own payoff, each player should observe and learn how other players are playing their games and adjust his/her strategy accordingly. To develop a robust collusion detection system, authors extend their previous work [9] on behavior forensics which assumed that the fingerprint detector has no information about the colluder. This paper investigates the impact of side information on the performance of traitor tracing problem. Authors have shown that if some information of collusion attacks can exploited during colluder identification process, intuitively this will improve the traitor tracing performance. Unlike side channels in digital communication, side information about collusion in multimedia fingerprinting systems can only be extracted from the colluded copy. This paper considers the worst case for the fingerprint detector, that is, only colluded copy is available at the detector.

This paper also studies user behavior in the multimedia fingerprint social networks by modeling the complex dynamics of the users in the social network using game theory and proposes optimal strategies of both players in the game. Authors have also shown that side information about collusion improves the traitor-tracing capability and influence the strategies of the colluders and the forensic detector. Authors also present techniques that enable the detector to probe and utilize side information and analyze its performance. This paper shows that the mean value of the detection statistics of each user is useful side information and can significantly improve the detection performance. Theoretical analysis and simulation results show that the side information about the means of the detection statistics can help the detector significantly improve the collusion resistance. Authors then propose a method for the detector himself/herself to probe such side information from the colluded copy. Simulation results demonstrate that the proposed self-probing detector has approximately the same performance as the optimal fingerprint detector, and the difference between these two can be ignored.

This paper also demonstrates that side information not only improves the fingerprint detector's collusion resistance but it also affects each colluder's probability of being detected and makes some colluders take a larger risk than others. Thus, it disturbs the collective fairness equilibrium between the colluders and the fingerprint detector, and they have to choose different strategies. This paper models the colluder-detector dynamics with side information as a zero-sum game and shows that under absolute fairness of the attack the assumption, the *min-max* solution achieves the equilibrium which is the optimal strategy of all users in the multimedia fingerprint social network. Neither the colluders nor the fingerprint detector can further increase their payoff and, therefore, they have no incentive to move away from this equilibrium.

The proposed behavior forensics with side information for multimedia fingerprinting method can be extended along several directions. For example, extending proposed game-theoretic framework to multimedia forensics where attacker manipulates forgery detector statistics to bypass detector.

References:

[1] X. Hei, Y. Liu, and K. W. Ross, "IPTV over P2P streaming networks: The mesh-pull approach," *IEEE Commun. Mag*., vol. 46, no. 2, pp. 86–92, Feb. 2008.
[2] E. Lua, J. Crowcroft, M. Pias, R. Sharma, and S. Lim, "A survey and comparison of peer-to-peer overlay network schemes," *IEEE Commun. Surveys Tutorial*, vol. 7, no. 2, pp. 72–93, Mar. 2004.
[3] X. Cheng, C. Dale, and J. Liu, "Statistics and social networking of YouTube videos," in Proc. *IEEE Int. Workshp Quality of Service (IWQoS),* pp. 229–238, Jun. 2008.
[4] C. Buragohain, D. Agrawal, and S. Sur, "A game theoretic framework for incentives in P2P systems," in Proc. 3rd *Intl. Conf. Peer-to-Peer Comput*., Sep. 2003, pp. 48–56.
[5] G. Tan and S. A. Jarvis, "A payment-based incentive and service differentiation mechanism for peer-to-peer streaming broadcast," in *Proc Int. Workshop Quality of Service (IWQoS)*, Jun. 2006.
[6] T. Locher, P. Moor, S. Schmid, and R. Wattenhofer, "Free riding in BitTorrent is cheap," in Proc. 5th *Workshop Hot Topics in Netw. (Hot-Nets)*, Irvine, CA, Nov. 2006.
[7] N. E. Baughman, M. Liberatore, and B. N. Levine, "Cheat-proof playout for centralized and peer-to-peer gaming," *IEEE/ACM Trans. Netw*., vol. 15, no. 1, pp. 1–13, Feb. 2007.
[8] G. Horng, T. Chen, and D. Tsai, "Cheating in visual cryptography," Designs, Codes, Cryptography, vol. 38, no. 2, pp. 219–236, Feb. 2006.
[9] H. Zhao, and K.J.R. Liu, "Behavior forensics for scalable multiuser collusion: Fairness and effectiveness," *IEEE Trans. Inf. Forensic Security*, vol. 1, no. 3, pp.311-329, Sep. 2006.

Hafiz Malik is an Assistant Processor in the Electrical and Computer Engineering (ECE) Department at University of Michigan – Dearborn. His research in multimedia forensics and security, wireless sensor networks, steganography/steganalysis, and biometric security is funded by the National Academies and other agencies. He has published more than 35 papers in leading journals, conference, and workshops. He is on the Review Board Committee of IEEE Technical Committee on Multimedia Communications (MMTC). He organized Special Track on Doctoral Dissertation in Multimedia, in the 6th IEEE International Symposium on Multimedia (ISM) 2006. He is serving as secretary of IEEE SEM, Chapter III since 2009 and is also serving on several technical program committees, including the IEEE AVSS, ICME ICIP, MINES, ISPA, CCNC, and ICC.

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