Joint Source and Channel (de)coding in 4G Networks: the PHOENIX Project

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Abstract— **The PHOENIX project main goal is to effectively exploit the available bandwidth on wireless links (WLAN, UMTS, 4G…) that is dynamic by nature, providing optimised solutions for multimedia transmission over IP-based wireless networks. To reach this goal, it is proposed to develop a scheme offering the possibility to let the application world (source coding, ciphering) and the transmission world (channel coding, modulation) talk together over an IPv6 protocol stack, so that they can jointly develop an end-to-end optimised wireless communication link.**

The design trade-offs of interactive wireless video systems will be studied, and performance comparisons provided both in the context of existing and future generations of wireless systems. Innovative schemes enabling joint optimisation over wireless links will be developed, that include the development of flexible channel coding and modulation schemes, the adaptation of existing source coding schemes with respect to their ability for Joint Source Channel Coding/Decoding (JSCC/D) and the specific development of new optimised ones. In parallel, efficient and adaptive optimisation strategies will be defined, that jointly control the coding blocks and realistically take into account the system limitations and specifications (*e.g.* presence of ciphering, presence of one or several wireless hops…). The design of a cross-layer protocol allowing the necessary information exchanges between the upper and lower layers of the protocol stack will also be investigated. Highly needed to improve the overall system performance in a joint approach, this cross-layer design will ensure compatibility with existing protocols in the wireless transmission domains.

Finally, a global network architecture based on joint optimisation for future wireless systems will be proposed, that includes the development of a transparent network communication approach allowing the application of PHOENIX optimisation strategies in any kind of 4G fully IP-based network.

I. INTRODUCTION AND OBJECTIVES

Following the path opened by GSM systems, the under-deployment UMTS system is leading to more and more configurable, dependable, adaptable, intelligent, secure but also complex wireless solutions. Aiming at handling digital data of different nature (text, voice, image, video ...) that will be used in various contexts (home, office, on the move, ...) these systems rely on inner software that make them more and more efficient and easy to use. The gap remains however still important between what the actual systems propose and what the users envision and could use. The lack is particularly noticeable in the domains of heterogeneous networks and systems interconnection, but also in the flexible management of resources, systems are targeting flexible re-configurable architectures and the 4th generation of wireless systems (4G) is foreseen as a globally integrated communication network interconnecting, in a transparent way, a multitude of heterogeneous networks and systems. The key challenge for such solutions, and firstly for Wireless Local Area Networks (WLANs) which can be seen as first stones for the edification of future wireless systems, is the definition of a flexible re-configurable network architecture that enables simultaneous optimisation of bandwidth as well as QoS management.

Major progresses were made throughout the last decades of 20th century that allowed to individually optimising each module in modern communication systems. Excellent results have been obtained, but this separate approach has shown limitations in the case of wireless communications. It is now time to focus on more integrated strategies and to design flexible beyond 3G wireless systems that interconnect, in a transparent way, a multitude of heterogeneous networks and systems. Indeed, optimal allocation of user and system resources may be effectively achieved with the co-operative optimisation of communication system components. This approach, following the already known joint source channel coding and decoding (JSCC/D) one, aims at developing strategies where the source coding, channel coding, modulation, ciphering, and, possibly, network parameters are jointly determined to yield the best end-to-end system performance.

PHOENIX is an FP6 IST European project started at the beginning of 2004 and will be finished at the end of 2006. It is a collaboration between industrial partners, industrial research laboratories, Small Medium Enterprise and specialised academic institutions of different countries.

The aim of the PHOENIX project is to develop a scheme offering the possibility to let the application world (source coding, ciphering) and the transmission world (channel coding, modulation) to talk to each other over an IPv6 protocol stack (network world), so that they can jointly develop an end-to-end optimised wireless communication link. To reach this goal, the following main axes will be pursued:

- development of innovative schemes to enable end-to-end joint optimisation over wireless links: flexible channel coding and modulation schemes, adaptation and development of source coding schemes with respect to their ability for JSCC/D, Quality of Service (QoS) and bandwidth optimisation. That is why beyond 3G
- establishment of efficient and adaptive optimisation strategies jointly controlling the coding blocks and realistically taking into account the system limitations,
- integration of those techniques in a global network architecture including the development of a cross-layer design approach which will allow to apply the optimisation strategies in any kind IP-based network.
- a real test-bed as a "proof-of-concept" will be also realised to demonstrate the correct functioning and the potentiality of the proposed architecture in different network and users scenarios. A complete set of tests will be executed in order to validate and assess the overall system.

II. STATE-OF-THE-ART

A. *Source coding schemes*

The newest video coding schemes improved the overall coding efficiency remarkably compared to previous video coding technology. Joining their efforts, ISO/IEC MPEG and ITU-T standardisation bodies have developed a new standard (MPEG-4 AVC and H.264 respectively), that outperforms the successor technology nearly by a factor of two in terms of compression efficiency. Unfortunately, this new standard does not offer scale capabilities in spatial resolution and does not incorporate SNR scalability. Both tools are yet prerequisites for adaptive video coding technologies, to allow for an adaptation to variable transmission conditions. On the other hand, existing adaptive technology (*e.g.* the SNR scalable MPEG-4 FGS) only has poor compression efficiency. In addition existing schemes are not designed to allow joint optimisations.

B. Channel coding schemes

The goal of channel coding is to ensure satisfactory error level by adding redundancy to the information bit stream, in order to combat noise and fading effects. By construction, channel coding is indeed the basis of a communication link, hence a place where research should help reduce the bit error rate (BER) and frame error rate (FER) for efficient transmissions to occur. This implies taking into account the different error protection needs various portions of a unique data stream may have. When transmission occurs over time-varying channels, this implies that more flexibility is required from the coding schemes. It is therefore desirable that the code rate may change, so that the correction power of the code can be adapted to the source and channel needs [7]. Rate Compatible Punctured Convolutional (RCPC) codes [6] meet such a goal and their simple implementation provides interesting gains, as was shown for Adaptive-Multi-Rate (AMR) in GSM systems or more recently with video sources [18][14]. A promising path is to use the recent family of codes such as turbo or LDPC codes, whose flexibility has been studied only to some extent [16][13].

The flexibility of the modulation segment should also be investigated. Several UEP schemes were already implemented through non-uniform signal constellations for terrestrial broadcasting of digital TV. Multi-resolution modulation techniques (MRM) have been considered for broadcast applications, in order to cope with different receivers, and also combined with multi-level coding. Next generation high speed wireless systems will consider multi-carrier, in particular orthogonal frequency division multiplexing (OFDM). Such schemes that are particularly suitable in frequency selective environments. Some studies have applied custom OFDM bit-loading algorithms to video sources in wired channels (DSL) [17]. Preliminary works studied the possibility to extend these techniques to wireless channels [15], showing interesting gains.

C. Ciphering tools and schemes

The major issue in securing transmission is to allow applications to build a secure end-to-end communication channel to prevent eavesdropping of the exchanged data. In future integrated networks, this service will rely on heterogeneous infrastructures, from network topology to fibre or wireless physical media. Furthermore new applications demand other security services, such as authentication for group communication, encryption or watermarking for multimedia data protection. These new services express different security requirements (different security policies) with respect to different types of data. In other words, versatile communication applications require the security services to be implemented in a versatile manner. In particular, these different security policies identify how different data types must be protected, possibly within a given format. For different channels, for different data types, the security functions (encryption, hashing, ...) must be optimised together with the joint coding approach. The integrated framework will thus offer a versatile and adaptable secure infrastructure which can satisfy the needs of rich multimedia based applications.

D. Joint source channel coding schemes

Traditionally, the two antagonistic encoding operations of compression and error correction are separated from each other, following Shannon's well-known separation theorem [1] which states that source coding and channel coding can, asymptotically with the length of the source data, be designed separately without any loss of performance for the overall system. However, many modern popular applications, such as audio/video streaming, do not meet these ideal hypotheses. They rather often require transmitting data with real-time constraints, operate on sources whose encoded data bit error sensitivity varies significantly or are wished to be as simple and low-power consuming as possible. Following the path opened by joint source and channel coding (JSCC) that showed that separation did not necessarily lead to the less complex solution [2], nor was always applicable [3], techniques that include a co-ordination between source and channel encoders were investigated, and solutions were developed [4][5][8] that improve both encoding and decoding processes while keeping the overall complexity at an acceptable level [9].

Given the above considerations, two different scenarios under which significant gains may be expected from joint source-channel coding have been identified. The first is a single-link scenario (for instance over WLAN), where both source and channel coding/modulation are (jointly) adapted to propagation conditions and QoS requirements. Such a scenario will consider the transmission of heterogeneous multimedia data and the evaluation of the gain introduced by the optimisation process. A second scenario includes more users in the same environment. Some of them exploiting the JSCC/D technique as well whereas others are traditional multimedia users. In this case, the focus will be put on the optimisation of the whole network performance with respect to the interference and radio resource allocation and the sharing.

After the analysis of these basic scenarios more complex ones will be considered: communication paths with multiple wireless hops and multicast distribution of the multimedia content. Furthermore the support of additional functionalities such as confidentiality and security will be taken into account. From an application point of view both streaming and interactive video services will be deployed.

E. Information exchanges over IPv6

To allow joint source and channel coding, methods for exchanging control information between the application layer (where source coding is located) and the physical or access layer (where channel coding and modulation are performed) are needed. At present, no such method exists to transfer detailed and real-time information between these layers for joint optimisation. First steps were however taken in order to provide control information between system layers in general. For example, QoS features in IPv6, namely Differentiated Services [19] and Integrated Services provide means for application to exploit network resources in a shared or dedicated manner, using IP signalling between application and network layers. Another example of the layer-to-layer signalling can be found from IEEE 802.11e standard where the QoS provisioning is performed between the application and the medium access layers. Still, the QoS information obtained by considering the IP packet priorities alone will not be sufficient for delivering all optimisation information between the source and physical layers. When more detailed information (such as the Decoder Reliability Information (DRI)) need to be delivered to fully optimise the transmission utilising both source and channel coding, a solution may be to generate extra packets to be inserted in the network communication streams, as discussed in [10]. Other solutions consist in extending existing protocols such as IPv6 [20] or ICMPv6 [21] through the definition of new options and messages respectively.

III. NEW APPROACHES

A. A new overall network architecture

Targeting the goal to offer the possibility to the source coding world to work as a team with the channel coding world and have them jointly optimise the end-to-end wireless communication link, the PHOENIX project proposes to consider the innovative global network architecture presented in Fig. 1. It consists of the key elements of the architecture to be deployed to develop suitable optimisation strategies for achieving adaptive/dynamic optimisation for secure multimedia information transmission over an IP-based wireless link. Co-ordinating tools, named joint controllers, which implement the controlling strategies are presented. Their task is to drive the whole communication chain by co-ordinating the scalable source encoder implementing data partitioning tools, the ciphering providing security and the adaptive channel encoder designed to provide selective protection of the transmitted data and the dynamic modulator fitted to the channel transmission conditions.

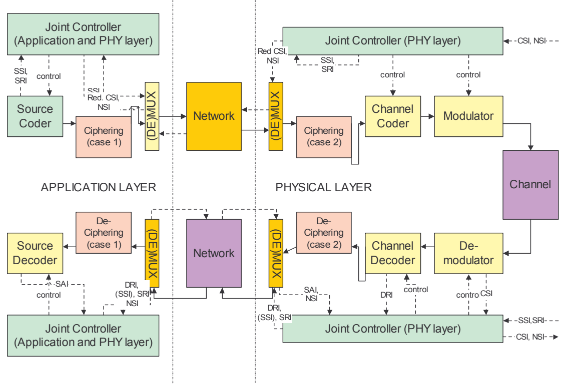

Fig. 1- End-to-end communication system over an IP based network.

The controlling activities are separated between the application and physical layers to provide the needed level of flexibility: the joint controller at physical level will deal with the bit-by-bit optimisation, whereas the application level one will operate at a higher level, selecting suitable configurations based on the user criteria, the considered application, the ciphering system and the intended network protocols. Note that the IP network may be restricted to a simple local stack in the case where one side of the transmission is only a terminal (*e.g.* a UMTS terminal).

More detail scheme of the relevant blocks in the transmission side in depicted in the Fig. 2.

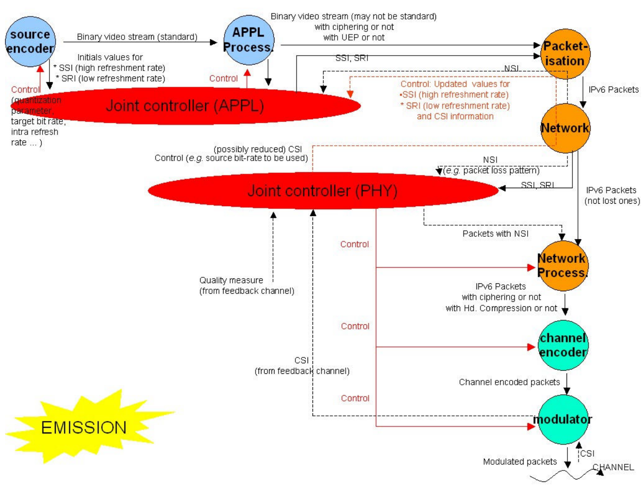

Fig. 2 – Detailed scheme of the transmission side.

At Application level, one sees first the action of the source encoder, which provides the actual source bitstream together with SSI and SRI information. This source encoder is a priori supposed standard compliant, or to correspond to a recommendation of PHOENIX project for adaptation of an existing standard with respect to joint optimisation techniques. This is followed by application processing (optional) which can include first unequal error protection techniques (if they are not directly included in the source encoder) and secondly security operations, when they are applied at application level. The format for these different application modules should consequently provide information on the binary video stream itself but also the extra control information. In practice, other information such as the size of the packet, or a time information allowing for delays derivation must also be present.

At Network level, one finds first the multiplexing operation, the packetisation according to OSI rule, with adhoc options for the RTP, UDP/UDP lite and IPv6 protocol layers, and then eventually network processing corresponding to header compression, with or without robustness, and eventually security. In reality, and ultimately, the format for these different network modules should correspond to IPv6 packets themselves, together with the control information transported. In a first approach however, it can be considered that the exchange format should simply provide information on the carried payload, together with the extra control information and time information.

SRI (source a priori information) is further information produced by the source coder that are exploited at the destination side and possibly also by the other entities concerned in the JSCC/D chain along the data path at the radio transmitter nodes, in order to optimise the QoS resulting from the decoding process of the video stream.

B. Coding techniques enabling joint optimisation and innovative schemes

In the video compression field the focus will be on the investigation of standard video codecs as well as proprietary modified solutions. These latter codecs will aim for maintaining various degrees of error resilience and/or compression, which are contradictory design factors. As a design alternative, the family of standardised video codecs will be invoked, and rendered robust against transmission errors for instance with the aid of post-coding arrangements. Hence these schemes will retain their standard-compatible structure.

In a first approach, existing source coding standards (*e.g.* MPEG-4, H.263/H.264) will be taken into consideration and their ability to enable joint optimisation will be studied and enhanced when possible through the development of new error resilience tools. In particular, the flexibility provided by scalability and the opportunity of choosing the most proper coding options according to channel conditions will be exploited.

The development of source coding schemes enabling JSCC/D will be done in two steps. The first step will analyse existing schemes and adapt promising ones to the needs of joint source channel coding. In the second step new approaches to joint source channel coding aware coding algorithms will be investigated, in particular adaptive coding techniques (*e.g.* scalable video coding [11][12]). Study and research on source significance information (SSI) that describes the sensitivity of different parts of the compressed data to transmission errors is also an important topic in this context.

Recent studies [18][13] have shown that remarkable gain may be achieved by unequally protecting a video source through convolutional codes. It is thus expected that even better results can be achieved through the use of more powerful state-of-the-art channel codes. In these investigations the entire suite of existing channel codecs will be invoked for protecting the video information. Examples of the schemes to be studied are: turbo codes, LDPC codes, trellis coded modulation (TCM), turbo trellis coded modulation (TTCM), bit interleaved coded modulation (BICM) and its iterative version known as BICM-ID. Similar studies should be performed for the modulation segment. In particular, source and channel adaptive multi-carrier modulation and joint source and channel coding/modulation for adaptive antenna (MIMO) systems will be considered. The best robustness versus throughput and complexity trade-offs will be sought in conjunction with transceiver schemes that should be backwards compatible with existing standard systems, but also capable of providing more flexible, future-proof features.

Naturally, the ability of such codecs to allow a joint optimisation should be deeply investigated. The impact of ciphering, whether at the application level or at the physical level will be investigated, and compromises between the offered security and its cost will be assessed. In particular, the possibility to cipher only given parts of the payload will be considered, following the path opened by JPEG 2000 security forum. The end-to-end video quality perceived by the user will then be optimised through rate-distortion evaluations.

C. *Controlling strategies for joint optimisation of end-to-end multimedia transmission*

The method for delivering between the JSCC/D components of control information, needed for the targeted cross-layer optimisation, is one of the key points as it affects the overall design of the system. The idea is to enable cross communication between layers, to make the system able to adapt to the changes and increase the overall performance.

In jointly optimised systems, the network layer finds itself between the communicating layers that exchange the control information. In order to provide full compatibility with the standardised IP based networks, the delivery of control packets over the network should be done in a way that is compatible with a standard IP stack. The control information should be delivered in IP network as ordinary packets or inside them, if possible. Because the IPv6 networks will be most likely the backbone of future systems, it is also reasonable to find solution that is IPv6 compatible. When looking at other ways to control the network, the QoS mechanisms proposed by IETF (*e.g.* differentiated services architecture) do not focalise on efficient use of network resources, but intend to support end-to-end QoS guarantees. This would emphasise the need of having not only the QoS mechanism, but to have additional ways to express different control messages.

Focussing precisely on the extra information to be transmitted, one finds the source significance information (SSI), source a priori information (SRI) and decoder reliability information (DRI), which are strictly coupled with data stream. To point out that SRI could be transmitted with a lower frequency than SSI. They consequently need to be synchronised with their associated video coding data, possibly by means of hop-by-hop or destination options headers, *i.e.* embedded in the same IP packet, or via extra packets [10]. The CSI (channel state information) and SAI (source a posteriori information) travel in the reverse path direction from the destination to the source nodes. Note that CSI control information does not require fine synchronisation to the data stream, hence does not need to be transmitted so frequently as SSI, DRI or SAI. For these reason they could be transferred by ICMPv6 messages, or extra packets [10]. The transferred control information are generated, collected and processed by the bit level controller that will similarly provide physical layer (de)coder and (de)modulator with configuration parameters and commands.

Application layer controllers will act as an application-level co-ordination tool. The main task for such elements is to select suitable criteria and parameter configurations for video coding and network delivery in order to optimise the usage of channel. They will also profit from Network State Information (NSI) which reports about the availability of resources across the data path and possibly also along the reverse direction. To fulfil this task the controller needs to have input about the quality of the transmission. Possible input parameters include bit-rate, delay, error characteristics, channel load *etc.*. Based on these parameters, the controller makes the decision about the transmission parameters such as delay requirements, priority *etc.*. The controller will also modify the video coding parameters for dynamically adjusting the source properties, so that the video stream characteristics are more suitable for given transmission path, end user terminal and used application.

D. *Expected results and their exploitation*

Research effort in the domain of end-to-end optimisation for multimedia transmission over wireless systems is expected to produce significant benefits for the whole community. From the industry point of view, efficient joint optimisation approaches will allow to propose general and powerful solutions for all type of networks that require bandwidth and QoS simultaneous optimisation strategies over heterogeneous networks and transmission schemes, hence providing improvement for any future wireless systems, and representing a first step for the 4G long-term targets. For the user, the proposed research activity should result in greatly enhanced quality for multimedia communication, potentially allowing the development of currently too complex and time consuming video over wireless systems. Indeed, preliminary results for prototype systems implementing some basic joint optimisation techniques [13][9] let hope for peak video quality gains of about 3 dB in PSNR for equivalent bandwidth occupation. Better results could also be envisaged. Work in progress shows that gains of up to 8 dB in PSNR could be expected in bad channel conditions with respect to conventional schemes. Finally, the questions raised to achieve this joint end-to-end optimisation are highly relevant in the context of new scientific and technological issues, whether on the flexibility aspects, the integration of different system modules for achieving high-quality transmission on the move, or the hope for seamless multimedia communication across heterogeneous networks.

Contributions to improve present (*e.g.* IEEE802.11…) or future wireless standards could be foreseen such as contributions to improve networking (via IETF) or video coding standards (*e.g.* H.264).

IV. CONCLUSIONS

The PHOENIX project main goal is to effectively exploit the available bandwidth on wireless links (4G, WLANs, UMTS…) that is dynamic by nature, providing optimised solution for multimedia transmission over 4G IP-based wireless networks. In this aspect, it is complementary to the QoS mechanisms proposed by IETF (*e.g.* Differentiated services [19] architecture), which do not focalise on efficient use of network resources and acts at IP layer only, although they intend to guarantee an end-to-end QoS. In the same way, the PHOENIX project allows for optimisation of the available bandwidth utilisation on the wireless channel in a dynamic way, supporting fine granularity up to application flow. This way, already deployed systems, like GSM or UMTS, could also be improved thanks to PHOENIX concept, by modifying the board driver to support runtime configuration by other control entities (PHY level controller).

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