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CONTENTS

Message from R-Letter Director

The publication of IEEE MMTC Review-Letter (R-Letter) has entered the second year! Thanks for all MMTC members' support and interest, R-letter keeps achieving the original goals to introduce disruptive and promising innovative concepts and ideas in multimedia and/or communication to MMTC members through thoroughly selecting and reviewing high-impact and pioneering papers from recent IEEE Communication Society and MMTC sponsored publications as well as other IEEE publications.

In this issue, we are pleased to introduce nine high quality papers, spanning four main areas: strategies for layered video transmission, diversity gain for multimedia communication, novel development of multimedia applications, and new perspectives for communication systems. The first paper, published in the *IEEE Transactions on Multimedia*, presents a layered video streaming approach with traffic information feedback from network nodes. The second paper, from the *IEEE Transactions on Multimedia*, presents a novel forward error correction method for layered video transmission. The third paper, from the *IEEE Transactions on Multimedia*, studies the cross-layer video streaming via optimizing the visual entropy. The fourth paper, published in the *IEEE Journal on Selected Areas in Communications,* explores the multi-dimension diversity for multi-stream video transmission scenario. The fifth paper, published in the *IEEE International Conference on Multimedia and Expo*, proposes a new performance metric to evaluate the remote rendering system. The sixth paper, from the *IEEE Transactions on Multimedia*, shows a novel video summarization method for multiview video. The seventh paper, published in *IEEE Transactions on Wireless Communications,* analyzes the tradeoffs between open access and close access for uplink femtocell. The eighth paper, published in *IEEE Transactions on Wireless Communications,* proposes a frequency-domain approach for frame detection and timing acquisition in OFDM system. The last paper, from the *IEEE Journal on Selected Areas in Communications*, presents an energy-efficient underwater sensor network via random access compressed sensing.

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: Hulya Seferoglu, Carl James Debono, Christian Timmerer, Ai-Chun Pang, Cheng-Hsin Hsu, Vladan Velisavljević, Walid Saad, Man-On Pun, and Hassan Mansour. I also would like to thank Nabil Sarhan for all his efforts.

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Bandwidth Allocation and Reliability for Layered Video

A short review for "Layered Internet video adaptation (LIVA): network-assisted bandwidth sharing and transient loss protection for video streaming"

Edited by Hulya Seferoglu

X. Zhu, R. Pan, M. S. Prabhu, N. Dukkipati, V. Subramanian, and F. Bonomi, "Layered Internet Video Adaptation (LIVA): Network-Assisted Bandwidth Sharing and Transient Loss Protection for Video Streaming", IEEE Transactions on Multimedia, vol. 13, no. 4, pages 720-732, August 2011.

Video traffic over the Internet is rapidly growing and it is expected to remain so in the foreseeable future [1]. For example, according to [1], Internet video is now approximately one-third of all consumer Internet traffic and will account for over 60% by the year 2013. Providing high quality video is a challenging problem over the Internet not only due to the sheer volume increase in video traffic. Video streaming applications also impose new challenges to the best-effort Internet, in that they require persistently high bandwidth and timely packet delivery to ensure continuous media playback. Furthermore, the compressed video streams are sensitive to packet losses, as error propagation at the decoder can severely degrade received video quality. These challenges make it crucial to design and analyze networks and video streaming applications considering the requirements of video as well as network related constraints to improve video quality.

This paper combines network utility maximization and rate-distortion optimization frameworks [2], [3] to provide design guidelines for bandwidth sharing and reliability. The combined scheme, layered Internet video adaptation (LIVA), provides a stable media-aware bandwidth sharing scheme that achieves fast convergence and efficient bottleneck utilization, and a proactive adaptive FEC scheme that provides reliability by correcting transient packet losses.

To develop the proposed scheme (LIVA), the authors first provide an optimization model which follows classical network utility maximization framework [4]. Differently, this paper uses a parametric model from [5] which characterizes rate-distortion (R-D) tradeoff in the objective function. This makes it possible to share the bandwidth among competing video streams in a media-aware fashion, by adapting the rate of each video stream according to its R-D characteristics.

The proposed optimization problem is decomposed into sub-problems following the same approach in [2]. In particular, the decomposed solutions involve calculation of a virtual congestion level at each network node and adaptation of the video rate at each sender (video source) based on observed maximum congestion level along the path. The decomposed solution lends itself for distributed implementation: the intermediate network nodes remain oblivious of the video R-D information while each video sender only needs the end-to-end maximum virtual congestion level. This makes the solution suitable for practical deployment.

The authors provide a guideline for practical implementation of the proposed solution. The following is the summary of the proposed implementation. (i) Each network node calculates virtual congestion level for each outgoing link based on the incoming and outgoing traffic as well as the current state of the congestion level. (ii) Virtual congestion level information is stamped on each packet (in a header field allocated for this information) traversing through the node if the packet's current congestion level information is lower than the one calculated at the node. This information is transferred from video packets to their corresponding acknowledgement packets, thus transmitted to a video source. (iii) Video sources determine optimal transmission rate based on the congestion level information. The effect of delayed information is eliminated by predicting current congestion level using recently received congestion level information. The proof of the stability of the proposed prediction and rate control algorithm is provided in the paper.

The second part of the proposed scheme (LIVA) is the adaptive forward error correction (FEC). LIVA uses FEC instead of re-transmission based mechanism to provide reliability due to time delivery requirements of video applications. The proposed adaptive FEC scheme arranges amount of redundancy at a video source based on the congestion feedback information. In particular, when congestion information increases, amount of redundancy should increase, because the large value of the congestion information is an indicator of upcoming packet drops. Finally, original video rate is determined using the optimal rate and redundancy which are calculated based on the congestion level information. In particular, the highest video rate (of a layered video) which is lower than the difference between the optimal rate and the redundancy rate is selected.

The simulation results show that LIVA can minimize the total distortion of all participating video streams and hence maximize their overall quality. At steady state, video streams experience no queueing delays or packet losses. In the face of transient congestion, the network-assisted adaptive FEC promptly protects video packets from losses. The authors also present a proof-of-concept system demonstration of LIVA.Furthermore, the authors show that LIVA can also be implemented using existing congestion signaling mechanisms such as the already standardized Explicit Congestion Notification (ECN), and that video streams using LIVA receives comparable bandwidth as competing TCP flows, without sacrificing media-awareness.

The authors provide an interesting aspect of combining network utility maximization and rate-distortion optimization frameworks to provide design guidelines for bandwidth sharing and reliability. It would be interesting to understand the TCP friendliness of the proposed scheme from a theoretical perspective. The other interesting extensions of this work would be to consider (i) wireless networks, (ii) multicast video streaming, and (iii) live video streaming.

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A Forward Error Correcting Scheme for Mobile Broadcasting of Layered Media

A short review for "Layer-aware forward error correction for mobile broadcast of layered media"

Edited by Carl James Debono

C. Hellge, D. Gómez-Barquero, T. Schierl, and T. Wiegand, "Layer-Aware Forward Error Correction for Mobile Broadcast of Layered Media", IEEE Transactions on Multimedia, vol. 13, no. 3, pp. 551—562, Jun. 2011.

Layered media formats such as Scalable Video Coding (SVC) [1] and Multiview Video Coding (MVC) [2] present new prospects for effective coding and distribution in Mobile TV services. The mobile environment is characterized by bandwidth constrains and high transmission error rates. Moreover, the industry requires that new broadcasting technologies still offer basic services compatible with older systems. These limitations can be eased through layered media formats that open up for features like graceful degradation and for the introduction of new services which are backward-compatible. Decoding some parts of the bitstream is only possible when the corresponding more important data is correctly reconstructed. This can be directly applied to layered media where unequal error protection (UEP) can be employed to protect the important parts of the bitstream. Mobile broadcasting techniques usually apply forward error correction (FEC) on the upper layers to handle transmission errors which are not corrected by the physical layer schemes. However, current FEC schemes are optimized for single layer video systems.

The original paper discusses a Layer-Aware FEC (LA-FEC) scheme, which considers the dependencies present in layered video systems when constructing the FEC codes. The base layer FEC is unaltered to ensure backward compatibility with existing systems. On the other hand, the FEC symbols of dependent layers are generated across all data on which the currently encoded layer depends upon. Thus, the scheme improves the error robustness of the more important layers. The LA-FEC approach can be applied to any linear FEC on the physical layer, like Low Density Parity Check (LDPC) codes [3], or upper layer, like Reed-Solomon [4], Raptor [5], and RaptorQ [6], while retaining the full correction performance of the extended FEC algorithm. Existing dependencies are maintained thereby tagging the worst performance to the standard techniques. At the decoder side, traditional FEC coding loses the frame if the lower layer information is corrupted in such a way that it cannot be corrected, even when higher layer codewords are received correctly. Conversely, with this technique the decoder uses the cross-layer information in the higher layers to try to reconstruct the corrupted lower layer codewords, thus improving the error resilience of the system.

The authors provide a short review of the history on related work, starting back from [7]. Furthermore, differences related to the current state-of-the-art are outlined, where the authors claim that LA-FEC manages to keep the original performance of the FEC codes across all involved media layers.

After outlining the principle of the LA-FEC, the authors employ a combinatorial analysis to show the potential gain in terms of FEC decoding probability of the media layers and the objective video quality assessment in terms of PSNR. The results presented with two media layers show that the proposed LA-FEC increases decoding probability for the more important layer, while the decoding probability of the other layer remains unchanged. The authors further analyzed the influence of the distribution of the FEC symbols across the media layers. They show, that for a normal FEC, where symbols are independently generated for each layer, a stronger robustness is beneficial. Contrary to that, a more equal FEC distribution seems is beneficial for LA-FEC, since the base layer is more protected. From the results, it can be concluded that the LA-FEC scheme generally gives better performances than normal FEC, with none of the tested scenarios giving weaker results.

The authors further describe an exemplary implementation of the concept with Raptor codes as a state-of-the-art FEC codes. A detailed description on the extensions required and references to a full standard specification are found in [8]. The authors also discuss the required means for signaling and transport of LA-FEC. All required resources for delivering layered media together with LA-FEC are specified at IETF within FECFRAME, RTP, and SDP protocols.

Finally, the authors show experimental results on the performance of layer-aware Raptor codes with SVC in a handheld Digital Video Broadcasting (DVB-H) environment. These confirm the outcome of the analytical analysis and show that the LA-FEC outperforms a Raptor implementation with independent FEC encoding for each layer.

This paper paves the way to further research into the exploitation of layer-media for better encoding and error protection. The testing was done on only two layers, yet further improves are expected when more information is exploited from other layers. A combination of coding schemes can also be applied to improve the protection of the important data while redundancies are reduced to save on bandwidth requirements.

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Improving Multimedia Communications over 4G Wireless Systems

A short review for "Cross-layer optimization for downlink wavelet video transmission"

Edited by Christian Timmerer

H. Lee, S. Lee and A.C. Bovik, "Cross-Layer Optimization for Downlink Wavelet Video Transmission", IEEE Transactions on Multimedia, vol.13, no.4, pp.813-823, Aug. 2011.

Over the past few decades, two dynamic forces have resulted in dramatically wider deployments of mobile video: firstly, substantial increases in wireless channel capacity using such next generation communication standards as Long Term Evolution (LTE) and Wireless Broadband (WiBro) / Mobile Worldwide Interoperability for Microwave Access (WiMAX) and secondly, significant advances in video compression methods as exemplified by the Advanced Video Coding (AVC) standard. Theoretical link capacity and compression gains are nearing Shannon's capacity bound and the entropy of reflecting intrinsic source randomness respectively. As such, it is highly desirable to seek new efficiencies in wireless video data transport. Towards this end researchers are studying new protocols for expressing visual data importance and radio resource allocation to deliver an optimized visual experience. Advances in these directions could yield substantial gains beyond traditional theoretical bounds, expressed in terms of perceptually driven Quality of Service (QoS), also referred to as Quality of Experience (QoE).

The authors present a visually optimized framework for cross-layer optimization and demonstrate its efficiency for 4th Generation (4G) wireless systems where channel capacity is close to theoretical upper bounds, yet rapidly decreases away from base-stations towards cell boundaries owing to multi-cell interference. Currently, QoS improvement in the cell edge region is actively discussed as a most pressing issue. From the service point of view, the unpleasant picture of interruptions or disconnections of streaming video and telephony services when channel capacity falls below an acceptable threshold is to be avoided.

The method taken in this paper follows a different paradigm: by imposing a new quality criterion called *visual entropy* as the objective utility function, rather than using the traditional log function, visual importance can be selectively imposed within the data stream. The visual entropy is defined as the expected number of bits required to represent image information in a perceptual manner, such as by a mapping onto foveated human visual coordinates assuming a gaze direction [1]. Thus, the authors aim to bridge gaps that exist between cross-layer optimization, principles of perceptual importance, and radio resource management.

Traditional cross-layer optimization methods rely on channel throughput enhancement by controlling the radio resource over the physical, data link, and network layers. Such approaches typically demonstrate that QoS is achieved using Peak Signal-to-Noise Ratio (PSNR) or the Structural SIMilarity (SSIM), rather than involving the application layer in the optimization. Utilizing the visual entropy, it is possible to describe cross-layer optimization more directly in terms of the QoE.

The authors employ wavelet transform-based Motion JPEG2000 – in previous studies the authors have also investigated Scalable Video Coding (SVC) [2] - which offers scalability wherein the video stream is divided into layered data enabling flexibility of spatial resolution, temporal resolution, and fidelity. By exploiting this multi-layer representation, each bitstream may be assigned a different degree of visual importance so that the corresponding radio resource can be optimized.

The authors describe interference mitigation from other cells by transmitting video data selectively as a function of visual importance. In this manner the most important visual data transmitted to edge users experiences minimal ICI from neighboring cells.

The procedure to mitigate inter cell interference (ICI) is straightforward. The total bandwidth is divided into subbands, where each scalable video stream is allocated to one subband with transmit power assigned in a stepwise pattern. A framework for coupling of multimedia and wireless channel layers using a reliable information-passing protocol is proposed for quality control in such diverse environments as

Orthogonal Frequency Division Multiple Access (OFDMA) networks, Multiple Input Multiple Output (MIMO) transmission, and multi-hop communication [2][3][4].

Although the proposed solution is optimal and has been demonstrated in the context of wavelet-based video coding, it is probably worth to investigate its applicability for Scalable Video Coding (SVC) [5] and a comparison thereof.

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Towards Finer Granularity of Multi-Dimension Diversity Video Streaming

A short review for "Distributed scheduling scheme for video streaming over multi-channel multi-radio multi-hop wireless networks"

Edited by Ai-Chun Pang

L. Zhou, X. Wang, W. Tu, G.-M. Muntean, and B. Geller, "Distributed Scheduling Scheme for Video Streaming over Multi-Channel Multi-Radio Multi-Hop Wireless Networks", IEEE Journal on Selected Areas in Comm., vol. 28, no. 3, pp. 409-419, April 2010.

With the introduction of multi-hop technology and multiple radio interfaces for multi-channel access in wireless networks [1], significant performance improvement has been demonstrated compared to single hop single radio interface [2]. Although the improved bandwidth and capacity network provides a potential better platform for larger volume of video communications, especially to provide multiple streaming scenario, the highly heterogeneous type of video streaming deviated among users will interfere each other if without any resource assignment. Therefore, it is fundamental to impose a scheduling policy dedicated to optimize video metrics in terms of perceived quality and fully utilize network utilization. The main issue of scheduling multi-user video over multi-channel multi-radio network is how to fully utilize the network resources, including channel assignment problem for each stream, rate allocation problem for each channel, routing problem for selecting channels and links for each stream; and fairness problem among all participated video streams in the network. It is a challenging cross-layer control problem if we want to resolve this problem by fully and jointly addressing the aforementioned problems. Besides, owing to user scaling issue, a distributed manner of scheduling policy is often preferred.

This work studies the problem of video streaming over multi-channel multi-radio multi-hop wireless networks, and develops fully distributed scheduling schemes with the goals of minimizing the video distortion and achieving certain fairness. Firstly, a general distortion model is constructed according to the network's transmission mechanism, as well as the rate distortion characteristics of the video. Secondly, the scheduling is formulated as a convex optimization problem, and a distributed solution is proposed by jointly considering channel assignment, rate allocation, and routing. Specifically, each stream strikes a balance between the selfish motivation of minimizing video distortion and the global performance of minimizing network congestions. Thirdly, the proposed scheduling scheme is extended by addressing the fairness problem. Unlike prior works that target at users' bandwidth or demand fairness, this work proposes a media-aware distortion-fairness strategy which is aware of the characteristics of video frames and ensures max-min distortion-fairness sharing among multiple video streams.

The main technical contributions or highlights of this paper are as follows:

1) It provides a novel distributed video scheduling scheme in the context of multi-channel multi-radio multi-hop wireless networks. The support for multi-user video streams in this network requires appropriate joint channel assignment, rate control and multi-path routing measure, ascertaining the reasonable routes for transmitting each stream and the rate of the video to be delivered over the chosen routes. Different from previous works on video scheduling in single-channel multi-hop wireless networks or multiple wireless networks in which channel assignment is not a concern [3], this paper considers the scheduling problem in the newly emerged networks and proposes an efficient assignment algorithm.

2) It takes into account the specific video characteristics in the routing and rate control scheme. Network congestion is considered in the channel assignment, rate allocation and routing metric, to meet the stringent delay requirement for video transmission. In addition, each video's rate-distortion characteristic is also taken into account in the joint routing and rate control procedure to provide multiple streams with various video contents. This work is the first one to deal with the video scheduling problem in the newly multi-channel networks.

3) It extends the scheduling scheme by proposing a strategy of media-aware distortion-

fairness, which is aware of the characteristics of video contents and ensures max-min distortion-fairness sharing among video streams. In particular, it does not employ any explicit utility function, but instead uses the importance of every frame which can be easily and explicitly calculated using the method in [4]. Furthermore, it is content awareness and is operated over both links and sources, so the proposed scheme belongs to per-stream performance guarantee, which is also different from existing architecture that offers application performance guarantees [5].

The results in this paper have some interesting implications on the practical use of multi-radio multi-channel multi-hop wireless networks, *i.e.*, multimedia sensor network is a good example. As we know, current sensor networks due to their limit transmit capacities can hardly transmit large amount of multimedia data concurrently. Multi-channel multi-radio technique is a direction to provide a satisfying multimedia service in wireless sensor networks. In addition, 3GPP LTE (Long Term Evolution) system using relay is also an example. As we know, LTE uses OFDM (Orthogonal Frequency Division Multiplexing) for the downlink, in which the transmitter sends information over a large number of sub-carriers [6]. So it can be viewed as a special type of multi-channel multi-radio multi-hop wireless system. Therefore, the proposed video scheduling scheme for multi-channel multi-radio multi-hop wireless networks is eager to have large application ground.

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A Novel Approach to Evaluate Interactivity

A short review for "Distortion over latency: novel metric for measuring interactive performance of remote rendering systems"

Edited by Cheng-Hsin Hsu

S. Shi, K. Nahrstedt, and R. Campbell, "Distortion Over Latency: Novel Metric for Measuring Interactive Performance of Remote Rendering Systems", in Proc. of 2011 IEEE International Conference on Multimedia and Expo (ICME'11), pp. 1-7, July 2011.

The recent interests in bringing computations into clouds make the term *remote rendering* popular again. The concept of remote rendering is to render the source contents on a server and display the resulting scenes on a client, which connects to the server through networks. Cloud gaming, represented by OnLive [1] is a good example, in which 3D video games are rendered in the cloud and the game scenes are encoded as 2D video streams and sent to game players through broadband networks. The control signals from game players are collected on the client side and sent back to the cloud server to interact with the game applications. Users of the remote rendering systems are sensitive to system *lags*, and thus the interactive performance is critical to the success of these systems.

Traditionally, latency has been the metric for system interactivity evaluation. The average latency for webpage rendering was used to compare the performance of different thin-client systems [2]. The study of network gaming indicates that long latency can significantly impair the user gaming experience. 100ms is reported as the largest tolerable latency for the first person shooting game [3]. However, latency may no longer be accurate to measure the actual interactivity of the remote rendering systems enhanced with *latency reduction* techniques. For example, some image based rendering algorithms [4] can help the client to synthesize a requested image immediately after the user interaction occurs (e.g., a gamer presses a button) and therefore significantly reduce the latency at the cost of lower rendering quality. More specifically, due to the limitation of algorithms, image based rendering may inevitably generate artifacts (e.g., holes caused by image warping) on the synthesized images. In this case, the interactivity evaluation should take both latency and rendering quality into consideration.

This paper addresses this problem. The authors proposed a new metric named DOL (Distortion Over Latency) to improve the evaluation of interactive performance. The metric is defined as the sum of product of the latency and the distortion during the latency period. The distortion is calculated as the pixel difference between the actually displayed image and the reference image, which can only be displayed in an ideal zero latency system. Therefore, the new metric DOL successfully combines both latency and rendering quality into one score.

Intuitively, larger DOL score indicates worse interactive performance because both long latency and bad post-rendering quality lead to large DOL scores. In the paper, three remote rendering systems with different latency reduction techniques are implemented to demonstrate that the new metric DOL can actually be used to indicate the interactive performance of real systems. The methodology is to build the system with already known performance and examine whether the metric data meets expectations. The experimental results clearly support the assumption: DOL score can distinguish the interactive performance of different system setups as expected while the traditional latency metric cannot.

While combining latency and rendering quality into the DOL score has never been proposed in the literature, the actual definition of DOL function might be adjusted to better match human perceived quality. For example, rendering quality metrics other than Mean-Square Error (MSE) may be adopted, and the relation between latency and user experience may not be linear.

Indeed, as the authors acknowledged in the paper, a more convincing demonstration of DOL metric requires quantitative studies to correlate the DOL score with the actual interactive performance that human can perceive. More human tests should be designed and carried out to better understand the connections between DOL scores and user satisfaction. Nonetheless, this paper points out a new research direction, and will stimulate many more studies along this direction.

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Higher-Dimensional Visual Summarization in Multiview Video

A short review for "Multi-view video summarization"

Edited by Vladan Velisavljević

Y. W. Fu, Y. W. Guo, Y. S. Zhu, F. Liu, C. M. Song and Z.-H. Zhou, "Multi-View Video Summarization", IEEE Trans. on Multimedia, Vol. 12, No. 7, pp. 717-729, July 2010.

"*The Media Internet...is at the crossroads of digital multimedia content and Internet technologies.*" [1] On one hand, the rapid development of Internet communications and hardware infrastructures enables video acquisition, archiving, cataloging and indexing much easier, which increase the usability of stored videos by "*professional and novice content prosumers*" [1]. On the other hand, the sheer volume of video data however blocks many practical multi-view video applications. Especially, the all-weather day and night multi-view surveillance systems record huge volume of video content for the same scene equipped in offices, banks, factories, and crossroads of cities for private and public securities. For example, in the London riots this year it was reported that surveillance cameras equipped in each corner of city streets had captured thousands of hours of multi-view surveillance videos. However, it still remains a challenging problem on how to make use of these multi-view videos due to the tremendous video size, which obstructs the browsing, retrieval, and storage of the video content.

To address this problem and better manage the volume of videos, recent advances in scalable video coding such as H.264/SVC and MVC (multi-view video coding) have paved the way for the blossom of adaptive and flexible media streaming applications. Video summarization, in contrast, recently emerges as a promising alternative solution to be integrated in various current and future Media Internet applications, such as interactive browsing and searching systems. However, previous video summarization studies focused on mono-view videos, and the results would not be good if they were applied to multi-view videos directly, due to problems such as the redundancy within and across multiple views.

Therefore, the authors for the first time present a promising approach, i.e. multi-view video summarization. This technique refers to the problem of summarizing multi-view video collection into informative video summaries to provide users an efficient overview and important details of the whole unfamiliar video collection. The multi-view video summaries are usually presented as dynamic video shots by considering content correlations within each view and among multiple views.

In this method, after firstly parsing videos into shots (e.g. [2]), a spatio-temporal shot graph (hypergraph) is constructed and then partitioned via random walks [3]. The final multi-level summary is chosen through multi-objective optimization, and presented by multi-view storyboard and event board. The main idea is to identify clusters of event-centered shots and to select those representative ones in the spatio-temporal shot graph. Meanwhile, to measure low-level shot information, the authors propose a Gaussian entropy fusion model inspired by nonlinear time series analysis [4]. From one shot, multiple modality features are extracted that can be taken as different "shot signals". Intrinsically, the more dramatic changes in these signals, the more information hidden, and the more important the shots are.

As a representation of the spatio-temporal structure of multi-view videos, the spatio-temporal shot graph encodes shot information and reflects intuitive correlations among multi-view shots. Hypergraph is used in the multi-view settings here [5] to systematically characterize the diverse visual, temporal and semantic correlations with different attributes among shots. A hypergraph is a graph in which an edge (named hyperedge) can connect more than two nodes. The hypergraph can overcome the severe parameter-sensitive problem existed in the ordinary graph structure which has been widely used in many previous mono-view summarization methods, when the different correlations for edge weights are fused in one graph.

The complex correlations in spatio-temporal shot graph make the summarization task challenging. The authors have considered selecting the most representative graph nodes by users' requirements. They formulate the multi-view video summarization as a graph labeling problem that partitions the graph in a random walks-based event-centered shot clustering, and then picks out the summary via multi-objective optimization. The authors have adopted random walks rather than other graph partition algorithm e.g. Graphcut, due to its effectiveness in handling large and complex graphs, and naturally K-way segmentation character. As for different user requirements, different summarization objectives, such as minimum summary length and maximum information coverage, are accomplished in this method. Moreover, multi-level summarization could be achieved easily by configuring the optimization parameters.

It is also non-trivial to display multi-view video summary. Different from previous mono-view summary, the summary has both multi-view spatial and temporal information and correlations. It requires a natural and intuitive way for the user to walk through and analyze the summary. To handle this problem, this work creatively presents two new display methods, i.e., multi-view video storyboard and event-board. Multi-view storyboard is a natural extension of storyboard in mono-view video summary by spatially listing multi-view videos and their summary in temporal order [7]. The storyboard naturally reflects correlations among multi-view summarized shots that describe the same important event. The event-board assembles event-centered multi-view shots in a temporal order. With the event-board, a single video summary that facilitates quickly browsing of the summarized video can be easily generated.

Multi-view video summary, especially multi-view storyboard, serves as an efficient and effective video navigation way for browsing multi-view video collection. The multi-view video summary here especially facilitates the browsing, retrieval, storage and consuming of these videos. Multi-view video summary provides a transformation of the given redundancy multi-view source videos that are less resource intensive and less time consuming (e.g. for users' individual mobile devices [8]) by only retaining the most important elements of the original content. It is also worth pointing out that the proposed framework has well defined the event-centered cluster structures which can be used not only in the display of summary but also in other applications, e.g., video indexing.

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Access Policies in the Uplink of Femtocell Networks: To Open or To Close?

A short review for "Open vs. closed access femtocells in the uplink"

Edited by Walid Saad

P. Xia, V. Chandrasekhar and J. G. Andrews, "Open vs. Closed Access Femtocells in the Uplink", IEEE Transactions on Wireless Communications, Vol. 9, No. 10, pp. 3798 - 3809, December 2010.

The recent surge in resource-demand wireless services such as multimedia and gaming led to an unprecedented increase in wireless data traffic. This increased load has strained existing networks and motivated researchers to seek innovative solutions for increasing the capacity and coverage of existing wireless networks. To this end, the deployment of a large number of small cells, serviced by low cost, low power, *femtocell access points (FAPs),* has emerged as a promising technique for enabling future wireless networks to meet the ever-increasing growth in wireless traffic [1]. Essentially, a femtocell access point can be seen as a plug-and-play device that operate over the licensed band and can be installed by end-users in their homes so as to benefit from better reception indoor.

The deployment of FAPs in future networks faces numerous challenges at different levels such as interference management and efficient co-existence with existing technologies [1][2][3]. In particular, due to the lack of coordination between the FAPs and the main macro-cell base stations, cross-tier interference is one of the major design challenges of femtocell-enabled networks. In order to overcome this challenge, three main femtocell access control schemes have been proposed [4]: closed access, open access, and hybrid access.

In closed-access mode, the FAPs service only a registered set of subscribers (e.g., home users) who are provided an exclusive access to the femtocell tier and its associated backhaul. While this access mode has many merits, in terms of preserving the privacy of the FAP owners and providing them with dedicated links, it can lead to severe uplink cross-tier interference from nearby macro-cell users. In contrast, by using an open-access mode, nearby macro-cell users can access and use the FAPs for their transmissions. Clearly, open access has two key advantages: (i)- it enables network operators to expand their coverage and benefit from the third party backhaul that interconnects the femtocells and (ii)- it enables femtocell owners to reduce uplink macro-to-femto interference by allowing the strong interferers to connect to the FAPs and coordinate with existing users through it. However, these advantages come at the expense of a potential decrease in the QoS of the femto home users, who must now share their femtocell resources with an unpredictable number of macro-cellular users. To combine the advantages of both closed and open access, a hybrid femtocell access mode has been proposed in which a restricted number of macro-cellular users can be handed over to the femto tier so as to maintain certain QoS targets of the femtocell owners.

As each access mode has its advantages and drawbacks, it is of interest to better understand the performance of each one of these modes as well as the preferences of the network operator and the femtocell owners. In this regard, the work in [5] provides an in-depth study of closed and open access modes (a general term including open and hybrid access mentioned above, since hybrid access is the same as open access but with an upper limit on the number of cellular users using the femtocell) in the uplink of a two-tier femtocell network.

Using techniques from stochastic geometry, the performance analysis done in [5] leads to very interesting insights on the suitability of each access mode. On the one hand, it is shown that in orthogonal access (e.g., OFDMA or TDMA), under the assumption of no base station coordination, the preferences of the network operator and the femtocell owners are highly dependent on the density of macro-cellular users and can be incompatible. For instance, while both parties prefer open access for medium density and closed access for high density networks, in low density, open access provides high rate gains for the femtocell home users at the expense of a decreased sum throughput for the macro-cellular users. Hence, as demonstrated in [5], for OFDMA-based 4G networks (e.g., LTE), adapting the access policy to the users' density is recommended.

For non-orthogonal multiple access such as CDMA (e.g., 3G networks), the results in [5] suggest that open access is the preferred mode of operation for both the femtocell owners and the network operator, as it leads to better rates at all user densities. In short, interference reduction is so important in CDMA networks, that even selfish femtocell owners should allow nearby macro-cellular users using their femtocells in the uplink.

In summary, the work in [5] has established that the choice of an access mode for femtocell networks is dependent on various factors such as multiple access technique and users' density. To this end, this work constitutes a stepping stone contribution towards deciding on the most suitable access mode for future femtocell networks. Note that, the tradeoffs of open and closed access are studied in [6] for the downlink and shown to be different from their uplink counterparts.

Several future extensions can be envisioned in order to gain a better understanding on the various femtocell access modes. On the one hand, it is of interest to study the impact of cooperation and coordination, at both tiers (e.g., inter-base station or inter-FAP coordination) on the access mode preferences. On the other hand, this work can be extended to investigate whether the presence of backhaul performance constraints (such as delay or congestion) would modify the access mode choices of the operator and femtocell owners. Finally, the effect of adaptive resource allocation strategies such as power control can be studied.

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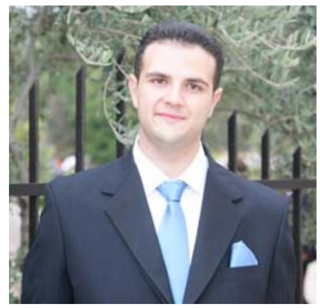

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A Fresh Look at Synchronization for OFDM under Strong Interference

A short review for "Frame detection and timing acquisition for OFDM transmissions with unknown interference"

Edited by Man-On Pun

L. Sanguinetti, M. Morelli, and H. V. Poor, "Frame detection and timing acquisition for OFDM transmissions with unknown interference", IEEE Transactions on Wireless Communications, vol. 9, no.3, pages 1226-1236, March 2010.

Orthogonal frequency-division multiplexing (OFDM) is well envisaged as one of the most promising technology for broadband digital transmissions. It has been successfully standardized into a wide range of commercial applications, including the digital video broadcasting, the IEEE 802.11 wireless local area network and the digital subscriber line. Despite of its many advantages, such as robustness against multipath fading environment and inherent flexibility in subcarrier assignment and power allocation, a major challenge in implementing practical OFDM systems resides in achieving accurate synchronization. Over the past decade, enormous research efforts have been devoted to the synchronization problem in OFDM. A very comprehensive overview of the results obtained in this area can be found in [1]. However, most of these results were derived by assuming an interference-free environment. As future wireless networks will be characterized by a reduced cell size in order to achieve higher spectral efficiency, such an interference-free assumption may appear too optimistic [2]. Thus, practical techniques to achieve synchronization under unknown interference will become indispensible for future OFDM-based wireless networks.

In this paper, the authors have taken a pioneering approach to develop practical and yet robust synchronization techniques for OFDM systems under unknown narrow-band interference (NBI). More specifically, the authors considered the two most critical steps in initial synchronization, namely the frame detection and the timing acquisition tasks. In the first step, this paper proposed to model NBI as a Gaussian process and devised general likelihood ratio test (GLRT)-based schemes by mitigating the uncertainty due to NBI and carrier frequency offset (CFO). After achieving frame detection, this paper proposed to accomplish timing acquisition by exploiting the time evolution of the test statistic derived in the first step and looking for its global maximum over a predefined timing window. Finally, extensive simulation results were shown to demonstrate the impressive performance of the proposed synchronization schemes even in the presence of strong unknown interference. In the following sections, key innovations in each of these two steps will be highlighted.

The synchronization operation in wireless communications systems, particularly those systems that use burst transmissions, begins with detecting the presence of the desired signal in the received waveform. Since OFDM systems organize data packets in a frame structure, this task is accomplished by frame detection. Most conventional methods work in the time-domain by exploiting known symbols (i.e. pilots) placed at the beginning of the OFDM frame. In sharp contrast, this paper argues that performing GLRT-based frame detection in the *frequency* domain offers more robustness under unknown NBI. However, to fully achieve this benefit, the frequency-domain detector has to cope with the uncertainty that unknown NBI and CFO introduce into the GLRT-based hypothesis testing. To this end, this paper considers the NBI power and the signal CFO as nuisance parameters. The former is eliminated from the likelihood function using a Bayesan approach [3], while the latter is replaced by its maximum likelihood (ML) estimate. More specifically, this paper proposes to model the unknown NBI as a zero-mean Gaussian process whose variance follows an inverse-gamma distribution. As a result, the uncertainty due to NBI can be removed by averaging the likelihood over the whole inverse-gamma distribution. The ML estimate of the CFO is subsequently derived by using both full ML as well as heuristic approaches. While the CFO estimate derived from the full ML approach can provide the optimal performance in the GLRT test, it requires a computationally expensive grid search over the whole CFO uncertainty range. Alternatively, two suboptimal practical detection schemes were presented in the paper. In the first one, the CFO estimate is obtained in closed-form and used in place of the ML CFO estimate. In the second approach, the dependence of the test statistic on the fractional CFO is removed by means of heuristic arguments. The resulting frame detection schemes make the GLRT become a straightforward comparison between a simplified test statistic and a predefined threshold.

After an OFDM frame is successfully detected, the next critical step in synchronization is to localize the beginning of the frame. This task is commonly known as timing acquisition. Timing synchronization is a critical task in OFDM transmissions as large timing errors can cause detrimental inter-block interference (IBI). Intuitively, the test statistic employed in the previous frame detection step can also be exploited for timing acquisition. However, the channel dispersion due to multi-path fading environment causes spurious indication signals, which reduces the acquisition accuracy. In order to mitigate the spurious indication signals, this paper proposed to pre-advance timing estimates so that the expected value of the resulting timing error falls within a pre-defined interval.

In the last part of the paper, extensive numerical simulations have been presented to assess the performance of the proposed synchronization schemes. The simulation results have revealed that the proposed schemes are inherently robust to NBI and outperform existing alternatives in the presence of strong interference at the price of a larger processing burden. However, the advantages of the proposed schemes over more conventional methods become less significant as the interference power subsides.

In summary, this paper has provided a very fresh look at achieving synchronization for OFDM systems under unknown interference. Unlike most conventional methods that perform frame detection in the time domain, this paper has demonstrated that a frequency-domain approach is more robust against unknown NBI, especially in the presence of severe interference. Furthermore, the test statistic can also be exploited for accurate timing acquisition. However, it is fair to say that the advantages of the proposed schemes are derived at the price of additional computational complexity. More computationally efficient implementation of the proposed schemes may deserve further investigation in the future. Nevertheless, it is expected that this paper will inspire more research efforts on exploring this new direction in OFDM synchronization.

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Energy-Efficient Networking: Random Access Compressed Sensing Approach

A short review for "Random access compressed sensing in energy-efficient underwater sensor networks"

Edited by Hassan Mansour

F. Fazel, M. Fazel, and M. Stojanovic, "Random Access Compressed Sensing in Energy-Efficient Underwater Sensor Networks", IEEE Journal on Selected Areas in Communications, vol. 29, no. 8, pp.1660-1670, September 2011.

Monitoring environmental phenomena is crucial in shaping our knowledge of the environment and its evolution. The wireless sensor network technology has enabled affordable large coverage and long term monitoring of the natural environment such as climate monitoring, ocean observation, etc. In a general setup, sensor nodes are scattered over a region and monitor a slowly varying field. Each node employs a sensing scheme, which dictates the frequency of measurements, then communicates its observation to a central processing unit, referred to as the fusion center (FC). Once the network is deployed, there can be little access to the sensors and recharging batteries becomes difficult. This is especially of concern in underwater networks, where sensor nodes are hundreds of meters below the surface, or in wireless networks where a hostile environment prohibits access to the sensor nodes. Therefore, enabling long term deployment calls for energy aware sensing and efficient communication architecture.

The paper that is reviewed here considers a two-dimensional spatially varying field, which has a sparse representation in the (spatial) frequency domain, as is the case in many natural fields. The authors propose an integrated sensing and communication, referred to as Random Access Compressed Sensing (RACS), which achieves overall efficiency in terms of the energy per bit of information successfully delivered. Considering the fact that most natural phenomena have a sparse representation in an appropriate domain, RACS capitalizes on integrating compressed sensing [1] with random channel access [2]. The former supports transmission of sensor data from only a random subset of all the nodes– thus reducing the overall energy consumption– while the latter supports a robust and simple implementation that eliminates the need for scheduling, and downlink feedback.
Because of the nature of random access, packets from different nodes may collide at the FC. The key idea in RACS is to let the FC simply discard the colliding packets. This approach is motivated by the compressed sensing theory, i.e., the fact that the FC does not care which specific sensors are selected as long as (i) the selected subset is chosen uniformly at random, and (ii) there are sufficiently many correct packets remaining to allow for the reconstruction of the field. Therefore, so long as the packet collisions occur randomly, discarding of those packets will not change the random nature of the correct packet arrival.

The FC thus discards the colliding packets and collects the remaining packets during a frame. The frame duration is assumed to be shorter than the coherence time of the measured process, such that the process can be approximated as fixed during that interval. It is assumed that packets which do not collide are correctly received. Here, assuming ideal channel conditions, collisions constitute the major source of packet loss. In [3] and [4] the authors extend this model to realistic channel conditions and study the effects of communication noise and channel fading on the system. In a given frame, reconstruction will be successful if sufficiently many correct packets are collected. Otherwise, reconstruction for that particular frame will fail. Using the theory of compressed sensing [5], in order to guarantee accurate recovery with very high probability, it suffices to ensure that the FC collects at least $\mathbf{N}_s = \mathbf{CS}\log\mathbf{N}$ correct packets per frame, where $\mathbf{N}$ is the size of the network (total number of nodes), $\mathbf{S}$ is the sparsity of the field and $\mathbf{C}$ is a constant independent of $\mathbf{N}$ and $\mathbf{S}$.

The proposed frame-based (slotted) RACS can be summarized as below:

1) At the beginning of a frame, each sensor participates in sensing with probability $\mathbf{p}$ or stays inactive with probability $1 - \mathbf{p}$ during that frame.
2) If a node is selected for sensing, it measures the process and encodes its measurement into a packet along with the sensor's location tag.
3) The selected node then picks a uniformly-distributed delay for the transmission of its packet.

4) The FC collects the packets received during one frame. If a collision is detected, it discards the colliding packets. Upon reception of a correct packet, the FC demodulates the signal and extracts the measurement information.
5) At the end of the frame, the FC uses the correctly received packets to reconstruct the field using ℓ_1 minimization or other sparse recovery methods.

Adopting a Poisson packet generation model, the above mentioned slotted RACS is extended to a continuous-time scheme in [6], obviating any synchronization requirements. To compensate for the packet losses and provide a sufficient number of measurements to the FC, the number of participating sensors (those that transmit a packet during a frame) has to be somewhat greater than the minimum number of packets required. This number of participating sensors is related to the per-node sensing probability $\mathbf{p}$, which is the key design parameter in the slotted RACS. Because of the random nature of the system architecture, a probabilistic approach to system design is necessary. The proposed design approach relies on the notion of sufficient sensing probability, the probability with which full field reconstruction is guaranteed in a certain interval of time. The principal figure of merit is the average energy required for field reconstruction. Setting the sufficient sensing probability to a desired target value, system optimization under the minimum energy criterion yields the necessary per-node sensing probability.

Conventional designs rely on deterministic sensing (querying all the nodes in a network) with deterministic channel access (e.g. TDMA). Numerical results obtained for RACS demonstrate that substantial savings in energy per bit– on the order of tens of dBs– are achievable over a conventional TDMA system for the same coverage. For example, for a network of $\mathbf{N} = 2500$ nodes deployed to monitor a field with sparsity $\mathbf{S} = 10$, with a frame duration of 800 seconds and a packet duration of 0.2 second, RACS offers 10 dB saving in energy compared to a conventional TDMA network. In addition, the per-node bandwidth required by the sensors to communicate their observations to the fusion center is several times lower than in the benchmark case. This is another important feature, not only for acoustic systems where bandwidth is fundamentally limited, but for any communication system where the total bandwidth must be traded off between the information rate and the coding rate. In addition to energy and bandwidth efficiency, RACS affords scalability and robustness against random (isolated) node failures. These features constitute an attractive scheme for large scale wireless sensor networks, deployed for long term monitoring of sparse phenomena (such as environmental monitoring applications).

The proposed sensing and communication scheme represents a new niche application to the emerging field of Compressed Sensing. As the number of deployed sensors in various monitoring applications increases and the amount of communicated information grows, the adoption of random access technology similar to the one proposed by the authors will become critical for the viability of these applications. Therefore, there is a need to initiate a standardization effort that unifies random access compressed sensing schemes for next generation communication protocols.

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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

Prospective authors should follow the IEEE Communications Magazine manuscript format described in the Authors Guidelines (http://dl.comsoc.org/livepubs/ci1/info/sub_guidelines.html). All articles to be considered for publication must be submitted through the IEEE Manuscript Central (http://commag-ieee.manuscriptcentral.com), according to the following timetable:

Submission Deadline:	**October 20, 2011**
Notification of Acceptance:	January 15, 2012
Final Manuscript Due:	March 10, 2012
Publication Date:	April 2012

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