Guest Editorial
Adaptive Media Streaming

Christian Timmerer, Carsten Griwodz, Ali C. Begen, Thomas Stockhammer, and Bernd Girod

RECENTLY, traditional TV services, Internet TV and mobile streaming services have started converging, and it is expected that this convergence trend will continue with other services. Additionally, new emerging multimedia services are being introduced. These developments in the multimedia arena mean that various content and services will be delivered over different networks, and the users expect to consume these services using those networks, depending on the availability and reach of the networks at the time of consumption. This massive heterogeneity in terms of terminal/network capabilities and user expectations requires efficient solutions for the transport of modern media in an interoperable and universal fashion. In particular, in recent years, the Internet has become an important channel for the delivery of multimedia. The Hypertext Transfer Protocol (HTTP) is widely used on the Internet and it has also become a primary protocol for the delivery of multimedia content.

Additionally, standards developing organizations (SDOs) such as MPEG have developed various technologies for multimedia transport and encapsulation, e.g., MPEG-2 Transport Stream and ISO base media file format. These technologies have been widely adopted and are heavily deployed by various providers and in different applications and services, such as digital broadcasting, audio and video transport over the Internet and streaming to mobile phones, etc. At the same time, many other SDOs such as the IETF, IEEE, and 3GPP have provided various protocols to deliver multimedia content packetized or packaged by such MPEG technologies.

This special issue is concerned with the latest developments in the state-of-the-art, and we can present you with a selection of 10 papers that cover a variety of aspects of modern adaptive streaming system research.

As all multimedia research, adaptive video streaming is primarily intended for the consumption by people. Thus, it is of major importance to understand human users' preferences in adaptation mechanisms that are applied.

The paper "Subjective Quality Study of Adaptive Streaming of Monoscopic and Stereoscopic Video" is therefore opening this special issue with the results of subjective experiments that have tested the end-user response to the video quality variations in adaptive HTTP streaming of monoscopic and stereoscopic video content.

We follow this user-oriented view on adaptive HTTP streaming with a series of papers that aim at this kind of systems as we see it most frequently in the present day. Adaptive HTTP streaming using non-scalable content supplants UDP-based streaming in most situations, but it is far from perfect.

The paper "An Evaluation of Bitrate Adaptation Methods for HTTP Live Streaming" refers to older work to take end-user preference into account, and uses them to evaluate the benefits of various typical adaptation methods for adaptive HTTP streaming. The authors carry out a comparative evaluation that takes not only bitrate and buffer behaviors into account, but also the perceptual impact on end users.

The authors of the paper "Optimal Delivery of Rate-Adaptive Streams in Underprovisioned Networks" consider distribution systems for adaptive HTTP streaming content in the live streaming case. The authors' approach aims at maximizing the quality of experience for end-users in spite of a possible under-provisioning in a CDN. They formulate both an optimal delivery method and design a practical system.

"Probe and Adapt: Rate Adaptation for HTTP Video Streaming At Scale" does not distinguish between live and on-demand content, but starts with the real-world observation that adaptive HTTP streams compete in an unexpected way for a bandwidth bottleneck whenever this kind of traffic contributes to a substantial portion of the total traffic. Clients have difficulty to correctly estimate their fair share, leading to oscillating bandwidth. As a response, the authors present PANDA, which implements a "probe and adapt" principle.

"Avoiding Quality Bottlenecks in P2P Adaptive Streaming" departs from the current state-of-the-art, and considers a peer-to-peer distribution system instead. Also encoding departs from non-scalable segmentation; the authors consider scalable video coding (SVC) for their adaptive streaming system. They coin the term "quality bottleneck" as a problem experienced by peers, which may have sufficient network capabilities to receive higher quality layers, but fail to do so due to other peers' limitations. The authors integrate overlay formation with data scheduling and content adaptation to reduce this quality bottleneck.

All remaining papers leave the wired Internet, and consider mobile receivers instead.

In "Control of Multiple Remote Servers for Quality-Fair Delivery of Multimedia Contents", this means mainly that receivers are wireless devices, which share a common wireless resource. They introduce a proposal for sharing of this wireless

C. Timmerer is with Alpen-Adria-Universität Klagenfurt and bitmovin GmbH, Austria (e-mail: christian.timmerer@itec.uni-klu.ac.at).
C. Griwodz is with Simula Research Laboratory AS and University of Oslo, Norway (e-mail: griff@simula.no).
A. C. Begen is with Cisco, Canada (e-mail: abegen@cisco.com).
T. Stockhammer is with NoMoR Research, Germany (e-mail: stockhammer@nomor.de).
B. Girod is with Stanford University, CA, USA (e-mail: bgirod@stanford.edu).
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resource that makes use of buffering in an aggregator, whose decision strategy considers communication delays between servers and aggregators. Their experimental results show a converging control system that provides a fairness improvement over the classical max-min bandwidth fairness approach.

Also “Impact of Execution Time on Adaptive Wireless Video Scheduling” considers mobile streaming beyond the timeframe of a single-hop mobile network. The authors observe that most wireless scheduling decisions assume on instantaneous adaptation, which is not generally possible. They develop a stochastic optimization problem that takes execution time into account, and prove that although the execution time is disadvantageous to the stability region, it is actually advantageous to the flow balance.

Delivering adaptive video streams to users in an extremely densely populated space, which exceeds the ability to deploy WiFi infrastructure effectively, is the topic of “Real-Time Network Coding for Live Streaming in Hyper-Dense WiFi Spaces”. The authors solve this challenge by creating a multicast model with network coding that requires a minimum of feedback from receivers. Not only initially sent packets are useful for many receivers, but also errors from multiple users are treated at once; all of this is done while taking packet deadlines into account.

The scenario that wireless nodes are the source of video transmissions is introduced in “Non-Stationary Resource Allocation Policies for Delay-Constrained Video Streaming: Application to Video over Internet-of-Things-Enabled Networks”. The challenge addressed here is that all video senders must have sufficient transmission opportunities to use before they reach their deadline. The authors solve this without detailed packet-level knowledge, but by translating delay deadlines into a weight distribution within the available time, and using this to prioritize transmission. They prove the optimality of their approach and have implemented it.

The final paper looks at multi-hop wireless networks. “A Real-Time Adaptive Algorithm for Video Streaming over Multiple Wireless Access Networks” assumes that mobile users want to stream video among each other. They make use of multiple wireless links for efficiency, cost-effectiveness and increasing streaming quality. The authors formulate a Markov Decision Process to optimize the video streaming process with a reward function that considers Quality of Service (QoS) for video, and then develop a feasible heuristic that has been implemented.

We hope that this short overview demonstrates the pervasiveness of adaptive media streaming in current research and practice, and invite our readers to proceed to the papers themselves.

Christian Timmerer is an assistant professor in the Department of Information Technology (ITEC) and a member of the Multimedia Communications Group at the Alpen-Adria-UniversitŁt Klagenfurt, Austria. His research interests include immersive multimedia communication, streaming adaptation, quality of experience, and sensory experience. Dr. Timmerer received an MSc (Dipl.-Ing.) and PhD (Dr.techn.) from the Alpen-Adria-UniversitŁt Klagenfurt. In addition to publishing more than 120 technical papers, he’s an associate editor for Computing Now responsible for social media technologies and he was the inaugurating chair of the Special Technical Community on Social Networking. He is also an editorial board member of the Encyclopedia of Multimedia, ACM/Springer International Journal on Multimedia Tools and Applications (MTAP) an area editor for the Elsevier journal on Signal Processing: Image Communication.

Dr. Timmerer has been actively participating in several EC-funded projects, notably the FP6-IST-DANAE, FP6-IST-ENTHRONE, FP7-ICT-P2P-Next, FP7-ICT-ALICANTE, COST-IC1003-Qualinet, and FP7-ICT-SocialSensor projects. He was the general chair of WIAMIS2008, QoMEX2013, and QCMan2014 and participated in the work of ISO/MPEG (the International Organization for Standardization and Motion Picture Experts Group) for more than 10 years. Hes a member of the IEEE Computer Society, IEEE Communications Society, and ACM SIGMM. Follow him on http://www.twitter.com/timse7 and subscribe to his blog http://blog.timmerer.com.

Carsten Griwodz leads the Media Department at the Norwegian research company Simula Research Laboratory AS, Norway, and is professor at the University of Oslo. He received his Diploma in Computer Science from the University of Paderborn, Germany, in 1993. From 1993 to 1997, he worked at the IBM European Networking Center in Heidelberg, Germany. In 1997 he joined the Multimedia Communications Lab at Darmstadt University of Technology, Germany, where he obtained his doctoral degree in 2000. He joined the University of Oslo in 2000 and Simula Research Laboratory in 2005. His research interest is the performance of multimedia systems. He is concerned with streaming media, which includes all kinds of media that are transported over the Internet with a temporal demands, including stored and live video as well as games and immersive systems. He was involved the organisation of various conferences, including General Chair of ACM MMSys 2013 and Area Chair of ACM MM 2014. He is currently editor-in-chief of the ACM SIGMM Records, and associate editor of ACM TOMCCAP and IEEE R-letters.

Ali C. Begen is with the Video and Content Platforms Research and Advanced Development Group at Cisco. His interests include networked entertainment, Internet multimedia, transport protocols and content delivery. Ali is currently working on architectures for next-generation video transport and distribution over IP networks, and he is an active contributor in the IETF and MPEG in these areas. Ali holds a Ph.D. degree in electrical and computer engineering from Georgia Tech. He received the Best Student-paper Award at IEEE ICIP 2003, the Most-cited Paper Award from Elsevier Signal Processing: Image Communication in 2008, and the Best-paper Award at Packet Video Workshop 2012. Ali has been an editor for the Consumer Communications and Networking series in the IEEE Communications Magazine since 2011 and an associate editor for the IEEE Transactions on Multimedia since 2013. He is a senior member of the IEEE and a senior member of the ACM. Further information on Ali’s projects, publications and presentations can be found at http://ali.begen.net.

Thomas Stockhammer received the Dipl.-Ing. and Dr.-Ing. degrees from the Munich University of Technology, Munich, Germany. He is cofounder and CEO of Novel Mobile Radio (NoMoR) Research, a privately owned company providing consulting and software development services as well as products on emerging communication networks such as HSPA, MBMS, LTE, LTE-Advanced as well as on Mobile TV, IPTV, and Web TV-related matters. Since 2009 he is a Consultant for Technical Standards for Qualcomm. Specifically, he is the active and has leadership and rapporteur positions in 3GPP, MPEG, IETF, DVB and the DASH-Industry Forum in the area of multimedia communication, TV-distribution, content delivery protocols and adaptive streaming.

Bernd Girod is Professor of Electrical Engineering in the Information Systems Laboratory of Stanford University, California, since 1999. Previously, he was a Professor in the Electrical Engineering Department of the University of Erlangen-Nuremberg. His current research interests are in the area of networked media systems. He has published over 500 conference and journal papers and 6 books, receiving the EURASIP Signal Processing Best Paper Award in 2002, the IEEE Multimedia Communication Best Paper Award in 2007, the EURASIP Image Communication Best Paper Award in 2008, the EURASIP Signal Processing Most Cited Paper Award in 2008, as well as the EURASIP Technical Achievement Award in 2004 and the Technical Achievement Award of the IEEE Signal Processing Society in 2011. As an entrepreneur, Professor Girod has been involved in several startup ventures, among them Polycom, Vivo Software, 8x8, and RealNetworks. He received an Engineering Doctorate from University of Hannover, Germany, and an M.S. Degree from Georgia Institute of Technology. Prof. Girod is a Fellow of the IEEE, a EURASIP Fellow, and a member of the German National Academy of Sciences (Leopoldina). He currently serves Stanfords School of Engineering as Senior Associate Dean for Online Learning and Professional Development.