Analysis and Optimization of a JSCC/D System on 4G Networks

C.Lamy-Bergot, G. Panza, A. Rotondi and L. Fratta

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.

***Index Terms* — 4G, JSCC/D, Modeling, Optimization, Performance, Evaluation, Simulation Analysis.**

I. Introduction

EFFICIENT and reliable wireless connection is crucial to meet the on-going demand for access "anywhere and anytime", but leads to facing the critical problem of band availability. Possible solutions for this problem are, at radio access level the flexible allocation of bandwidth, and on the overall transmission chain the joint adaptation of source and channel (de)coding, as analyzed in previous research works such as [9][10]. The generalized joint approach allows for strategies in which the choice of channel code, modulation, or network parameters varies with the source characteristics, as presented in [11]. One of the main drawbacks and implementation difficulty of the joint approach is that it requires the exchange of a variety of information between the systems blocks. Such information is used to perform the system optimization, which lead to have joint approach is often discarded as impractical for real systems. However, the IST FP6 PHOENIX project has proposed an original JSCC/D (Joint Source and Channel Coding/Decoding) system [1][8], that was declined in a real test-bed proving the feasibility of the architecture. To be extended in IST FP7 OPTIMIX project begun in March 2008 in a point to multi-point context, PHOENIX approach proposed innovative solutions enabling enhanced video streaming in a point-to point IP based wireless heterogeneous system.

The efficient communication and feasibility of the joint optimisation is made possible in PHOENIX system via the use of joint controllers which collect quality feedbacks (channel state information (CSI), network state information (NSI) …) and update the working parameters of modules at the transmission side accordingly. This paper presents PHOENIX architecture, and provides a short description of the key modules before giving a detailed analysis of the signalling proposed to perform the cross-layer exchanges transparently for the network and assessing the application controller behaviour with respect to the video quality perceived by the user at the receiver side. The analysis is made for two different radio technologies (UMTS and WiMAX), and trade-offs for the configuration parameters are investigated with the aim of maximizing the visual quality. Finally, a comparison with traditional and other JSCC/D systems is also provided, before conclusions and future work.

II. PHOENIX System Architecture Overview

PHOENIX JSCC/D functional architecture [1][8], is shown in Fig. 1. The depicted architecture includes the transmitter side, in the upper part, and the receiver side in the lower one, as well as the control/signalling messages which are shown transferred between the modules and with the controllers. Differently from the traditional architecture in use for multimedia transmission (i.e. without joint adaptation of source and channel coding), there are two different controllers: the Application Controller and the Physical Controller. These controllers manage the (de)coders, (de)modulators and the (de)compression modules adapting them to the network conditions (both wired and wireless links). Information about the network and radio access, such as jitter, packet loss, packet error rate (PER), bit error rate (BER), etc., is carried by the control/signalling messages and provided to the controller for optimisation.

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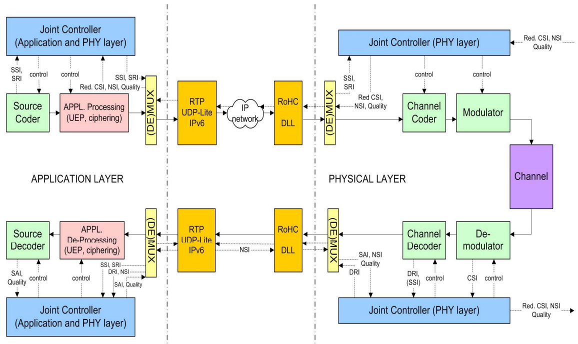

Fig. 1. PHOENIX JSCC/D architecture.

A. *Signalling and controllers*

The achievement of the end to end joint optimisation proposed in PHOENIX is ensured by the control/signalling information messages that inform the controllers of the current communication link states, allowing to dynamically update the source codec, channel codec and modulator settings in order to improve the system overall performance. These control messages are the cost to pay to achieve the adaptation of the system to the transmission conditions, and are of four different types (see Table I for respective transmission mechanisms). Firstly, the Channel State Information is sent through the network by each wireless receiver and contains information about the radio channel conditions such as BER and PER. Then, the Network State Information (NSI) contains information about the IP network such as delay and packet loss. Then are found the Source a-priori Information (SRI) messages, and Source Significance Information (SSI) message, which are generated by the source coder to respectively help the source decoding process or allow unequal error protection techniques. Two more specific information signals (Decision Reliability Information (DRI) and Source A-posteriori Information (SAI) messages, sent via IP packets) have also been specified, whose role is to allow the implementation of soft-input soft-output decoding at the receiver side for channel and source decoder and foreseen to be used only if the wireless receiver is the end-point of the communication.

With such information available in a unique monitoring equipment (application controller), it was proposed to implement in said equipment more or less complex optimisation strategies to select the best parameters to be used jointly by the different modules of the chain for the current time step (see [8] for more details). The final criterion being the end-user video quality, the objective is to provide the best possible performance for given transmission conditions, typically by adapting source coding parameters and packetization, in conjunction with a given ciphering method [8]. In the simple setting considered in the following, the application controller role is to adapt the video coding parameters at the beginning of each operation cycle. In a more complex configuration would be optimised jointly the insertion of protection at transport or radio access level (more compression when more protection is needed and conversely to meet fixed bandwidth usage). In practice, reading the feedback information about the packet loss, the application controller decides on reducing at minimal rate the transmission (if a threshold is overcome) or on estimating the PSNR (Peak Signal-to-Noise Ratio) of the received video by a simple non-linear model, whose coefficients are calculated employing the least square method, and having channel BER and PER as parameters. Then, if the difference between PSNR of transmitted video and its estimate exceeds a given threshold (hence, an approximation of the PSNR is sufficient), the source quantization parameters are changed to increase or decrease the source rate. In the general PHOENIX model, for the case where the radio access is not a standardized one, another controller is also added, called physical layer controller, which is subordinated to the application layer. This even more complex configuration allows to further enhance the tuning by driving the radio access (channel coding and modulation) parameters on a short time basis (order of tens of milliseconds), while the application controller works over long scale phenomena (up to seconds). In the following tests, standard UMTS and WiMAX access were considered, hence the physical controller was not deployed.

B. *IP network and radio access*

In the transmission chain, the presence of a wired IPv6 network is also included in the system analysis. It is modelled as an IP cloud composed of a configurable number of nodes introducing delay and loss. This wired network allows to take into account the presence of a LAN or an autonomous system crossing. More specifically, the modelling of loss and delay is based on statistical distributions (Uniform and Gamma, respectively), properly parameterised to fit well real world empirical data [2]. Below the Internet layer, the packets are handed to the radio access, which includes data-link and PHY layers. In said layers, no complete physical controller was introduced, but critical modifications were done to ensure that the joint approach for multimedia streaming is taken into account: namely, packets with errors only in the payload are not discarded thanks to the limitation of the MAC CRC (cyclic redundancy check) to the packet header, including the extended header carrying control information such as SRI and SSI. In practice, the link layer provide in this manner unequal error detection, as in the solution enhancing the IEEE 802.11 standard in the multimedia delivery case proposed in [5]. Such modifications were applied to 802.16 (WiMAX) [6] and UMTS [7], radio access technologies considered in our work.

C. *Other supported features*

Finally, it must be noted that the PHOENIX global architecture has been designed to be compatible with different source coding schemes. That feature was validated using either MPEG-4 Part 2 video codec or the more recent H.264/AVC standard [3], including with its new temporal scalability functionality [13] and partial ciphering [12] extensions developed to ensure a more resilient and more secure source coder. It was also shown that the sensitivity models developed for both H.264/AVC and MPEG-4 Part 2 to apply efficient unequal error protection were compatible with application controlling strategies thanks to the SRI messages distribution. In the following an MPEG-4 Part 2 codec is considered.

III. Numerical Results

The model relying on PHOENIX architecture described in previous section was implemented and run under the OPNET simulator modeller environment [4] to provide an assessment at architectural level. Using a low mobility setting, corresponding to a use case 'video conferencing from a café [1], results were collected over several simulation runs (about 10 different seeds were used) for each configuration settings. The IP network was composed of 8 IP routers introducing each an average delay of 17.775 ms (a bound of about 150 ms was imposed for the 99th percentile of the network delay, as suitable for real-time applications) and a loss of 1800 ppm (resulting in an acceptable end-to-end network loss for video transmission) at the output interface. A single wireless hop was present, with either an UMTS or a WiMAX radio channel.

Table I. Control/signalling message overheads (for different refreshing periods, when applicable)

Message	Size (*Bytes*)	Transmission mechanism	Overhead
CSI	20	ICMPv6	560 *Byte/s* for 50 *ms*; 140 *Byte/s* for 200 *ms*; 28 *Byte/s* for 1 *s*
NSI	36	Report RTCP/ICMPv6	215 *Byte/s* for 250 *ms*; 80 *Byte/s* for 1 *s*; 60 *Byte/s* for 2 *s*
SSI/SRI	8	IPv6 Extension Header	2,5 *KByte/s* for 448 *Kbps*, 30 *fps*; 1,46 *KByte/s* for 271 *Kbps*, 15 *fps*; 1,3 *KByte/s* for 189 *Kbps*, 7,5 *fps*

A non- selective block (slow) fading channel with additive white Gaussian noise (corresponding to fading samples uncorrelated and log-normal distributed), with 10 ms of coherence time, sample period for fast fading gain of 1 ms, Doppler frequency for time correlated Rayleigh fading of 5 Hz and mean SNR ranging from 1 to 8 dB, was implemented. The source was MPEG-4 Part 2 encoded with a maximum average coding rate of 448 Kbps. To properly evaluate the Quality of Service (QoS) perceived by the user the following statistics were collected.

- Throughput (Byte/s): (average) rate of traffic received by end users.
- E2E Packet Loss: amount of total losses in the network.
- E2E Delay: overall delay from transmitter to receiver.
- PSNR: PSNR of the received video that is an objective quality estimation.

These statistics were then used to evaluate the overall system behaviour over the issue of control/signalling overhead cost, Application Controller reaction time, impact of loss and delay of feedback information and crossing of multiple wireless hops, in order to propose the best trade-offs to optimize the achieved performance.

A. Control/Signalling overhead

The control/signaling overhead introduced by the specific control messages and headers is the cost to pay for using a JSCC/D system instead of a traditional one to transmit multimedia data and does not depend on the specific wireless technology. The recommended encapsulation methods and the resulting overhead for each control/signalling information are reported in Table I.

Analysing such data, some remarks can be made.

1) Concerning CSI and NSI messages. Generated periodically, they are sent uplink from the wireless receiver and were tested with different refreshing periods corresponding to a different amount of overheads. From Table I, it appears that a good compromise is 200 ms for CSI and 250 ms for NSI, which entail nearly negligible overheads of 140 and 215 Bytes/s, respectively, and allow for a sufficiently accurate updating of the channel and overall network conditions. Setting shorter refreshing periods, in order to better follow the channel and network variations, would not help, because the transmission delay could make the information received out of date.

2) Concerning SSI and SRI messages. Strictly related to the multimedia stream, for instance providing the coding rate and characteristics of the source, this information is encapsulated in IPv6 extension headers to be easily used for differentiated services or unequal error protection. It is worthwhile to point out that in IPv4, the header options could be used for the same purpose. The gathered results indicate an overhead not greater than 5% of the traffic transmitted over the network, which is a small cost to pay with respect to the improvement such differentiation information can provide [11].

B. Application Controller reaction time

Application Controller reaction time is the time interval between two adaptation processes. This adaptation speed is consequently a primordial parameter that affects the overall system performance as the modules (here source encoder only) settings are unchanged during the interval, even if the transmission conditions vary.

In the considered case, the good QoS for the user was defined as a compromise between high throughput and PSNR. To establish good trade-offs for this reaction time values, 60 s simulations were run, during which the channel quality (i.e. the Gaussian SNR) changed according to the pattern reported in Table II.

Fig. 2, Fig. 3 and Fig. 4 were obtained with transmission over UMTS radio technology, and first confirm that the shorter the reaction time, the higher the adaptation ability to the channel status. Moreover, it is observed that the 5 s reaction time achieves the highest throughput (~70 KByte/s), while the 2 s reaction time is a better compromise as it allows to gain of more than 3 dB in PSNR with a loss of just about 6 KByte/s of throughput when compared to the 5s case.

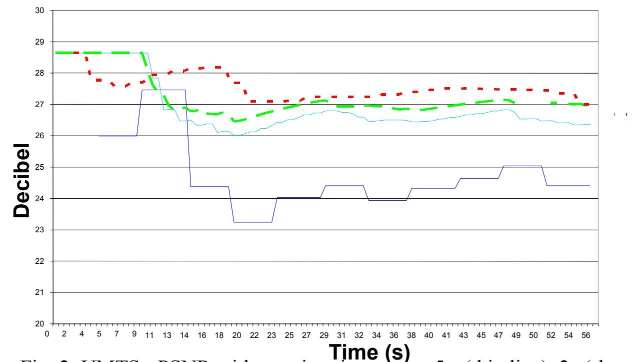

Fig. 2. UMTS - PSNR with reaction time set to: 5 s (thin line), 2s (short dashed line), 1 s (thick line), and 0.5 s (long dashed line).

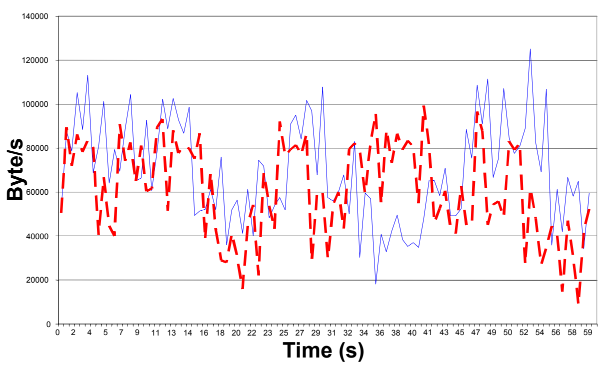

Fig. 3. UMTS - Throughput with reaction time set to: 5 s (solid line), 2 s (dashed line).

Table II. Channel status along 60 s simulations

Sim.Time (s)	0–10	10-20	20–30	30–40	40–50	50–60
Ch. Status (dB)	8	1	8	4	8	1

Table III. System performance for both UMTS and WiMAX technologies

Radio Techn.	React time (s)	Thr. (KB/s)	Loss (%)	Delay (ms)	PSNR (dB)
UMTS	5	69.24	0.19	172	24.5
	2	61.54	0.24	170	27.4
	1	50.68	0.22	164	26.7
	0.5	45.25	0.24	165	27.0
WiMAX	5	70.51	0.19	148	24.5
	2	64.13	0.20	145	26.8
	1	49.62	0.23	142	26.0
	0.5	37.05	0.21	143	27.0

With shorter reaction times (e.g. 100 and 200 ms reaction times), the loss of frames due to quite fast quantization parameter variations causes a reduction in throughput (2.2 KByte/s and 15 KByte/s respectively). Furthermore, a short reaction time does not always allow to average the bursty losses (in particular on the radio interface or in case of network congestion), which in our model leads to choose minimum rate due to threshold on packet loss.

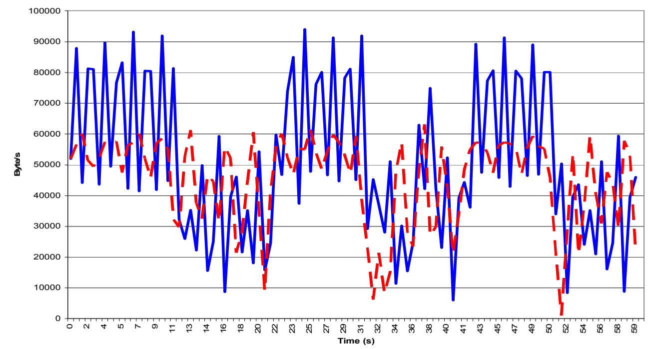

Fig. 4. UMTS - Throughput with reaction time set to: 1 s (solid line), 0.5 s (dashed line)

Table III reports the values of interest for both UMTS and WiMAX technologies. It is worthwhile to highlight that the highest value of PSNR is reached when the source codec operates with 2 s of reaction time, when just negligible details of the original video are eliminated.

C. *Effect of CSI and NSI loss and delay*

The availability of feedback information is a main basis for a JSCC/D system. Indeed, lost or excessively delayed CSI and NSI make the Application Controller unable to adapt to actual network conditions. Table IV presents the set of different configurations settings used for the wired part of the telecommunication infrastructure (i.e. loss and delay at the single router interface) and wireless channel (modelled by a non-selective block fading channel with additive Gaussian noise) that have been used for tests. The reaction time of the controller was set to the optimal value of 2s, and CSI and NSI refreshing periods to 200 and 250 ms respectively (see section III-A). Table V reports the resulting CSI and NSI loss and delay for each case under analysis. As expected, the system performance decreases when loss or delay on CSI and NSI increase, in particular when the radio channel status becomes bad. In such a case, the Application Controller should really adapt fast. Also when channel conditions improve the source coding rate should be augmented rapidly in order to well exploit the available transmission resources and maximize the QoS. Results collected are shown in Table VI and Table VII.

From those two tables, it appears that the main difference between WiMAX and UMTS is on the End-to-End packet delay. WiMAX technology allows to obtain an average reduction of about 22 ms, which is beneficial for both the control/signalling and data traffic. However, just a slightly higher PSNR, about 0.5 dB on average, is obtained with WiMAX, as the slotted reaction time of the Application Controller tends to smooth the difference in performance.

D. *Crossing of Multiple Wireless Hops*

In general, multiple wireless hops might be crossed into the network, for example corresponding to the radio interfaces of two user terminals (e.g. UMTS handsets) attached each to a different base stations which are interconnected by a fully wired telecommunication infrastructure. This scenario differs from the single wireless hop case mainly for the way the control/signalling information has to be processed by the Application Controller. While there are no issues for NSI (it depends only on the whole route through the network), in such a scenario it is necessary to decide how to manage the different CSI messages coming from the two wireless hops. Indeed, a CSI message is referred to a single wireless channel status. Application Controller has consequently to manage properly the state information of two different radio interfaces. Furthermore, CSI messages follow different path segments into the network and experience different delays.

Two different approaches were considered for the CSI information processing, corresponding to the Application Controller taking either the worst value or the average value of the two CSI, in terms of impact on the estimated PSNR of the received video while of course the actual PSNR quality results from the effect of both the concerned wireless hops. IP interfaces and UMTS channels were as in previous section, with here a larger IPv6 wired network composed of 10 routers. Simulations were run for several configuration settings (using about 10 seeds for each of them), with different combinations for the status of the two wireless channels and maximal rate of the MPEG-4 Part 2 source set to either 256 or 448 Kbps.

The collected results, including also the packet loss and error rates on the wireless hops, as expected show that the way Application Controller processes feedback information has a dramatic impact on the system behavior only when the two wireless channels are substantially different in SNR. On the whole, the achieved performance doesn't change significantly. Nevertheless, in most cases the best choice is to take the average value, mainly to allow the Application Controller to increase the coding rate when possible. Moreover, such a choice contributes to filter short-term variations of the network status.

The system performance with or without the Application Controller was also analyzed for the considered multiple wireless hop scenario, with reaction time and updating periods of feedback messages set to the optimal values as specified in sections III-A and III-B.

Table IV. Scenario specification

Scen.	Ch. Status (*dB*)	Interf. Delay (ms)	Interf. Loss (%)
1	1	10	0.0001
2	1	20	0.001
3	1	50	0.01
4	1	100	0.1
5	4	10	0.0001
6	4	20	0.001
7	4	50	0.01
8	4	100	0,1
9	8	10	0.0001
10	8	20	0.001
11	8	50	0.01
12	8	100	0.1

Table V. Loss and delay for CSI and NSI messages

	CSI			NSI		
Scen.	Mean delay (*ms*)	Delay Std.Dev. (*ms*)	Loss (*pkt/s*)	Mean delay (*ms*)	Delay Std.Dev. (*ms*)	Loss (*pkt/s*)
1	85	50	0	90	50	0
2	200	80	0	200	60	2.5
3	480	90	0.26	480	80	0.33
4	1000	100	0.016	950	150	1.2
5	85	60	0	85	30	0
6	208	60	0	210	50	0.16
7	500	70	0.33	490	70	0.41
8	1000	60	0.83	980	100	1.2
9	100	50	0	110	40	0
10	220	40	0.16	215	30	0.16
11	600	40	0.26	600	50	0.33
12	1000	100	1	1100	100	1.2

Table VI. System performance for WiMAX

Scen.	Thr. (KByte/s)	Loss (*%*)	Delay (*ms*)	PSNR (*dB*)
1	40.34	$8.67 \cdot 10^{-4}$	104	23.4
2	32.23	$8.50 \cdot 10^{-3}$	210	20.5
3	23.45	$8.32 \cdot 10^{-2}$	515	22.0
4	12.67	1.14	1010	24.2
5	42.54	$8.32 \cdot 10^{-4}$	102	25.5
6	45.34	$8.31 \cdot 10^{-3}$	202	24.0
7	24.32	$8.22 \cdot 10^{-2}$	504	23.5
8	11.02	1.02	1007	25.3
9	70.13	$8.13 \cdot 10^{-4}$	105	28.2
10	68.56	$8.46 \cdot 10^{-3}$	212	28.0
11	50.03	$8.58 \cdot 10^{-2}$	525	27.5
12	12.20	0.91	1010	27.1

Table VII. System performance for UMTS

Scen.	Thr. (KByte/s)	Loss (*%*)	Delay (*ms*)	PSNR (*dB*)
1	40.00	$8.69 \cdot 10^{-4}$	110	23.3
2	31.36	$8.41 \cdot 10^{-3}$	220	20.0
3	22.54	$8.32 \cdot 10^{-2}$	530	22.5
4	10.15	1.20	1012	23.2
5	42.32	$8.36 \cdot 10^{-4}$	108	25.4
6	44.34	$8.30 \cdot 10^{-3}$	215	24.5
7	22.34	$8.24 \cdot 10^{-2}$	516	22.0
8	12.00	1.00	1010	25.5
9	70.10	$8.04 \cdot 10^{-4}$	108	27.5
10	68.02	$8.20 \cdot 10^{-3}$	220	28.0
11	49.82	$8.14 \cdot 10^{-2}$	540	27.0
12	12.00	0.91	1012	27.0

Concerning the End-to-End Delay, its standard deviation in presence of the Application Controller is lower and jitter is an important issue for real-time multimedia applications. Moreover, the average value is 100 ms with the Application Controller, instead of 105 ms. The benefit of the adaptation is even more evident looking at the End-to-End Packet Loss, which is about either 0.18 pkt/s or 0.45 pkt/s, with or without the Application Controller, respectively. Similar considerations can be made for the packet loss and error rates on the wireless channels. Throughput with the Application Controller was lower than in absence of it (about 25 and 35 KByte/s on average, respectively) due to the fact that in case of poor channel conditions, the Application Controller reduces the coding rate with the aim to achieve a better quality for the end-user (improved PSNR of several dBs).

IV. Comparison with a Traditional and Other JSCC/D Systems

To better evaluate the improvement provided by our joint optimisation approach, a comparison with other systems on the basis of a similar test scenario with an MPEG4 coded source, a single wireless channel (as described in section III) and UMTS radio technology, is proposed. For our system, results were collected with the optimal parameter settings (as specified in sections III-A and III-B). Being the assessment based on user perceived quality, PSNR is the reference parameter for performance comparison. The signalling overhead is not considered since it is negligible for all the issued JSCC/D systems.

Table VIII reports the PSNR results obtained when comparing our system in adaptive and traditional (i.e. not adaptive, with fair fixed source coding settings) modes, with source coding rate of either 448 or 256 kbps in bad, fair and good channel conditions. As expected, the benefit of a JSCC/D system is more evident with a bad channel, when it is really effective to adapt the coding rate. It is worth noting that the 256 kbps source achieves higher PSNRs than the 448 kbps source due to the different impact of errors on the channel.

In [9], an analysis is provided for a JSCC/D proposal on a channel of 6 dB of SNR. Collected results show a maximum value of 22.5 dB, while in our system 27.9 dB is registered for PSNR with only 4 dB of SNR. With 10 dB of SNR, 28 dB of PSNR is achieved, value that is reached with PHOENIX proposal already at 8 dB of SNR.

In [10], performance statistics for a different JSCC/D system are reported. In that proposal, a fair channel status of 4 dB allows a PSNR of 25.6 dB, lower of 2.3 dB than the value achieved by the system assessed in this work in the same conditions. Such difference reaches nearly 2 dB, when considering a better channel status (SNR of 8 dB).

Table VIII. PSNR (dB) for a traditional system and our JSCC/D proposal

Ch. Status (*dB*)	1	4	8
448 *kbps* source	20.3	24	26
256 *kbps* source	21.1	26.5	27
JSCC/D System	25.7	27.9	28.6

Typically, a 2 s Application Controller reaction time and refreshing periods of respectively 200 and 250 ms for CSI and NSI messages have shown to provide a good PSNR in both good and bad network conditions. Such a choice ensures robustness to delay and loss of feedback messages, thanks to the implicit filtering process (NSI and CSI feedback messages being transmitted at higher rates, some can be lost without critical impact). This 2 s reaction time also allows to average sensitivity measurements over the time and avoids trying source coding adaptation based on micro-variations of the transmission conditions (e.g. shadowing effects), that would result in degraded video quality due to fast quantization parameters variation. Regarding the crossing of multiple wireless hops, a better choice is to average the CSI of the concerned radio channels, rather than taking the worst status as the reference. Indeed, a less conservative system can more effectively and fully exploit the available network resources for an improved PSNR.

V. Conclusions and Future Work

This paper presents the innovative JSCC/D system designed into the framework of IST FP6 PHOENIX project [1][8] and provides assessment on several critical issues that are feedback information overhead, reaction time of the Application Controller, impact of loss or delayed feedbacks and crossing of multiple wireless hops. This analysis allows to evaluate fairly the cost and benefits of the joint source and channel coding PHOENIX system, as well as propose some trade-offs between the configurations parameters in the considered application controller mode, to optimize the QoS and resource utilization.

With the optimal configuration settings of reaction time and refreshing periods, comparisons in terms of PSNR for video transmission show that our system outperforms other JSCC/D proposals [9][10], as well as itself in non-adaptive mode. Collected results have shown better performance with WiMAX technology, in particular in terms of delay. Finally, similar results and conclusions have been obtained for different application and network scenarios tested in the PHOENIX project framework, corresponding to more or less demanding solutions in terms of mobility and bandwidth.

Future work, foreseen in the frame of IST FP7 OPTIMIX project, will include such assessments of critical messages and adaptation means in a point-to-multipoint scenario. In particular, a novel critical issue to be considered is the way feedback information related to different users are generated, transmitted, possibly aggregated into the network and processed by the Application Controller.

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