



## Joint source-channel coding for 4G multimedia streaming

Leonardo Camiciotti, Catherine Lamy, Lisa Meilhac, Stefano Olivieri, PierGiorgio Verdi

Philips Research  
Via G. Casati 23, 20052 Monza (MI) - Italy  
stefano.olivieri@philips.com

EC-IST JOCO Project Consortium<sup>1</sup>  
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### Subject Area: Air Interface and Spectrum

#### 1. Objectives of the required research

The 4<sup>th</sup> generation of wireless systems (4G) will be a globally integrated communication network interconnecting, in a transparent way, a multitude of heterogeneous networks and systems. A key challenge for the deployment of 4G will be the definition of a flexible re-configurable network architecture that enables simultaneous optimisation of bandwidth as well as Quality of Service (QoS) management.

Significant progress has been made throughout the last decades of 20<sup>th</sup> century to individually optimise each module in modern communication systems. Although excellent results have already been obtained, this separate approach has shown limitations in the case of wireless communications. The challenging task for 21<sup>st</sup> century system designers is to focus on a more integrated strategy. Indeed, optimal allocation of user and system resources may be effectively achieved with the co-operative optimisation of communication system components. This approach, known as joint source channel coding and decoding (JSCC/D), allows for strategies where the source coding, channel coding, modulation, and, possibly, network parameters are jointly determined to yield the best end-to-end system performance. This idea represents the next natural step for 4G system designers. The added value in such a scheme is the possibility offered to let the source coding world and the channel coding world talk to each other, so that they can jointly develop an optimised end-to-end wireless communication link.

The main research items to reach this goal are:

- to develop innovative schemes to enable JSCC/D. This includes the development of flexible channel coding and modulation schemes, the adaptation of existing source coding schemes with respect to their ability for JSCC/D and the development of new ones specifically optimised for this purpose,
- to build a global network architecture based on JSCC/D for future 4G systems. This objective includes the development of JSCC/D controllers which will jointly control the coding blocks and the development of the *network transparency* approach which will allow JSCC/D techniques to be applicable in any kind of network and especially in the 4G ones that should be fully IP-based.

This research activity here described will be the main objective of the EC-IST JOCO project.

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<sup>1</sup> The partners of the JOCO consortium are: Philips, Siemens, VTT, University of Southampton, University of Bologna, CEFRIEL.



## 2. State of the art in the area

### Joint source channel coding schemes

The Shannon '*separation*' theorem [1] proves that, under idealised conditions, source coding and channel coding can, asymptotically with the length of the source data, be treated separately without any loss of performance for the overall system. However, most of the popular applications (e.g. audio/video streaming) do not meet such ideal hypotheses. Indeed, they often require transmitting data with real-time constraints, as well as the bit error sensitivity of source encoded data varies significantly.

Therefore, in order to avoid performance loss, source and channel (de)coders should be designed jointly, which implies they should be able to exchange and use mutual information for co-operative setting of their parameters. As a result, there have been lots of studies made on JSCC, but in practice researchers limited themselves to the case where the source model was precisely defined, so as to be able to attune as finely as possible their source encoder. This attitude was due to the common necessity to show that joint schemes could be simpler or give better performance for the same complexity than separate ones. Various proposed methods in the literature for JSCC are hence in general classified with respect to the underlying communication model they consider. An example of them is the speech codec in GSM (Adaptive Multi-Rate (AMR)) designed just for its specific source and specific channel. Many solutions for audio and video sources are presented in [2], [3].

It has to be noted that none of the schemes, present in literature, does take into account the fact that the data, whether audio or video, will be transmitted over a network and the limitation this one could imply. Beside the obvious problems that may arise due to the possible inadequacy of the optimised scheme to the chosen/existing type of network, other problems may happen, like, for instance, the not foreseen need to transmit additive information through the network protocol layers.

### Source coding schemes

Unequal Error Protection (UEP) can be viewed as a particular type of source and channel coding combination. In particular, the channel encoder uses knowledge of the source bit error sensitivity as well as channel state information, and the source encoder frees up portions of the global bit rate for error protection, in order to adaptively add redundancy. Hence, whenever the data to be transmitted has different importance levels or different sensibilities to noise, UEP is useful.

When the different bit error sensitivities are known for each bit position in the source generated data stream, as it is the case in the majority of the existing audio and video coding, a channel encoder can in theory easily be mapped to the source one. However, adapting in practice the channel coding on such a fine granularity to audio or video sources is complex, partially due to the use of variable length encoding. This explains why schemes describing the bit stream in a hierarchical way have been thoroughly analysed. As a result, the main video coding standards existing today (MPEG-2, MPEG-4, H.263 and H.26L) propose two tools enabling hierarchical bit stream representation:

- *scalability*, which is the property of a source coded representation that allow decoding of appropriate subsets of this representation to generate complete pictures of a reduced resolution (temporal/spatial) and/or quality that commensurate with the proportion of the decoded bit stream,
- *data partitioning*, which is a hierarchical representation for which the more critical parts of the bit stream (such as headers, motion vectors, low frequency DCT coefficients...) are separated



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from the less critical parts (such as higher frequency DCT coefficients...).

The video coding standards existing today thus fit the requirements to enable UEP to some extent, with the restriction that only few levels of protection can be applied. Clearly further work is needed on this topic for a full adaptation to JSCC/D to achieve the potential gain offered by this technique.

## **Channel coding and modulation schemes**

Widely used in the wireless communications world, channel coding ensures satisfactory error level by adding redundancy to the information bit stream to combat noise and fading effects. The design of an error-correction code usually consists in selecting a fixed code with a rate and correction capability matched to the protection requirement of the data to be transmitted and adapted to the channel conditions. However, if the data stream has different error protection needs and is transmitted over time varying channels, more flexibility is required from the coding schemes. Therefore, it is desirable that the code rate may change, so that the correction power of the code can be adapted to the source and channel needs. Rate Compatible Punctured Convolutional (RCPC) codes [4] meet such a goal, since they allow both the transmission of incremental redundancy in ARQ/FEC schemes and continuous rate variation to change from low to high error protection within a data frame.

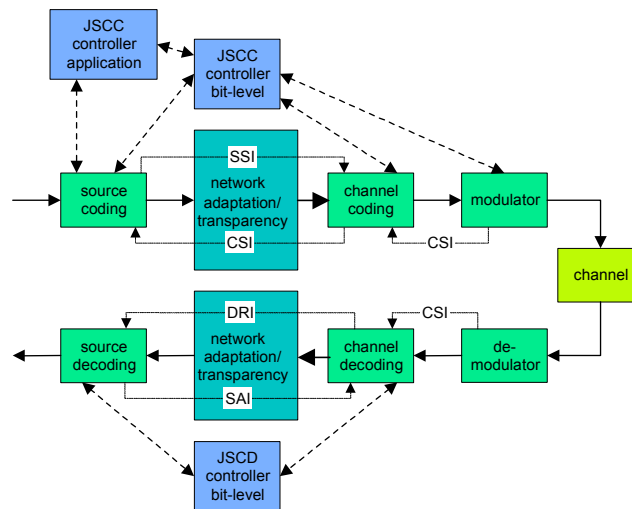
Another option to exploit the redundancy of the source encoder output is to use a suitable coded modulation (possibly using non-uniform or asymmetric signal constellations) combined with multistage decoding. For example, in [5] either multiplexing different coded signal constellations or partitioning the signal constellation into disjoint subsets is considered. In some cases, it is also possible to treat source coding, channel coding and modulation as a whole. For instance, in [6] Joint Trellis Coded Quantisation and Modulation (TCQ, TCM) is investigated, where the same trellis is used for the TCQ and the TCM. Multi-resolution modulation techniques (MRM) have also been considered for broadcast applications, in order to cope with different reception conditions for different receivers, as well as their possible combination with multilevel coding.

## **3. Possible approach**

### **Overall Network Architecture Concept**

An *innovative global network architecture* introducing the concept of *JSCC/D controllers* may be specified. Its modules are the key elements of the architecture deploying a strategy suitable to achieve adaptive/dynamic optimisation (efficiency/robustness) of end-to-end wireless links for the transport of interactive high-speed packet data (*i.e.* video and graphics). Developing such a system concept represents a significant innovation, as multiple components must be jointly controlled (see Figure 1).

A co-ordinating tool, named *JSCC controller*, which implements suitable controlling strategies, is necessary. Its task is to drive an efficient and powerful transmission scheme, which involves a scalable source encoder implementing data partitioning tools, a dynamic channel encoder designed to provide selective protection of the transmitted data, and a modulation adapted as much as possible to the channel.



**Figure 1:** Information exchange between source (de)coding and channel (de)coding/(de)modulation.

Source Significance Information (SSI), giving information about the sensitiveness of the source to channel errors, may be passed from the source encoder to the channel encoder to enable UEP. In order to adapt source and channel coding rates to channel conditions, it appears necessary to provide Channel State Information (CSI).

At the receiver side, the source decoder may use Decoder Reliability Information (DRI) received from the channel decoder to perform soft input variable-length codes (VLC) decoding [7], and Source *A posteriori* Information (SAI) may be passed from the source decoder to the channel decoder to perform source-controlled channel decoding, and, possibly, iterative channel-source decoding.

It is expected that separating the controller into an application-level one and a bit-level one will provide the needed level of flexibility. The latter deals best with a bit-by-bit optimisation, while the former selects suitable criteria and parameter configurations based on the application and the intended network protocols. The design of flexible source and channel (de)coding/(de)modulation schemes in conjunction with network transparency tools are enabling conditions to achieve the goal.

### Source coding enabling JSCC/D techniques

Both existing and new source coding schemes have to be investigated as far as their suitability for JSCC/D is concerned. It is indeed not clear whether existing source coding schemes (MPEG-2, MPEG-4, H.26L) are suitable for a full deployment of JSCC/D. Source coding schemes should be able to adapt themselves fast to channel conditions in terms of bandwidth. That would ensure that some faulty bits do not incur a significant quality loss. To enable a fine granular adaptation of the channel encoder to the bit error sensitivity, hierarchical data representations (scalable video coding and data partitioning techniques) must be developed without sacrificing performance too much. This includes efficient descriptions of the relevant side information. Coding schemes with monotonically decreasing sensitivity or analytically describable sensitivity curves are envisioned. Fine granular UEP schemes must be developed and employed as a means to adapt varying bit error sensitivities to the powerful channel encoders.

### Innovative flexible channel coding and modulation schemes

Several coded modulation schemes should be investigated, in order to evaluate the suitability of



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their use in the JSCC/D framework. Besides the flexibility aspects, it should be taken into account the fact that data have to be transmitted over a wireless radio channel heavily conditioned by propagation effects. The additive information (*e.g.* synchronisation, equalisation, *etc.*) generally required to ensure a reliable transmission, diminishes drastically the gain initially obtained by JSCC schemes. The challenge is then to jointly consider the necessity of this additional information together with the JSCC optimisation. Another important challenge is to investigate the possibility to use UEP in the modulation domain, not only by considering multilevel modulations but also the opportunities offered by multi-carrier modulations, like OFDM, and the eventuality to develop systems compatible with the existing 2<sup>nd</sup> and 3<sup>rd</sup> generation CDMA systems.

## **Network transparency**

For implementing JSCC/D, information must be exchanged between the different blocks of the transmission scheme. Therefore, information must be passed through different levels of networking protocols without interfering with the regular use of the network. Called network transparency, this capability constitutes a key research item. An important objective is to keep the system backward compatible. The same situation holds for the decoder side, where soft values of the channel decoding process should be passed to the source decoder to improve decoding performance along with source *a posteriori* information back to the channel decoder.

## **Suitable controlling strategies for joint source channel decoding**

It is critically important to develop suitable controlling strategies on the decoder. Indeed, on one side *a priori* and *a posteriori* knowledge of the source must be made available to the channel decoder and on the other side all received information along with soft information must be passed to the source decoder (allowing it to implement soft input VLC decoding). This implies the transmission through the network layers of soft information, and hence the use of network transparency at the source decoder too. This technique may improve the performance of the variable length codes, which was limited until very recently by their classical hard decoding. As a consequence, whereas the packets with false CRC are currently discarded, causing a procedure of error concealment to take place, the source decoder will have the possibility to exploit the soft information generated by the encoders. This could be achieved by using a network protocol that will let the damaged packets go through the layers together with their corresponding soft information frames down to the decoding layer. Beside the gain this technique will achieve, another interesting point is that it is backward compatible.

## **4. Expected results**

Research effort in this field (*e.g.* the JOCO project) is expected to produce benefits of great importance for the community.

### **For the industry**

Significant benefits are expected for the European industry of communication equipment, going even beyond the 4G long-term target. In fact the JSCC/D approach is a general and powerful solution for all type of networks that require bandwidth and QoS simultaneous optimisation. Therefore, it would help to the evolution of the 3G system and would also be applicable in all type of wireless networks including domestic networks.

### **For the user**

The proposed research activity is supposed to have significant impact on the quality of multimedia communication, which has not widely accepted yet in the mobile environment because of quality limitations. Preliminary simulations of prototype systems adopting some basic JSCC techniques for





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the transmission of coded video over wireless channels show peak video quality gains of about 3db in terms of PSNR, or alternatively, the possibility to double the spectral efficiency.

Also, a seamless multimedia communication across different networks has still not become true. Therefore, in the context of 4G, targeting flexibility aspects and the integration of different system modules moves to the direction of allowing high-quality ubiquitous multimedia communication for a large number of applications.

### **For the scientific and technical advancement**

The scientific and technological issues to be studied, being highly relevant, would teach and mentor a large number of experts for European companies and universities.

### **5. Time frame to get the expected results**

The experience of previous generation wireless systems shows that today is a very appropriate time to elaborate new concepts and architecture for 4G. As a matter of fact, second and third generation European wireless communication systems standards (GSM and UMTS) were released more than 10 years after the introduction of the first concepts. It is expected that the development of such an innovative JSCC/D optimised global network architecture will be ongoing for the same timeframe, in order to follow changing scenarios and to incorporate any new developments relating to sources, network protocols and physical layers.

It is our belief that the development of a new architecture around the network transparency layer concept and the JSCC/D optimisation would have strong commercial outputs in terms of licensing of standardised technology and through the marketing of innovative and timely available 4G products.

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