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Robust Rate control for Video Streaming in Heterogeneous Environment

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ABSTRACT—the problem of scalable video streaming from a server to multinetwork clients over heterogeneous access networks, with the goal of minimizing the distortion of the received videos. This problem has numerous applications. They are: 1) mo- bile devices connecting to multiple licensed and ISM bands, and 2) Cognitive multiradio devices employing spectrum bonding. In this paper, we ascertain how to optimally determine which video packets to transmit over each access network. We present models to capture the network conditions and video characteristics and develop an integer program for deterministic packet scheduling. Solving the integer program exactly is typically not computationally tractable, so we develop heuristic algorithms for deterministic packet scheduling, as well as convex optimization problems for randomized packet scheduling. We carry out a thorough study of the tradeoff between performance and computational complexity and propose a convex programming-based algorithm that yields good performance while being suitable for real-time applications. We conduct extensive trace-driven simulations to evaluate the proposed algorithms using real network conditions and scalable video streams. The simulation results show that the proposed convex programming-based algorithm: 1) outperforms the rate control algorithms defined in the Datagram Congestion Control Protocol (DCCP) by about 10–15 dB higher video quality 2) reduces average delivery delay by over 90% compared to DCCP; 3) results in higher average video quality of 4.47 and 1.92 dB than the two developed heuristics 4) runs efficiently, up to six times faster than the best-performing heuristic

KEYWORD—Quality optimization, rate control, stream adaptation, video streaming

I. INTRODUCTION

Market research indicates that mobile data traffic will increase 39 times over a span of five years, and 66% of the increase will be due to mobile videos. In fact, cellular service providers are having a hard time coping with the huge increase in mobile data traffic, and will have to care- fully engineer their systems to support high-quality real-time video streaming. In wireless networks, one way to achieve the best possible streaming quality is to leverage all available wire- less spectra by connecting the streaming server to each client via multiple access networks. We refer to the clients capable of connecting to multiple access networks as multinetwork or multihomed clients. Potential application scenarios of multinetwork clients include streaming videos to: 1) multiradio wireless devices connected to different Industrial, Scientific, and Med ical (ISM) bands; 2) cognitive multiradio clients employing spectrum bonding; and 3) multiradio clients connected to both licensed band (such as 3G cellular network) and ISM band (such as IEEE 802.11 networks) multihoming can also be viewed as an alternative to multipath video streaming. Multipath streaming, although studied in the literature, e.g., is not widely deployed. This is partially due to the additional requirements on designated Network equipment. In contrast to multipath video streaming, multihomed video streaming works on the current Internet infrastructure

An approach of arbitrarily splitting a video stream into multiples substreams and sending each substream over an access network may lead to degraded video quality and playout glitches. This is because transmitting a substream at a low rate may underutilize the network resources, while transmitting at a rate close to the available bit rate may lead to network congestion, which in turn causes packet drops and late packet delivery. To this end, rate control based on measurements of available bit

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rate (ABR) and round-trip time (RTT) needs to be performed to achieve a good tradeoff between throughput and delay In this paper, we present a mathematical formulation of the joint rate control and scalable stream adaptation problem for multiple clients[1] concurrently competing for the same access networks . We abstract the problem of streaming videos to multinetwork clients and formulate an optimization problem to determine, for each client: 1) the streaming rate over each access network; 2) the video packets to be transmitted; and 3) the access network over which each transmitted video packet is sent. Due to the discrete nature of the considered optimization problem, and its NP-completeness, we formulate it as an integer program in order to derive the global-optimal solutions.

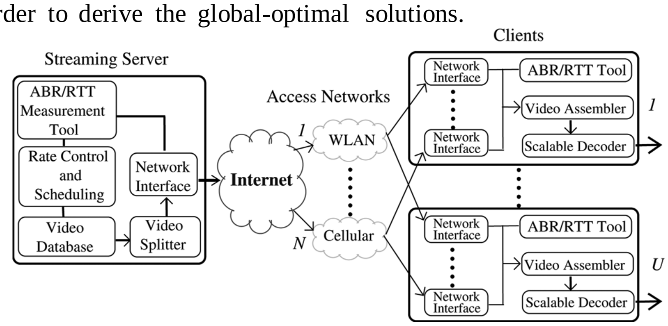

Fig.1. Sample system architecture of a scalable video streaming system with clients and access networks.

Our contributions can be summarized as follows.

- We formulate the joint rate control and packet scheduling problem as an integer program where the objective is to minimize a cost function of the expected video distortion. We suggest different cost functions in order to provide service differentiation and address fairness among users.
- We propose heuristic algorithms for packet scheduling, analyze their complexity, and study their performance through trace-driven simulations.
- We consider randomized packet scheduling by relaxing the integer program into a real-valued optimization problem. We derive convex programming approximations to this problem.

We analyze, both analytically and experimentally, the performance versus computational complexity tradeoff of the proposed optimization programs and recommend one that yields good performance while being suitable for real-time applications.

II. PROBLEM FORMULATION

Video Model We consider H.264/SVC video streams coded with medium-grained quality scalability (MGS). Each stream Is divided into multiple layer called network administration layer(NALs)

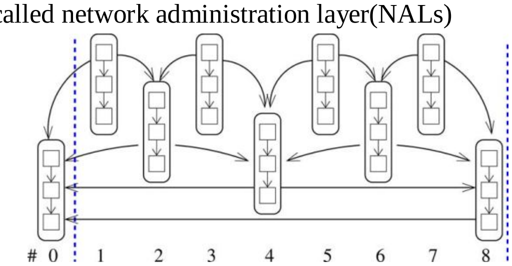

Fig. 2. Dependency among NALUs of H.264/SVC streams. Each square rep- resents an NALU belonging to an MGS layer, and each rounded rectangle rep- resents a video frame.

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We allow for a packet to be sent over at most one access network; this is because efficient link-layer error control mechanisms, such as Forward Error Correction (FEC) and Automatic Repeat Re- quest (ARQ), are widely applied in wireless networks to re- duce packet losses , hence sending an NALU over multiple access networks does not lead to significant improvements on video quality, while it increases the network load.

III. EVALUATION

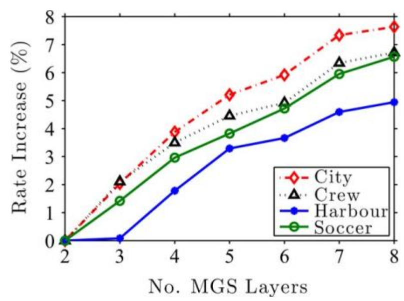

Fig. 3. Rate increase for different numbers of MGS layers.

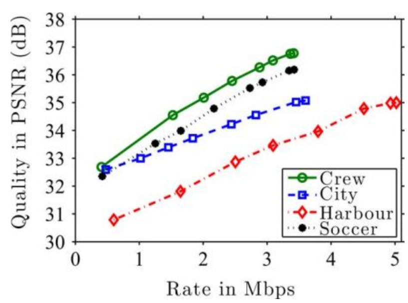

Fig. 4. R-D curves of the scalable video streams.

We use Abing to periodically measure ABR and Stanford University (Stanford, CA). We collect the network traces on weekdays with dozens of hosts on each network generating background traffic. At Deutsche Telekom Laboratories, Abing was run over three access networks: Eth- ernet, 802.11b, and 802.11g. Parts of the network traces were used further details can be found therein.

We consider four 10-s 4CIF (704x576) video sequences: City, Soccer,Crew, and Harbour, encoded as H.264/SVC scalable streams using H.264/SVC baseline profile of JSVM Reference Software. We tested different numbers of MGS layers and found that does not affect coding efficiency substantially. Fig. 3 illustrates that only results in 5%–7.5% higher bit rate than . This shows that additional MGS layers do not lead to severe coding inefficiency. Therefore, each video sequence is encoded into a scalable stream with eight MGS layers for higher flexibility. To illustrate the video characteristics of individual videos, we plot the R-D curves in Fig. 4.

We estimate the video model parameters by extracting and decoding 32 random substreams from each scalable stream

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and measuring the video quality. Knowing which video packets were successfully delivered as well as truncation distortion and drifting distortion, we estimate the model parameters of the video model using standard least-squares fitting in MATLAB. To evaluate the accuracy of the video model, we randomly extracted another 32 substreams from each video stream, com- puted the empirical per-frame video quality, and compared it to the video quality estimated by the video model. Fig. 5 shows the actual and estimated video quality of Soccer and Crew: The proposed video model is quite accurate, and the average abso- lute errors for City, Soccer, Crew, and Harbour were measured to be 2.82%, 1.38%, 0.74%, and 1.65%, respectively.

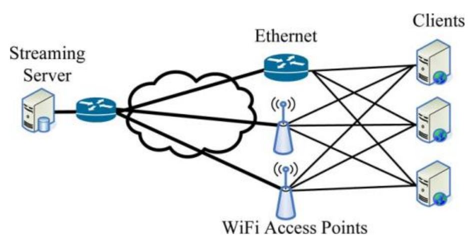

Fig. 5. Simulation setup for multiple user scenarios.

For comparison, we also encode the same video sequences into nonscalable streams using the H.264/AVC baseline profile of JM Reference Software. We configure the H.264/AVC encoder as close to the H.264/SVC configuration as possible, e.g., we use the prediction structure of IPPPPPPP in both cases. Compared to nonscalable streams,10%–50% bit rate increases of scalable streams are acceptable. To be conservative, we use rate control mechanism to generate H.264/AVC streams with 20% lower bit rates than H.264/SVC streams; we found that the H.264/AVC streams still achieve 0.1–1 dB better quality than H.264/SVC streams.

We implemented a multinetwork streaming server in NS-2, which supports the SRDO, PRDO, and HC al- gorithms, implemented as MATLAB subroutines. The HC algorithm uses CVX [2] to numerically solve the convex program given in Remark 2. We report runtime values cor- responding to a 2.8-GHz PC with MATLAB R2010a. For comparison, we also implemented a multinetwork DCCP streaming server based on an open-source DCCP implementation [26] that supports two standard rate control algorithms: TCP-like and TCP-Friendly Rate Control (TFRC). The DCCP streaming server sets up a connection over each access net- work and assigns NALUs to each connection from lower- to higher-quality layers until reaching the rate limit computed by the rate control algorithms. The DCCP streaming servers with TCP-like and TFRC rate control algorithms are referred to as DCCP-TCP and DCCP-TFRC, respectively.

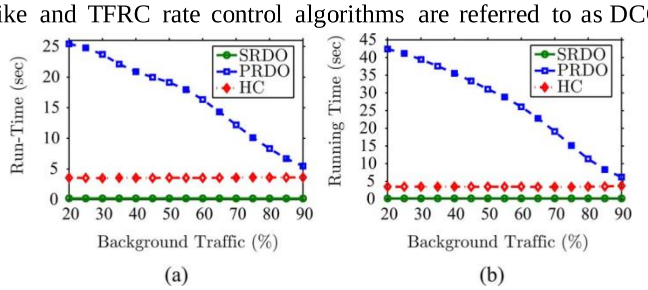

Fig. 6. Runtime under different background traffic loads. Sample results from (a) harbour and (b) Crew.

Fig. 6 presents the runtime of the proposed algorithms for Harbour and Crew; the HC algorithm has an up to 10-fold

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lower runtime as compared to PRDO. SRDO runs fast, less than 200 ms on average, but it results in lower video quality Therefore, we propose using the HC algo- rithm for good performance as well as reasonable runtime. Note that the runtime of HC is constant independent of background traffic since it is a convex program that takes the same time to solve numerically irrespective of background traffic. The run- time of PRDO decreases substantially with background traffic since there are much fewer packets that can be sent before ca- pacity is reached. The same is true for SRDO, but it is not as apparent because SRDO does not perform the time-costly func- tion evaluation (which PRDO does). Lastly, although our proposed proactive algorithms outper- form responsive DCCP-TCP and DCCP-TFRC, we need to point out that DCCP algorithms still have several advantages over the proposed algorithms. First, DCCP algorithms are simple and easy to deploy. Second, DCCP algorithms have very low computational complexity. Third, DCCP works in the considered system architecture (Fig. 1) as well as others, while our proposed algorithms only run on streaming servers.

IV. LIMITATIONS AND FUTURE WORK

This paper considers multihomed, multiple-client video streaming from a server's perspective (Fig. 1). This results in server-driven adaptation solutions, which may incur too much overhead on the server for many clients. Therefore, each server might only be able to serve a small number of clients. While the streaming service providers may deploy multiple streaming servers in a server farm, as well as exploit increased computational power via grid or cloud computing, the centralized nature of our solution could still render the proposed algorithms less efficient in such deployments. For example, probing traffic from multiple servers to infer available bit rate and round-trip time could interfere with each other. To tackle this, we can control the number of decision variables by simplifying the scalable stream structures, hence trading streaming optimality for shorter running time for the case of many users. Trans- forming our current architecture toward client-driven solutions is one of our future tasks. Techniques such as Lagrangian decomposition could be used to develop distributed algorithms for more scalable solutions. The resulting distributed algorithms will be more suitable to client-driven HTTP streaming, such as 3GPP/MPEG DASH, which is getting more and more popular nowadays.

V. CONCLUSION

In this paper, we have addressed various usage scenarios of video streaming from a server to multinetwork clients over heterogeneous access networks. More precisely, we have formally abstracted the problem of joint rate control and stream adaptation as an optimization problem of minimizing the expected distortion of the received videos subject to constraints based on net- work conditions. We have formulated this problem as an integer program for joint rate control and stream adaptation in order to determine, for each client: 1) the streaming rates over individual access networks; 2) the video packets to be transmitted; and 3) the access network each transmitted video packet is sent over, so as to minimize a cost function of the expected distortion at the receiver side. We have proposed using different cost functions to account for service differentiation and fairness among users. We have proposed two heuristic algorithms for packet scheduling, namely SRDO and PRDO. In addition, we have derived convex programming approximations to the randomized packet scheduling problem and have studied the tradeoff between performance and runtime; one of our randomized algorithms (TTC) has a better runtime at the cost of lower performance, while the other one (MC) has better performance at the cost of exponential complexity. We have proposed a hybrid algorithm (HC) that yields good performance for a low number of access networks while being suitable for real-time applications.

We have conducted extensive simulations to compare the performance of HC against SRDO, PRDO, and the rate control algorithms defined in the DCCP standard. The simulation results have shown that the HC algorithm: 1) outperforms the rate control algorithms in the DCCP standard by about 10–15 dB in video quality; 2) reduces average delivery delay by over 90% compared to DCCP; 3) results in an average quality improvement of 4.33 dB versus SRDO, and 1.84 dB versus PRDO, under different background traffic loads; 4) runs efficiently, up to six times faster than PRDO; and 5) indeed provides service differentiation among users.

REFERENCES

1 .M.Megna, "AT&T faces 5,000 percent surge in traffic," Oct. 2009 [Online]. Available: http://www.internetnews.com/mobility/article.php/3843001

2. "CVX: Matlab software for disciplined convex programming," 2009 [Online]. Available: http://www.stanford.edu/~boyd/cvx/

3. K. Fitchard, "T-Mobile's growth focusing on 3G," 2009 [Online]. Available: http://connectedplanetonline.com/wireless/news/t-mobile-3g-growth-0130

4 Cisco Systems, San Jose, CA, "Cisco Visual Networking Index forecast Web site," 2010 [Online]. Available: http://www.cisco.com/go/vni

5 Stanford University, Stanford, CA, "Abing project page," 2004 [On- line]. Available: http://www-iepm.slac.stanford.edu/tools/abing/

6 T. Alpcan, J. Singh, and T. Basar, "Robust rate control for heteroge- neous network access in multihomed environments," IEEE Trans. Mobile Comput., vol. 8, no. 1, pp. 41–51, Jan. 2009.

7 I. Amonou, N. Cammas, S. Kervadec, and S. Pateux, "Optimized rate- distortion extraction with quality layers in the scalable extension of H.264/AVC," IEEE Trans. Circuits Syst. Video Technol., vol. 17, no. 9, pp.

8 D. Bertsekas, Convex Optimization Theory. Belmont, MA: Athena Scientific, 2009.

9 S. Boyd and L. Vandenberghe, Convex Optimization, 1st ed. Cam- bridge, U.K.: Cambridge Univ. Press, 2004.

10 J. Chakareski and B. Girod, "Rate-distortion optimized packet sched- uling and routing for media streaming with path diversity," in Proc. DCC, Snowbird, UT, Mar. 2003, pp. 203–212.

11 P. Chou and Z. Miao, "Rate-distortion optimized streaming of pack- etized media," IEEE Trans.n Multimedia, vol. 8, no. 2, pp. 390–404, Apr. 2006.

12 K. Evensen, T. Kupka, D. Kaspar, P. Halvorsen, and C. Griwodz, "Quality-adaptive scheduling for live streaming over multiple access networks," in Proc. ACM NOSSDAV, Amsterdam, The Netherlands, Jun. 2010, pp. 21–26.

13 N. Freris, C. Hsu, X. Zhu, and J. Singh, "Resource allocation for mul- tihomed scalable video streaming to multiple clients," in Proc. IEEE ISM, Taichung, Taiwan, Dec. 2010, pp. 9–16.

14P. Fuxjager, H. Fischer, I. Gojmerac, and P. Reichl, "Radio resource allocation in urban femto-WiFi convergence scenarios," in Proc. Euro-NF NGI, Paris, France, Jun. 2010, pp. 1–8.

15 D. Gross, J. Shortle, J. Thompson, and C. Harris, Fundamentals of Queueing Theory, 4th ed. Hoboken, NJ: Wiley-Interscience, 2008.