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An Architectural Analysis and Evaluation of a JSCC/D System on 4G Networks

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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 undertaken 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.

Keywords: 4G, JSCC/D, Modelling, Optimization, Performance, Evaluation.

1. 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 [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 a joint approach is often discarded as impractical for real systems. However, the FP6 PHOENIX project has proposed an original JSCC/D system [1][8], that was declined in a real test-bed proving the feasibility of the architecture. To be extended in the FP7 IST OPTIMIX project in a point to multi-point context beginning in 2008, the 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 the 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 the 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 visual quality. Finally, a comparison with traditional and other JSCC/D systems is also provided, followed by conclusions and future work.

2. PHOENIX System Architecture Overview

The PHOENIX JSCC/D system concerns both the transmitter and receiver sides, and requires the transfer of control/signalling messages between relevant modules, such as the controllers (see [1][8] for a basic functional scheme and details). Differently from the standard architecture in use for multimedia transmission, there are two additional controllers: Application Controller and 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.

2.1 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 it 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 1 for respective transmission mechanisms). Firstly, the Channel State Information is sent to the controllers 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) processed by the Application Controller only contains information about the IPv6 network such as delay and packet loss. The Source a-pRiori Information (SRI) messages, and Source Significance Information (SSI) message, which are generated by the source coder 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 IPv6 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. 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 the packetization, in possible conjunction with ciphering. 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 the insertion of protection at transport or radio access level (more compression when more protection is needed and conversely to meet fixed bandwidth usage) would be optimised jointly. 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 by employing a model of the decoded video quality with channel BER and PER as parameters. Then, if the difference between PSNR of transmitted video and its estimate exceeds a given threshold (hence, a good 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 adaptation by driving the radio access (channel coding, modulation) parameters on a short time basis, while the application controller works over long scale phenomena. In the following tests, standard UMTS and WiMAX access were considered so the physical controller was not deployed.

2.2 IP network and radio access

In the transmission chain, the presence of a wired IPv6 network, modelled as an IP cloud composed of a configurable number of nodes crossed by IPv6 packet that introduce delay and loss, is also included in the system analysis. It facilitates taking 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 (Gamma distribution for the latter) properly parameterised to fit real world empirical data [2] well. Below the Internet layer, the packets are handed to the radio access, which includes data-link and PHY layers. In these 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, the packets with errors only in the payload are not discarded due 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 provides unequal error detection in this manner, as in the solution enhancing the IEEE 802.11 standard in the multimedia delivery case [5]. Similar modifications were applied to 802.16 (WiMAX) [6] and UMTS [7] radio access technologies considered in the tests.

2.3 Other supported features

Finally, it must be noted that the PHOENIX global architecture has been designed to be compatible with different source coding schemes, such as the most 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, as well as the second most recent one MPEG-4 Part 2. 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 due to the SRI messages distribution. In the following section an MPEG-4 Part 2 codec is considered.

3. Numerical Results

The model relying on the PHOENIX architecture described in the previous section was implemented and run under the OPNET simulator modeller environment [4] to provide an assessment at an 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 for each configuration settings.

The IP network was composed of 8 IP routers introducing each an average delay of 17.775 ms and a loss of 1800 ppm at the output interface. A single wireless hop was present, with either an UMTS or a WiMAX radio channel. A non-selective block (slow) fading channel with additive white Gaussian noise (the fading samples are 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 MPEG4 coded 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): amount of traffic received by end users;
- End-to-End Packet Loss: amount of total losses in the network;
- End-to-End 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 and impact of loss and delay of feedback information, in order to propose the best trade-offs to optimize the achieved performance.

3.1 Control/Signalling overhead

This is the drawback of using a JSCC/D system instead of a traditional one to transmit multimedia data and does not depend on the specific wireless technology. The most suitable encapsulation method and the related overhead for each control/signalling information are reported in Table 1.

Table 1 - 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*

Following those observations, 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 1, 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 quite 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. 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.

3.2 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 case considered, the good QoS for the user was measured as a compromise between high throughput and PSNR.

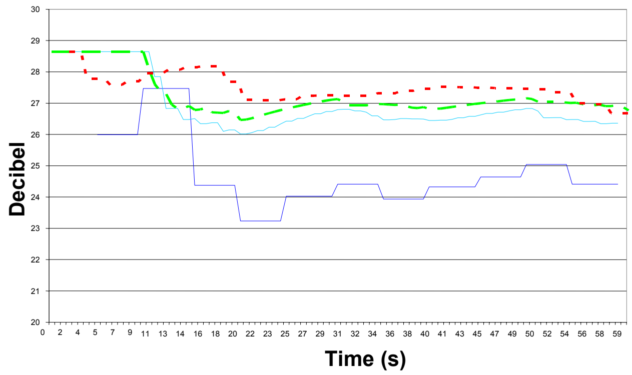

Figure 1 – UMTS – PSNR with reaction time set to: 5 s (thin line), 2s (short dashed line), 1 s (thick line), and 0.5 s (long dashed line).

Table 2 - 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

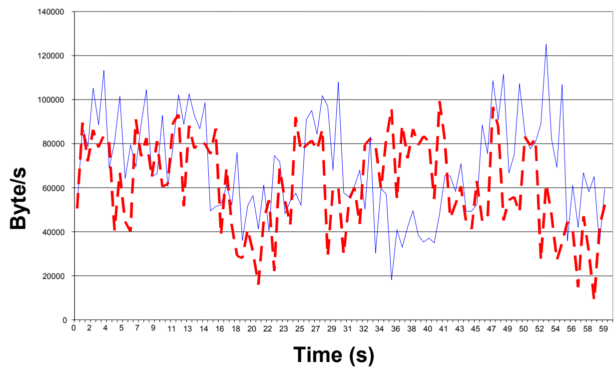

Figure 2 - UMTS - Throughput with reaction time set to: 5 s (solid line), 2 s (dashed line).

Table 3 - Scenario specification

Scen.	Ch. status (*dB*)	Interf. delay(%)	Interf. loss(*ms*)
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

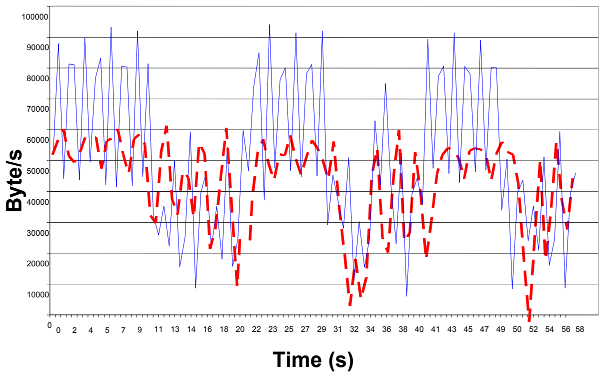

Figure 3 – UMTS - Throughput with reaction time set to: 1 s (solid line), 0.5 s (dashed line)

Table 4 – 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

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 2. Figure 1, Figure 2 and Figure 3 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 (72 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 2 KByte/s of throughput when compared to 5s case. 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 radio interface or in case of network congestion), which in our model leads to choosing minimum rate due to threshold on packet loss. Table 4 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 in a good compromise between reasonable settings (i.e. just negligible details of the original video eliminated) and network condition adaptation speed (i.e. 2 s of reaction time).

3.3 Effect of CSI and NSI loss and delay

Feedback information is a main basis for a JSCC/D system. Lost or excessively delayed CSI and NSI make the Application Controller unable to adapt to actual network conditions.

Table 3 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.

Table 5 – 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

Table 6 – Loss and delay for CSI and NSI messages

	CSI			NSI		
Scen.	Mean delay (*ms*)	Delay Std.Dev. (*ms*)	Loss (*pck/s*)	Mean delay (*ms*)	Delay Std.Dev. (*ms*)	Loss (*pck/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

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 3.1). Table 6 reports on 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 7 and Table 8.

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 (the slotted reaction time of the Application Controller tends to smooth the difference in performance).

Table 7 – System performance for WiMAX

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

Table 8 - System performance for UMTS

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

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

To better establish the improvement provided by the 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 2.2) and UMTS radio technology, is proposed, and results will be compared with those obtained with the optimal parameter settings for our system (as defined in section 3.3). 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 5 reports first comparison values for our system in adaptive and traditional (i.e. not adaptive, with settings set to fair conditions) system 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. Results collected 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).

5. Conclusions and Future Work

This paper presents the innovative JSCC/D system designed into the framework of the FP6 IST PHOENIX project [1][8] and provides assessment on several critical issues, which are feedback information overhead, reaction time of Application Controller and impact of lost or delayed feedbacks. This analysis allows to evaluate fairly the cost and benefits of the end to end 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.

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 to try 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.

With these 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. Results collected have shown better performance with WiMAX technology, in particular in terms of delay. Finally, similar results and conclusions have been obtained for other application and network scenarios tested in the PHOENIX project framework, corresponding to more or less demanding solutions in terms of mobility and bandwidth usage.

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

A key point in the realization of our proposal, with adaptation at application level only, is that no modifications in the existing video, network or radio access standards are required. Indeed, the system relies on the addition of just JSCC/D controllers in the video delivery architecture, which will then drive the encoding process on the basis of the network RTCP reports. Typical clients for such a solution are Telcos and SPs, who could begin to integrate now this first realization for point to point adaptation in their own video delivery system. For the more challenging problem of point to multi-point adaptation, the OPTIMIX project is expected to propose first solutions in three year time, offering first implantable versions by 2011.

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