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CONTENTS

Message from the E-Letter Director

Welcome to the March Issue of E-Letter! This is the first issue published by the new editorial team, whose names can be found at the end of this issue. We would like to thank the former E-letter Editor-in-Chief Dr. Haohong Wang and MMTC Chair Prof. Qian Zhang for their strong support. Thanks!

First I would like to call your attention to page 3 for the Call for Paper of IEEE GLOBECOM 2010, to be held during December 6-10 in Miami. The deadline for regular paper submission is **March 15**. Do not miss this great opportunity for planning your winter vacation at this beautiful beach city.

I would like to thank Dr. Xiaoqing Zhu (Cicso, USA), who puts together an excellent special issue on *Network Coding for Multimedia Communications*, with three invited papers contributed from world top scientists in the field. Please check out this special issue starting from Dr. Zhu's Guest Editorial on page 4.

In the Technology Advances Column, Dr. Kai Yang (Bell Labs, Alcatel-Lucent, USA) puts together a fantastic collection of four invited papers from world-renowned experts with the common theme of *Distributed Multimedia Networking*. Please check out the column starting from Dr. Yang's Editorial on page 17.

In the Editor Recommended Paper Column on page 37, a paper published in IEEE Trasactions on Image Processing in November 2009 is recommended by the Column Editor, Dr. Guan-Ming Su (Marvell Semiconductors, USA).

Finally, Prof. Xinbing Wang (Shanghai Jiaotong University, China) has worked closely with the MMTC officers to provide us with the minutes of the last MMTC meeting as well as the Call for Papers of the IEEE GLOBECOM CSSMA Symposium that is sponsored by MMTC. Details can be found on pages 38 - 41.

I would like to thank all the editors and authors for the hard work and great contributions. I also look forward to valuable feedback and suggestions from our MMTC members.

Best Regards,

Jianwei Huang
Director, IEEE MMTC E-Letter
http://home.ie.cuhk.edu.hk/~jwhuang/

IEEE GLOBECOM 2010 @ MIAMI

MIAMI: Moving Into the Age of Mobile Interactivity!!!

Message from the GC'10 TPC Chair

One of the most prestigious conferences in communications, IEEE Global Communications Conference (IEEE GLOBECOM 2010), will be held on 6-10 December 2010 in Miami, Florida, USA.

Other than the traditional 11 symposia and business forum in the technical program, IEEE GLOBECOM 2010 brings in many new features:

- **FREE Tutorial Program**, it is the first time in history that GLOBECOM opens its high-quality tutorial sessions to all conference attendees for free;
- **Plenary Feature Talks**, 12 world-top scientists are committed to deliver 45-minute position talks to GC10 audiences;
- **Plenary Panels,** 6 panels (on different topics) with 24 world-top scientists and executives of major industries are invited to deliver their vision for future technologies to GC10 audiences;
- **Funding Forum**, officers from NSF, DARPA, and others funding agencies are invited to share information and tips to GC10 faculty audiences;
- **Early-Bird Student Award**, a new award for students to recognize representatives of productive student authors who also have good paper submission manners;
- **Expanded Workshop Program**, the double-sized (in total of 23) workshop program have a good coverage of research topics that are complimentary to GC10 symposia;
- **Fair Review Process**, GC10 enforces fair review process by executing conflict-of-interest rules that not allow TPC Chairs and symposium Chairs to submit papers to 11 symposia. This way, the papers from general authors would not compete directly with papers from any people who has power in GC10 review process.

More information regarding to GC10 can be found at www.ieee-globecom.org/2010

It is really amazing to notice that more than 15 MMTC members will speak at GC10 and I just name a few of them in below:

- Mung Chiang, Princeton University
- Philp Chou, Microsoft Research
- Adam Drobot, Telecordia Technologies
- Andrea Goldsmith, Stanford University
- Peter Grant, University of Edinburgh
- Lajos Hanzo, Univ. of Southampton
- Jenq-Neng Hwang,Univ. of Washington
- Aggelos Katsaggelos, Northwestern University
- P. R. Kumar, UIUC
- C. C. Jay Kuo, University of Southern California
- Jin Li, Microsoft Research
- H. Vincent Poor, Princeton University

We encourage all MMTC members to submit papers to IEEE GLOBECOM 2010 by **March 15, 2010**. Please submit your papers to **CSSMA Symposium** (Communications Software, Services and Multimedia Applications), which is sponsored by MMTC.

At the end, we welcome you to enjoy the winter sunshine and beach activities at Miami during the GLOBECOM 2010 conference in December.

Thank you and best regards,

Haohong Wang
Technical Program Chair,
IEEE GLOBECOM 2010

SPECIAL ISSUE ON NETWORK CODING

Network Coding for Multimedia Communications

Guest Editor: Xiaoqing Zhu, Cisco Systems Inc., USA
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The main challenges in designing multimedia communication systems include how to efficiently utilize available network capacity, how to provide loss protection for contents sent over unreliable channels, and how to best coordinate the behavior of multiple senders and viewers in a complex network topology. Recent advances in the research of network coding reveal a promising avenue of new solutions for these challenges. In particular, with the development of practical network coding techniques, researchers are now investigating how network coding can be leveraged in the design of multimedia systems, for a wide range of application scenarios.

This special issue provides a few sampling points in this direction. Our first article, "Network Coding and Multimedia Content Delivery" by Dah Ming Chiu, gives a very nice overview on the progress and promises of network coding theory and applications. The author then discusses how network coding can be applied to multimedia content delivery networks, argues why the approach is attractive for enhancing peer-to-peer content delivery schemes, and also highlights the potential gains that can be achieved via Random Network Coding. The article also offers the author's view on where open research problems lie in this area.

In the second article, "Network Coding for Peer-Assisted Multimedia Streaming" by Chen Feng and Baochun Li, the authors investigate the design challenges in peer-assisted video streaming systems. Here, peers also contribute their uplink bandwidth for content distribution, thereby offloading the burden of the streaming server. The authors briefly describe a scheme called R^2, based on random push operations with random network coding. The R^2 scheme provides a concrete illustration, showing how network coding can help to address issues such as efficient utilization of peers' uplink bandwidth, limited buffering delay, and scalability in a video peer-assisted streaming system.

The potential merits of network coding for wireless video are investigated in the third article, "Network Coding for Wireless Video Communications" by Thinh Nguyen. The article describes two promising schemes, one designed for scheduling video packets for multiple users at a wireless base station; the other designed for flow mixing in a time-slotted WiMax base station. In both examples, network coding allows information across multiple users to be mixed before transmission at the base station, hence ensures that the transmission is at least useful for a subset of the receivers. The author shows in the article that this approach achieves higher spectrum efficiency than transmitting packets individually, and argues for a video-aware optimization framework to fully leverage network coding techniques.

Once again, we thank our contributing authors in sharing their gems of thoughts and recent achievement of their own research in this field. Our selection is by no means comprehensive. Rather, we hope the articles presented in this special issue can help to ignite further explorations in various open research problems in this exciting field.

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University, Beijing, China, in 2001. She earned both the M.S. and Ph.D. degrees in Electrical Engineering from Stanford University, California, in 2002 and 2009, respectively. She was awarded the Stanford Graduate Fellowship from 2001 to 2005, and was recipient of the best student paper award in ACM Multimedia 2007. She interned at IBM Almaden Research Center in 2003 and at Sharp Labs of America in 2006. Her research interests include wireless video streaming, distributed resource allocation for video streaming systems, and adaptation techniques in Internet video delivery.

Network Coding and Multimedia Content Delivery

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1. CDN and P2P

Multimedia content delivery is projected to be by far the biggest bandwidth consumer of the Internet. For many years, the mechanism for content delivery envisioned by the networking community is network multicast. It is arguably the most efficient. By having the network routers provide the duplication of content and distribute it along a tree spanning all the nodes wanting to receive the content, the least amount of bandwidth can be expended. But multicast has some major limitations. (1) Although it economizes on bandwidth usage, it may not provide the necessary bandwidth to all users. To accommodate users' needs better, multiple distribution trees may be needed, and the determination of the most suitable trees depends on topology and other policy parameters – it is a combinatorial optimization problem too challenging for routers to solve. (2) The multicast service, by requiring a group of users to receive the content simultaneously, is too restrictive. Due to these limitations, it plays only a limited role as a kind of TV services in carefully provisioned edge networks, referred to as IPTV [1].

A more flexible service for users is Video-on-Demand (VoD). The most primitive VoD is based on a client-server solution, which does not scale as the number of users grows. To deal with the scalability problem, Content Delivery Network (CDN) is created. Users are redirected from the main server to servers less busy, or closer to them. CDN plays an extremely important role in providing content delivery in today's Internet, as a large percentage of web-based content are delivered through CDNs [2]. The drawback is that it does not dynamically adapt to user needs, so careful planning and management is needed.

In the last few years, Peer-to-Peer (P2P) systems for content delivery have been developed and been proven to work quite well. BitTorrent (BT) [3] is the most well-known (and perhaps the first) P2P system for file downloading. Coolstreaming [4], perhaps the first to demonstrate the feasibility of P2P streaming, is the work of an MPhil student in my department. By new, many P2P systems using slightly different algorithms have been deployed. It is reported that P2P traffic has become the dominant bandwidth consumer in many networks.

The most distinct feature of BT-like P2P systems is that they divide the content into multiple sub-streams, or if the content is a file then it is divided into multiple chunks. These different sub-streams or chunks will then follow different distribution trees to reach all the receivers, simultaneously. BT and many systems form these distribution trees adaptively and on-demand, depending on who has what, and who is available and willing to serve. In fact, [3] likens the BT mechanism of matching up the providers and receivers (of chunks) with the mechanisms of bringing together sellers and buyers in an economic system. The problem of setting up these multiple distribution trees and allocating bandwidths to different peers is of the same nature, and more complicated than the combinatorial problem network multicast has to solve; the difference is that the peers, not situated at the network cross-roads like routers, are in a better position to handle it.

Although P2P systems have often been used to distribute content violating copy-right protection, this technology is indeed a new paradigm for network content delivery. I like to compare it to the advent of *packet switching*. In the early days of contemporary data networks, the prevailing idea was based on circuit switching – emulating the telephone network service that had been in service for many years. It was the seemingly chaotic packet switching that helped the development of a whole generation of new applications with more bursty traffic for relatively limited bandwidth, including emails and the web. For content distribution networks, multicast is like virtual circuits, emulating the broadcast TV service that have existed for many years, and efficient. But it is the adaptive, multi-path P2P technology that is more likely to satisfy the on-demand, high definition/demand nature of future content distribution needs.

2. The Promise of Network Coding

Network coding is another idea that has clearly caught people's fancy. In less than ten years, more than 1800 papers have been published on the subject. Network coding was invented in my department by Raymond Yeung (who works on Information Theory) and Bob Li (who works on switching theory), among others helping them [5]. Claud Shannon's Information Theory elegantly characterized the capacity of a communication channel. Network Coding is the counterpart of Information Theory applied to a network. It addresses a very important question for network content distribution, albeit in a highly abstract and theoretical manner; that is, what the capacity (or maximum throughput) of a network for distributing content to a set of users is, and how this capacity can be achieved.

It is well-established that the maximum throughput cannot exceed the min-cut between the source of the content and all the receivers. However, it is easy to construct an example to show that no matter how well-informed and smart you are, there are networks for which you cannot arrange the routing and scheduling to achieve this theoretical throughput. The theory of network coding proves that you can always achieve this theoretical throughput by making the intermediate nodes apply some coding (e.g. construct linear combinations of different pieces of the original content). The most often used example is the butterfly network, explained in many papers on network coding including a paper I co-authored several years ago [6].

3. The Application of Network Coding

For evaluating P2P content distribution systems, as explained in [6], the most useful model is the *uplink sharing model,* the term originally coined in [7]. This model assumes the network is not the bottleneck in content distribution. Rather, the bottleneck is the total uplink available at the edge – the content server and the peer nodes (including cache servers). For today's Internet, this model more or less reflects the reality. Based on this model, it is possible to determine the capacity (maximum throughput) for P2P content distribution, assuming we have perfect routing and scheduling, as explained in [6]. The result can be expressed as a simple closed form equation. This capacity, it is shown, happens to correspond to the min-cut from the content server to the set of given receivers. This result is a negative result, as it implies network coding does not help to increase the capacity for networks satisfying the uploading sharing model conditions.

Does this mean network coding will remain a theoretical curiosity only in its application to network content distribution? Not necessarily so. Although network coding may not increase the maximum throughput in common network scenarios, it may be still be quite helpful. It turns out that achieving the maximum throughput in a P2P network is not an easy feat. Peers must exchange lots of information about what they each have, and carefully schedule how they serve others. Random Network Coding, a special algorithm to randomly mix different chunks of a file, can greatly ease the maneuvers needed to maximize throughput. This is because by spreading randomly mixed chunks around, it helps to increase the likelihood that peers have the content others need, hence making the scheduling job easier. This may prove to help achieving higher throughput by minimizing routing overheads and scheduling complexity. A prototype of such a system (called Avalanche) was built by Microsoft Research a few years ago, and the reported results were interesting [8]. The Random Network Coding scheme can also be helpful for networks not satisfying the uplink sharing model. When there are bottlenecks inside the network (not the uplinks), it is not obvious how to program the network to apply network coding targeting the bottlenecks. It is suggested that Random Network Coding can automatically target network bottlenecks to help maximize throughput [9]

4. Conclusions

Network content distribution, both in streaming or non-streaming forms, is envisioned to be the most important types of network activities. CDN and P2P types of platforms will enable many new applications to allow different ways of generating/publishing contents. Network coding is an intriguing new idea for helping to realize these systems. There are many challenges, and many research opportunities.

A new Institute of Network Coding (INC) will be formed at CUHK, located in my department, with the original inventors as directors. I use this occasion to wish this effort to be a great success.

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Network Coding for Peer-Assisted Multimedia Streming

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Introduction

Peer-assisted multimedia streaming has recently witnessed unprecedented growth on the Internet, delivering live streaming content to millions of users in real-world applications. The essential advantage of peer-assisted streaming is to dramatically increase the number of peers a streaming channel may sustain with a limited pool of available bandwidth provided by dedicated streaming servers. Intuitively, as participating peers contribute their own upload bandwidth to serve one another in the same channel, the load on dedicated streaming servers is significantly mitigated.

There are a number of fundamental performance metrics that characterize "good" peer-assisted streaming systems. First, if streaming content does not arrive in a timely fashion, it has to be skipped at playback, which degrades the *playback quality*. Second, an *initial buffering delay* must be experienced by a peer when it first joins or switches to a new streaming channel. Third, how do we encourage maximum bandwidth contribution from participating peers, which in turn minimizes *server bandwidth costs*—a sizable operational expense? Finally, how do we design a system that *scales* well to accommodate a large flash crowd and a high degree of peer dynamics?

Network coding [1, 2] has been originally proposed in information theory, and has since emerged as one of the promising information theoretic approaches to improve multicast session throughput. In peer-to-peer content distribution systems, Avalanche [3, 4] has demonstrated that network coding may improve the overall performance. The intuition was that, with network coding, all pieces of information are treated equally, without the need to identify and distribute the "rarest piece" first. Can network coding be useful in peer-assisted media streaming systems as well?

Compared to peer-to-peer content distribution, peer-assisted media streaming has unique timing requirements. The advantages of network coding are less obvious. In fact, it has been shown in [5] that the success story of applying network coding in content distribution cannot be simply replicated in multimedia streaming. To take full advantage of network coding, a complete redesign of peer-assisted streaming protocols is required. This motivated the design and implementation of R^2 [6], an entirely new streaming algorithm with network coding.

R^2: random push with random network coding

A traditional peer-assisted streaming protocol—one similar to CoolStreaming [7] and PPLive [8]—utilizes a pull-based streaming mechanism (later called "pull" for ease of reference). In pull, the streaming content to be served is divided into a series of data blocks, each representing a short duration of playback. Every peer periodically exchanges block availability information (often called *buffer maps* in the literature) with its neighbors. Based on such information, data blocks are pulled from appropriate neighbors, in order to meet their playback deadlines. Data blocks that are not received on time are skipped during playback, leading to a degraded playback quality.

In R^2, the streaming content is divided into *large* segments, with each segment further divided into *smaller* data blocks. Whenever a peer is able to serve a downstream peer *p*, it randomly chooses a segment that *p* has not completely received and then sends a coded block in that segment using random network coding [2]. Since all coded blocks are equally useful, a missing segment on a peer in R^2 can be served by *multiple* neighbors simultaneously without any explicit coordination, as illustrated in Fig. 1. In this way, participating peers in R^2 are able to perform *push* rather than pull operations, thereby fully utilizing available bandwidth resources.

Before pushing coded blocks, an upstream peer should obtain the precise knowledge of the missing segments on its downstream peers at any time. This requires participating peers in the system to exchange their buffer maps in a timely fashion. Since large segments are used in R^2, buffer maps that indicate segment availability information—as opposed to block availability information—can be exchanged, which may be an order of magnitude smaller. In addition, it takes much longer to playback or finish downloading a large segment, leading to a less frequent need to update buffer maps. As such, R^2 can afford "on-demand" exchanges of buffer maps without additional overhead (or even with less overhead!): buffer maps are sent to all neighboring peers immediately upon a local update.

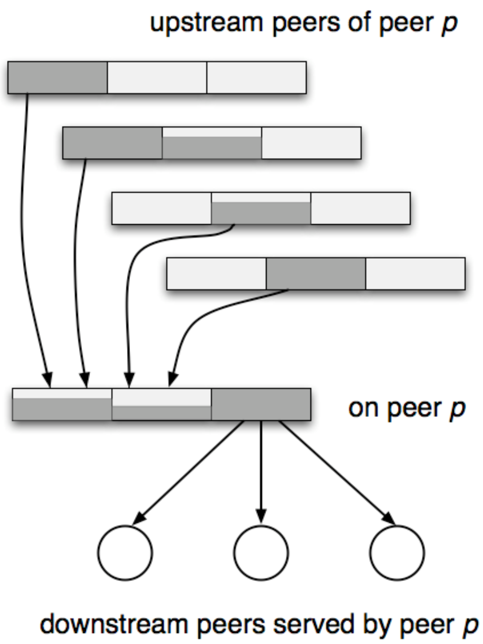

Figure 1: An illustration of R^2.

Why is network coding helpful in R^2?

We intuitively explain why the use of network coding in R^2 helps provide a good overall performance for streaming systems. A detailed theoretical analysis can be found in [9].

First, with network coding, R^2 is able to use large segments and small blocks. On one hand, the use of large segments naturally leads to "on-demand" exchanges of buffer maps. With up-to-date buffer maps in R^2, participating peers are able to serve more to one another. In contrast, buffer maps in pull are exchanged periodically with a longer time interval, in order to avoid excessive overhead. As shown in [10], the lack of timely exchanges of buffer maps may be a major factor that separates the performance of pull from optimality.

On the other hand, with random push operations performed on small coded blocks, the probability of saturating peer upload bandwidth capacities is much higher than that in pull. In particular, even slow connections may be fully utilized in R^2 due to much finer granularity in data blocks, which are generally impossible in pull [11]. Both of these factors contribute to a better utilization of peer bandwidth resources, leading to a higher playback quality and reduced server bandwidth costs.

Second, with network coding, robustness to peer departures in R^2 has been significantly improved. Since multiple upstream peers are serving each segment at the same time, the departure of a few of them does not constitute a challenge. In contrast, a missing block in pull can only be served by one upstream peer at a time. Whenever an upstream peer leaves the system, the downstream peer has to detect such a departure and request the missing block again. If this block is close to its playback deadline, the unlucky downstream peer is indeed under the risk of missing a deadline.

Finally, with network coding, R^2 scales well to accommodate a large flash crowd. Due to the use of large segments, it is easier to synchronize playback buffers so that participating peers are playing the same segment at approximately the same time. With playback buffers overlapping as much as possible, newly arrived peers during a flash crowd are able to serve one another immediately after they have received a few coded blocks. This allows fully utilization of upload bandwidth from newly arrived peers, which in turn reduces initial buffering delays and improves system scalability.

Conclusion

Inspired by the success of network coding in peer-to-peer content distribution applications, it is natural to explore the potential benefits of network coding in peer-assisted media streaming applications. Due to the strict timing requirement, however, the advantage of network coding is less obvious, and would certainly justify an in-depth study. In this article, we briefly illustrate the design principles behind R^2, a new set of streaming protocol design principles that combine random network coding with random push operations. With its intuitive advantages, we believe R^2 has shown the clear potential to help improve user experience and server costs in current-generation peer-assisted streaming systems.

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Engineering since August 2005. In 2000, he was the recipient of the IEEE Communications Society Leonard G. Abraham Award in the Field of Communications Systems. In 2009, he was a recipient of the Multimedia Communications Best Paper Award from the IEEE Communications Society, and a recipient of the University of Toronto McLean Award. His research interests include large-scale multimedia systems, peer-to-peer networks, applications of network coding, and wireless networks. He is a senior member of IEEE, and a member of ACM.

Network Coding for Wireless Video Communication

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Recent years have witnessed a phenomenal growth in wireless technologies. Yet, the vision of an ubiquitous communication strata that can support anywhere-anytime wireless video applications still seems to remain just a vision for years to come. A main challenge in building such a ubiquitous wireless network is to efficiently use the limited shared spectrum in such a way that guarantees throughput and delay for real-time video applications. While many emerging underlying technologies such as software-defined radio and MIMO can take advantage of space, time, and frequency diversity to increase wireless capacity, it is not clear whether these technologies alone are sufficient, economical, or always available to address a wide range of wireless video communication scenarios. In this letter, we take a look at the parallel development of Network Coding (NC), its potential to further improve wireless capacity, and some promising techniques for integrating NC techniques to wireless video transmissions. It should be noted that NC techniques are often orthogonal to the MIMO and software-defined radio technologies, and thus automatically inherits their benefits without any trade-off.

Network Coding

NC is the generalized routing approach that allows an intermediate router to encode an outgoing packet by mixing multiple incoming packets appropriately [1]. In this way, it is possible to significantly improve the throughput of a multicast session over that of the current store-and-forward routing scheme. Figs. 1 and 2 show two classical examples, illustrating the throughput improvement of using NC techniques in wired and wireless scenarios.

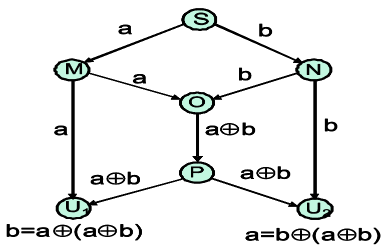

Figure 1: Improving multicast throughput in a butterfly network. Router O performs NC, i.e. mixing packets a and b to produce $a \oplus b$, transmits it to P. P broadcasts the mixed packet to U_1 and U_2. U_1 and U_2 recover packets b and a by performing bitwise-exclusive-or.

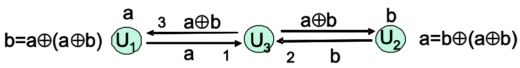

Figure 2: Wireless information exchange. Node U_3 broadcasts $a \oplus b$. Nodes U_1 and U_2 recover packets b and a by performing bitwise exclusive-or with the received packets, respectively.

In Fig. 1, the goal is to transmit both packets *a* and *b* to both receivers U_1 and U_2 as fast as possible. Assuming that the capacity of each link is 1 packet per unit time, then it is not hard to see that link OP is the bottleneck which results in an aggregate rate of no greater than 1.5 packets per unit time for U_1 and U_2. However, when router O is allowed to mix packets *a* and *b*, and the receivers decode packets *a* and *b* by performing bitwise exclusive-or as shown in the Fig. 1, then the total throughput is now 2 packets per unit time.

In Fig. 2, the goal is for U_1 and U_2 to exchange their packets *a* and *b* in fewest number of transmissions [2,3]. In current wireless networks, four transmissions in four time slots are necessary to complete the exchange. With NC, only 3 transmissions are sufficient. Specifically, in the first 2 time slots, U_1 and U_2 send their packets to U_3. In the third time slot, U_3 broadcasts $a \oplus b$ which can be heard by both U_1 and U_2. This mixed packet is then used to recover packet *a* at U_1 and *b* at U_2.

Network Coding for Wireless Video

Much literature on NC have been focused on throughput improvement in a rather application-agnostic way. Such throughput increase would indirectly improve quality for many video applications. However, it is doubtful that such a video-agnostic approach to NC will be as good as ones that jointly optimize NC techniques and video specificity. A case in point is the successful approach of integrating scalable video coding techniques with network protocols that allows the source to adapt its sending rate to the current available bandwidth [4,5,6]. In the same spirit, we advocate the integration of NC techniques and videos by describing the following two scenarios.

1) NC-based Scheduler at a Wireless BS

For simplicity, we consider the scenarios in which a wireless BS that broadcasts 2 packets *a* and *b* to both receivers R_1 and R_2. Let us first examine an existing broadcast scheme in which, packets are sent in time slots. Assuming the availability of ACK, using this scheme, a packet loss at any receiver will require the BS to retransmit that packet. If there are two distinct lost packets at two different receivers, the BS will need at least two retransmissions, resulting in a total of 4 transmissions to transmit both packets *a* and *b* to receivers $R_1$and R_2 as seen in Figure 3.

Time slots	1	2	3	4
Packet sent	a	a	b	b
Events at R1,R2	Lost at R1, received at R2	Received at R1	Lost at R2, received at R1	Received at R2

Figure 3: Traditional wireless broadcast requiring a total of 4 transmissions.

Time slot	1	2	3
Packet sent	a	b	a⊕b
Events at R1,R2	Lost at R1, received at R2	lost at R2, received at R1	Received at both R1, R2

Figure 4: Wireless broadcast with network coding requiring only 3 transmissions.

We now consider a network coding technique that requires only one retransmission to recover two lost packets at both receivers. Using this scheme, the BS does not retransmit the lost packet *a* at R_1 immediately. Instead, the BS continues to broadcast the next packet until there is a lost packet *b* at receiver R_2. At this time, the BS broadcasts the new packet $a \oplus b$ to both receivers. If R_1 has packet *b* but not *a*, and R_2 has packet *a* but not *b*, then both receivers will be able to reconstruct their missing packets by XORing the packet they have with the packet $a \oplus b$ as shown in Figure 4. It is straightforward to extend the network coding technique to unicast setting.

The key to improving bandwidth efficiency in the above scenario is the efficient generation of XOR packets to enable all the receivers to recover their lost packets quickly. However, one important constraint with this network coding scheme is that for higher bandwidth efficiency, longer delay of some packets may be necessary due to longer waiting time of packets in the queue, to allow packet losses to occur at other receivers, leading to more opportunities for the BS to generate more XOR packets. This might not be acceptable for video streaming applications where every packet has a playback deadline. In general, the BS must consider the trade-off between the delay and bandwidth efficiency based on the application requirements. Given K receivers with different packet loss rates, an optimal scheduling algorithm would determine which packets, including pure and mixed packets to send at which times, in order to maximize the video quality.

One approach to solving this optimal scheduling problem is to use the Markov Decision Process (MDP) framework [7,8,9]. The main idea of the MDP approach is that based on the observations at every time slots, the BS chooses the best action to perform, i.e. what packets to send in order to maximize the *expected* video quality under the time constraint. This implies that the BS must consider the contributions of different packets and their appropriate mixing to the overall video quality. It has been shown that such joint optimization between video packets and NC via the MDP framework improves the overall video quality for streaming applications. Further detail on this approach can be found in [7,8,9].

2) NC-based Flow Mixing at a Wireless BS

In this scenario, we consider a WiMax BS that broadcasts M different TV programs to M receivers. Given the fixed capacity and the need to maintain the minimum throughput and packet jitter for each TV program, the BS needs to deliver N packets to each of the M receivers in a total of T time slots. Assuming that $T > M\times N$, then the BS can use $T-M\times N$ time slots for sending Forward Error Correcting (FEC) packets to increase the transmission reliability. An existing time-sharing approach is to allocate T/M time slots for each receiver. For example, if T = 8, M = 2, and N = 3, then the BS can use RS(4,3) for each flow separately. Here, one additional packet is sent every 3 packets of each receiver. In many situations, this approach is not as efficient as the one that allows the BS to mix packets from different flows, i.e., imagine mixing your video traffic with your friend's web traffic. Using this mixing approach, instead of separately encoding packets from each flow, we concatenate packets from different flows together, treat them as though they are from one long flow, and code them together. The requirement is that each receiver has to listen and cache all packets sent by the BS. In the example above, we can use RS(8,6) which has the same redundancy as RS(4,3). We can theoretically show that, under typical channel conditions, the mixing approach is better than the time-sharing approach [10,11]. The intuition is that when using the time-sharing approach, if the channel of a receiver is bad at the moment the packet intended for that receiver is sent, the transmission of this packet is wasted, since the other receiver has no interest in this packet. On the other hand, when mixing is used, any packet will carry information that can be used by both receiver. Thus, it is still useful for the other receivers with good reception quality. Effectively, as long as both channels are not bad simultaneously, useful information is received by either receiver.

It is not always advantageous to blindly mix packets from either inter or intra flows. Preliminary investigations showed that for video applications, mixing should be done based on the QoS requirements of different flows and packet types [10]. Specifically, most important packets and inelastic flows should not be mixed with least important packets and elastic flow, respectively. In general, mixing should be done in the consideration of dependency structure of video packets. Further details on how to mix packets of different types can be found in [11,12] and an brief introduction to current NC techniques for multimedia networking applications in [13].

Conclusions

This letter argues for a video-aware approach to applying NC techniques for wireless video applications. Two examples were given to demonstrate the benefits of such an integrated approach. Much more research is needed to provide a deep understanding of the benefits and trade-offs in employing NC techniques for multimedia networking applications.

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TECHNOLOGY ADVANCES

Distributed Multimedia Networking

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Multimedia computing and communications have become ubiquitous in our daily life, thanks to the rapid growth of Internet and advance of compression and communication technologies. In many cases, a multimedia networking system, such as the home multimedia wireless network and the large-scale multimedia stream mining system, is physically distributed, and multimedia information over such a networking system needs to be processed and transmitted in a decentralized manner. In this Technology Advances Column, we bring together four research articles from top-notch researchers to offer their valuable insights into design, implementation, and optimization of distributed multimedia networking systems.

The first article "Multiuser Rate Allocation Games for Multimedia Communications" by Yan Chen, Beibei Wang, and K. J. Ray Liu proposes a framework of multiuser rate allocation game for fair and efficient bandwidth allocation among many multimedia users. The authors first introduce a novel utility function to characterize the gain and cost of transmitting a video sequence by a multimedia user. The authors then formulate a multiuser rate allocation game using the proposed utility function and show that a unique Nash Equilibrium (NE) exists under a proper price. In addition to this, their analysis reveals that this unique UE attains fairness across multiple users. Based on the theoretic analysis, authors present a decentralized rate allocation scheme which is guaranteed to converge to the unique NE. The theoretic claims are also substantiated by extensive simulation studies.

The rapid increase of the scale of online multimedia streaming systems poses significant challenges on the real-time multimedia steam mining, as the multimedia data sources are naturally distributed and the high-volume multimedia processing often requires a substantial amount of computational resources. The second article "Resource-Adaptive Multimedia Analysis on Distributed Stream Mining Systems" by D. S. Turaga, R. Yan, O. Verscheure, B. Foo, F. Fu, H. Park, and M. van der Schaar summarizes some of their recent progresses on designing and building distributed stream mining systems for resource-adaptive multimedia analysis. In particular, the applications of semantic concept detection are modeled as topologies of networked binary concept detectors, and the performance utility function is defined as the tradeoff between the probabilities of successful detection and false alarm. A network optimization problem is formulated and both centralized and distributed algorithms are presented. Simulation studies show significant performance improvement over the conventional approaches.

The intelligent tutoring and e-learning system constitutes one of the most important applications in distributed multimedia networking. The third paper "Web-enabled Intelligent Tutor for Multimedia Distance Education System" by Yan Liu, Yang Liu, and Shenghua Zhong discusses a multimedia intelligent tutoring system. The goal of this tutoring system is to simulate the functions of human tutor, such as reproducing the course lecture, explaining the difficulties, summarizing the emphases, via utilizing the advanced semantic-based multimedia data mining techniques. It is shown through empirical studies that the online tutoring system could considerably enhance the learning outputs.

Research on home multimedia networking over unlicensed TV band has drawn significant research interest recently due to its distinct advantages such as free spectrum usage, good coverage performance, and low power level it can operate at. The fourth article "Home Multimedia Networking using TV White Space" by Jianfeng Wang, Vasanth Gaddam, Monisha Ghosh, Kiran Challapali discusses their recent work on using wireless unlicensed TV band for home networking. The authors first give three major challenges for this technology, i.e., protection of incumbent users, QoS provision, and interference avoidance. After that, PHY/MAC standards are presented. Authors finally conclude this paper by presenting a complete cognitive radio prototype on multimedia wireless home networking.

Collectively, four invited articles in this column illustrate that distributed multimedia networking has attracted significant research interest. With many new and emerging applications, we believe it will continue to be a vibrant and active research area. Finally, we would like to thank all authors for their valuable contributions. We strongly hope this column can further stimulate research on distributed multimedia networking.

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Dr. Yang was awarded the Eliahu Jury Award for his Ph.D. thesis work from Columbia University. He is also the recipient of the Chinese Government Award for Outstanding Students Abroad**.** He serves and has served as TPC members of various conferences, such as GLOBECOM and ICC.

Multiuser Rate Allocation Games for Multimedia Communications

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Nowadays, due to the explosive growth of the Internet and the advance of compression technologies, delay-sensitive multimedia networking applications such as multimedia streaming and multi-camera surveillance become more and more popular. Therefore, a fundamental problem in these applications, how to fairly and efficiently allocate the rate among many users who share the same network bandwidth, becomes more and more important and draws great attention recently.

The simplest multi-user rate allocation is the constant bit-rate allocation (CBR), where the available network bandwidth is equally assigned to each user. A major problem of CBR is that it does not consider the variable bit-rate characteristics of the video sequences. One way to overcome this disadvantage is to optimize a global objective function that involves the characteristics of all video sequences using conventional optimization methods such as Lagrangian or dynamic programming [1]. For example, a commonly adopted method is for the rate controller to minimize the weighted sum of the distortions or try to maximize the weighted sum of the PSNRs, i.e., the optimization problem becomes

$$\min_{R_i} \sum_{i=1}^{N} w_i D_i(R_i),\ s.t.\ \sum_{i=1}^{N} R_i \le R \qquad (1)$$

or

$$\max_{R_i} \sum_{i=1}^{N} w_i PSNR_i(R_i),\ s.t.\ \sum_{i=1}^{N} R_i \le R \quad (2)$$

where R is the available network bandwidth, w_i is the weight, D_i is the distortion, and $PSNR_i$ is the PSNR of the i^{th} user.

Notice that the solution to the above optimization-based methods is highly related to the selection of the weights. However, in the literature, the weights are usually heuristically determined, e.g., the weights are set to be uniform [2]. Moreover, such a formulation can only address the efficiency issue, e.g., how to maximize the weighted sum of the PSNRs or minimize the weighted sum of the distortions. As such, the fairness issue, which is an important problem for multi-user rate allocation, has been generally ignored in the image/video/multimedia community.

To efficiently and fairly allocate the available network bandwidth to different multimedia users, we propose a multi-user rate allocation game framework in [3]. The utility/payoff function of each user/player is defined according to the characteristics of the transmitted video sequences and the allocated bit-rate. Specifically, motivated by the intuition that the quality difference in the low PSNR region is easier to be distinguished than that in the high PSNR region, we define the gain as a logarithm function of the PSNR. We also introduce a cost term in the utility function, which is linear in the allocated rate, to guide users' behaviors. In this way, the users will be more rational in choosing bit-rate since transmitting data with a higher bit-rate in this case does not necessarily result in a higher payoff, especially when the transmitted video sequence is a fast motion and complex scene sequence. With such a utility function, the objective of the users is to maximize their own utility subject to some constraints, i.e., the game can be formulated as

$$\begin{aligned} &\max_{R_i}\ U_i(R_i) = \ln(\gamma_i + \beta_i R_i) - aR_i \\ &s.t.\ R_i^{min} \le R_i \le R_i^{max},\ \forall i \\ &\quad \sum_{i=1}^{N} R_i \le R, \end{aligned} \qquad (3)$$

where γ_i and β_i are two content-dependent parameters, a is the unit price of the rate, R_i^{min} and R_i^{max} are the minimal and maximal rate requirement.

Then, we discuss the Nash equilibrium (NE) of this rate allocation game. We show that with a proper price, a unique NE, which is proportionally fair in terms of both utility and PSNR, can be obtained, based on which the rate controller can efficiently and fairly allocate the available rate. Moreover, we propose a decentralized cheat-proof rate allocation scheme for the users to converge to the unique NE using alternative ascending clock auction [4]. With the proposed scheme, all players will report their true demands since any deviation will lead to a loss in terms of utility. We also show that the traditional optimization-based method in (2) is a special case of the game-theoretic framework if the utility function is defined as an exponential function of PSNR. This fact indicates that the game-theoretic approach offers a more general and unified solution, especially in a multi-user setting.

To demonstrate the efficiency and effectiveness of the proposed game-theoretic multi-user multimedia rate allocation method, we compare its performance with three approaches: the Absolute Fairness in Rate (**AFR**), which equally divides the available bandwidth to all the users, the Absolute Fairness in Distortion (**AFD**), which minimizes the maximal distortion of all the users, i.e., min-max fairness, and the approach Maximizing the Sum of the PSNRs (**MSPSNR**), i.e. the traditional optimization-based approach shown in (2) with uniform weights. From the results shown in Figure 1 and 2, we can see that with the proportional fairness criterion, the proposed game-theoretic method can efficiently and fairly allocate bit-rate to different users by allocating more bit-rate to the sequence with slower motion and/or simpler scene while keeping an eye on the fast motion and/or complex scene sequence. We also find that, with the proposed distributed cheat-proof rate allocation scheme, reporting the true optimal demand at every clock is the mutual best response for every user.

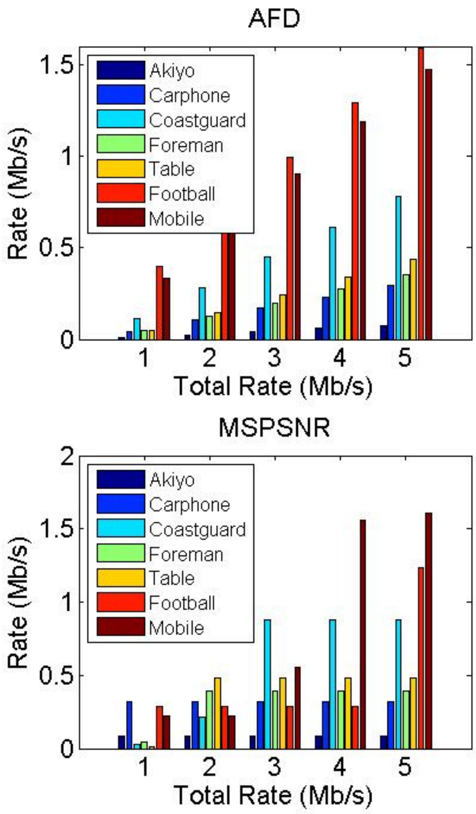

Figure 1. Allocated rates for Akiyo, Carphone, Coastguard, Foreman, Table, Football, and Mobile using different methods.

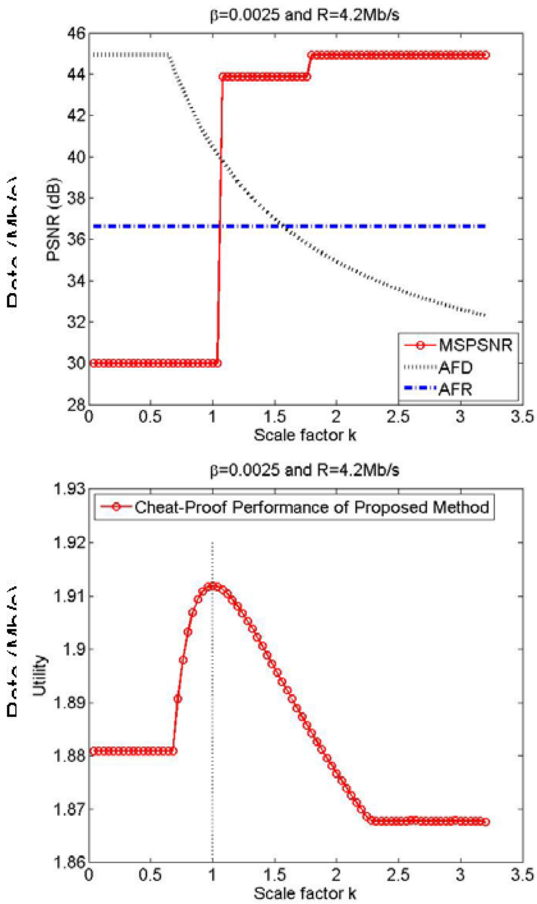

Figure 2. Cheat-proof performance of different methods.

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Resource-Adaptive Multimedia Analysis on Distributed Stream Mining Systems

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Abstract
Large-scale multimedia semantic concept detection requires real-time identification of a set of concepts in streaming video or large image datasets. The potentially high data volumes of multimedia content, and high complexity associated with individual concept detectors have hindered the practical deployment of many current solutions. In this paper, we present a summary of our work in building systems and applications for resource adaptive semantic concept detection in multimedia using large-scale distributed stream mining systems. We construct such concept detection applications as a hierarchical topology of individual concept detectors, and deploy them on distributed processing infrastructure. We then focus on dynamically configuring individual concept detectors to meet system imposed resource constraints while minimizing a penalty defined in terms of the misclassification cost. We develop multiple centralized and distributed algorithms for this configuration, and describe the implemented application and system. We verify through simulations that significant improvement in classification accuracy can be achieved through our approach.

1. Introduction

Recently, there has been the emergence of several applications that require processing and classification of continuous, high volume multimedia streams. These include online photo and video streaming services, search engines, spam filters, security services, etc. Each application may be viewed as a processing pipeline that analyzes streaming data from a set of raw data sources to extract valuable information in real time. In order to handle the naturally distributed set of data sources and jobs, as well as high computational burdens for the analytics, distributed stream mining systems have been recently developed [1]. An example architecture of a stream mining system is shown in Figure 1. These systems leverage computational resources from a set of distributed processing nodes and provide the framework to deploy and run different stream mining applications on various resource topologies. In such systems, complex jobs are decomposed into a network of operators performing feature extraction, classification, aggregation, and correlation.

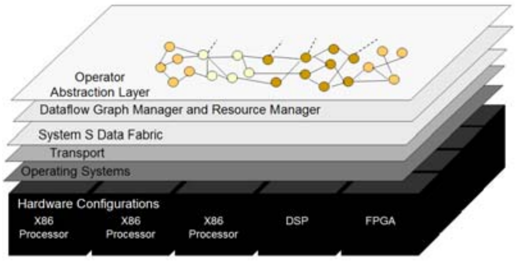

Figure 1. Stream Mining Architecture

Applications are then composed as topologies of distributed operators are deployed on middleware that maps them onto different transport, OS and hardware configurations. Such decomposition and distributed deployment has significant merits in terms of scalability, reliability, and performance objectives of large-scale, real-time stream mining applications.

In this paper we focus on applications for real-time semantic concept detection in multimedia streams, deployed on a stream mining system. We construct these applications as topologies of networked binary concept detectors, where the detectors are organized into topologies based on the semantic relationships between concepts of interest. A key research challenge lies in the management of limited system resources (e.g. CPU, memory, I/O bandwidth etc.) while providing desired application performance. Prior research in this area relies mostly on load-shedding, where algorithms determine a discard policy given the observed data characteristics e.g. burst, and the desired Quality of Service (QoS) requirements [5] [2]. These approaches are limited by their assumption that the impact of load shedding on performance is known a-priori. They also impose significant overheads for computing any relevant metrics, and consider only locally available information, which may lead to sub-optimal end-to-end performance.

Instead of deciding on *what fraction* of the data to process, as in load-shedding based approaches, we determine *how* the available data should be processed given the underlying resource allocation. Hence, we allow individual classifiers in the topology to operate at different *performance levels* given their resource allocation. We define application utility, as a function of these performance levels, in terms of the end-to-end accuracy, i.e. desired tradeoff between probability of detection and probability of false alarm. Using this utility, our resource management problem may be formulated as a network optimization problem (NOP). However, unlike traditional NOPs, that determine a transmission rate per job, we configure the performance level, in terms of the operating point, of each classifier to meet system resource constraints while maximizing the end-to-end performance. We develop solutions for this problem using both a centralized optimization solution using Sequential Quadratic Programming (SQP), as well as distributed solution based on game theoretic principles. We then implement a multimedia analysis application, for semantic concept detection in sports images, along with our optimization algorithms on IBM's distributed stream mining system [1], and benchmark performance in terms of scale as well as misclassification penalty.

The paper is organized as follows. We introduce our application of interest in Section 2. In Section 3 we describe our models for classifiers, classifier tree topologies, and define utility and resource constraints for the optimization problem. In Section 4, we include a summary of our designed centralized as well as distributed solutions. We conclude in Section 5.

2. Semantic Concept Detection Application

We consider an application for hierarchical semantic concept detection [3] in streaming sports images. Incoming images of several different types are classified into six classes of interest: *Little League Baseball*, *Basketball*, *Cricket*, *Skating*, *Skiing* and *Tennis*, each of which identifies a specific type of sport. By introducing a set of additional intermediate concept detectors, we may then construct a hierarchical topology of classifiers such that not all classifiers need to process all the images. For instance, using a set of additional classifiers for *Team Sports*, *Winter Sports*, *Ice Sports*, *Racquet Sports* and *Baseball* concept identification, we can build a classifier tree as shown in Figure 2.

The *Team Sports* classifier filters data relevant to the *Little League*, *Cricket* and *Basketball* classifiers, the *Winter Sports* classifier filters data relevant to *Skating* and *Skiing* and the *Racquet Sports* classifier filters data relevant to *Tennis*. The mutually exclusive nature of concepts *Team Sports*, *Winter Sports* and *Racquet Sports* allows identifying these in series, i.e. passing only data that does not belong to a class to the next class. Using this hierarchy, the amount of data each classifier needs to process is significantly lower than the total data volume - depending on the a-priori probability of concept occurrence, leading to savings in resource consumption.

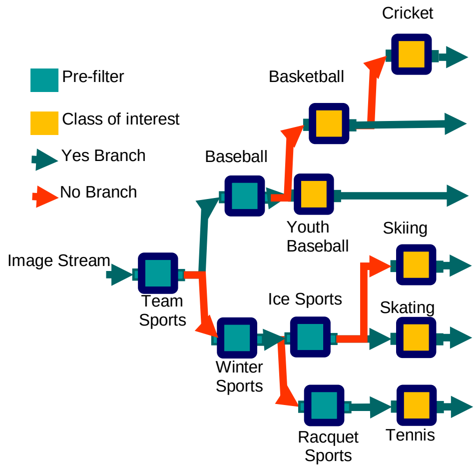

Figure 2. Hierarchical Classification Tree

Consider a binary classifier (concept detector) C_i in the topology shown in Figure 2. The total data rate or throughput entering C_i is labeled t_{i-1}. Since classifiers make mistakes, the throughput consists both of correctly labeled as well as incorrectly labeled data. The total goodput entering C_i is labeled g_{i-1}, where the goodput consists only of that portion of the throughput that is correctly classified. Classifier C_i labels data as belonging to the *yes* class or the *no* class, and forwards data appropriately. The apriori conditional probability of the data belonging to the *yes* class for C_i (given the processing the data has already undergone) is labeled ϕ_i. Correspondingly, the probability for the *no* class is $1-\phi_i$.

The operation of C_i is characterized by its Receiver Operating Characteristic (ROC) curve, a curve that represents a tradeoff between probability of detection p^D and probability of false alarm p^F. Note that a binary classifier may have two such curves, one corresponding to the *yes* class and the other to the *no* class. These are coupled together when the classifier uses, for instance, one score threshold, i.e. if the classification score for a data object falls above a threshold it is labeled *yes* otherwise it is labeled *no*. In the most general case though, the classifier may use two separate thresholds, i.e. when the score falls above the first threshold, the data is labeled *yes*, and if the data falls below the second threshold the data is labeled *no*. The use of two independent thresholds allows the classifier to both replicate data across its output branches (i.e. say both *yes* and *no*), as well as discard data (i.e. say neither *yes* nor *no*) as required. Hence the performance of the classifier is represented by two curves $\left(p_i^D, p_i^F\right)$ for the *yes* class, and $\left(\bar{p}_i^D, \bar{p}_i^F\right)$ for the *no* class.

Given these definitions, the output of C_i, i.e. t_i and g_i on the *yes* and $\bar{t}_i$ and $\bar{g}_i$ on the *no* may be determined recursively as:

$$\begin{bmatrix} t_i \\ g_i \end{bmatrix} = \begin{bmatrix} p_i^F & \phi_i\left(p_i^D - p_i^F\right) \\ 0 & \phi_i p_i^D \end{bmatrix} \begin{bmatrix} t_{i-1} \\ g_{i-1} \end{bmatrix} \quad (1)$$

$$\begin{bmatrix} \bar{t}_i \\ \bar{g}_i \end{bmatrix} = \begin{bmatrix} \bar{p}_i^D & \left(1-\phi_i\right)\left(\bar{p}_i^F - \bar{p}_i^D\right) \\ 0 & \phi_i \bar{p}_i^D \end{bmatrix} \begin{bmatrix} t_{i-1} \\ g_{i-1} \end{bmatrix} \quad (2)$$

Given this recursive relationship, we can then determine the throughputs and goodputs at each terminal (leaf) classifier to determine the end-to-end throughput and goodput. The misclassification penalty may be defined in terms of the two types of errors – a penalty c^M per unit rate of missed detection, and a penalty c^F per unit rate of false alarm. For the *yes* output of leaf classifier C_i, we can compute this as:

$$u_i = c_i^M \left(t_0 \pi_i - g_i\right) + c_i^F \left(t_i - g_i\right) \quad (3)$$

where π_i represents the apriori probability of data actually belonging to the yes class of C_i and t_0 represents the input data rate into the tree. We can similarly define the penalty for the *no* class. The end to end misclassification penalty may thus be defined as a sum of these penalties for all the leaf classifiers. Note that this misclassification penalty function is non-concave in nature.

In addition to the misclassification penalty, we also need to model the resource consumption of each classifier. Given an underlying model, e.g. SVM, Bayes, Decision Tree etc., the computational requirements of a classifier C_i may be modeled as being directly proportional to the rate of data entering it. Hence we use a linear model: $\rho_i = t_{i-1}\alpha_i$, where α_i represents the per-unit rate computations for the classifier. Finally, given a set of N classifiers, organized into a tree topology, placed on M resource-constrained nodes, the configuration problem involves determining the right set of operating points for each classifier such that the end to end misclassification penalty is minimized, while the M resource constraints are satisfied.

3. Summary of Solutions

In [6] we solve the topology configuration optimization problem using a centralized approach based on Sequential Quadratic Programming. In this case, the central optimizer has access to all the ROC curves, apriori probabilities as well as resource placements and constraints. The gradient descent based approach guarantees convergence to local minima for the utility, however, the non-concavity of the utility as well as the constraints do not allow analytical bounds on performance. The approach clearly outperforms load-shedding approaches - leading to savings of a factor 1.5-2 times the misclassification penalty. This factor increases with increasing cost for false alarms, and with tightening resource constraints. Alternately, in [4] we propose a distributed solution to this problem, using game-theoretic principles. In this solution, individual classifiers select their operating points to maximize a local utility function. The utility may be purely local to the current classifier, corresponding to a *myopic* strategy, or may include the impact of the classifier actions on successive classifiers in the tree, corresponding to a *foresighted* strategy. We analytically show that foresighted actions of any classifier improve end-to-end performance over myopic strategies, and derive an associated probability bound. We then evaluate our algorithms on our application for hierarchical sports scene classification. We compare centralized, myopic and foresighted solutions and show that foresighted strategies outperform myopic strategies, and also asymptotically approach the centralized optimal solution as the number of actions available to each classifier increase.

4. Conclusions

Large-scale stream mining requires the development of systems and algorithms that support dynamic tradeoffs between data characteristics, result accuracy, and available computational resource. As part of this paper, we summarize our approaches for centralized and distributed configuration of classifier topologies for hierarchical semantic concept detection in multimedia. There are several directions for future work in this area. We would like to extend these ideas for more complex processing topologies, including arbitrary acyclic graphs. We also need to examine the related problem of constructing such topologies, distributing them across machines, etc. given the individual classifiers, and the end-to-end resource constraints and requirements. Further, we would like to revisit the training of individual classifiers given that they are likely to be used in such semantically hierarchical processing topologies. We are also validating these on large-scale distributed infrastructures using different multimedia collections and streams. We intend to examine game theoretic approaches that combine decentralized information exchange, and dynamic multi-agent learning to design scalable, dynamic, and failure-resilient optimization strategies. We would like to use these insights to provide design guidelines for the development and deployment of real multi-stage classification based applications.

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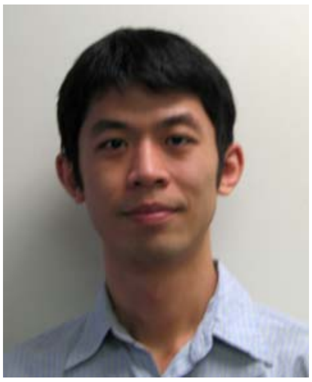

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Web-enabled Intelligent Tutor for Multimedia Distance Education System

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1. Abstract
An Intelligent tutoring system simulates the functions of a human tutor, such as reproducing the lecture, explaining the difficulties, summarizing the emphases, providing the exercises, and interacting with the students. This paper proposes a web-enabled intelligent tutor for our human-centered multimedia e-learning system, which includes an online virtual classroom and offline self-review web. To reproduce the lecture, we provide a multi-level browsing with slides, web, handwriting, audio and video based on automatic multimedia segmentation and reconstruction. For the review of the difficult parts, visual content based retrieval and course content based retrieval are proposed respectively. For the emphasis summarization, the importance of the topics is evaluated using novel multimedia mining techniques. Interactive exercise is provided based on semantic indexing. This system has proved helpful in enhancing the engagement and extent of understanding under the real teaching environment in our university.

2. Introduction
A tutor is an indispensable member of the course delivering team, whose primary purpose is to ensure that the learners get the most comprehension out of the instructional languages and facilities, by providing additional academic interpretation and backup to the instructor's material [1].

A learning process takes a different form in the digital era from the conventional education. E-learning is such a learning behavior that a learner receives education or training via the Internet. Compared with the traditional learning behavior, e-learning systems are superior in terms of convenience, independence, adaptation, and interaction [2].

Intelligent tutoring systems, which have been existed for decades, play an important role in the e-learning system. It integrates the technologies of education, artificial intelligence, data mining, networking and multimedia to enhance the learner's engagement and the depth of the learning [3]. The "intelligence" in these systems can be seen from the way these systems adapt themselves to the characteristics of the students, such as the initial speed of learning, specific areas in which the student excels or not, and the rate of learning as more knowledge is learned [4]. In recent years, web-enabled tutoring system has received significant attention with the rapid growth of the Internet technology and the wide application of distance education [5]. Web-enabled learning is beneficial for students who prefer learning by flexible interaction with a teacher online, as a supplement to the face-to-face instruction [4]. It also provides a means for students to access the learning content at a distance regardless of the location and time constraints.

The usage of multimedia techniques such as animation, video transmission, and multimedia retrieval improves the teaching outputs of the e-learning system obviously, especially for the distance education [6], such as interactive virtual classroom [7][8]. Graesser et. al. proposed a 3D animated conversational agent to simulate the conversation between a human tutor and the learner in [3]. Butz et. al. provides an interactive multimedia intelligent tutoring to assist electrical engineering undergraduate students taking their first circuit course [9].

In this paper, we propose a web-enabled multimedia intelligent tutoring system for distance education system by integrating our previous work of real-time virtual classroom [10] with our latest work of semantic-based multimedia data indexing, retrieval, and mining. Here, we summarize the particular functions that are expected to be supported by the proposed tutoring system as below:

- Reproduce the lecture
- Explain the difficulties
- Summarize the emphases
- Provide the exercise and evaluation
- Interaction with the students and sometimes with the lectures

3. Multimedia Intelligent Tutoring System
For the convenience of the further processing, we incorporate a real-time recording module to store the useful information with timestamps in

virtual classroom [10]. We propose an intelligent tutoring system as show in Figure 1 based on the real-time authored multimedia data: an operation log file, text of the lecture notes, handwriting images, and audio/video file.

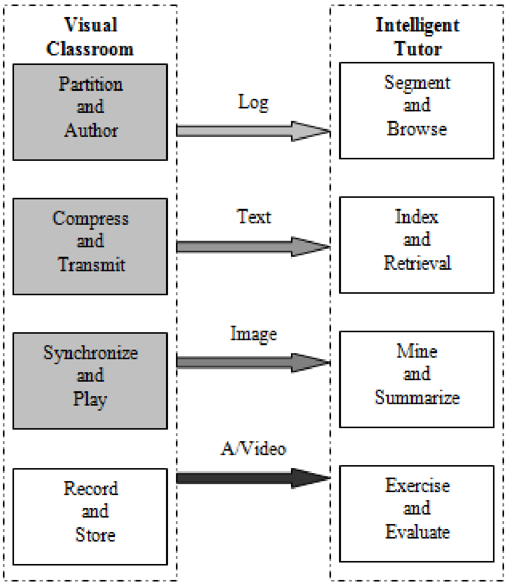

Figure 1. Multimedia Distance Education System

As the first step for video analysis, video segmentation is very important, because it has a great influence on the accuracy of the further processing. In our intelligent tutoring system, we use the timestamps recorded in the operation log file to segment the video/audio file. Based on the segmentation results of the multimedia data, the web-based intelligent tutoring system has three levels of browsing interface: course level, lecture level, and slide level. Course level interface lists all available courses and provides automatic summarization result, such as the most popular topics and the most important contents. The lecture level interface lists the lectures given in this course and provides the information of the lecture topic and date. The slide level browsing interface for the web-based intelligent tutoring system extracts the title part of the slides as the index for browsing. If the users are interested in that topic, they can access the content by clicking on that topic. The system also provides the lecture notes, instructional video, available figures and drawings on the tablet, respectively.

For visual content and course content analysis, we first generate the semantic indexing for the video clips. For visual content, we use the operation log file to classify the video clips to four categories: presentation with slides, viewing the web, drawing the figure, and others. For course content indexing, we process different kinds of video clips using different methods. To generate course content indexing for the video clips with visual content index of "presentation with slides", we will create a reference list of the keywords (generally the index of the textbook). Then, the system will list all words displayed in the slides as a table with the entry of: word, filename, and slide number. The system filters out the irrelevant words, such as "and", "or", based on the matching of the reference list. Finally, the system will generate the course content indexing for each video clip based on the appearance of the keyword in the slide shown in that clip. For the course content index of the video clips with a visual content index of "drawing the figure", we will use the course content index of the video clips that are just presented before them, generally a course content index of the slides presentation. For the course content index of the webpage, we can generate the index based on the head of the web page.

We will summarize the course content with "Most important topics" and "Most popular topics" based on multimedia data mining. Figure 22 shows the outputs of the summarization The output for the most important topics shows the emphasis parts from the instructor's perspective and the output for the most popular topics shows the interesting points from the student's perspective. The system is designed to detect the important topics of the course based on the text mining and instructional video mining. For each instructional video clip, we generate course content indexing using keywords, as discussed previously. Then, we calculate the importance of the key words based on the following equation:

$$KI = D * II*KP' \quad (1)$$

where KI is the keyword importance in this slide, D is the duration of the video clip, II is the importance of the items, high-level, high importance. KP is the vector that indicates the appearance of the keyword in each item. If the keyword does not appear in that item, the value of the corresponding component is equal to 1. If there are n keywords appearing in one item, the value of the component is equal to 1/n. The system scans all slides for all lectures to calculate the score of each keyword and chooses the highest ones as the important topics.

Different with important topics detection, the detection of popular topics intends to find out which part the students are most interesting based on their operation on the web. Instead of calculating score of importance, we will consider the effective click times. If the user not only views the slide but also view the corresponding instructional video clip and/or written board image longer than a pre-defined duration, such as 30 seconds, it implies that the user may be interested in that topic. Repeatedly viewing will also be account. Mining of popular topics based on user's preference provides asynchronous interaction among the users. Instructor can found out that which topics students are interested in or feel difficult to understanding. Students also can achieve the information, such as which slides other classmates focus on by viewing this part.

Interactive exercise provides the way to the students to learn, practice and evaluate by themselves. The system stores the user's operations of each question, such as click count, duration, how many people have done it correctly and how many people have done it wrongly. When the student accesses the exercise, he can view the statistical information of this question. The system matches the string in the question with the semantic indexing generated previously, and provides corresponding course material, such as slide, instructional video to help the understanding.

4. Empirical Validation and Conclusions

This paper proposes a web-enabled intelligent tutoring system to simulate the functions that a human tutor can provide: reproduce the lecture by multi-level multimedia browsing, explain the difficult by providing related learning material based on course content retrieval, summarize the emphases by multimedia mining, self-exercise and self-review are provided based on the semantic indexing. The efficient interaction between the tutoring system and the students/instructors has been integrated in each part of the system.

We test our intelligent tutoring system using one course for part-time student and one course for full time student at The Hong Kong Polytechnic University for one semester. There are totally 83 students attending the evaluation. When asking about how intelligent tutoring system affect the effectiveness of learning, 97.6% of students agree that the intelligent tutor is helpful to enhance the engagement and extent of understanding and 81.9% of students agree that it can greatly enhanced the learning outputs. The feedback also shows that the intelligent tutoring system is especially helpful for the part-time students.

We want to investigate our work further from two aspects. First, we will explore the intelligent exercise system by integrating multimedia technology to provide more effective self-study and self-evaluation tools. Second, we intend to use concept ontology to provide more effective understanding and representation of the instructional video.

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Home Multimedia Networking using TV White Space

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1. Background

On February 17, 2009, the Federal Communications Commission (FCC) adopted the final rules for "Unlicensed Operation in the TV Broadcast Bands" [1]. This was the culmination of many years of deliberations on the subject, starting with the first NPRM in May 2004 [2] and followed by laboratory and field testing of devices through 2007 and 2008 [3]. Meanwhile, Ofcom, the regulatory body in the UK, has also made significant progress in developing regulations for the TV white spaces with a first consultation released on February 16, 2009. The ECC has just begun working on cognitive radio in the TV bands [4] within its newly created group SE 43 which is tasked with defining the technical and operational requirements for devices operating in the TV White Space (TVWS) with a first report on the subject due in May 2010.

Applications benefitting from using TVWS include whole-home high definition multimedia streaming, community mesh networking, rural internet access, smart grid, and public safety communication networks. The primary benefit of TVWS comes from the better propagation characteristics and therefore increased range and robustness, in comparison to higher frequencies. The ability to operate at lower power-levels for a given range would result in better energy efficiencies as well. Additional spectrum in the TVWS helps deal with overcrowding of ISM bands. In addition, ready availability of hardware components such as radio frequency tuners, makes these frequency bands especially appealing.

2. Challenges

The challenges for enabling whole-home High Definition Television (HDTV) video streaming using TVWS can be three-fold: (a) to access TVWS, incumbent services using these bands must be robustly detected and protected; (b) to efficiently fit an HDTV stream within a single television channel, QoS provisioning with respect to limited bandwidth must be supported; and (c) to ensure co-located or neighboring networks co-exist, self-coexistence and interference mitigation in personal/portable environments must be designed-in. In terms of incumbent protection, the first issue is the ability to robustly detect incumbents without adding too much Medium Access Control (MAC) and Physical (PHY) layer overhead (assuming single radio used for both communication and sensing). For example, it is not trivial to detect incumbent signal at -114dBm or even lower using spectrum sensing [5]. Note sensing is either mandatory by regulatory rules or desired by personal/portable use cases. The second issue is to enable seamless video streaming while doing dynamic channel switch to protect incumbents.

Protocol efficiency and QoS provisioning is another design challenge. Assuming the achievable spectral efficiency (with single antenna) is 3.96 bit/s/Hz on a 6 MHz channel, the protocol overhead including PHY and MAC layer has to be less than 20% in order to satisfy the rate for full HDTV. In addition to effective throughput, the delay jitter and packet loss rate have to be low for real-time video streaming.

Moreover, user mobility and lack of central coordination between neighboring networks in personal/portable environments impose challenges for self-coexistence and interference avoidance.

3. Standard

Ecma 392 [6][7] is a recently completed international PHY/MAC standard which mainly targets personal and portable wireless devices. It aims to serve a broad range of applications, including multimedia distribution and internet access. The standard is carefully designed to support whole-home HDTV streaming.

Ecma 392 supports flexible network formation with three types of devices: master devices, slave devices, and peer devices. A network can be formed as master-slave, or peer-to-peer, or as a mesh-network. The interoperability of the three device types is built-in due to the fact that all devices follow the same beaconing and channel access protocols.

The PHY design is based on a 128 point FFT Orthogonal Frequency Division Multiplexing (OFDM) structure. Some key differentiators have been included in order to improve the performance compared to IEEE 802.11a. The enhancements include an FEC scheme using the concatenation of a Reed-Solomon (RS) outer code and a convolutional inner code, two types of preamble structure, an improved retransmission scheme and support for multiple antennae.

The MAC is based on a hybrid medium access architecture that allows both reservation based channel access and contention based channel access. Medium access slots may be reserved periodically for QoS-demanding audio/video stream. All unreserved medium access slots may be used for Prioritized Contention Access (PCA), according to four Access Categories (ACs): background (BC), best effort (BE), video (VO) and voice (VI). The MAC also specifies a number of frame processing mechanisms including frame aggregation, burst transmission and block-acknowledgement (B-ACK) to support very high efficiency data transmission. These mechanisms, together with channel reservation, can provide a data rate (measured at MAC SAP) of more than 20 Mbps, which is sufficient for one HDTV stream. In addition, several mechanisms are proposed to address self-coexistence and to mitigate interference between neighboring networks, including beacon exchange, superframe merge, and link quality measurement and adaptation.

Considering regulations for the protection of incumbents may vary from one region to another, Ecma 392 takes a toolbox approach and specifies a number of incumbent protection mechanisms including Dynamic Frequency Selection (DFS), Transmit Power Control (TPC), and spectrum sensing, that may be adapted based on the regulatory requirements of a particular region. While sensing algorithms are out of the scope of the standard, coordinated sensing (including quiet period schedule) is specified to reduce the impact of secondary interference on the reliability of sensing. A sender is able to adjust transmission power as well as data rate based on the feedback from its receiver such as the signal-to-noise-ratio, received signal strength, frame error ratio or other parameters. Such TPC mechanisms are designed to avoid/minimize interference to incumbents as well as neighboring networks. In addition, to help devices recover quickly from interruption caused by the emergence of incumbents on a channel, solutions such as the use of backup channel and coordinated reservation re-establishment have been developed.

4. Prototype

We developed a complete cognitive radio prototype [9] to demonstrate whole-home HDTV streaming for personal/portable devices operating in the TVWS. The prototype is built using a combination of custom algorithms implemented on a Field Programmable Gate Array (FPGA) and commercial off-the-shelf components. Figure 1 shows the top-level block diagram of the CR prototype.

A standard WiFi card is used to realize the MAC and PHY functionality. The PHY is operated in 5 MHz mode. A frequency conversion module is used to down/up convert the WiFi signals to the desired channel in the UHF band. The MAC is extended to include some basic features built in Ecma 392 for cognitive operation and QoS support. The extended MAC is based on superframe structure. Each superframe consists of a beaconing period, a data transfer period and a quiet period. Each superframe lasts about 100ms.

The sensing module consists of a standard TV tuner and an FPGA evaluation board. The TV tuner is used to down-convert the signals to an IF, which are then sampled by the ADC on the FPGA board at 100 MHz. The digitized signals are processed in real-time in the FPGA to sense for digital and analog TV signals and wireless microphone signals. A version of this sensing prototype was submitted to the FCC as part of the TV band device testing exercise. The sensing device was tested in the laboratory and in the field environments. The sensing prototype was able to reliably detect the incumbent signals with levels down to -114 dBm. The results of the prototype tests were summarized in FCC's phase I and phase II test reports [3]. The more detailed description of the sensing prototype can be found in [8].

The cognitive MAC provides automatic device discovery and automatic set-up of home channel and backup channels, and provides mechanisms to determine channel occupancy based on the information available from itself and from the other nodes in the network. The MAC also schedules synchronized periodic and on-demand quiet periods to enable incumbent sensing in the home channel and/or backup channels. A unique feature of the proposed sensing algorithms is the ability to distribute sensing time into multiple superframes. Each sensing period needs only 5 ms. This feature enables the CR device: a) to support applications that have strict QoS requirements, and b) to quickly vacate a channel when an incumbent appears thus minimizing the interference to the incumbents.

The MAC also provides protocols to move to the backup channel when an incumbent is detected on the home channel without causing interference to the incumbent while maintaining QoS for the application that it is supporting.

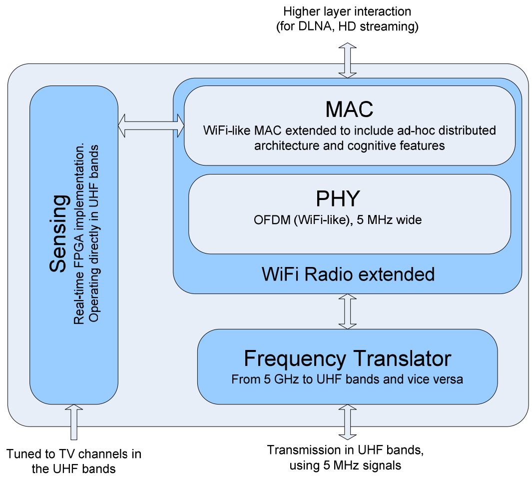

Figure 1 – Top-level block diagram of the CR prototype

5. Conclusions

The opening of TV white space for unlicensed operation will foster a whole range of innovative applications. Using TVWS for home multimedia networking is one of such appealing applications. This paper described the design challenges and introduced Ecma 392 – the first standard that is carefully designed to support whole-home HDTV streaming over the TVWS. We also presented a prototype including some basic features of Ecma 392 for cognitive operation and QoS support.

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Jianfeng Wang is a senior member research staff at Philips Research North America since 2006. He received B.E. and M.E. in electrical engineering in 1999 and 2002 respectively from Huazhong University of Science and Technology, and Ph.D. in electrical and computer engineering from University of Florida in 2006. His research has been focused on wireless networks and innovative applications. His work led to over thirty publications in journals and conferences. He actively participates in the standardization of cognitive radio networking. He currently serves as the technical editor of Ecma 392 – PHY/MAC standard for cognitive radio operating in TV White Space. He also leads the coexistence group within IEEE 802.22.

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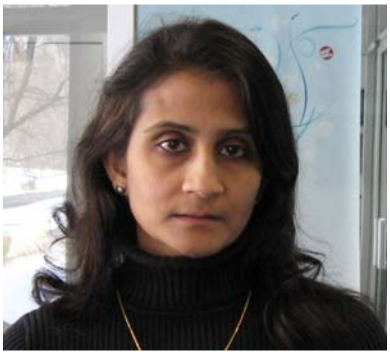

Monisha Ghosh is a Principal Member of Research Staff at Philips Research currently working on cognitive and cooperative radio networks. She received her B.Tech. in Electronics and Electrical Communication Engineering from the Indian Institute of Technology in 1986, and M.S. and Ph.d. in Electrical Engineering in 1988 and 1991 respectively from the University of Southern California. From 1991 to 1998 she was a Senior Member of Research Staff in the Video Communications Department at Philips Research, where she was involved with digital transmission of high data rate signals over terrestrial, satellite and cable channels. From 1998 to 1999 she was at Lucent Technologies, Bell Laboratories, working on wireless cellular systems. Her research interests include estimation and information theory, error-correction and digital signal processing for communication systems.

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Editor's Selected Paper Recommendation

Editor: Guan-Ming Su, Marvell Semiconductors, USA

Z. Li, P. Ishwar, and J. Konrad, "Video Condensation by Ribbon Carving," IEEE Transactions on Image Processing. vol. 18, no. 11, Nov. 2009, pp. 2572-2583.

Millions of video cameras deployed around the world continuously monitor transportation hubs, streets, and office buildings. This mountain of visual data contains a wealth of information, but mining it is a tremendous challenge. Although automated solutions that can scan, detect, and classify specific items of interest exist today, their ability to process diverse scenes with unscripted complex scenarios still cannot match that of a human. Hence, most surveillance systems continue to involve people to monitor and make critical decisions. However, continuous monitoring by human operators is impractical due to high cost and reliability issues (e.g., operator fatigue). Therefore, of interest are automatic algorithms which can increase the efficiency of human operators.

Video digest, a new class of such algorithms, computes a digest of video activities in the field of view of a surveillance camera. A video digest is a time-abbreviated video that preserves the most important activities (motion events) while removing relatively static segments. A video digest enables fast browsing through recorded surveillance material thus increasing efficiency in video forensics, but it can also be used in animal habitat monitoring and even in the broad consumer market for quick review of motion pictures and home videos.

A good video digest system should simultaneously achieve the four distinguishing characteristics, namely simplicity, real-time implementation capability, ability to maintain the spatial relationships of events, and limited ability to control the temporal ordering of events. To achieve the aforementioned goals, the authors proposed a video condensation algorithm built upon the idea of seam carving [1] used for multimedia dimension resizing. A video ribbon is a connected surface within the spatio-temporal volume of a segment of video frames which partitions the video into "past" and "future" volumes. Ribbons are rigid along either the horizontal or vertical dimension but are flexible along the other dimension. Deleting a ribbon reduces the duration of a video segment by exactly one frame but, unlike frame skipping, ribbons allow for the removal of pixels across frames. To decide which parts of the video to remove, the authors associate an activity-based cost with each ribbon. The activity costs are computed by state-of-the-art background subtraction algorithms. The condensation algorithm recursively carves out ribbons by minimizing the activity-based cost function using dynamic programming. Furthermore, it permits control of the tradeoff between the degree of anachronism of events and the condensation ratio, thus allowing adaptation to varying surveillance scenarios (fast-moving versus slow-moving objects).

The output of video condensation is a high-fidelity shortened video, as much as 50-fold shorter compared to the original (depending on content), that preserves the integrity and relative timing of all moving objects by shifting them in time in a controlled manner while preserving their spatial positions. Therefore, an earlier car in one lane and a later truck in an adjacent lane may both appear in the same frame in the condensed video but without spatially interfering with each other. The authors also proposed algorithm for handle video streaming scenarios.

Further research on joint video condensation and transmission, such as works done on video summarization [2], can be explored. The feasibility and algorithms for multi-view/scalable video condensation can also be studied, too [3].

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MMTC COMMUNICATIONS & EVENTS

Technical Committee on Multimedia Communication Meeting Minutes

IEEE GLOBECOM 2009, Honolulu, Hawaii
Date: December 2, 2009
Time: 12:00-13:30
Room: Ilima Boardroom, Hilton Hawaiian Village
Organized by Committee Chair Qian Zhang, minutes by Bin Wei

Agenda

0. Informal discussion and networking time
1. Welcome new members / introduction
2. Last meeting minutes approval (ICC 2009)
3. MMTC Best Paper Award 2009 winner announcement
4. Call for nomination for Best Paper Award for 2010
5. Report on conference activities
6. MMTC IGs reports - all IG chairs
7. Sub-committee report (report for the recent changes)
8. Report for News Letter activities (Haohong)
9. Publication Report (e.g., activities in terms of special issues on IEEE journals/magazines)
10. Suggestions & discussions – everyone
11. Adjourn

Meeting High-Lights

1. Welcome new members and introduction: the meeting started at 12 noon local time on 12/2. Around the table introductions were made.

2. Last meeting minutes approval (ICC 2009): Committee approves it.

3. MMTC Best Paper Award 2008 winner announcement:

   Award subcommittee - Chair: Dr. Dapeng Wu; Members: Prof. Oliver Wu (University of Florida), Prof. K.P.Subbalakshmi (Stevens Institute of Technology), Prof. Marco Roccetti (University of Bologna), Prof. Chang Wen Chen (University of Buffalo), Dr. Olivier Verscheure (IBM Research), Prof. Zhihai He (University of Missouri Columbia), Dr. Jin Li (Microsoft Research), and Prof. Jianchuan Liu (Simon Fraser University).

   Two papers are selected for the best paper award:
   - Mea Wang and Baochun Li. “R2: Random Push with Random Network Coding in Live Peer-to-Peer Streaming,” *IEEE Journal on Selected Areas in Communication*s, Special Issue on Advances in Peer-to-Peer Streaming Systems, Vol. 25, No. 9, pp. 1655-1666, December 2007.
   - B. Li, S.-S. Xie, G. Y. Keung, J.-C. Liu, I. Stoica, H. Zhang and X.-Y. Zhang, “An Empirical Study of the Coolstreaming+ System,” *IEEE Journal on Selected Areas in Communications*, Special Issue on Advances in Peer-to-Peer Streaming System, 25(9):1627-1639, December 2007.

Dr. Mea Wang is present to receive the award at the announcement.

4. Report for the conference activities
   a. CCNC 2010 (Zhu Li's slides)
   b. Globecom 2010 (Bin Wei, Haohong Wang)
   c. CCNC 2010 (Gary Chan's slides)
   d. ICME (Wenjun, commented by Jin Li and Mark Karol)
   e. ICC 2010 (Zhu Li's slides)

5. MMTC IGs reports – all IG chairs
   Mobile and Wireless Multimedia (Dapeng Wu), Cross-layer Multimedia Communication (Haohong Wang). Contact existing IG or Qian Zhang if you want to set up a new group.

6. Subcommittee reports
   MMC TC established the following subcommittees:
   Membership Development Subcommittee Chair: Chang Wen Chen;
   Conferences Subcommittee Chair: Pascal Frossard;
   Publications Subcommittee Chair: Gary Chan;
   Awards Subcommittee Chair: Dapeng Wu;
   Standards Subcommittee Chair: Stanley Moyer;
   TC Promotion & Improvement Sub-Committee: Chair: Haohong Wang.

7. E-Letter:
   Editor-in-chief: Haohong Wang. Associated Editor-in-Chief: Chonggang Wang, NEC Laboratories.

   Thanks to Haohong's great leadership. He will finish his last issue at the end of the year and hand the E-Letter to a new leadership team. The new team members will be finalized by the end of the year. E-Letter will become bi-monthly issues in 2010.

8. Suggestions & discussions – everyone
   Need our member's strong support and contribution to our major TC conferences: ICC/GlobeCom, CCNC and ICME. The goal of the E-Letter is to make it a Journal of our TC.

9. Adjoin
   We would also like to thank the committee chair, Dr. Qian Zhang, for organizing this meeting and providing the information for our MMTC members. Dr. Zhang was unable to come to the meeting and Dr. Bin Wei made the presentation.

10. Attendee list (according to sign-in sheet)

Name	Affiliations	Email
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Call for Papers of Selected Conferences

IEEE Globecom 2010
Communications Software, Services and Multimedia Applications Symposium (CSSMA)
Dec 6 - 10, 2010 Miami, FL USA
http://www.ieee-globecom.org/2010

The Communications Software, Services and Multimedia Applications Symposium covers challenges and advances for service delivery and management in fixed and mobile communication networks. These topics are particularly relevant for researchers, developers and industries in the areas of networking and services covered by many Technical Committees. The symposium will follow GC2010 instructions for paper submission, review, and session construction.

Papers offering novel research contributions in any aspect of Communications Software and Services are solicited for submission to the symposium.

Topics of Interest:

- Next Generation Services and Service Platforms
  - ✓ Mobile Services and Service Platforms including IMS
  - ✓ Home Network Service Platform
  - ✓ VoP2P and P2P-SIP Services
  - ✓ Converged Application/Communication Servers and Services
  - ✓ Location-based Services
  - ✓ Social Networking Communication Services
  - ✓ Advanced Communication Services and Feature Interaction
- Multimedia applications and services including VoIP, IPTV, Gaming
  - ✓ Multimedia delivery over wired and wireless networks
  - ✓ Cross-layer optimization for multimedia service support
  - ✓ Multicast, Broadcast and IPTV
  - ✓ Media streaming
  - ✓ Peer-to-Peer services
- Software and Protocol Technologies for advanced service support
  - ✓ Web Services and distributed SW technology
  - ✓ Distributed systems and applications, including Grid Services
  - ✓ Peer-to-Peer technologies for communication services
  - ✓ Service overlay networks
  - ✓ Context Awareness and Personalization
- Network and Service Management and Provisioning
  - ✓ Multimedia QoS provisioning
  - ✓ Quality of Experience for End-to-End Communications
  - ✓ End-to-End Quality of Service Routing algorithms
  - ✓ Service Creation, Delivery, Management
  - ✓ Network Management
  - ✓ Virtual Home Environment
  - ✓ Charging, Pricing, Business Models
  - ✓ Triple Play Services
  - ✓ Security and Privacy in Network and Service Management
  - ✓ Service Overlay Networks
  - ✓ Cooperative Networking for Streaming Media Content

Submission deadline: March 15, 2010
Notifications due: July 1, 2010
Final version due: Aug 1, 2010

For more information about IEEE Globecom 2010, please see http://www.ieee-globecom.org/2010

Symposium Co-Chairs
- John Buford, Avaya Labs Research, USA (buford at avaya.com)
- Mohammad S. Obaidat, Monmouth university, USA (obaidat at monmouth.edu)
- Joel Rodrigues, University of Beira Interior, Portugal (joelr at ieee.org)
- Bin Wei, AT&T Research, USA (bw at research.att.com)

Sponsoring Technical Committees:
- Communications Software
- Multimedia Communications