OPTIMIX:

OPTIMISATION OF MULTIMEDIA OVER WIRELESS IP LINKS VIA X-LAYER DESIGN

DELIVERABLE D1.4
INTERMEDIATE REPORT ON DISSEMINATION AND STANDARDISATION ACTIVITIES

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Executive Summary

This deliverable gives a summary of dissemination and standardisation activities done in the first half of the OPTIMIX project.

The dissemination part mainly covers the project web page, publications and participation at conferences, workshops, symposiums, concertation meetings and other ICT project meetings. The web page serves as a central information point for the project. Among others, especially information about the main project objectives and the technical approach followed can be found. Publications were submitted to different conferences (e.g., Globecom, VTC IEEE ICC, ICASSP) and journals (e.g., IEEE TCSVT, IEEE Journal on Selected areas in Communications (JSAC)). This is also reflected in the participation of project members at different conferences, workshops, etc.

Standardisation was done by the industrial partner Siemens. Participation at the standardization bodies ISO/IEC MPEG and ITU-T VCEG and their Joint Video Team (JVT) was in the focus here. The relevant meetings were attended. A proposal on rate-distortion optimisation for the scalable video coding (SVC) extension of H.264/AVC was submitted to and accepted by the JVT for inclusion into the reference software JSVM (Joint Scalable Video Model).

Finally, the industrial partners Thales and Siemens have applied for patents on their technology. Naturally, the number of patents is expected to increase in the course of the project.

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1 Introduction

This deliverable summarizes dissemination and standardization activities of the OPTIMIX project in the first 18 months of the project lifetime. It mainly gives information on

- the project web page (brief review),
- publications in books, journals, conference proceedings, workshops, etc.,
- participation at conferences, concertation meetings, standardization meetings, etc.,
- standardization contributions (mainly in the video domain),
- the generation of IPR, and
- project liaisons.

2 Dissemination activities

2.1 Project web page

The project website can be found at http://www.ict-optimix.eu. It is a Wiki based website to support concurrent editing and versioning. This specific system is powered by the so called MediaWiki engine. The site is administrated and operated by the Hungarian project partner BME. Since the beginning the website has been filled with much useful and interesting content.

A more detailed description of the website can be found in deliverable D1.1 ("Project Web Site and FTP Site Setting Up").

2.2 Books or book chapter contributions

[1] T. Zahariadisi, C. Lamy-Bergot, T. Schierl, K. Gruneberg, L. Celetto, C. Timmerer, "Content Adaptation Issues in the Future Internet", appeared in IOS Press "Towards the Future Internet - A European Research Perspective", 2009, ISBN 978-1-60750-007-0

Abstract—Future Media Internet is envisaged to provide the means to share and distribute (advanced) multimedia content and services with superior quality and striking flexibility, in a trusted and personalized way, improving citizens' quality of life, working conditions, edutainment and safety. Based on work that has taken place in projects ICT SEA and ICT OPTIMIX, and the Media Delivery Platforms Cluster of projects, we try to provide the challenges and the way ahead in the area of content adaptation.

[2]C. Lamy-Bergot and G. Panza."Cross-layer joint optimization of multimedia transmissions over IP based wireless networks". Book chapter by IGI Global Press 2009 (submitted and being published)

Abstract—The traditional approach consisting in separately optimizing each module of a transmission chain has shown limitations in the case of wireless communications where delay, power limitation and error-prone channels are experienced. This is why modern designers focus on a more integrated strategy to establish the heterogeneous 21st century networks, such as 3G (i.e. UMTS) system and its evolutions (i.e. Beyond 3G or 4G like LTE or future 5G systems). Indeed, it was shown in several studies that optimal allocation of user and system resources could be effectively achieved with the co-operative optimization of communication system components. In this chapter, an innovative Joint-Source Channel Coding and Decoding (JSCC/D) system is described and its performance over an IPv6-based Network infrastructure is assessed. A particular focus is put on the application controller, the key component to realize the adaptation strategies. Conclusions and considerations about the system implementation are also proposed, and the interest of a possible extension to a point-to-multipoint scenario is explained.

2.3 Publications in journals and magazines

[3] X. Li, P. Amon, A. Hutter, A. Kaup, "Analysis of Inter-Layer Prediction and Hierarchical Prediction Structures in Scalable Video Coding", submitted to IEEE Transactions on Broadcasting (TBC).

Abstract—Abstract—In scalable video coding (SVC), there are two important coding tools, i.e., inter-layer prediction and hierarchical prediction structures, which greatly affects the overall coding efficiency of SVC encoder. In this paper, the performance of these two tools is thoroughly investigated by simulations and model based emulations. First, it is shown that the efficiency of interlayer prediction is highly dependent on the correlation between layers. However, to cover a certain scalable range while keeping a high inter-layer correlation normally results in a relatively big layer overhead, which will finally degrade the overall performance. Therefore, a proper tradeoff between the coding efficiency and scalability has to be established. Second, among the three inter-layer prediction methods, inter-layer residual prediction contributes most gains in simulations. Nevertheless, all of them are comparatively less efficient in spatial scalability due to the lowpass filtering effect of the upsampling process. Third, the hierarchical prediction structures with a proper quantization parameter cascading is a quite efficient coding scheme, where the error propagation within a group of picture is discovered as a key factor which determines the performance of such a scheme. Based on these analyses, possible approaches for further improving the efficiency of the two coding tools are presented, i.e., frequency sub-band prediction and multi-layer joint rate-distortion optimization will improve the performance of inter-layer prediction while an adaptive quantization parameter cascading algorithm based on the factor of error propagation will contribute gains to hierarchical prediction structures.

[4] Kyungchun Lee and L. Hanzo "MIMO-Assisted Hard Versus Soft Decoding-and-Forwarding for Network Coding Aided Relaying Systems", IEEE Transactions on Wireless Communications, Volume 8, Issue 1, January 2009, Pages:376-385.

Abstract—This paper proposes two types of new decoding algorithms for a network coding aided relaying (NCR) system, which adopts multiple antennas at both the transmitter and receiver. In the NCR system, the relay station (RS) decodes the data received from both the base station (BS) as well as from the mobile station (MS) and combines the decoded signals into a single data stream before forwarding it to both. In this paper, we consider the realistic scenario of encountering decoding errors at the RS, which results in erroneous forwarded data. Under this assumption, we derive decoding algorithms for both the BS and the MS in order to reduce the deleterious effects of imperfect decoding at the RS. We first propose a decoding algorithm for a hard decision based forwarding (HDF) system. Then, for the sake of achieving further performance improvements, we also employ soft decision forwarding (SDF) and propose a novel error model, which divides the error pattern into two components: hard and soft errors. Given this error model, we then modify the HDF decoder for employment in SDF systems. We also derive estimation algorithms for their parameters that are required for the efficient operation of the proposed decoders. Our simulation results show that the proposed algorithms provide substantial performance improvements in terms of the attainable packet error rate as a benefit of our more accurate error model.

[5] O. Alamri, S. X. Ng, F. Guo, S. Zummo and L. Hanzo "Non-Binary LDPC-Coded Sphere-Packed Transmit Diversity", IEEE Transactions on Vehicular Technology, Volume 57, Issue 5, September 2008, Pages:3200-3205.

Abstract—A recently proposed space–time block-coding (STBC) signal construction method that combines orthogonal design with sphere packing (SP) (referred to here as STBC-SP) has shown useful performance improvements over Alamouti's conventional orthogonal design. In this paper, we propose a purely symbol-based, low-density parity-check (LDPC)-coded scheme, demonstrating that the performance of STBC-SP systems can further be improved by concatenating SP-aided modulation with nonbinary LDPC and by performing symbol-based turbo detection between the nonbinary LDPC decoder and a rate-1 nonbinary inner precoder. We also investigate the convergence behavior of this symbol-based concatenated scheme with the aid of novel nonbinary extrinsic information transfer (EXIT) charts. Finally, we demonstrate that in the investigated scenarios, it requires 1–2 dB lower power in comparison with the equivalent effectivethroughput 0.5-, 0.75-, and 1-bit/symbol systems employing bit-based turbo detection.

[6] O. Alamri, J. Wang, S. X. Ng, L.-L. Yang and L. Hanzo "Near-Capacity Three-Stage Turbo Detection of Irregular Convolutional Coded Joint Sphere-Packing Modulation and Space-Time Coding", IEEE Transactions on Communications, Volume 57, Issue 5, May 2009, Pages 1486-1495.

Abstract—Conventional two-stage turbo-detected schemes typically suffer from a Bit Error Rate (BER) floor, preventing them from achieving infinitesimally low BER values, especially, when the inner coding stage is of non-recursive nature. We circumvent this deficiency by proposing a three-stage turbo-detected Sphere Packing (SP) aided Space-Time Block Coding (STBC) STBC-SP scheme, where a rate-1 recursive inner precoder is employed to avoid having a BER floor. The convergence behaviour of this serially concatenated scheme is investigated with the aid of 3D Extrinsic Information Transfer (EXIT) Charts. Furthermore, the capacity of the STBC-SP scheme is determined and an algorithm is proposed for calculating a tighter upper bound on the maximum achievable bandwidth efficiency, based on the EXIT charts of the STBCSP demapper. The proposed three-stage turbo-detected scheme operates within about 1.0 dB of the capacity and within 0.5 dB of the maximum achievable bandwidth efficiency limit.

[7] M. El-Hajjar, O. Alamri, R.G. Maunder and L. Hanzo "Layered Steered Space-Time Spreading Aided Generalised MC DS-CDMA", to appear in IEEE Transactions on Vehicular Technology.

Abstract—We present a novel tri-functional Multiple-Input Multiple-Output (MIMO) scheme, that intrinsically amalgamates Space-Time Spreading (STS) for achieving a diversity gain, Vertical Bell Labs Layered Space-Time (V-BLAST) scheme for attaining multiplexing gain in the context of generalised MultiCarrier Direct Sequence Code Division Multiple Access (MC DS-CDMA) as well as beamforming. Furthermore, the proposed system employs both time and frequency domain spreading for increasing the number of users, which is also combined with a user-grouping technique for reducing the effects of multi-user interference. Further system performance improvements can be attained by serially concatenating our proposed scheme with an outer code amalgamated with a unity-rate code for the sake of improving the convergence behaviour of the proposed system, which is evaluated with the aid of EXIT charts. We also propose a novel Logarithmic Likelihood Ratio (LLR) postprocessing technique for improving the iteratively detected system's performance. Explicitly, after I=10 decoding iterations and employing an interleaver depth of Dint = 160, 000 bits, the proposed system supporting K=1 user attains a BER below 10−5 for Eb/N0 values in excess of −2.8 dB. On the other hand, at a BER of 10−5 our K=8-user system having a normalised system throughput of 4 bits/sec/Hz requires an Eb/N0 of only about 0.45 dB higher than the single-user benchmark system employing a time domain spreading factor of Ne=4 and V =4 subcarriers.

[8] E. Piri and K. Pentikousis, "IEEE 802.21: Media-Independent Handover Services", The Internet Protocol Journal, Volume 12, Number 2, June 2009, pages 7-27.

Abstract—Popular mobile devices now ship with several integrated wired and wireless network interfaces. Personal Digital Assistants (PDAs) and smartphones, for example, are increasingly supporting communications through both cellular technologies and Wireless LANs (WLANs); laptops typically come with built-in Ethernet, Wi-Fi, and Bluetooth. As multiaccess devices proliferate, we move closer to a network environment that is often referred to as "beyond 3G" (B3G). Key success factors for cellular third-generation (3G) communications include better cell capacities, increased *data rates, transparent mobility within large geographical areas, and global reachability. For B3G, the next frontier lies beyond transparent mobile connections within the same access technology because users will expect to be globally reachable anytime, anywhere, and remain "always best-connected" (ABC). In order to select the best possible connectivity option (anytime, anywhere), mobile devices and access networks will have to work together in order to enable users to take full advantage of all available options. The IEEE 802.21 working group (see www.ieee802.org/21) recently finalized the first standard for dealing with handovers in heterogeneous networks, also called Media-Independent Handovers (MIH). The standard is expected to allow mobile users (and operators) to take full advantage of overlapping and diverse access networks. It provides a framework for efficiently discovering networks in range and executing intelligent heterogeneous handovers, based on their respective capabilities and current link conditions. This article aims to serve as a primer for those interested in the IEEE 802.21 standard. After introducing the IEEE 802.21 reference model, we present the MIH services and provide illustrative use cases that highlight the benefits of employing the Media-Independent Handover Services standard in heterogeneous networks.*

[9] R. S. H. Istepanian, N. Philip, M.G. Martini, *"Medical QoS Provision based on Reinforcement Learning in Ultrasound Streaming over 3.5G Wireless Systems"* in 'IEEE Journal on Selected Areas in Communications', vol, 27, no. 4, May 2009 , pp. 566-574.

Abstract—The design of an efficient mobile healthcare system using 3.5G and 4G wireless networks is a challenging problem, in particular for bandwidth demanding telemedical applications. In this paper we propose a novel multi-objective rate-control mechanism for the optimized delivery of diagnostically acceptable ultrasound video images over 3.5G wireless networks. The rate control policy of wireless video streaming is regarded as a discrete-time Markov Decision Process (MDP) problem and the application-layer controller is based on a reinforcement learning algorithm known as Q-learning, tailored for such a specific purpose. The algorithm exploits information on the video quality (such as image quality index and frame rate) and on the buffer occupancy. The performance of the proposed algorithm has been evaluated via both simulations and experimental studies for H.264 medical video transmission over HSDPA. The proposed rate control algorithm provides dynamic adaptation achieving performance improvements with respect to the standard H.264 rate control methodology.

[10] Arpad Huszak, Sandor Imre, "*Multipath Video Streaming Using GRA Network Ordering Algorithm without Rank Inconsistency*", Journal on Information & Communications Technologies, Special issue on Tools, Modelling Techniques and Analysis Aspects of Heterrogeneous Networks, "Research, Development and Application on Information and Telecommunication Technology", Volume E-1, Number 1 (5), pp. 43—58, 6 August 2009

Abstract—Simultaneous connection to several networks through multiple interfaces is possible with today's mobile terminals. In order to efficiently utilize the interfaces' capabilities and increase video frame not only lead to reduced quality of the frame but the quality of the streamed video, multipath streaming can be used. Resource intensive applications can deliver high bitrate streams over multiple paths by cumulating the available bandwidth of the different subpaths. In this paper we propose a multipath streaming method that chooses a set of paths maximizing the overall quality at the client. While the available paths have different bandwidth, delay and loss probability constrains, the packet distributor must take the video packet importance and the dependencies between packets into account. In order to efficiently distribute the packets, the link must be ordered based on the network attributes. Grey Relational Analysis (GRA) is a promising algorithmic approach that can realize dynamic interface ordering with multiple alternatives (interfaces) and attributes (network parameters). However similarly to some other decision methods, GRA also suffers from rank reversal phenomenon. Transmitting the reference video frames on the most reliable links will decrease the loss probability of important data packets and increase the measured video quality. The change of link order can lead to frequent handovers causing the degradation of the observed video quality.

[11] Andrea Conti, Velio Tralli and Marco Chiani, "Pragmatic Space-Time Codes for Cooperative Relaying in Block Fading Channels", EURASIP Journal on Advances in Signal Processing, vol. 2008, Article ID 872151, 11 pages, 2008.

Abstract—We address the problem of construction of space-time codes for cooperative communications in block fading channels. More precisely, we consider a pragmatic approach based on the concatenation of convolutional codes and BPSK/QPSK modulation to obtain cooperative codes for relay networks, for which we derive the pairwise error probability, an asymptotic bound for frame error probability, and a design criterion to optimize both diversity and coding gain. Based on this framework, we set up a code search procedure to obtain a set of good pragmatic space-time codes (P-STCs) with overlay construction, suitable for cooperative communication with a variable number of relays in quasistatic channel, which outperform in terms of coding gain other space-time codes (STCs) proposed in the literature. We also find that, despite the fact that the implementation of pragmatic space-time codes requires standard convolutional encoders and Viterbi decoders with suitable generators and branch metric, thus having low complexity, they perform quite well in block fading channels, including quasistatic channel, even with a low number of states and relays.

[12] E. Paolini, G. Liva, B. Matuz and M. Chiani, "Generalized IRA Erasure Correcting Codes for Hybrid Iterative/Maximum Likelihood Decoding", IEEE Communications Letters, vol. 12, no.6, pp.450-452, Jun. 2008

Abstract—The design of low-density parity-check (LDPC) codes under hybrid iterative / maximum likelihood decoding is addressed for the binary erasure channel (BEC). Specifically, we focus on generalized irregular repeat-accumulate (GeIRA) codes, which offer both efficient encoding and design flexibility. We show that properly designed GeIRA codes tightly approach the performance of an ideal maximum distance separable (MDS) code, even for short block sizes. For example, our (2048, 1024) code reaches a codeword error rate of 10E−5 at channel erasure probability e= 0.450, where an ideal (2048, 1024) MDS code would reach the same error rate at e = 0.453.

[13] E. Paolini and M. Chiani, "Construction of near-optimum burst erasure correcting low-density parity-check codes", IEEE Trans. Commun., vol. 57, no.5, pp.1320-1328, May 2009

Abstract—In this paper, a simple and effective tool for the design of low-density parity-check (LDPC) codes for iterative correction of bursts of erasures is presented. The design method consists of starting from the parity-check matrix of an LDPC code and developing an optimized parity-check matrix, with the same performance over the memoryless erasure channel, and suitable also for the iterative correction of single erasure bursts. The parity-check matrix optimization is performed by an algorithm called pivot searching and swapping (PSS) algorithm. It executes permutations of carefully chosen columns of the parity-check matrix, after a local analysis of particular variable nodes called stopping set pivots. This algorithm can be in principle applied to any LDPC code. If the input parity-check matrix is designed to achieve a good performance over the memoryless erasure channel, then the code obtained after the application of the algorithm provides a good joint correction of independent erasures and single erasure bursts. Numerical results are provided in order to show the algorithm effectiveness when applied to different categories of LDPC codes.

[14] E. Paolini, M. Chiani and M. Fossorier, "Doubly-generalized LDPC codes: Stability bound over the BEC", IEEE Trans. Inform. Theory, vol. 55, no.3, pp.1027-1046, Mar. 2009

Abstract—The iterative decoding threshold of low-density parity-check (LDPC) codes over the binary erasure channel (BEC) fulfills an upper bound depending only on the variable and check nodes with minimum distance 2. This bound is a consequence of the stability condition, and is here referred to as stability bound. In this paper, a stability bound over the BEC is developed for doubly-generalized LDPC codes, where variable and check nodes can be generic linear block codes, assuming maximum a posteriori erasure correction at each node. It is proved that also in this generalized context the bound depends only on the variable and check component codes with minimum distance 2. A condition is also developed, namely, the derivative matching condition, under which the bound is achieved with equality. The stability bound leads to consider single parity-check codes used as variable nodes as an appealing option to overcome common problems created by generalized check nodes.

[15] A.Conti, D.Panchenko, S.Sidenko and V.Tralli, "Log-Concavity Property of the Error Probability with Application to Local Bounds for Wireless Communications" IEEE Trans. Inform. Theory, vol. 55, no.6, pp.2766-2775, Jun. 2009

Abstract—A clear understanding of the behavior of error probability (EP) as a function of signal-to-noise ratio (SNR) and other system parameters is fundamental for assessing the design of digital wireless communication systems. We propose an analytical framework based on the log-concavity property of the EP which we prove for a wide family of multidimensional modulation formats in the presence of Gaussian disturbances and fading. Based on this property, we construct a class of local bounds for the EP that improve known generic bounds in a given region of the SNR and are invertible, as well as easily tractable for further analysis. This concept is motivated by the fact that communication systems often operate with performance in a certain region of interest (ROI) and, thus, it may be advantageous to have tighter bounds within this region instead of generic bounds valid for all SNRs. We present a possible application of these local bounds, but their relevance is beyond the example made in this paper.

2.4 Publications at conferences, symposiums and workshops

[16] Catherine Lamy-Bergot, Maria G. Martini, Pierre Hammes, Peter Amon, Janne Vehkapera, Gianmarco Panza, Lajos Hanzo, Marco Chiani, Gabor Jeney, Gabor Feher and David Tarrant, "Optimisation of Multimedia over wireless IP links via X-layer design," in Proceedings of the 2nd European Symposium on Mobile Media Delivery, Oulu, Finland, July 2008.

Abstract—Following the path opened by FP6 IST PHOENIX project which was shown allowing an optimised allocation of resources for multimedia transmission over wired/wireless links in a joint source-channel coding approach for unicast transmissions, OPTIMIX project proposes to study innovative solutions enabling enhanced video streaming in a point to multi-point context for an IP based wireless heterogeneous system, based on cross layer adaptation of the whole transmission chain. Expected applications for such improvements are numerous in a time where users demand to have at their disposal on the move the same services they are already experiencing since a few years at home or in their offices. Typically, visiophony, video on demand on the move, access to streaming Internet websites, access to mobile *personal television, location based services for educational, health, transport and environment purposes, virtual meetings or security professional applications are envisioned applications for the European end-user.*

[17] C. Lamy–Bergot, R. Fracchia, J. Vehkaperä, T. Sutinen, E. Piri, M. Mazzotti, G. Panza, G. Feher, G. Jeney, and P. Amon, "Optimisation of Multimedia over wireless IP links via X-layer design: an end-to-end transmission chain simulator," accepted to 3rd European Symposium on Mobile Media Delivery, London, UK, September 2009.

Abstract—End-to-end optimised Quality of Service (QoS) and its specific declination for multimedia applications with the enduser Perceived Quality of Service (PQoS) is a topic that is more and more discussed in the literature. Many different techniques and approaches have been proposed, which are in general focusing on specific weak technical aspects of the transmission chain in the considered scenario. The end-to-end optimisation from a system point of view, i.e., to be transparently integrated in existing legacy systems and not perturbate their operation, is more complex and its practical realisation is yet to be achieved. In this paper, we propose an architecture set-up within the ICT FP7 OPTIMIX project to study innovative solutions enabling enhanced multimedia streaming in a point to multi-point context for an IP based wireless heterogeneous system, based on cross layer adaptation of the whole transmission chain. The corresponding simulation chain architecture is detailed with the description of the existing and/or future features of each module.

[18] Xiang Li, Peter Amon, Andreas Hutter, André Kaup, "One-pass multi-layer rate-distortion optimization for quality scalable video coding", in Proceedings of IEEE International Conference of Acoustics, Speech, and Signal Processing 2009 (ICASSP 2009), Taipei, Taiwan, April 2009.

Abstract—In this paper, a one-pass multi-layer rate-distortion optimization algorithm is proposed for quality scalable video coding. To improve the overall coding efficiency, the MB mode in the base layer is selected not only based on its rate-distortion performance relative to this layer but also according to its impact on the enhancement layer. Moreover, the optimization module for residues is also improved to benefit inter-layer prediction. Simulations show that the proposed algorithm outperforms the most recent SVC reference software. For eight test sequences, a gain of 0.35 dB on average and 0.75 dB at maximum is achieved at a cost of less than 8% increase of the total coding time.

[19] Xiang Li, Peter Amon, Andreas Hutter, André Kaup, "Lagrange multiplier selection for rate-distortion optimization in SVC", in Proceedings of Picture Coding Symposium 2009 (PCS 2009), Chicago, IL, USA, May 2009.

Abstract—The Lagrangian multiplier based rate-distortion optimization (RDO) has been widely employed in single layer video coding. During the development of scalable video coding (SVC) extension of H.264/AVC, it was directly applied in a multilayer scenario. However, such an application is not very efficient since the correlation between layers is not considered in the Lagrange multiplier selection. To improve the overall performance, in this paper a new selection algorithm is presented for RDO in SVC. Simulations show that the proposed method outperforms the recent SVC reference software. With a tiny computational cost, average gains of 0.22 dB and 0.35 dB were achieved in the tests of four-layer quality scalability and three-layer spatial scalability, respectively.

[20] Xiang Li, Peter Amon, Andreas Hutter, André Kaup, "Model Based Analysis for Quantization Parameter Cascading in Hierarchical Video Coding", in Proceedings of IEEE International Conference on Image Processing (ICIP 2009), Cairo, Egypt, November 2009.

Abstract—Originally, the hierarchical coding structure was proposed to achieve temporal scalability. Soon after, it was realized that with a proper quantization parameter cascading (QPC) scheme the general performance can be significantly improved by hierarchical coding. However, the theory behind the gain has not been explored so far. In this paper, the QPC in hierarchical coding is investigated by model based emulations. From the analysis, it is noticed that a parameter β*, which represents the error propagation in a group of pictures greatly affects the performance of QPC in hierarchical coding. Based on* β*, a simple adaptive QPC algorithm is designed. Simulations verify the efficiency of this algorithm: a gain up to 0.5 dB is obtained over the most recent SVC reference software.*

[21] G. Panza, C. Lamy-Bergot, A. Rotondi and L.Fratta, "An Architectural Analysis and Evaluation of a JSCC/D System on 4G Networks". ICT Mobile Summit 2008

Abstract—Foreseen as an effective transparent interconnection of heterogeneous, wired and wireless, networks with critical requirements on bandwidth, 4G telecommunication infrastructures are a challenge for the design of transmission optimisation. In this paper, the FP6 IST PHOENIX project system, which was shown allowing an optimised allocation of resources for multimedia transmission over wired/wireless links is presented, and its architectural choices are analysed, with a particular focus on the signalling used for joint source channel coding, and the optimisation modules called joint controllers. The analysis of the achieved performance is done with respect to three critical issues: cost of the control/signalling overhead, joint controller at application level reaction time and effect of loss or delay of feedback information. The goal of the study is to assess the practical feasibility and effectiveness of *the original PHOENIX approach designed to maximize the end-user quality in 4G networks scenarios comprising UMTS and WiMAX technologies.*

[22] C. Lamy-Bergot, G. Panza, A. Rotondi and L. Fratta, "Analysis and Optimization of a JSCC/D System on 4G Networks", in Proceedings of the *IEEE International Symposium on Spread Spectrum Techniques and Applications (ISSSTA'08)*, Bologna, Italy, August 2008.

Abstract—Foreseen as an effective transparent interconnection of heterogeneous, wired and wireless, networks with critical requirements on bandwidth, 4G telecommunication infrastructures are a challenge for the design of transmission optimization. In this paper, the IST FP6 PHOENIX project system, which was shown allowing an optimized allocation of resources for multimedia transmission over wired/wireless links is presented, and its architectural choices are analyzed, with a particular focus on the signalling used for joint source channel coding, and the optimization modules called joint controllers. The analysis and optimization of the achieved performance is done with respect to four critical issues: cost of the control/signalling overhead, reaction time of joint controller placed at application level, effect of loss or delay of feedback information and impact of crossing multiple wireless hops. The goal of the study is to assess the practical feasibility and effectiveness of the original PHOENIX approach, while maximizing the end-user quality, in 4G networks scenarios comprising UMTS and WiMAX technologies.

[23] G. Panza, G. Bigini, M. Prudente and P. Belotti, "Addressing robust Next-Generation Networks". IEEE NETWORKS 2008

Abstract—Next-Generation Networks (NGNs) employ the Internet Protocol (IP) over a wide variety of packet-switching technologies, which often lack in fault resilience enabling features. An overlay MPLS infrastructure with its fast-reroute mechanisms can be deployed to overcome such an issue. Addressing NGNs robust to single link and node failures, an off-line method to effectively calculate working and recovery paths for highly demanding services, is proposed and analyzed. The strength of our work is the ability to address two recovery techniques in a very simple manner, by formulating an Integer Linear Programming (ILP) problem, optimizing the bandwidth allocation while limiting the recovery time.

[24] G. Panza, A. Capone, D. Pinarello and P. Belotti. "Engineering Robust Next-Generation Networks". IEEE IFIP IM2009-BCN

Abstract—Next-Generation Networks (NGNs) employ the Internet Protocol (IP) over a wide variety of packet-switching technologies, which often lack in fault resilience enabling features. An overlay MPLS infrastructure with its fast-reroute mechanisms can be deployed to overcome such an issue. Addressing NGNs robust to single link and node failures, an off-line method to effectively calculate working and recovery paths for highly demanding services, is proposed and analyzed. The strength of our work is the ability to address two recovery techniques in a very simple manner, by formulating an Integer Linear Programming (ILP) problem, optimizing either the overall switching delay experienced by the user in case of failure or the bandwidth allocation thanks to a shared protection, while limiting the recovery time to some tens of ms as in SONET/SDH networks.

[25] G. Panza, A. Capone, D. Pinarello and P. Belotti. "Robustness in Next-Generation Networks". ICT Mobile Summit 2009 (awarded as runner up paper)

Abstract—Next-Generation Networks (NGNs) employ the Internet Protocol (IP) over a wide variety of packet-switching technologies, which often lack in fault resilience enabling features. An overlay MPLS infrastructure with its fast-reroute mechanisms can be deployed to overcome such an issue. Addressing NGNs robust to single link and node failures, an off-line method to effectively calculate working and recovery paths for highly demanding services, is proposed and analyzed. The strength of our work is the ability to address two recovery techniques in a very simple manner, by formulating an Integer Linear Programming (ILP) problem, optimizing either the overall switching delay experienced by the user in case of failure or the bandwidth allocation, thanks to a shared protection, while limiting the recovery time to some tens of milliseconds as in SONET/SDH networks.

[26] S. X. Ng, Y. Li and L. Hanzo, "Distributed Turbo Trellis Coded Modulation for Cooperative Communications", Proceedings of IEEE ICC 2009, Dresden, Germany, 14 - 18 June 2009.

Abstract—In this contribution, we propose a Distributed Turbo Trellis Coded Modulation (DTTCM) scheme for cooperative communications. The DTTCM scheme is designed based on its decoding convergence with the aid of non-binary Extrinsic Information Transfer (EXIT) charts. The source node transmits TTCM symbols to both the relay and the destination nodes during the first transmission period. The relay performs TTCM decoding and re-encodes the information bits using a Recursive Systematic Convolutional (RSC) code regardless whether the relay can decode correctly or not. Only the parity bits are transmitted from the relay node to the destination node during the second transmission period. The resultant symbols transmitted from the source and relay nodes can be viewed as the coded symbols of a three-component parallel-concatenated TTCM scheme. At the destination node, a novel three-component TTCM decoding is performed. It is shown that the performance of the DTTCM matches exactly the EXIT chart analysis. It also performs very closely to its idealised counterpart that assumes perfect decoding at the relay.

[27] Lingkun Kong, S. X. Ng and L. Hanzo, "Near-Capacity Three-Stage Downlink Iteratively Decoded Generalized Layered Space-Time Coding with Low Complexity", IEEE GLOBECOM 2008, New Orleans, LO, 30 Nov.-4 Dec.2008.

Abstract—This paper presents a low complexity iteratively detected space-time transmission architecture based on Generalized Layered Space-Time (GLST) codes and IRregular Convolutional Codes (IRCCs). The GLST combines the benefits of the Vertical Bell-labs LAyered Space-Time (V-BLAST) scheme and Space- Time Coding (STC). The GLST is serially concatenated with a Unity-Rate Code (URC) and an IRCC which are used to facilitate near-capacity operation with the aid of an Extrinsic Information Transfer (EXIT) chart based design. Reducedcomplexity iterative Successive Interference Cancellation (SIC) is employed in the GLST decoder, instead of the significantly more complex Maximum Likelihood (ML) detection. For the sake of approaching the maximum achievable rate, iterative decoding is invoked to achieve decoding convergence by exchanging extrinsic information across the three serial component decoders. Finally, it is shown that the SIC-based iteratively detected IRCC-URCGLST system is capable of providing a trade-off between the affordable computational complexity and the system throughput.

[28] M. F. U. Butt, R. A. Riaz, S. X. Ng and L. Hanzo, "Near-Capacity Iteratively Decoded Binary Self-Concatenated Code Design Using EXIT Charts", IEEE GLOBECOM 2008, New Orleans, LO, 30 Nov.-4 Dec.2008.

Abstract—In this treatise Extrinsic Information Transfer (EXIT) charts are used to design binary Self-Concatenated Convolutional Codes employing Iterative Decoding (SECCC-ID) for communicating over both uncorrelated Rayleigh fading and Additive White Gaussian Noise (AWGN) channels. Recursive Systematic Convolutional (RSC) codes are selected as constituent codes, an interleaver is used for randomising the extrinsic information exchange of the constituent codes, while a puncturer assists us in increasing the achievable bandwidth efficiency. At the receiver, self-iterative decoding is invoked for exchanging extrinsic information between the hypothetical decoder components. The convergence behaviour of the decoder is analysed with the aid of bit-based EXIT charts. Finally, we propose an attractive system configuration, which is capable of operating within about 1 dB of the information-theoretic limits.

[29] W. Liu, S. X. Ng and L. Hanzo, "Multicell Cooperation Based SVD Assisted Multi-User MIMO Transmission" IEEE VTC Spring 2009, Barcelona, Spain, 26-29 April 2009

Abstract—In this treatise, we investigated the application of singular value decomposion (SVD) assisted multiuser transmission in a multicell scenario. The SVD based scheme is capable of completely removing the cochannel interference, similarly to the classic zero forcing (ZF) based and block diagonalization (BD) aided schemes. Two different power allocation schemes are investigated for both SVD, ZF and BD based multicell transmssion. The SVD scheme achieves a suboptimal performance, but at a reduced complexity. Nonetheless, it always outperforms the ZF based scheme due to the joint reception of the transmitted symbols.

[30] Chun-Yi Wei, Du Yang, Lie-Liang Yang and Lajos Hanzo, "Iterative Detection Aided DL SDMA Systems Using Quantized Channel Impulse Response," on VTC Spring 2009, pp. 1-5.

Abstract—An iterative detection aided Down-Link (DL) Space Division Multiple Access (SDMA) system is proposed. The Base Station (BS) requires DL Channel State Information (CSI) estimated by and fed back from the Mobile Station (MS) for separating the DL signals to be transmitted to the MSs from the BS. We investigate the achievable system performance subject to finite-accuracy CSI feedback. The achievable receiver performance is investigated with the aid of Extrinsic Information Transfer (EXIT) Charts. Furthermore, a novel concept, referred to as the EXIT-Chart Optimized CSI Quantization (ECO-CQ) is proposed. The ECO-CQ scheme assists the system in maintaining the lowest possible CSI feedback overhead, while ensuring that an open EXIT-tunnel is still attainable, which implies maintaining an infinitesimally low BER. Furthermore, we demonstrate that the proposed ECO-CQ may reduce the normalized feedback overhead compared to the conventional CQ. For instance, the ECO-CQ aided iterative DL-SDMA system using an average of q = 2.7 quantization bits per CIR coefficient achieves a 10% normalized overhead reduction at Eb/N0 = 5dB, compared to the conventional CQ aided benchmarker system.

[31] Ahmed, S. Lie-Liang Yang and Hanzo, L.,"Soft Metrics and EXIT Chart Analysis of Noncoherent MFSK with Diversity Reception," on VTC Spring 2009, pages 1 – 5.

Abstract—A convolutionally encoded noncoherent M-ary Frequency Shift Keying (MFSK) modulation scheme using diversity reception is considered. A novel technique of generating soft information for Soft Decision Decoding (SDD) of noncoherent MFSK is derived assuming various channel models. Furthermore, the convergence behavior of the MFSK receiver is investigated using EXtrinsic Information Transfer (EXIT) charts. It is demonstrated that the expressions derived for SDD of the MFSK scheme are useful for exploiting the beneficial effects of diversity. We also demonstrate that EXIT chart analysis provides accurate insights into the system's iterative decoding behavior.

[32] Shinya Sugiura_, Nan Wu and Lajos Hanzo, "Improved Markov Chain MBER Detection for Steered Linear ispersion Coded MIMO Systems," on VTC Spring 2009, pp.1-5.

Abstract—In this paper, we present an iterative Markov Chain Minimum Bit Error Rate (MC-MBER) detection aided steered Linear Dispersion Coded (LDC) structure, which amalgamates three different Multiple-Input Multiple-Output

(MIMO) functions, namely Space-Time Coding (STC), Spatial Division Multiplexing (SDM) and beamforming. Furthermore, the concept of using a novel a priori Log Likelihood Ratio (LLR) threshold based technique is invoked for the sake of further reducing the computational complexity imposed. Both the EXtrinsic Information Transfer (EXIT) chart analysis and our BER performance results demonstrate that the achievable performance is substantially improved upon increasing the number of iterations I. At BER = 1×10−4, the required Eb/N0 value is about 2.4 dB apart from that of the maximum achievable rate in conjunction with I = 10 in a rank-deficient system transmitting Q = 6 QPSK modulated streams in T = 2 symbol durations with the aid of N = 2 receive antennas and M = 2 transmit antenna arrays, where each of the arrays is composed of L = 2 antenna elements. It is demonstrated that the computational complexity of the a priori LLR threshold assisted MC-MBER detector is reduced by 38% at BER = 1.4×10−5 in comparison to that of the MC-MBER detector dispensing with the thresholding technique.

[33] R. G. Maunder and L. Hanzo, "Concatenated Irregular Variable Length Coding and Irregular Unity Rate Coding," VTC Spring 2009, pp. 1-5.

Abstract—In this contribution we propose the novel serial concatenation of Irregular Variable Length Coding (IrVLC) and Irregular Unity Rate Coding (IrURC), where we matched the corresponding EXtrinsic Information Transfer (EXIT) functions to each other. This approach facilitates a higher degree of design freedom than matching the EXIT function of an irregular codec to that of a regular codec. As a result, a narrow EXIT chart tunnel can be created, facilitating operation at Eb/N0 values that are closer to the channel's capacity bound. The computational complexity and Bit Error Ratio (BER) performance of our IrVLC-IrURC scheme is favourable in comparison to the bench markers that replace either one or both of the irregular codecs by the equivalent-rate regular codec.

[34] R. A. Riaz, M. F. U. Butt, R. G. Maunder, S. X. Ng, S. Chen and L. Hanzo, "Optimized Irregular Variable Length Coding Design for Iteratively Decoded UltraWideBand Time-Hopping Spread-Spectrum Impulse Radio," VTC Spring 2009, pp.1-5.

Abstract—Irregular Variable Length Coding Design for serial concatenated and iteratively decoded Time-Hopping (TH) Pulse Position Modulation (PPM) UltraWideBand (UWB) Spread- Spectrum (SS) Impulse radio system is considered. The proposed design is capable of low Signal-to-Noise Ratio (SNR) operation in Nakagami-m fading channel amalgamated with joint source and channel coding schemes. A number of component Variable Length Coding (VLC) codebooks with different coding rates are being utilized by IVLC scheme for encoding specific fractions of the input source symbol stream. The EXtrinsic Information Transfer (EXIT) charts are used to select these fractions in order to shape inverted EXIT curve of IVLC according to EXIT curve of the inner decoder match. The proposed scheme can achieve near-zero bit error ratio at low SNR values. This IVLC based scheme provides a gain of up to 0.45 dB over the identical-rate single class VLC based scheme.

[35] T. Keränen and J. Vehkaperä, "Error Concealment for SVC Utilizing Spatial Enhancement Information," MobiMedia'08 conference, Oulu, Finland, July 2008.

Abstract—In this paper packet loss error concealment for video sequences compressed using spatial scalability is investigated. Slice support is implemented into the JSVM reference codec of the scalable extension to H.264/AVC video coding standard. The non-normative error concealment scheme introduced in the codec is developed further, adding to it the capability to also consider correctly received slice information from the same frame to conceal lost frame areas. In case of lost base layer slices further improvement on the reconstruction is then achieved by using the correctly received spatial enhancement information for the same frame. The proposed enhancements focus on packet loss concealment on the base layer of I and P-coded frames, where the greatest improvements to the original scheme were identified. Simulation results for given packet loss model indicate on average 2dB improvement over original scheme in the target error scenario.

[36] E. Piri, J. Pinola, F. Fitzek and K. Pentikousis, "ROHC and Aggregated VoIP over FixedWiMAX: An Empirical Evaluation," IEEE ISCC'08 conference, Marrakech, Morocco, July 2008.

Abstract—WiMAX has been at the center of attention in wireless communications during the last years. Nonetheless, very few testbed or field trial measurement accounts have been reported in the peer-reviewed literature. We fill this gap by exploring scenarios where fixed WiMAX is employed for VoIP traffic. VoIP packets typically exhibit large header overheads and small total packet sizes. The actual codec payload per packet is very small compared to the total length of headers appended to each voice frame. Robust Header Compression (ROHC) can significantly decrease header size by capitalizing on static or rarely changing header fields. Aggregating multiple voice frames into one packet is another attractive and effective way to increase application goodput and overall bandwidth utilization. We study the effect of ROHC and application layer aggregation on VoIP performance in a fixed WiMAX testbed consisting of one base station and two subscriber stations. We find that ROHC increases the number of simultaneous bidirectional emulated VoIP flows by 6% when compared to plain VoIP. When aggregation and ROHC are employed in unison, they allow for 86% more flows than standard VoIP to be sustained in our testbed.

[37] E. Piri and K. Pentikousis, “Towards a GNU/Linux IEEE 802.21 Implementation,” IEEE ICC’09 conference, Dresden, Germany, June 2009.

Abstract—Multiaccess mobile devices and overlapping wireless network deployments have emerged as a next generation network fixture. To make the most of all available networks, mobile devices should be capable of handing over between heterogeneous networks seamlessly and automatically. At the same time, operators should be able to steer network attachment based on their criteria. Although several cross layer mechanisms have been proposed in recent years, only the Media Independent Handover (MIH) Services framework has advanced in any of the established standardization bodies. This paper presents a blueprint for a GNU/Linux implementation of IEEE 802.21. We review the salient points of the standard, introduce our software implementation architecture, detail information gathering in GNU/Linux, and show how our prototype implementation can be used in practice. In contrast with prior published work, this paper presents a real IEEE 802.21 implementation, not an abstracted or reduced MIH-like framework, tested and empirically evaluated over real heterogeneous networks.

[38] E. Piri, T. Sutinen and J. Vehkaperä, “Cross-layer Architecture for Adaptive Real-time Multimedia in Heterogeneous Network Environment,” European Wireless ‘09 conference, Aalborg, Denmark, May 2009.

Abstract—Despite the interest, the utilization of today’s networking environment including a variety of access technologies and various services and capabilities is still minimal. Multi-access mobile devices already on the market provide a capability to hand over the heterogeneous networks but so far there has not been any commonly approved way to efficiently capitalize on this feature. Media Independent Handover Services standard specified by the IEEE 802.21 working group is expected to establish the basis for heterogeneous handovers. Although IEEE 802.21 has usage also beyond heterogeneous handovers, its capabilities do not fulfil all the requirements of adaptive multimedia transmission. For example, upper layer events and end-to-end traffic control communication are outside the scope of IEEE 802.21. In this study, we introduce information service architecture for adaptive multimedia which enables to collect and disseminate events and information from the different layers of the protocol stack locally and also between network entities regardless of their location in the network. Our architecture presents a Triggering Engine on top of the IEEE 802.21 services in order to introduce upper layer events, flexibility to event distribution, and end-to-end event based signalling to adaptive multimedia transmission.

[39] M. Uitto and J. Vehkaperä, “Spatial Enhancement Layer Utilisation for SVC in Base Layer Error Concealment,” to appear in MobiMedia’09 conference, London, UK, September 2009.

Abstract—The Scalable Video Coding (SVC) has been recently added as an extension to H.264/AVC standard. This extension allows both bit rate and device capability adaptation which are desirable features especially in error-prone wireless heterogeneous networks. This paper investigates the spatiotemporal error concealment techniques for packet losses in wireless IP networks, where scalable video coding (SVC) can be used. Three methods are introduced: pixel-value interpolation, frame copy and a new method, which utilises the correctly received spatial enhancement layer information if the corresponding base layer is missing. Unlike the new method, the traditional methods discard the corresponding enhancement layer data in the case when the base layer is missing. The simulation results indicate that enhancement layer utilisation provides better results in the case of a missing base layer than the traditional error concealment methods improving the image quality on average 2 dB.

[40] M.G. Martini, M. Chiani, *"Optimum Metric for Frame Synchronization with Gaussian Noise and Unequally Distributed Data Symbols"*, IEEE International Workshop on Signal Processing Advances for Wireless Communications (SPAWC 2009) , Perugia, Italy, June 2009.

Abstract—The problem of frame synchronization with equiprobable data symbols is widely analyzed in the literature and the optimum metric for AWGN channels has been derived both in the case of periodically and aperiodically embedded data symbols. On the contrary, the case of non-equiprobable data symbols has not been studied, although it can occur in many practical situations. An optimum metric for frame synchronization in data streams with non-uniformly distributed data symbols is derived in this paper and its performance is investigated through simulation. A performance comparison with the metric which is optimal for equiprobable data symbols is also provided. Results show that the derived optimum metric results in an evident gain, at the expense of a small additional complexity.

[41] M.G. Martini, V. Tralli, *"Video Quality Based Adaptive Wireless Video Streaming to Multiple Users"*, IEEE International Symposium on Broadband Multimedia Systems and Broadcasting, Las Vegas, US, April 2008.

Abstract—The paper proposes a cross-layer approach for joint closed-loop control of wireless video transmission to multiple users. The approach allows joint adaptation of the multiple users source rates by maximizing the worst served user video quality, taking into account the characteristics of the video sources, channel conditions and user requirements. The controlling task is split in two parts: resource allocation between users and cross-layer optimization for each user. A novel simple algorithm for dynamic resource allocation between users is proposed in conjunction with single user rate control.

[42] M. Zoffoli, J. Gibson and M. Chiani, "On Strategies for Source Information Transmission over MIMO Systems" Proc. IEEE Globecom 2008 Wireless Communications Symposium (GC'08 WC), New Orleans, LA, USA, Nov/Dec 2008.

Abstract—We consider strategies for the lossy transmission of a zero mean Gaussian source over a 2×2 MIMO channel with Rayleigh fading. The source is represented either using a single description or a multiple description code, depending on each strategy characteristic. Performance is evaluated using normalized expected distortion at the receiver, as a function of outage probability. The first strategy employs repetition coding over the two transmit antennas for the transmission of a single description representation of the source. The second strategy uses a time-shared approach to the two transmit antennas, allowing for the transmission of a multiple description representation of the source. The third and fourth strategies are based on, respectively, the Alamouti scheme and spatial multiplexing, and both of these strategies are used for the transmission of a single description representation of the source. The results show that the spatial multiplexing strategy is able to achieve the lowest distortion, and also that it is possible, with the Alamouti strategy, to obtain similar performance at a lower complexity. We finally consider the outage rates of the different strategies and observe that if a system is designed to maximize the outage rate, the corresponding distortion observed at the receiver will not be minimized.

[43] A. Conti, Velio Tralli and Marco Chiani, "Cooperative Relaying with Pragmatic Space-Time Codes" Proc. of International Conference on Communications (ICC'08), Beijing, China, May 2008.

Abstract—The construction of space-time codes for wireless cooperative communications is investigated by considering a pragmatic approach based on the concatenation of convolutional codes and BPSK/QPSK modulation to obtain cooperative codes for relay networks. We also derive the pairwise error probability, an asymptotic bound for frame error probability and a design criterion to optimize both diversity and coding gain. This framework is useful to characterize the behavior of Cooperative Pragmatic Space-Time Codes (CP-STC) and to set up a code search procedure to obtain good pragmatic space-time codes (PSTC) with overlay construction (COP-STC) which are suitable for cooperative communication with a variable number of relays in quasi static channel. We find that P-STCs perform quite well in block fading channels, including quasi-static channel, even with a low number of states and relays, despite the fact that the implementation of P-STC requires common convolutional encoders and Viterbi decoders with suitable generators and rates, thus having low complexity.

[44] E. Paolini, M. Chiani and M. Fossorier, "On the Growth Rate of Irregular GLDPC Codes Weight Distribution" Proc. of 2008 Int. Symp. on Spread Spectrum Techniques and Applications (ISSSTA 2008), Bologna, Italy, Aug. 2008.

Abstract—In this paper the exponential growth rate of irregular generalized low-density parity-check (GLDPC) codes weight distribution is considered. Specifically, the Taylor series of the growth rate is expanded to the first order with the purpose of studying its behavior in correspondence with the small weight codewords. It is proved that the linear term of the Taylor series, and then the expected number of small linear-sized weight codewords of a randomly chosen GLDPC code in the irregular ensemble, is dominated by the degree-2 variable nodes and by the check nodes with minimum distance 2. A parameter is introduced, only depending on such variable and check nodes, discriminating between an exponentially small and exponentially large expected number of small weight codewords.

[45] E. Paolini, M. Varrella, M. Chiani, G. Liva and B. Matuz, "Low-Complexity LDPC Codes with Near-Optimum Performance over the BEC" Proc. of 2008 Advanced Satellite Mobile Systems Conference (ASMS 2008), Bologna, Italy, Aug. 2008.

Abstract—Recent works showed how low-density parity-check (LDPC) erasure correcting codes, under maximum likelihood (ML) decoding, are capable of tightly approaching the performance of an ideal maximum-distance-separable code on the binary erasure channel. Such result is achievable down to low error rates, even for small and moderate block sizes, while keeping the decoding complexity low, thanks to a class of decoding algorithms which exploits the sparseness of the parity check matrix to reduce the complexity of Gaussian elimination (GE). In this paper the main concepts underlying ML decoding of LDPC codes are recalled. A performance analysis among various LDPC code classes is then carried out, including a comparison with fixed-rate Raptor codes. The results confirm that a judicious LDPC code design allows achieving achieving a near-optimum performance on the erasure channel, with very low error floors. Furthermore, it is shown that LDPC and Raptor codes, under ML decoding, provide almost identical performance in terms of decoding failure probability vs. overhead.

[46] E. Paolini, M. Fossorier and M. Chiani, "Pseudo-Binomial Degree Distributions from Derivative Matching" Proc. of 2008 Int. Symp. on Turbo Codes & Related Topics, Lausanne, Switzerland, Sept. 2008.

Abstract—In this paper, a method to design check concentrated LDPC degree distributions for the erasure channel is proposed. This method is obtained taking a derivative matching approach. It consists of matching the first and high-order derivatives of the variable node decoder EXIT function and inverse check node decoder EXIT function in order to reduce the gap between the two curves in the EXIT chart. A sufficient condition for a check-concentrated distribution to achieve derivative matching up to some order is first developed, and then a design algorithm is proposed exploiting this *sufficient condition. A comparison with other deterministic design techniques is provided, revealing the potentialities of the proposed algorithm.*

[47] G. Liva, E. Paolini and M. Chiani, "Packet Loss Recovery in the Telemetry Downlink via Maximum-Likelihood LDPC Decoding" Proc. of 2008 ESA Int. Workshop on Signal Processing for Space Comunications, Rhodes Island, Greece, Oct. 2008.

Abstract—In this paper a novel framework for packet loss recovery in the Consultative Committee for Space Data Systems (CCSDS) telemetry downlink is presented. The framework relies on packet-level LDPC erasure correcting codes and on low complexity maximum likelihood (ML) decoding. A code design tailored to the ML decoder is presented, which is based on Generalized Irregular Repeat-Accumulate (GeIRA) codes. An optional concatenation with outer BCH codes is proposed as a mean for lowering the error floors. Numerical results show that for short block lengths the outer concatenation gives rise to very low error floors, while already for moderate-length codes the concatenation can be skipped without sacrificing the erasure recovery performance. Remarkably, all the proposed schemes under ML decoding tightly approach the performance of ideal maximum distance separable (MDS) codes down to low error rates.

[48] Mark F. Flanagan, "Codeword-Independent Performance of Nonbinary Linear Codes Under Linear-Programming and Sum-Product Decoding" Proc. of IEEE International Symposium on Information Theory (ISIT 2008), Toronto, Canada, Jul. 2008.

Abstract—A coded modulation system is considered in which nonbinary coded symbols are mapped directly to nonbinary modulation signals. It is proved that if the modulator-channel combination satisfies a particular symmetry condition, the codeword error rate performance is independent of the transmitted codeword. It is shown that this result holds for both linear programming decoders and sum-product decoders. In particular, this provides a natural modulation mapping for nonbinary codes mapped to PSK constellations for transmission over memoryless channels such as AWGN channels or flat fading channels with AWGN.

[49] Velio Tralli, Andrea Conti and Marco Chiani, "On the Design of Space-Time Trellis Codes for Cooperative Relaying" Proc. of 1st COST2100 Workshop on MIMO and Cooperative Communications, Trondheim, Norway, Jun. 2008.

Abstract—The design of space-time codes for wireless communications with relays is investigated by considering a pragmatic approach based on the concatenation of convolutional codes and BPSK/QPSK modulation to obtain cooperative codes. We propose a design criterion which aims at optimizing both diversity and coding gain, based on an asymptotic bound for frame error probability. This framework is useful to characterize the behavior of Cooperative Pragmatic Space-Time Codes (CP-STC) and to set up a code search procedure to obtain good CP-STC with overlay construction (COP-STC) which are suitable for cooperative communication with a variable number of relays and have improved performance with respect to cooperative codes in the literature. The CP-STCs result to perform quite well in block fading channels, including quasi-static channel, even with a low number of states and relays. Moreover, their implementation is low complexity, since it requires common convolutional encoders and Viterbi decoders with suitable generators and rates.

[50] M. Flanagan, E. Paolini, M. Chiani and M. Fossorier, "On the growth rate of the weight distribution of irregular doubly-generalized LDPC codes" Proc. of 2008 Allerton Conference on Communications, Control and Computing, Monticello, IL, USA, Sept. 2008.

Abstract—In this paper, an expression for the asymptotic growth rate of the number of small linear-weight codewords of irregular doubly-generalized LDPC (D-GLDPC) codes is derived. The expression is compact and generalizes existing results for LDPC and generalized LDPC (GLDPC) codes. Assuming that there exist check and variable nodes with minimum distance 2, it is shown that the growth rate depends only on these nodes. An important connection between this new result and the stability condition of D-GLDPC codes over the BEC is highlighted. Such a connection, previously observed for LDPC and GLDPC codes, is now extended to the case of D-GLDPC codes.

[51] E. Paolini, M. Chiani and M. Fossorier, "On the design of irregular GLDPC codes with low error floor over the BEC" Proc. of 2008 International Symposium on Information Theory and its Applications, Auckland, New Zealand, Dec. 2008.

Abstract- The design of GLDPC codes for the binary erasure channel with low error floor under iterative decoding is investigated. Both bounded distance and maximum a posteriori decoding at the check nodes are considered. For both check node decoding algorithms a key parameter is identified, discriminating between an exponentially small and exponentially large expected number of small size stopping sets. A code design technique is proposed based on this theoretical investigation.

[52] A. Benvegnu, S. Bandi and V. Tralli, "On the performance and the optimization of LDPC codes on fading channels with imperfect CSI" Proc. of IEEE International Symposium on Wireless Communication Systems (ISWCS 2008), Reykjavik, Iceland, Oct. 2008.

Abstract—In this article we investigate low-density parity check (LDPC) codes for fast Rayleigh fading channels in the *presence of channel estimation error. The analysis is carried out using the Density Evolution technique assuming a belief propagation (BP) decoder. After having derived the analytical framework for decoding analysis, we prove that there exists a simple approximated relationship, function of estimation error variance, between the signal-to-noise ratio thresholds in the two cases of perfect and imperfect channel state information (CSI). This approximation follows the exact computation tightly, when the variance of the estimation error is sufficiently smaller than the fading variance. As a result, optimized codes for perfect CSI are also approximately optimized for imperfect CSI.*

[53] E. Paolini, M. Chiani, G. Liva, B. Matuz and Z. Katona, "On Construction of Moderate-Length LDPC Codes over Correlated Erasure Channels" Proc. of 2009 IEEE International Conference on Communications (ICC 2009), Dresden, Germany, Jun. 2009.

Abstract—The design of moderate-length erasure correcting low-density parity-check (LDPC) codes over correlated erasure channels is considered. Although the asymptotic LDPC code design remains the same as for a memoryless erasure channel, robustness to the channel correlation shall be guaranteed for the finite length LDPC code. This further requirement is of great importance in several wireless communication scenarios where packet erasure correcting codes represent a simple countermeasure for correlated fade events (e.g., in mobile wireless broadcasting services) and where the channel coherence time is often comparable with the code length. In this paper, the maximum tolerable erasure burst length (MTBL) is adopted as a simple metric for measuring the code robustness to the channel correlation. Correspondingly, a further step in the code construction is suggested, consisting of improving the LDPC code MTBL. Numerical results conducted over a Gilbert erasure channel, under both iterative and maximum likelihood decoding, highlight both the importance of the MTBL improvement in the finite-length code construction and the possibility to tightly approach the performance of maximum distance separable codes.

[54] Ana I. Perez-Neira, Pol Henarejos, Velio Tralli and Miguel A. Lagunas, "A low complexity space-frequency multiuser resource allocation algorithm" Proc. of International ITG/IEEE Workshop on Smart Antennas (WSA 2009), Berlin, Germany, Feb. 2009.

Abstract—This work presents a resource allocation algorithm in K-user, M-subcarrier and NT-antenna systems for on-line scheduling. To exploit temporal diversity and to reduce complexity, the ergodic sum rate is maximized instead of the instantaneous one. Dual optimization is applied to further diminish complexity together with a stochastic approximation, which is more suitable for online algorithms. Weighted sum rate is considered so that users can be either prioritized by higher layers or differentiated by proportional rate constraints. The performance and complexity of this algorithm is compared with well-known benchmarks and also evaluated under real system conditions for the MIMO Broadcast channel.

[55] László Bokor, Szabolcs Nováczki, László Tamás Zeke, Gábor Jeney: "Design and Evaluation of Host Identity Protocol (HIP) Simulation Framework for INET/OMNeT++", in the proceedings of the 12-th ACM International Conference on Modeling, Analysis and Simulation of Wireless and Mobile Systems (MSWIM 2009), Tenerife, Canary Islands, Spain, Oct. 26. 2009.

Abstract—Host Identity Protocol (HIP) decouples IP addresses from higher layer Internet applications by proposing a new, cryptographic namespace for host identities. HIP has great potential in means of mobility and multihoming support, security, and performance, such making it quite a promising candidate as the basic architecture of the Future Internet. However, HIP is still in development and very early standardization phase: the protocol is continuously evolving due to its adaptivity to functional changes and extensions. Aiming to completely understand the protocol's behavior, its applicability to wide-scale usage and to analyze current and future improvements and enhancements, it is crucial to develop a proper, RFC-compliant, extensible simulation model for Host Identity Protocol. In this paper we present the structural design and the functional details of our HIP simulation framework (called HIPSim++) integrated into the INET/OMNeT++ discrete event simulation environment. In order to evaluate the accuracy and preciseness of HIPSim++, we designed a real-life HIP testing environment and compared the simulation outcomes with the reference results obtained from the HIP testbed. Our comparison results show excellent accuracy and consistent operation of the simulation framework in terms of handover metrics (latency, packet loss, throughput) and behavior experienced in the real HIP testbed.

[56] László Bokor, László Tamás Zeke, Szabolcs Nováczki, Gábor Jeney: "Protocol Design and Analysis of a HIP-based Per-Application Mobility Management Platform", in the proceedings of the 7-th ACM International Symposium on Mobility Management and Wireless Access (MobiWAC 2009), Tenerife, Canary Islands, Spain, Oct. 26. 2009.

Abstract— Rapid evolution of wireless networking has provided wide-scale of different wireless access technologies like Bluetooth, ZigBee, 802.11a/b/g, DSRC, 3G UMTS, LTE, WiMAX, etc. The complementary characteristic of the above architectures motivates next generation network operators to integrate them in a supplementary and overlapping manner. Recent wireless devices are equipped with multiple interfaces, thus enabling concurrent communication sessions. With the advance of such heterogeneous structures – and considering that users are often running applications simultaneously – the traditional per-host mobility management approach cannot be the optimal solution *for handling connection changes. Instead, the concept of per-application mobility management is to be introduced, where a dedicated interface (i.e. access network) is selected by each application according to its QoS prerequisites and the actual networking conditions. Aiming to benefit from this novel concept in practice, in this paper we designed and evaluated a HIP-based per-application mobility management platform founded on the promising Host Identity Protocol (HIP) and the cross-layer building blocks closely incorporating with it.*

[57] Arpad Huszak, Sandor Imre: "DCCP-based Multiple Retransmission Technique for Multimedia Streaming", 6th International Conference on Advances in Mobile Computing & Multimedia, MoMM2008, In Cooperation with ACM SIGMM, ISBN 978-1-60558-269-6, pp. 21—28., Linz, Austria, 24—26 November 2008

Abstract—Retransmission-based error recovery is the simplest technique to minimize the overall packet loss ratio in order to increase the quality of the applications. Multimedia applications are becoming increasingly popular in IP networks, while in mobile environment the limited bandwidth and the higher error rate arise in spite of its popularity. Retransmission can be also used for loss recovery in media applications, but the number of retransmissions is limited by the playout buffer and the recent network delay. In this paper we present a source controlled adaptive multiple retransmission scheme for multimedia streaming. Due to source controlling the receiver do not need to send additional retransmission request messages. The other advantage of transmitter controlled decision algorithm is that the necessary RTT information is available at the source due to DCCP transport protocol. Therefore no additional network measurements and probe messages are needed. With the knowledge of the actual network delay and the playout buffer delay, we are able to determine the number of retransmission attempts without late delivery of packets.

[58] Arpad Huszak, Sandor Imre, "Content-aware Interface Selection Method for Multi-Path Video Streaming in Best-effort Networks", 16th International Conference on Telecommunications, ICT 2009, ISBN 978-1-4244-2937-0, pp. 207—212, Marakech, Morocco, 25—27 May 2009

Abstract—Today's mobile devices are equipped with multiple interfaces to make the connection possible to different type of networks. In order to efficiently utilize the interface capabilities, multi-path streaming can be used. Resource intensive applications can deliver high bitrate streams over multiple paths by cumulating the available bandwidth of the different subpaths. In this paper we propose a multi-path streaming method that chooses a set of paths maximizing the overall quality at the client. While the available paths have different bandwidth, delay and loss probability constrains, the packet distributor must take the video packet importance and the dependencies between packets into account. Transmitting the reference video frames on the most reliable links will decrease the loss probability of important data packets and increase the measured video quality. Both analytical and simulation analysis were performed to examine the behavior of the presented content-aware interface selection model. The results show that the proposed solution provides higher video quality than common scheduling algorithms.

[59] Vikor Gergely, Gábor Fehér, "Enhancing Progressive Encryption for Scalable Video Streams", EUNICE 2009 Broadband for all, Barcelona, September 7-9

Abstract—The technique called progressive encryption is used in many areas of content security. However, the plain algorithm itself is only applicable in real transmission scenarios where no packet loss occurs, otherwise additional error correction techniques need to be used in order to achieve maximum decodeability of network packets. The cipher-stepping method (CSM) described in this article adds error correction to progressive encryption in the case where stream ciphers are used to encrypt stream data. It is also explained how the CSM method along with progressive encryption can be used in the encryption of scalable video streams.

2.5 *Participation at conferences, symposiums, and workshops*

Dates	Conference, symposium, or workshop	Type of audience	Countries addressed	Size of audience	Partner(s) involved
September 2009	IGI Global Press	Research	World	Several thousands	CEFRIEL, THALES
June 2009	IEEE Signal Processing Advances for Wireless Communications (SPAWC) 09	Research	World	~300	KU
September 2009	eMobility General Assembly	Research/Industry	Europe	~300	KU
September 2009	5th International Mobile Multimedia Conference (Mobimedia 2009)	Research	World	~200	KU
September 2009	ICT-PEACE Workshop	Research	World	~80	KU
September 2009	IMS Workshop	Research	World	~80	KU
September 2009	3rd European Symposium on Mobile Media Delivery (EUMOB 2009)	Research	Europe/World	~80	KU

Dates	Conference, symposium, or workshop	Type of audience	Countries addressed	Size of audience	Partner(s) involved
April 2009	IEEE International Conference of Acoustics, Speech, and Signal Processing 2009 (ICASSP 2009)	Research	World	~2000	Siemens
May 2009	Picture Coding Symposium 2009 (PCS 2009)	Research	World	~200	Siemens
June 2008	ICT Mobile Summit	Research	World	~1000	CEFRIEL
August 2008	International Symposium on Spread Spectrum Techniques and Applications (ISSSTA 2008)	Research	World	~800	CEFRIEL, CNIT
October 2008	IEEE NETWORKS	Research	World	~600	CEFRIEL, BME
November 2008	6th International Conference on Advances in Mobile Computing & Multimedia, (MoMM2008)	Research	World	~200	BME
June 2009	IFIP IM-BCN	Research	World	~400	CEFRIEL
June 2009	ICT Mobile Summit	Research	World	~1000	CEFRIEL
				~200	
April 2009	Vehicular Technology Conference Spring	Research	World	~600	Soton
May 2009	Communication Networks and Services Research Conference, Moncton, Canada	Research	World	~150	Soton
June 2009	International Conference on Communications (ICC'09)	Research	World	~2000	Soton, VTT, BME, CNIT
July 2008	European Symposium on Mobile Media Delivery	Research	Europe/World	~100	CNIT/Thales/KU /VTT
July 2008	IEEE Symposium on Computers and Communications	Research	World	~800	VTT
July 2008	4th International Mobile Multimedia Communications Conference (MobiMedia'08)	Research	World	~200	VTT, CNIT/KU, Soton
May 2009	European Wireless '09	Research	World	~800	VTT
October 2009	12-th ACM International Conference on Modeling, Analysis and Simulation of Wireless and Mobile Systems (MSWIM'09)	Research	World	Tbc.	BME
May 2009	International Conference on Telecommunications (ICT 2009)	Research	World	~400	BME
September 2009	EUNICE 2009	Research	Europe/World	~200	BME
May 2008	International Conference on Communications (ICC 2008)	Research	World	~2000	CNIT
June 2008	1st COST2100 Workshop on MIMO and Cooperative Communications	Research	Europe	~200	CNIT
March/April 2008	IEEE International Symposium on Broadband Multimedia Systems and Broadcasting	Research	World	~200	CNIT
July 2008	International Symposium on Information Theory (ISIT 2008)	Research	World	~800	CNIT
August 2008	Advanced Satellite Mobile Systems Conference	Research	World	~200	CNIT
September 2008	International Symposium on Turbo Codes and Related Topics	Research	World	~300	CNIT
October 2008	International Workshop on Signal Processing for Space Communications	Research	World	~200	CNIT
December 2008	IEEE Globecom 2008 (GC'08)	Research	World	~2000	CNIT
September 2008	Allerton Conference on Communications, Control and Computing	Research	World	~300	CNIT
December 2008	International Symposium on Information Theory and its Applications (ISITA)	Research	World	~300	CNIT
October 2008	IEEE International Symposium on Wireless Communication Systems (ISWCS 2008)	Research	World	~200	CNIT
February 2009	International ITG Workshop on Smart Antennas (WSA 2009)	Research	World	~200	CNIT

2.6 Participation at concertation meetings and other (ICT) project meetings

Partner		Date (Start/ End)	Meeting place	N° of persons	Details
1	THALES	16/04/08 18/04/08	Faro, Portugal	1	WP5 : ICT concertation meeting, “networked media” cluster
1	THALES	15/10/08	St Malo, France	1	WP5 : ICT concertation meeting, “networked media” cluster

2.7 Lectures and courses

The JSCC/D concepts and issues, with particular reference to the solution being proposed within the framework of OPTIMIX project, have been explained and discussed during some lectures since the start of the project (March 2008). More specifically, the lecture entitled “Quality of Service over IP networks” has been given to master and university students, as well as to newly employed personnel of IT company which were sent to CEFRIEL for training courses in telecommunications and other fields of ICT.

In Kingston University the concepts of (network-aware) joint source and channel coding and cross-layer design are part of the program of the “Multimedia Communications” and “Wireless Communications” modules in the M.Sc. course on “Networking and Data Communications”, Dr. Martini is also coordinating the development of a new M.Sc. course on “Wireless Communications” in her University, where these concepts are embedded in the curriculum. Dr. Martini held several seminaries on the topic in Kingston University and other Universities in UK and abroad.

In Budapest University of Technology and Economics there are several courses, where the project outcomes are taught. The so called “Mobile Internet” is an optional subject started recently. “Optional” means that students are not obliged to learn it: if they are interested, they can join. Both B.Sc. and M.Sc. level students are permitted to join this lecture. HIP related knowledge and transport protocols are also detailed in the Mobile Internet lectures and exercises. There are plenty of other subjects related to mobile communications and infocommunication technologies, where the OPTIMIX project related results are also mentioned.

Also in the University of Bologna, both JSCC/D and cross-layer concepts are taught (especially in the form of seminars directed to M.Sc. level students) and constitute important training elements for Ph.D. students.

3 Standardization activities

3.1 Contributions to standardization bodies

The following contribution was presented to the Joint Video Team (JVT) of ISO/IEC MPEG and ITU-T VCEG and accepted for inclusion into the SVC reference software.

[60] Xiang Li, Peter Amon, Andreas Hutter, André Kaup, "Lagrange Multiplier Selection for Rate-Distortion Optimization in SVC", ISO/IEC JTC1/SC29/WG11 and ITU-T SG16 Q.6, Document JVT-AD021, Geneva, Switzerland, January/February 2009.

Abstract—The Lagrange multiplier based rate-distortion optimization (RDO) has been widely employed in single layer video coding. During the development of scalable video coding (SVC) extension of H.264/AVC, it was directly applied in a multi-layer scenario. However, such an application is not very efficient since the correlation between layers is not considered in the Lagrange multiplier selection. To improve the overall performance, in this contribution a new selection algorithm is presented for RDO in SVC. Simulations show that the proposed method outperforms the most recent SVC reference software (JSVM 9.15). With small computational cost, average gains of 0.22 dB and 0.35 dB were reportedly achieved in the tests of four-layer quality scalability and three-layer spatial scalability, respectively.

3.2 Participation at standardisation meetings

Partner	Date	Meeting Place	Standardization Body and Group
Siemens	23/04/08-29/04/08	Geneva, Switzerland	ITU-T VCEG, JVT
Siemens	16/07/08-18/07/08	Berlin, Germany	ITU-T VCEG
Siemens	08/10/08-10/10/08	San Diego, CA, USA	ITU-T VCEG
Siemens	28/01/09-03/02/09	Geneva, Switzerland	ITU-T VCEG, JVT
Siemens	01/07/09-03/07/09	London, U.K.	ITU-T VCEG, ISO/IEC MPEG

4 Intellectual property rights (IPR)

In this section, a list of IPR generated during the first half of the OPTIMIX project is given. The two industrial partners Siemens and THALES were active here.

[61] C. Lamy-Bergot and B. Gadat, “Outil et procédé de détermination des paramètres de codage pour une allocation de débit disponible entre une opération de codage de source et une opération de protection par ajout de codage correcteur d'erreur”, French patent application, Oct. 6th, 2008.

[62] J. Pandel, P. Amon, "Verfahren zum codieren einer Folge von digitalisierten Bildern.", German patent application, Nov 2008.

5 Project liaisons

One of OPTIMIX' goals is to establish links with other related ICT projects to have both a better knowledge of existing solutions and to take advantage of possible advances in the considered fields.

This action has in particular been carried through the participation to Networked Media clusters meetings, and in particular the participation to Multimedia Delivery Platform (MDP) cluster. Participation to the cluster white paper (Multimedia Delivery Platform cluster white paper) has also linked to participating to the global think tank on multimedia delivery over the future internet.

The participation to the cluster allowed also collaboration with the ICT SEA project. In the first year of the OPTIMIX project a common publication with SEA has been realized (Theodore Zahariadis, Catherine Lamy-Bergot, Thomas Schierl, Karsten Grneberg, Luca Celetto, Christian Timmerer "Content Adaptation Issues in the Future Internet", accepted for EC Future Internet Assembly book).

Finally, in order to disseminate the project results to the public knowledge and to present them to the ICT community, OPTIMIX participated both to the ICT Mobile Summit 2008 and 2009. It has to be noted that OPTIMIX obtained the runner-up Best Paper Award with the paper "Robustness in Next-Generation Networks" at the ICT Mobile Summit 2009.

Dr. Martini (first as CNIT, then as Kingston University) organised EUMOB 2008 (European Symposium on Mobile Multimedia, Oulu, Finland, July 2008), where a number of European projects were invited to present and discuss their achievements. OPTIMIX was also presented. The symposium was a good opportunity to network with representatives from other European projects in similar areas.

Kingston University has organised this year the 5th International Conference on Mobile Multimedia Communications (Mobimedia 2009) in London (7th-9th September) and the EUMOB workshop will be repeated, embedded in the conference program.

Kingston University is part of the eMobility platform and members of Kingston University attended the General Assembly in 2008 and KU will attend the General Assembly 2009 (chairing a panel session on Mobile Applications). Also, University of Bologna/CNIT is part of the eMobility platform. In particular, Prof. Marco Chiani is a member of the eMobility Expert Group and attended the group meeting in 2008 (PIMRC 2008).

Kingston University is also an active member of WWRF (chairing a WG).

Discussions with members of the ICT-PEACE project are ongoing (Kingston University participates to both projects).

6 Conclusion

The work in the OPTIMIX project on dissemination and standardization is in line with the project objectives. Numerous publications have been made by all project partners. Joint publications indicate the collaborative spirit in the project. The project shows an active participation at conferences, workshops, etc. and also participates at concertation meetings and other ICT project meetings. The industry partners Thales and Siemens work actively on the generation of intellectual property rights (IPR). Siemens also participates at different standardisation bodies in the video coding domain (ISO/IEC MPEG, ITU-T VCEG).

Deliverable D1.5 will complement this listing of dissemination and standardisation items by activities performed in the second half of the project duration.