ERCIM "Alain Bensoussan" Fellowship Scientific Report

Fellow: Jari Korhonen
Visited Location : Norwegian University of Science and Technology (NTNU)
Duration of Visit: 12 months, 01/02/2008-31/01/2009

I - Scientific activity

The scientific activities during the fellowship period can be divided into several subtopics. Part of the work carried out is direct continuation of the earlier research work started during the previous term of employment. The research activities can be divided roughly in three different subtopics: a) forward error/erasure correction (FEC) for video streaming applications, b) congestion control in wireless links, based on delivery of partially damaged packets, and c) analysing the perceptual impact of data losses in video streaming. The outcomes of the research activities provide several options for future research.

First, different aspects in FEC coding have been studied and reported in papers [1-2,4]. The major focus has been on unequal erasure protection in video streaming. More specifically, we have studied the performance of FEC codes with different characteristics, such as sparse Reed-Solomon generator matrices (ie. some of the coefficients of the Reed-Solomon generator matrix have been turned into zeros), coding block length and the maximum size for the source media unit. The results show that sparse codes can be used to allow different protection levels in a flexible manner even within short coding blocks [1]. On the other hand, when equal FEC is more desired, the behaviour of the code can be made smoother by using short video media units and spreading them evenly with redundancy units in transport packets [2].

Second, the possibility to alleviate congestion in wireless links by allowing delivery of packets containing bit errors has been studied. Traditionally, congestion control in streaming applications is performed by reducing the media transmission rate, which in turn decreases the media quality. We have studied an alternative strategy, where the system first omits the bit errors in the least vulnerable parts of packets in case of the congestion is partially derived from physical transmission errors in a wireless channel. With this strategy, media quality is impacted by bit errors instead of lower transmission rate. We have found evidence that this mechanism may improve the perceived overall quality, compared to traditional rate control mechanisms [3].

Third, we have analyzed the perceived impact of data unit losses in realistic compressed video streams. Complex spatial and temporal prediction mechanisms make it difficult to estimate the relative perceptual significance for different individual data units. However, this information is vital in order to define the optimal protection levels for different media units, when unequal FEC codes are used to protect the media data. We have experimentally studied the quality degradation caused by losses of individual media units and concluded that in realistic video streams it is not straightforward to evaluate the relative significance directly from the frame type [5]. The results of this study have been used to develop a theoretical framework for evaluating and optimizing different unequal FEC codes for realistic video streams [4].

II- Publication(s) during your fellowship

[1] J. Korhonen, and P. Frossard: Flexible Forward Error Correction Codes with Application to Partial Media Data Recovery. *Signal Processing: Image Communications*. Elsevier. In press.

Abstract. Conventionally, linear block codes designed for packet erasure correction are targeted to recover all the lost source packets per block, when the fraction of lost data is smaller than the redundancy overhead. However, these codes fail to recover any lost packets, if the number of erasures just exceeds the limit for full recovery capability, while it can still be beneficial to recover part of the symbols. In addition, common linear block codes are not well suited for unequal error protection, since different block codes with different rates must be allocated for each priority class separately. These two problems motivate the design of more flexible FEC codes for media streaming applications. We first review the performance of short and long linear block codes. Long block codes generally offer better error correction capabilities, but at the price of higher complexity and larger coding delay. Short block codes can be more appropriate in media streaming applications that require smooth performance degradation when the channel loss rate increases. We study a new class of linear block codes using sparse generator matrices that permit to optimize the performance of short block codes for partial recovery of the lost packets. In addition, the proposed codes are extended to the design of unequal erasure protection solutions. Simulations of practical video streaming scenarios demonstrate that the flexible sparse codes offer a promising solution with interesting error correction capabilities and small variance in the residual loss rate. They typically represent an effective trade-off between short block codes with limited flexibility, and long block codes with delay penalties.

[2] J. Korhonen, and P. Frossard: Error Control for Video Streaming with Small Data Units. 4th International Mobile Multimedia Communications Conference (*MobiMedia'08*), Oulu, Finland, July 2008. Awarded with the best paper award.

Abstract. In multimedia streaming, small errors are typically easier to mask with common error concealment strategies, but small packet size increases the overhead caused by network header information. To reduce the header overhead, large packets are typically favored. In multimedia streaming applications, every packet comprises ideally one individually decodable data unit only. Unfortunately, large packets penalize the error concealment performance at the decoder, which may lead to large and fluctuating distortion. In this paper, we propose an error control mechanism based on efficient packetization of small independent decoding units. Instead of using erasure correction codes to protect packets as such, it gathers several small source data units in each transport packet together with redundancy data units, and the distribution of the units is chosen in order to minimize the distortion at the decoder. The proposed technique has been evaluated by simulating an H.264/AVC video streaming system and comparing the performance against conventional erasure protection scheme involving large data units. The results show that in the presence of packet losses the proposed mechanism provides smoother perceived video quality degradation performance than the conventional packetization and generic forward error correction mechanisms.

[3] J. Korhonen, and A. Perkis: Wireless Congestion Control based on Delivery of Erroneous Packets. IS&T/SPIE Electronic Imaging: Visual Communications & Image Processing (*VCIP'09*), San Jose, USA, January 2009.

Abstract. Traditional mechanisms for congestion control in multimedia streaming systems reduce the data transmission rate when congestion is detected. Unfortunately, decreasing the rate of the media stream also decreases the media quality, but it is the only way to combat

congestion when it is caused by overwhelming traffic that exceeds the capacity of the network. However, if the bottleneck is a wireless link, congestion is often derived from retransmissions caused by bit errors in the radio link. If this is the case, it might be beneficial not to reduce the transmission rate, but allow delivery of packets containing bit errors up to the application layer first. In this scenario, the quality of media will be impacted by bit errors instead of lower coding rate. In this paper, we propose a system concept allowing bit errors in packets in order to relieve congestion. We have built a simulation to compare the performance of the proposed system against traditional congestion control. The results show that the proposed approach can improve the overall performance both by increasing the throughput over the wireless and improving the perceived video quality in terms of *peak signal-to-noise ratio* (PSNR).

[4] J. Korhonen, and A. Perkis: Optimizing Unequal Erasure Protection for Video Streaming. Packet Video Workshop (*PV'09*), Seattle, USA, May 2009. Submitted for review.

Abstract. Several different schemes for unequal error/erasure protection (UEP) in video streaming have been proposed during the past years. However, it is not a trivial task todefine the optimal relative protection levels for different media units (ie. the basic units handled individually by media decoder) in practical streaming applications. In this paper, we use a theoretical packet loss-distortion model toevaluate the performance of different UEP schemes in realistic video streaming scenarios with fixed media data rate and redundancy overhead budget. Our results confirm the observation that equal error/erasure protection (EEP) is desired when the packet loss rate is low compared to the redundancy overhead. The higher the packet loss rate, the more segregating UEP should be used to gain optimal performance. Our results suggest that comparable or even better performances can be obtained by using two protection levels only (protected part and unprotected part) instead of more complex UEP schemes.

[5] J. Korhonen, and A. Perkis: Loss-Distortion Estimation for H.264/AVC Ns. IEEE International Conference on Multimedia and Expo (*ICME'09*), Cancun, Mexico, June 2009. Submitted for review.

Abstract. Even though several unequal erasure protection (UEP) schemes have been proposed for video streaming, it is still a challenge to define the optimal relative protection level for different data units. In this paper, we study loss-distortion modeling in realistic video streaming scenarios. The main observation is that when video streams with complex hierarchical structures and error resilience features are concerned, the relative significance of different units cannot be reliably estimated from mutual dependencies between units as suggested in related research. Therefore, media units can be classified reliably into different perceptual priority classes via analysis by synthesis only. For estimating the co-impact of multiple losses, we propose a simple loss-distortion model. The model can be used for optimizing UEP in practical streaming systems.

III -Attended Seminars, Workshops, and Conferences

The events in which I have participated during my fellowship period are listed below.

1. IEEE International Conferenece on Multimedia and Expo (*ICME'08*), June 23-26, 2008, Hanover, Germany.

2. 4th International Mobile Multimedia Communications Conference (*MobiMedia'08*), July 7-9, 2008, Oulu, Finland.

3. IS&T/SPIE Electronic Imaging: Visual Communications & Image Processing (*VCIP'09*), January 18-22, 2009, San Jose, CA, USA.

IV – Research Exchange Programme (12 month scheme)

First period: 2 weeks at INRIA, Sophia-Antipolis, France, October 6-18, 2008
Project team: PLANÈTE
Contact person: Dr. Thierry Turletti (E-mail: turletti@sophia.inria.fr)

During my visit at INRIA, I participated in the work of PLANÈTE project team. The project focuses on networking research with emphasis on designing, implementing and evaluating Internet protocols and applications. INRIA has an active role in the development of ns-3 network simulator, and I had a fruitful discussion with Mathieu Lacage, who is one of the key developers of the simulator. Ns-3 has the ambitious goal to finally replace older simulator, ns-2, that has currently a very strong position in the academic world.

As a practical part of my visit, I implemented UDP Lite functionality and Gilbert-Elliot bit-error model on the physical layer of the ns-3 network simulator. This work was largely based on the ideas and my earlier work with ns-2, concerning delivery of packets containing bit errors in order to relieve congestion in wireless links. It is supposed that the outcome of this work can be used for research purposes when ns-2 gets gradually replaced by ns-3 in the future.

On the October 15, I gave a seminar presentation about my work regarding congestion control for video streaming over a wireless link. The presentation was covering the ideas and results published in [3] and the topic was also closely linked to the implementation work performed during the visit. After the presentation I received relevant feedback for my work.

Second period: 1 week at CWI, Amsterdam, The Netherlands, November 24-30, 2008
Department: SEN5 (Distributed Multimedia Languages and Infrastructures)
Contact person: Prof. Dick Bulterman (E-mail: Dick.Bulterman@cwi.nl)

During my visit at CWI, I visited the research group at the department of Distributed Multimedia Languages and Infrastructures. The research work of the group focuses on fundamental problems in the fields of creating, encoding, distributing and modelling of digital media content. I had several informal discussions with the researchers at the institute to exchange ideas and find possible areas of future collaboration. The most relevant discussions I had with PhD student Ishan Vaishnavi, whose research interests cover scheduling of multimedia packets in network nodes, fitting closest to my own research interests.

On the November 26, I gave a seminar presentation that was covering my background as a researcher as well as some highlights of my recent work, especially in the field of bit-error resilient video streaming and performance issues in unequal erasure protection. The presentation awakened lively discussion and many good comments and suggestions.