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Message from R-Letter Co-Director

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

In this issue, we are pleased to introduce eight high quality papers, spanning four main broad areas: P2P video streaming, video quality assessment, wireless communication, and multiview imaging. The first paper, published in the *IEEE Transactions on Multimedia*, presents an approach for reducing startup delays and playout glitches in live P2P streaming. The second paper, from the *IEEE Journal on Selected Areas in Communications*, presents an analytical approach that characterizes the relationship between the probability of playout interruption and the initial buffering in P2P video streaming systems. The third paper, from the *IEEE Communications Magazine*, studies the use of home routers as proxies for energy-efficient mobile P2P applications. The fourth paper, published in the *IEEE Global Communications Conference,* analyzes how many neighbors are sufficient to maintain most of the advantages of peer assistance. The fifth paper, published in the *IEEE International Conference on Multimedia and Expo*, proposes assessing video quality by balancing the global and local quality components. The sixth paper, from the *IEEE Global Communications Conference*, shows that the use of cooperative transmission in multihop wireless networks can help maintain the quality-of-service required by applications while minimizing energy consumption. The seventh paper, published in *IEEE Transactions on Wireless Communications,* discusses how a balance between performance and complexity in multiple-input multiple-output (MIMO) beamforming can be reached in wireless communication. The last paper, from the *IEEE International Conference on Multimedia and Expo*, proposes a compressed-sensing reconstruction algorithm for multiview image sets which utilizes the strong correlation among the images within the set.

In addition to the aforementioned introduced papers, this issue of the R-Letter includes Call for Papers for two special issues of IEEE magazines and journals that are currently being organized by MMTC interest groups.

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

Finally, I would like thank all the editors of this issue for their great work: Cheng-Hsin Hsu, Hulya Seferoglu, Christian Timmerer, Ai-Chun Pang, Tao Liu, Walid Saad, Man-On Pun, and Vladan Velisavljević. I also would like to thank the R-Letter Director Guan-Ming Su for all his efforts.

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Reducing Startup Delays and Playout Glitches in P2P Live Streaming

A short review for "SPANC: optimizing scheduling delay for peer-to-peer live streaming"

Edited by Cheng-Hsin Hsu

K. Chan, S. Chan, and A. Begen, "SPANC: Optimizing Scheduling Delay for Peer-to-Peer Live Streaming", IEEE Transactions on Multimedia, vol. 12, no. 7, pp. 743—753, Nov. 2010.

While Peer-to-Peer (P2P) live video streaming services, such as PPLive, PPStream, and SopCast, have been popular for some time, a measurement study [1] points out that many P2P users suffer from long start-up delays and playout lags, which are largely due to the nature of their *mesh-pull* design. More precisely, peers in mesh-pull systems periodically exchange buffermaps and explicitly request for data chunks from each other. The buffermap changes and pulling process considerably increase the transmission delay over each overlay link, which quickly accumulates along overlay paths and results in very long delay [2].

To cope with this limitation, a hybrid *push-pull* approach has been recently proposed to incorporate the benefits of low-delay of pushing and high-bandwidth utilization of pulling [2,3]. In these systems, each video is first divided into multiple substreams, which are pushed to peers. To recover from missing (or late) data chunks, peers explicitly pull data chunks from their neighbors upon timeout events. While push-pull systems achieve short startup delays and fewer playout glitches under good network conditions, they fall back to the mesh-pull systems when packet loss rates are non-negligible.

The authors propose a system, called Substream Pushing and Network Coding (SPANC), to address the aforementioned shortcomings. Different from push-pull systems [2,3], SPANC employs network coding as a Forward Error Correction (FEC) scheme to eliminate (or at least minimize) the needs of pulling data chunks. In particular, SPANC considers an arbitrary (given) mesh overlay and focuses on efficient scheduling between multiple senders and a single receiver to minimize startup delays and playout lags. A schedule is computed by each receiver, and used in recurring windows until the network topology or the network conditions are changed considerably. Each schedule specifies: (i) the substream assignments, i.e., which sender pushes which substream to the receiver and (ii) the strength of the FEC codes, i.e., number of parity packets for each substream. More details are given in the following paragraphs.

The authors formulate the Substream Assignment (SA) problem as a delay minimization problem, in which the delay of assigning substream j to sender i is captured by an input parameter $C(i, j)$. Although the authors do not assume any form of $C(i, j)$, they present a polynomial-time algorithm to optimally solve the SA problem. More specifically, the SA problem is transformed to a Max-Weighted Bipartite Matching problem, and solved by the Hungarian algorithm [4], which runs in cubic time.

The authors also formulate a Fast Recovery with Network Coding (FRNC) problem, which minimizes the recovery time by determining the number of network coding packets a sender should push to its receiver in each time window. A peer generates random network coding packets [5,6] upon receiving enough packets to reconstruct the original data chunks. The FRNC formulation models the complex relationship between packet loss rate and residue bandwidth. Solving it optimally enables us to eliminate (minimize) the needs for explicitly pulling data chunks. The authors present an optimal algorithm, based on dynamic programming, to solve the FRNC problem, which runs in quadratic time.

The authors evaluate the proposed SPANC system using simulations, and compare it against three existing systems: mesh-pull, hybrid pull-push, and network coding. The simulation results indicate that the SPANC algorithm substantially reduces the startup delays under the same degree of playout glitches measured by residue packet loss rates. Furthermore, SPANC results in small traffic overhead compared to existing systems, and is very reactive to network dynamics including peer churns.

This paper proposes a new approach for P2P live streaming systems, and abstracts two research problems in these systems. The authors present polynomial-time yet optimal algorithms to solve these two problems. These algorithms can also be leveraged in other video streaming services, including multipath and multihomed video streaming [7], which are becoming increasingly important as the cellular service providers are under the pressure of tremendous amount of mobile video traffics [8,9]. Last, despite the extensive simulations conducted by the authors, it will be very interesting to see whether the proposed system can be implemented, and how it performs in real P2P live streaming as well as on-demand services.

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How to Reduce Interruptions in Streaming Video?

A short review for "Avoiding interruptions — A QoE reliability function for streaming media applications "

Edited by Hulya Seferoglu

A. ParandehGheibi, M. Médard, A. Ozdaglar, and S. Shakkottai, "Avoiding Interruptions — A QoE Reliability Function for Streaming Media Applications", IEEE Journal on Selected Areas in Communications, vol. 29, no.5, pp. 1064-1074, May 2011.

The demand for video applications over current networks (especially over mobile networks) is exponentially increasing and it is expected to remain so in the foreseeable future [1]. This demand makes it crucial to design and analyze networks and video streaming applications considering the requirements of video as well as network related constraints to improve mobile/wireless users' Quality of Experience (QoE).

Peer-to-peer (P2P) systems have been successful in distribution of video content including video on demand and live media streaming (*e.g.*, PPLive [2] and QQLive [3]). As the demand for video applications are increasing, P2P for video streaming is likely to gain significance, especially over wireless mobile networks. In P2P video streaming systems, an important question is the selection of playout delay. In particular, it is necessary to have a steady, even stream of packets to reproduce video without interruptions. However, delivery of packets is often irregular due to congestion, server (peer) occupancy, and the time varying nature of wireless channels. A natural solution to this problem is to buffer packets in advance (*i.e.*, before playing out) in order to provide a level of protection against interruptions in video. Yet, the selection of the initial buffer size is crucial for the end users' video experience due to the trade-off between the initial buffer size, *i.e.*, delay, and the probability of interruptions.

This paper presents an analytical approach to characterize the relationship between the probability of interruption and the initial buffering considering the system parameters such as packet arrival rate and size, and different channel models. The main objective is to characterize the amount of buffering needed for a target probability of interruption over the duration of a video stream.

The authors consider a system in which a media file of finite size is transmitted from multiple servers (peers) to a single user. It is assumed that the media file is divided into blocks, and each block is divided into multiple packets. Each server uses random linear network coding [4] to combine packets in a block, and sends these packets to the receiver. The rationale behind using network coding is to increase packet diversity and make coordination for packet transmissions easier. In particular, requesting each packet from multiple peers introduces the need to keep track of packets, and causes duplicate reception problem. This problem is solved with network coding, because network coding makes coded packets equally beneficial at the user [5], [6].

First, it is assumed that each server transmits packets according to a Poisson process. The choice of using network coding simplifies the packet requests. Furthermore, it allows modeling the buffer at the receiver side as an M/D/1 queue. For this model, upper and lower bounds on the minimum initial buffering size required so that the interruption probability is below a desired level are formulated. It has been observed from these bounds that the optimal trade-off between the initial buffering and the interruption probability depends on the file size as well as the playback rate compared to the arrival rate of the packets. It is also shown that the bounds are tight as the file size goes to infinity. Moreover, when the arrival rate and the play rate match, it is shown that the minimum initial buffer size grows as the square-root of the file size. However, if the arrival rate is slightly larger than the play rate, the minimum initial buffering for a given interruption probability remains bounded as the file size grows.

The authors extend this analysis to systems with memory, modeled by Markov modulated arrival processes where the packet arrivals follow two-state continuous time Markov process. Similar trade-offs observed for Poisson processes are observed for Markov processes. An interesting behavior of the optimal trade-offs (for Markov processes) is that the interruption probability reduces as the state transition probabilities increase. The reason is that when the state transition probabilities increase, the arrival process tends to be deterministic (*i.e.*, packet arrivals become steadier) which is required for video stream without interruptions.

The numerical calculations confirm that the trade-off curves between the initial buffering delay and the interruption probability have similar behavior to that of the predicted results by the upper and lower bounds derived in the analytical part.

The authors provide an analytical framework for studying media streaming applications in an unreliable environment. They take the first steps by characterizing the QoE trade-offs (*i.e.*, initial buffering versus interruption probability) for video streaming applications using network coding. In another work [7], the authors use this framework to design network association policies when a user received a media stream from multiple cost-heterogeneous access points such as WiFi and 3G. It would be interesting to extend the provided analysis for more general classes of arrival processes as well as considering dynamic behavior of joining and leaving servers (peers) in the system. Another interesting extension of this work would be to understand the potentials and the limitations of the provided analysis for real networks, *e.g.*, for a system in which one or multiple smartphones demand a video stream from multiple servers (peers) in a P2P system.

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Home Routers as Proxies for Energy-Efficient Mobile P2P Applications

A short review for "Using home routers as proxies for energy-efficient BitTorrent downloads to mobile phone"

Edited by Christian Timmerer

I. Kelényi, Á. Ludányi, and J. K. Nurminen: "Using Home Routers as Proxies for Energy-efficient BitTorrent Downloads to Mobile Phone", IEEE Communications Magazine, vol.49, no.6, pp.142-147, June 2011.

BitTorrent is currently the most popular peer-to-peer content sharing solution, which is used all around the Internet. It can be used for sharing and downloading any kind of data as there are no restrictions on the size or content type. The key idea is that clients downloading the content, which is often referred to as torrent, also become provides, and share the downloaded pieces with the other peers. This enables scalability and robustness, making BitTorrent a serious alternative for centralized content sharing solutions such as FTP or HTTP.

Using BitTorrent on mobile platforms has already a history. The first mobile BitTorrent client was released for Symbian in 2004. Since that, clients have been released for all major mobile platforms. The increased storage capacity and network connectivity of today's smartphones make them first class BitTorrent peers. However, BitTorrent on mobile devices has one serious shortcoming: active BitTorrent usage can drain an average phone's battery in less than five hours. This can be enough for downloading a couple of audio files or smaller videos, but the user must always be aware of the battery status if he/she does not want to get it completely depleted after leaving the client running in the background.

Thus, improving the energy-efficiency of BitTorrent clients is a key requirement for widespread mobile usage. The following three factors are responsible for more than 90% of the energy consumption of a mobile device: display usage, processing using CPU/GPU, and wireless radio usage [1]. It turns out that from these three, the energy cost of the wireless communication is the most significant. One approach to address this issue is improving the speed with which the mobile device receives the content. Measurement studies show that higher bitrate improves the energy-efficiency of file transfer [2]. The speed of BitTorrent download can vary considerably and is often well below the capacity of the wireless interface. A recipe for energy saving is thus to increase the download speed to fill the whole capacity of the wireless channel.

Prior work on energy-efficient BitTorrent took advantage of this phenomenon and shaped the traffic arriving to the mobile device to increase the bitrate it experiences. The essence of the work was to alternate between idle time intervals when the mobile device can the enter power saving state and active intervals when the bitrate is as high as possible. In one case, the mobile peers negotiated a transfer schedule with regular peers [3]. The system guaranteed that during active periods the available peers would dedicate their upload capacity to the use of the energy-sensitive mobile peers. In this way, the mobile peers can save up to 50% of the energy. However, the drawback of this approach is that it requires modifications to the BitTorrent protocol and, thus, was not compatible with existing peers. Another study used an intermediate server, a BitTorrent proxy, hosted in the Internet to split content download into two parts [4]. Regular BitTorrent was used to download the content to the server, which was hosted on Amazon EC2, and normal HTTP was used to transfer the complete file from the server to the phone. Because Amazon EC2 provides a high uplink speed the mobile device received the content with high bitrate. In this way, the energy-limited peers, in comparison to standard BitTorrent, can achieve 40% energy savings without significantly affecting the download speed of regular peers.

As an alternative solution to hosting the proxies in the Internet, the reviewed paper focuses on the use of broadband routers as platforms for running BitTorrent proxies. Broadband routers that connect the home to the Internet via ADSL or other technology are commonly available. Please note that even if these routers are located at homes they do not limit the mobility of the users because they are accessible with mobile devices from everywhere through Internet connections. If the mobile users happen to be at home they could use the local WiFi connection but they can also connect to the router remotely (acting as proxy) through a cellular data connection or through a public WiFi access point. The router platform is attractive for a number of reasons. Since most homes are equipped with broadband routers the installed base is large and the routers are frequently idle when their owners are on the move. Routers are typically powered up all the time and their energy consumption is almost constant no matter how actively they are used. Thus, there is plenty of spare capacity that could be exploited without additional costs for the users. From the technical point of view, the firmware of many routers can be altered, which allows extending the original functionality of the router.

However, a number of new problems arise because broadband routers have limited resources. Although some models allow hardware extensions with USB devices, and some high-end router models even have built-in support for torrent download, typically, the memory size is limited and no mass memory is available. The storage space available on the router is usually not enough to a fit the whole torrent, thus after downloading some pieces and pushing them to the mobile device, the router must delete the content to free some space. This is problematic, since the BitTorrent protocol assumes that if a peer has downloaded a piece, it can also upload it from that point; there is no way to revoke an already downloaded piece. The solution introduced by the paper keeps some of the downloaded pieces forever (these pieces are put into a so called “upload buffer”), and the other pieces are deleted after being pushed to the mobile. This way the peer always has some pieces to share with the others but can also free space if needed.

Both simulations and measurements were carried out to evaluate the solution. For the measurements a prototype system was created using a Linux based router and a mobile client written in Java ME. As a comparison, measurements were performed with a standalone mobile BitTorrent client and a cloud-based solution. The router proxy outperformed both alternatives. Compared with the stand-alone client, using the router-based proxy consumes 40% less energy with 3G and 55% less with WiFi. As expected, doing transfers at higher speeds significantly improves the energy efficiency. In addition to better bandwidth utilization, shorter download times and lower protocol overhead contribute to energy savings.
The simulations investigate the effects of the proxy-based peers on the BitTorrent community. The results show that although the higher percentage of proxy peers in the network also resulted in somewhat slower performance for the standard peers, they had only a minor effect. Furthermore, storing only a few pieces on the routers for sharing was enough to maintain good service quality.

Currently the main drawback of the solution is that it requires installing custom software on the router and on the mobile device. The mobile client is a relatively simple application which can be easily ported to any mobile platform. The router software, however, requires a router with a modifiable Linux-based OS. Installing the software might be too difficult for the average user.

Both the simulation and measurement results indicate that the solution works well in practice, and would worth further investigation. In particular, the scalability of the router itself, i.e., how many streams/downloads can be handled in parallel and how many (mobile) devices can be served within the home network as these numbers are increasing recently. In comparison to downloading the torrent directly to the phone with a native client, up to 40-50% energy can be saved according to some of the measurement results.

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Gossip Protocol Analysis for Peer-assisted Video Streaming

A short review for "Are a few neighboring peers good enough?"

Edited by Ai-Chun Pang

L. Zhong, J. Dai, B. Li, B. Li, and H. Jin, "Are a Few Neighboring Peers Good Enough?", in Proc. of IEEE Global Communications Conference (GLOBECOM), pp. 1-5, Dec. 2010.

It has been well known that the peer-assisted streaming has the most competitive advantage in the robustness and the scalability of the system. This efficient technique can engage millions of users as resource contributors by fully utilizing their upload capacities to assist neighboring peers in the topology. The bandwidth load on dedicated streaming servers is effectively alleviated which therefore supports more demands in the system.

To obtain consecutive video streams in a time-varying environment, one peer has to frequently exchange its state information with other peers periodically, and sends requests for segments on-demand. This refers to a gossip protocol design in peer-assisted video streaming, in which each peer connects to a small number of neighbors in the mesh topology. This "limiting" number of connected neighbors represents the tradeoff between the communication overhead brought by the state information exchange, and the limited local knowledge associated with system-wide efficiency.

An important question discussed in this paper is how many neighbors are sufficient to maintain most of the advantages of peer assistance, including a consistent level of quality, the ability to scale to millions of peers, and robustness against peer churn. Surprisingly, this problem is not yet well understood and addressed. Boyd et al. [1] and Massoulie et al. [2] have modeled gossip protocols as a decentralized broadcasting problem. While studying the process of segment dissemination to participating peers, the maximum sustainable rate and minimum buffering delay of a system without peer churn and with global knowledge have been analyzed in [3] and [4], respectively. Feng et al. [5] have investigated the adverse effects without global knowledge. It has been shown that a performance gap exists between decentralized and centralized scheduling.

This work focuses on exploring the number of neighbors required for "gossiping" protocols to work effectively in a mesh topology. Generally speaking, it introduces an analytical model to examine the influence of the neighbor size while the newest first scheduling strategy is used in the basic model, as it was demonstrated to have the minimum performance gap with the optimal centralized scheme

Authors first analyze the fundamentals of partial knowledge by investigating the distribution pattern of video segments. It proposes several difference equations to describe the global behavior of the system. The theoretical analysis shows that the neighbor size plays an important role in the spreading pattern of segments. A very small neighbor size slows down the propagation of an individual segment due to a limited scope of content availability information obtained from neighbors. This phenomenon may be referred to as the effect of partial knowledge in "gossiping" protocols. However, this effect is significantly subdued with a slight increase in the number of neighbors to more than 5. Such an effect of partial knowledge is not affected by the scale of the system, as long as the number of neighbors a peer has is beyond the examined small threshold. The similar analysis conducted with the random scheduling strategy also proves that a small neighboring size is good enough to alleviate the effect of partial knowledge. The effect is not limited to a specific scheduling strategy.

The dynamics of the system with the peer churn are then presented as a result of the difference equations. Intuitively, a larger neighbor size can alleviate the fluctuation caused by peer churn, by maintaining the network topology [6] and helping the video segment distribution. However, contrary to the intuition, setting the number of neighbors beyond the threshold does not help alleviate challenges caused by peer churn. In fact, peers experience a similar degree of performance degradation under different neighbor size settings.

In summary, with numerical results based on an analytical model consisting of a system of difference equations, authors explore the pattern of distributing media segments to gain important insights into the number of neighbors required for peer-assisted streaming systems that employ "gossiping" protocols. The communication overhead of typical "gossiping" protocols can be substantially contained and mitigated, by using a very small number of neighbors, with which state information is periodically exchanged. The results are appealing and can very useful in the practical system design.

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Role of Human Visual System Attributes in Video Quality Modeling

A short review for "Attention modeling for video quality assessment: balancing global quality and local quality"

Edited by Tao Liu

J. You, J. Korhonen, and A. Perkis, "Attention Modeling for Video Quality Assessment: Balancing Global Quality and Local Quality" , in Proc. of IEEE International Conference on Multimedia and Expo (ICME), pp. 914-919, July 2010.

Perceived video quality assessment plays an important role in video based multimedia services, due to the fact that the signal quality can be degraded by different errors in the process of the signals [1], e.g., compression errors introduced in lossy coding schemes, transmission errors common in wireless communication, and packet losses common in IP networks. Subjective quality assessment conducted by showing distorted video to human observers and asking them to judge the quality quantitatively or qualitatively is the most reliable way to evaluate the perceived quality [2]. However, subjective assessment is very expensive in time and labor. Thus, development of objective video quality metrics that can automatically evaluate the perceived distortions is a reasonable alternative approach. Mean-squared-error (MSE) and peak signal-to-noise ratio (PSNR) are two common metrics widely used in video compression and transmission schemes [3]. Most objective video quality metrics take into account the characteristics of the human visual system (HVS), as it is the final receiver of video stimuli.

HVS is an extremely complex system and not entirely understood so far. An important attribute of the HVS is visual attention mechanism, while its capability in video quality assessment has not been explored adequately. Active selection of the attentive information from entire stimuli might occur early or late in the processing of the stimuli. The late attention selection theory assumes that the selection of attentive information does not occur before categorization and semantic analysis of all the input visual stimuli [4]. Therefore, not only the attended stimuli, but also the unattended stimuli, make contributions to the understanding of the entire visual stimuli. In this paper, considering the late selection mechanism of attentive information, the authors proposed to divide the perceived video quality into two components: global quality and local quality. The global quality is evaluated by viewing an entire video sequence which is a coarse impression; whilst the local quality is a result from subjects allocating their attention to certain stimuli, namely the attended stimuli. Finally, the overall video quality can be a combination of the global quality and the local quality, based on an appropriate fusion method, e.g., linear combination.

In order to evaluate the global quality and local quality, the authors employed four image quality metrics: PSNR, SSIM, multi-scale SSIM, and a modified PSNR based on the HVS. These image metrics were performed on each frame in a video sequence to evaluate the image quality degradation. Under an assumption that the global quality is the result of subjects equally allocating their attention to all regions in a frame and all frames in a video sequence, the global quality is calculated by an averaging operation on the image quality maps derived from the image quality metrics over all spatial regions/pixels and temporal frames.

On the other hand, evaluation of the local quality is more complicated. The authors assume that the local quality is influenced by the distortions on attended regions only, and can also be affected temporal quality change. In order to detect attended regions, an improved video attention model was proposed in this paper. The authors first adopted a well-referenced saliency model proposed by L. Itti et al. [5] to generate a single topographical saliency map. Additionally, because viewers usually pay more attention to moving objects in a sequence and those regions with high contrast, a motion attention model [6] and the contrast information were also employed in the video attention model. Based on the saliency map, the motion attention map, the contrast map of a video frame, an attention map was finally derived to depict the distribution of visual attention region over a video frame.

The local quality assessment is performed on individual frames first by weighting the quality map derived from image quality metrics and the attention map. In addition, some quality assessment experiments have demonstrated that the frequency of quality variations over time also influence the overall quality [7]: (1) the more frequent quality variation, the worse perceived quality; (2) the frames in the beginning and the end of a video sequence have more significant impact on the overall quality, and the tendency that increasing the quality of frames in the end leads to a better perceived quality. Therefore, the authors proposed an appropriate temporal pooling scheme to combine the quality values of individual frames into a single local quality measure.

Finally, the overall video quality is derived from a linear combination between the global quality and the local quality. The authors have used 11 different weights to test their performance. In order to evaluate the performance of the proposed scheme for video quality assessment, two publicly available video quality databases, including the EPFL-PoliMI and LIVE data sets, were employed in the experiments. The experimental results demonstrated that the combination of global quality and local quality achieves a promising performance in evaluation of the perceived video quality. Therefore, in accordance with the late attention selection theory, those visual stimuli, even though they might not attract adequate attention from viewers, cannot be completely ignored in assessing video quality. However, the influence of unattended stimuli on the overall quality assessment might be much weaker than attended stimuli. This work shows a good example how video quality modeling can benefit from HVS characteristics, which become an indispensable component in modern video quality metrics.

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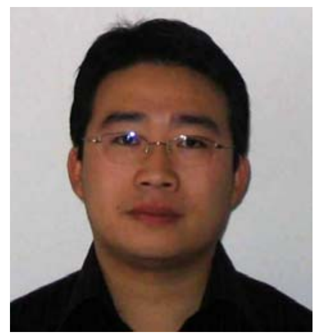

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Energy-Efficient Cooperative Transmission in Multihop Wireless Networks

A short review for "Energy-efficient space-time coded cooperative routing in multihop wireless networks"

Edited by Walid Saad

B. Maham, R. Narasimhan, and A. Hjørungnes, "Energy-Efficient Space-Time Coded Cooperative Routing in Multihop Wireless Networks", in Proc. of IEEE Global Communications Conference (GLOBECOM), pp. 1-7, Dec. 2009.

Wireless sensor networks are characterized by small devices with limited capabilities and energy supply. As a result, energy efficiency is a crucial design component for extending the lifetime of such networks. This work studies, using cooperative transmission, the problem of routing in multihop wireless networks that are outage-restricted. In such networks, a source node wants to transmit messages to a single destination using other nodes in the network that may choose to operate as relay nodes. To study this problem, a new cooperative routing protocol is introduced using the Alamouti space-time code for the purpose of energy savings, given a required outage probability at the destination node. Two efficient power allocation schemes are derived, which depend only on the statistics of the wireless channels. In the first scheme, each node needs to know only the local channel statistics, and can be implemented in a distributed manner. In the second scheme, a centralized power control strategy is proposed, which has higher energy efficiency, at the expense of additional complexity and signaling overhead. Compared to non-cooperative multihop routing, an energy saving of 80% is achievable in line networks with 3 relays and an outage probability constraint of 10^{-3} at the destination.

In general, energy consumption in multihop wireless networks is a crucial issue that needs to be addressed at all the layers of a communication system, from the hardware up to the application. The focus of this work is on energy efficiency when messages may be transmitted via multiple wireless hops. After substantial research efforts in the last several years, routing for multihop wireless networks has become a broadly investigated problem [1], [2]. Nevertheless, with the emergence of new multiple-antennas technologies, existing routing solutions in the traditional radio transmission model are no longer energy.

For instance, it is feasible to coordinate the simultaneous transmissions from multiple transmitters to one receiver simultaneously.

As a result, simultaneous transmitter signals from several different nodes to the same receiver are not considered a collision, but instead could be combined at the receiver to obtain a stronger received signal. In [3], the concept of multihop diversity is introduced where the benefits of spatial diversity are achieved from the concurrent reception of signals that have been transmitted by multiple previous terminals along the single primary route. This scheme exploits the broadcast nature of wireless networks where the communications channel is shared among multiple terminals. On the other hand, the routing problem in the cooperative radio transmission model over static channels is studied in [4], where it is allowed that multiple nodes along a path coordinate together to transmit a message to the next hop as long as the combined signal at the receiver satisfies a given SNR threshold value. In [4], it is assumed that transmitting nodes adjust their phases in such a way that the coherent reception of signals at the receiving node is possible. However, the knowledge of the instantaneous channels at the transmitting nodes is difficult to realize.

In this work, a cooperative multihop routing scheme is proposed for Rayleigh fading channels. The investigated system can achieve considerable energy savings compared to non-cooperative multihop transmission, when there is an outage probability quality-of-service (QoS) requirement at the destination node. Two power control schemes, i.e., distributed and centralized power allocations are derived to minimize the total transmission power given the outage probability constraint. Using tight approximations, simple closed-form power allocation schemes are presented without requiring the knowledge of instantaneous channel state information (CSI); hence, the proposed schemes can be implemented in real wireless systems. Numerical results show that the proposed power allocation strategies provide considerable gains compared to non-cooperative multihop transmission.

In summary, this work shows that the use of advanced communication techniques such as cooperative transmission can help maintain the quality of service required by applications (e.g., outage) while minimizing energy. Relaying is expected to lie at the heart of forthcoming wireless networks such as LTE-Advanced, and, thus, the need for energy efficient techniques such as those proposed in this paper should have significant practical importance. Several extensions for this paper can be envisioned. On the one hand, it is of interest to study energy efficiency in multi-hop networks while considering practical quality of service measures for multimedia services. These measures include cross-layer metrics such as throughput, delay, and traffic modeling. On the other hand, it is of interest to assess the impact of mobility and fast channel variations on the energy consumption and cooperative techniques adopted by the wireless nodes. In a nutshell, the study of energy efficiency using cooperative techniques is expected to become a key design criterion in future wireless communication networks

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Towards Balance between Performance and Complexity in MIMO Beamforming

A short review for "Equal gain transmission with antenna selection in MIMO communications"

Edited by Man-On Pun

S.-H. Tsai, " Equal Gain Transmission with Antenna Selection in MIMO Communications", IEEE Transactions on Wireless Communications, vol. 10, no.5, pp. 1470-1479, May 2011.

Multiple-input multiple-output (MIMO) beamforming has been well-known for its outstanding capability of providing significant performance improvement in wireless communication systems. Among the many existing beamforming techniques, equal gain transmission (EGT) has recently drawn considerable attention, thanks to its excellent balance between performance and complexity. In contrast to the optimal beamforming technique known as maximum ratio transmission (MRT) whose beamforming vector allocates different power magnitudes and phases adaptively across transmit antennas, the suboptimal EGT only adjusts the phase of transmitted signal from each antenna while keeping transmitted power equal across all antennas. This simplification has rendered EGT some significant advantages over MRT in implementation. First of all, since all antennas transmit at the same power level in EGT, it lessens the peak-to-average-ratio (PAPR) problem in EGT and subsequently, provides more flexibility in power amplifiers (PA) design. Second, in frequency division duplex (FDD) systems, the beamforming vector is computed and returned to the transmitter from the receiver. As a result, EGT exempts the receiver from providing magnitude information of beamforming vectors to the transmitter, which is particularly valuable for FDD systems with limited feedback.

Despite the many benefits of EGT, three fundamental questions about EGT have remained open. First, there are little analytical results on the performance loss of EGT as compared to MRT for MIMO systems. Second, it is unclear if there exists any effective and yet simple EGT design to minimize the aforementioned performance loss. Third, if such a design exists, what are the strategies to apply it to systems with limited feedback? Through rigorous mathematical analysis and extensive simulation, this paper has successfully provided fundamental insights into the above three questions. In short, the main results reported in this paper can be summarized as follows. First, the performance loss between EGT and MRT is about 1.05 dB in MIMO systems. Second, this performance loss can be reduced by carefully selecting only the "good" transmit antennas. Finally, this paper has proposed a number of design strategies for EGT with antenna selection (AS) in limited-feedback systems. In the rest of this review, we will first provide a more formal problem statement before highlighting these main results.

We consider a MIMO system equipped with N_t and N_r transmit and receive antennas, respectively. Given a transmit symbol x, the transmitter multiplies x by an $N_t \times 1$ unit- energy beamforming vector $\mathbf{f}$ before sending it over the channel. For EGT, the *i*-th element of $\mathbf{f}$ is given as $f_i = e^{j\theta_i} / \sqrt{N_t}$. Denote by $\mathbf{H}$ of $N_r \times N_t$ the MIMO channel matrix with each element being modeled as independent and identically distributed (i.i.d) zero-mean complex Gaussian random variables. Upon receiving signals from all antennas, the receiver linearly combines the N_r receive symbols to derive an estimate of x. Denote by $\mathbf{g}$ the $N_r \times 1$ unity-energy combing vector. It has been shown that the optimal MRT is achieved if $\mathbf{g}$ and $\mathbf{f}$ are the left and right singular vectors of $\mathbf{H}$ corresponding to the largest singular value [1]. For EGT, the beamforming and combining vectors can be obtained by maximizing the resulting signal-to-noise ratio (SNR), subject to the constraint that each element of the beamforming vector is unit-energy. Unfortunately, deriving the optimal $\mathbf{g}$ and $\mathbf{f}$ is a non-trivial task. This paper proposes to approximate the solution by utilizing the phases of right singular vector of $\mathbf{H}$ as the beamforming vector. It was shown later in the simulation section that this approximation is indeed rather accurate. After invoking the Cauchy-Schwarz inequality, this paper has shown that the maximum performance loss between EGT and MRT is asymptotically bounded by approximate 1.05 dB as N_t approaches ∞.

To reduce the performance loss, the authors proposed to first perform antenna selection before EGT. More specifically, rather than transmitting from antennas suffering from poor channel conditions, it is better to unselect those antennas and re-allocate the energy to other good antennas. The authors developed several low-complexity selection algorithms for EGT. Furthermore, the authors derived a new upper bound for the performance loss between EGT with AS and MRT. Through simulation, the authors showed that the performance loss upper-bound of EGT with AS is substantially reduced as compared to the conventional EGT. In addition to the benefit of low PAPR inherited from EGT, the newly proposed EGT with AS requires fewer radio frequency (RF) components than the conventional EGT.

For FDD systems, beamforming vectors have to be returned from the receiver to the transmitter. This may become an issue of concern for FDD systems with limited feedback. To address this issue, the authors considered various feedback schemes to efficiently return the most essential information to the transmitter with the minimum feedback overhead. More specifically, the author considered feedback comprised of two items: (1) antenna indices of the selected antennas and (2) quantized beamforming vectors. For the first item, the author proposed corresponding efficient index representations for the proposed antenna selection algorithms so that the required numbers of bits are minimized. For the second item, the beamforming vector can be first quantized using either the optimal Lloyd codebook [2] or scalar quantization [3] before being fed back to the transmitter.

In the last part of this paper, extensive simulation results have been shown to confirm the reported analytical results as well as the performance of the proposed EGT with AS. It has been demonstrated in simulation that the SNR loss between the conventional EGT and MRT is indeed about 1.05 dB. Furthermore, it has been shown that EGT with AS can reduce the SNR loss with respect to MRT to as low as 0.45-0.65 dB for MIMO systems with 4 to 8 transmit antennas. Finally, the BER performance of EGT with AS using different limited-feedback schemes was compared.

In summary, this paper has provided a very comprehensive study on both theoretical and practical aspects of EGT by addressing some of the most fundamental and long-standing open questions about EGT. By striking a good balance between performance and complexity, the EGT with AS scheme proposed in this paper is expected to play an important role in future wireless MIMO systems.

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Multiview Imaging: Compressed Sensing Recovery using Disparity Compensation

A short review for "Disparity-compensated compressed-sensing reconstruction for multiview images"

Edited by Vladan Velisavljević

M. Trocan, T. Maugey, J. E. Fowler and B. Pesquet-Popescu, "Disparity-Compensated Compressed-Sensing Reconstruction for Multiview Images", in Proc. of IEEE International Conference on Multimedia and Expo (ICME), pp. 1225-1229, July 2010.

Many systems today use multiple cameras to capture information about a specified scene, such as 3D reconstruction, creation of virtual environments, and surveillance applications. In these multiple-perspective – or multiview – situations, the correlation between images is often high due to similar content. Compression, restoration, or other data-processing tasks can benefit greatly by exploiting this redundancy to improve performance. An efficient way to reduce the redundancy is to estimate disparity (or depth) of objects in the 3D scene, which is related to their displacement when switching between the images taken at different viewpoints.

The process of acquisition of multiview images can still be costly due to a relatively new technology. However, this cost could be significantly reduced if only a few of the multiviews are captured at high resolution or fidelity, whereas the other views are acquired at a worse quality and, consequently at a lower acquisition cost. Such a concept can be implemented using a compressed-sensing (CS) recovery of these latter images.

The recently proposed CS technique (e.g., [1]) for signal representation allows for a signal to be sampled at sub-Nyquist rates with a corresponding methodology of lossless recovery. The CS theory holds that this is achievable under the assumption that the original signal can be described sparsely in either its ambient domain or in some other basis Ψ. The core of the signal-acquisition step commonly involves a projection onto a random basis Φ, which must exhibit a high level of incoherence with the sparse domain [1]. Physical implementations of this methodology have been made, such as the well-known single-pixel camera [2], and many methods have been proposed for the recovery of signals acquired in this manner [3,4,5,6,7,8].

In this work, the authors propose a CS reconstruction algorithm for multiview image sets which takes advantage of the strong correlation between images within the set. The authors start with an efficient algorithm called block-compressed sensing with smoothed projected Landweber reconstruction (BCS-SPL) which was proposed in [4] for the CS reconstruction of a single image. The main idea of the presented paper is to extend the BCS-SPL algorithm to the multiview setting through the use of inter-image disparity compensation during the reconstruction process. Therefore, CS is applied to multiview image sets and inter-image disparity compensation (DC) is incorporated into image reconstruction in order to take advantage of the high degree of inter-image correlation common to multiview scenarios. Instead of recovering images in the set independently from one another, two neighboring images are used to calculate a prediction of a target image, and the difference between the original measurements and the CS projection of the prediction is then reconstructed as a residual and added back to the prediction in an iterated fashion.

To adapt the BCS-SPL algorithm of [4] to the multiview scenario, the authors assume that the images adjacent to the CS-sampled view – i.e., those that anchor the DC prediction – are known. However, in [9,10] the method has been further developed, and the disparity-based CS-recovery has been realized in a more complete, multi-image CS-acquisition system.

The proposed algorithm is partitioned into two phases: in the first phase, a predictor for the current image is created by bidirectionally interpolating the closest (i.e., adjacent) views. The residual between the original observation and the observation resulting from the projection of this prediction (using the same measurement matrix Φ is BCS-SPL reconstructed. In the second phase, the reconstructed residual is further refined in an iterative manner with reverse DC in order to obtain the final reconstruction. The quality of the DC-based reconstruction obtained after the first phase is several dBs higher than that obtained by direct BCS-SPL reconstruction, regardless of the transform Ψ employed. Given the efficiency of the residual-based CS-recovery of phase one, the second phase needs only a small number of iterations. At each iteration, the prediction is obtained by DC between the current reconstructed image and its neighbors; the improvement in reconstruction quality is due to the refinement of the disparity vectors, leading to a smoother residual at each step which is more easily reconstructed by BCS-SPL.

This new method for the CS recovery of multiview images takes advantage of the high degree of inter-frame correlation which is characteristic of multiview applications. The authors include disparity estimation and compensation and employ a reconstruction involving a residual rather than an image. The experiments reveal a substantial increase in reconstruction quality for the DC-based algorithm as opposed to a simple, direct CS reconstruction driven by only the random measurements of the image while ignoring the surrounding views.

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Special Issue on QoE management in emerging multimedia services

The realization of the paradigm of Internet anywhere, anytime and any-device and the diffusion of end-user multimedia devices with powerful and user-friendly capabilities such as smartphones, tablets pc, mobile gaming terminals and ebooks, are leading to the proliferation of a significant amount of emerging multimedia services: immersive environments, mobile online gaming, 3D virtual world, book/newspaper consumption, social networking, IPTV applications, just to cite a few. Some of these services have already reached a major market success, such as the case of newspaper/magazine mobile readers and smartphone multimedia apps. Their success could be achieved especially because a user-centered approach has been followed to design the whole process of content production, service activation, content consumption, service management and updating. Indeed, the quality of the user experience, the perceived simplicity of accessing and interacting with systems and services, and the effective and acceptable hiding of the complexity of underlying technologies are determining factors for success or failure of these novel services, as well as graceful degradation.

The management of the Quality of Experience (QoE) is then undoubtedly a crucial concept in the deployment of future successful services, and it is straightforward to be understood as well complex and stimulating to be implemented in real systems. The complexity is mainly due to the difficulty of its modeling, evaluation, and translation in what for more than a decade we have been mainly dealing with (partially in its substitution), that is the Quality of Services (QoS). Whereas QoS can be now easily measured, monitored and controlled at both the networking and application layers and at the end-system and network sides, the quality of experience is something that is still quite intricate to be managed. The practice in evaluating the QoS can be exploited in evaluating the QoE, but it is just a starting point for a complete QoE management procedure, which should encompass at least the following activities: monitoring of the experience of the user when consuming the service, adapting the provisioning of the content on the basis of the varying context conditions (e.g. network status, user behavior, user profile, environment), predicting potential experience level degradation, and masking quality degradation due to abrupt system changes. To have a complete control of the final user experience, all these tasks need to be performed in a coordinated way and their real effectiveness depends on the validity of the adopted user perception model.

Objectives

The purpose of this special issue is to present to the magazine's audience a concise, tutorial oriented reference of the state-of-the-art, current and future research challenges and trends on the management of QoE in emerging multimedia services. To achieve this goal the special issue seeks original research and review papers that survey and present new ideas, leading-edge research prototype development, trials and early deployment, and performance evaluations in the following areas:

- Definition of QoE (Quality of Experience) for emerging services
- Relationship between QoE and QoS
- Architectures for the management of the QoE in emerging multimedia services
- Offline and online prediction and evaluation of QoE
- QoE-oriented multimedia traffic management
- QoE-oriented multimedia source and channel coding
- Testbeds for performance evaluation of QoE-oriented systems
- Middleware solutions for QoE management
- Adaptive and self-configuring solutions for QoE management
- Advanced, scalable service-aware QoE-oriented traffic control and management
- QoE management in heterogeneous networks

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Call for Papers: IEEE Communication Surveys and Tutorials

Special Issue on Energy-Efficient Multimedia Communication

Background

Multimedia has gained immense popularity in a variety of applications related to education, entertainment, business, and location-based services. Recent advances in networking and display technologies have enabled the dissemination of multimedia to a variety of devices, from cellular telephones to tablet PCs to wall-size screens. The proliferation of media hosting services and social networks have allowed users to easily share multimedia content with a much wider audience. Digital cameras and camcorders have replaced films and tapes, making it simpler to generate multimedia. Users can easily view, process, analyze, publish, retrieve, or modify multimedia on these devices. However, energy consumption is still a major challenge in the dissemination of multimedia. Energy is consumed during various stages - processing, communication, and storage - of multimedia. In addition, data centers where media services are hosted have also seen a rapid increase in energy consumption in recent times. This trend is not sustainable. Significant progress must be made to save energy and slow down the rate of energy consumption in all these stages. This special issue aims to provide researchers and professionals in the communication, networking, multimedia, and computing communities with insightful papers that present an overview of new approaches to making multimedia communications more energy-efficient.

Scope

Topics of interest include (but are not limited to):

- Energy-efficient network/communication protocols for multimedia data transmission
- Energy-efficient multimedia communication architectures
- Low-power hardware, software, or both for different stages of multimedia processing, such as acquisition, coding, compression, storage, transmission, and reception
- Energy-efficient techniques for content analysis, indexing, searching, and retrieval in resource-constrained (such as mobile and embedded) systems
- System-level energy-efficient design and implementation for multimedia communication
- Energy conservation for multimedia on mobile devices
- Tools for measuring and analyzing energy consumed during multimedia communication

Manuscript Submission

IEEE Communications Surveys & Tutorials is a ComSoc publication. It is an ideal venue for researchers and other communications professionals to publish tutorials and surveys reachable to a large global audience. Articles should be written in a style comprehensible and appealing to readers outside the specialty of the article. Authors are encouraged to visit the "Call for Papers" and "Information for Authors" pages at the IEEE Surveys and Tutorials web site at http://dl.comsoc.org/surveys/. Please submit manuscripts via the ManuscriptCentral website at http://mc.manuscriptcentral.com/comst-ieee (the entry name: Special Issue: Energy Efficient Multimedia Communication)

Important dates

Manuscript due:	**September 30, 2011**
Acceptance Notification:	January 31, 2012
Publication date:	Q2, 2012

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

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

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

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