Optimisation of Multimedia over wireless IP links via X-layer design: an end-to-end transmission chain simulator*

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ABSTRACT

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

Keywords

end-to-end optimisation, Quality of Service, joint source channel (de)coding, multimedia transmission, point to multi-point video delivery, cross–layer design, IPv6 mobility, adaptive medium access control

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1. INTRODUCTION

In wireless communications over power- and band-limited channels, the main concern of engineers is to define an acceptable compromise for the contradictory requirements of low bit rate offer, requests of high robustness against channel errors, low delay, and low complexity for a given target Quality of Service (QoS). The minimum bit-rate at which distortion-less communications is possible is determined by the entropy of the multimedia source message. However, in practical terms the source rate corresponding to the entropy is only asymptotically achievable as the encoding memory length or delay tends to infinity. Any further compression is associated with information loss or coding distortion. An ideal and optimum source encoder generates a perfectly uncorrelated source-coded stream, where all the source redundancy has been removed; therefore, the encoded symbols are independent, and each one has the same significance. Having the same significance implies that the corruption of any of the source-encoded symbols results in identical source signal distortion over imperfect channels. Under these ideal conditions, according to Shannon's pioneering work[1], the best protection against transmission errors is achieved if source and channel coding are treated as separate entities. This work, together with the obvious interest of designing separately source and channel standards explain why source and channel coding have historically been separately optimized.

However, as highlighted, among others, by Hagenauer [2], in practical situations the scenario is usually different. Mobile radio channels are indeed subjected to multipath propagation and so constitute a more hostile transmission medium than AWGN channels, typically exhibiting path-loss, log-normal slow fading and Rayleigh fast-fading. Furthermore, if the signalling rate used is higher than the channel's coherence bandwidth, over which no spectral-domain linear distortion is experienced, then additional impairments are inflicted by dispersion, which is associated with frequency-domain linear distortions. Under these circumstances the channel's error distribution versus time becomes bursty, and an infinite-memory symbol interleaver is required in order to disperse the bursty errors and hence to render the error distribution random Gaussian-like, such as over AWGN channels. For mobile channels, many of the above mentioned, asymptotically valid, ramifications of Shannon's theorems have limited applicability. Furthermore, most com-

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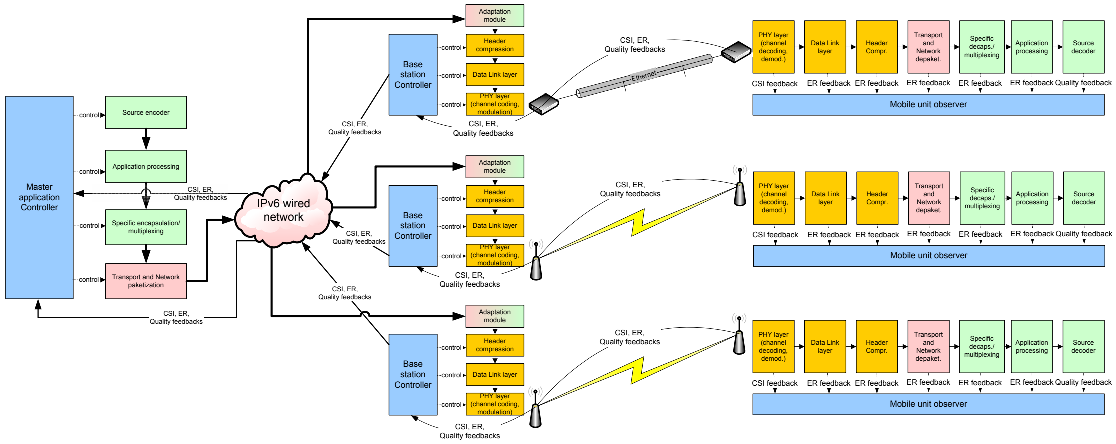

Figure 1: OPTIMIX detailed functional architecture.

munications nowadays include transmission over a network, if only to use the available infrastructure interconnecting most wired and wireless equipments and to be compatible with all IP-based systems. The impact of the network-layer should consequently be taken into consideration, which often led to further promote Shannon approach, by leading to further separate source and channel coding that cannot directly communicate as defined by the OSI paradigm. Cross layer design has been introduced to allow the evolution of the concept of joint source and channel coding, by targeting the joint design of the classically separated OSI layers at any layer. Characteristics and limits of such approaches are described *e.g.*, in [3, 14].

The approach followed by the ICT OPTIMIX project can be summarised as follows: "Controlling the different modules of the transmission chain to optimise the communication". Figure 1 presents the detailed functional architecture considered, which aims at modelling point to multi-point multimedia transmission, from the source signal (audio or video) to the display of the received and decoded signal for the end-user, in between standard and optimised representations of the different OSI layers of interest in such a multimedia streaming context, as detailed further in Section 2.

In order to benefit from joint source channel coding (JSCC) and cross-layer communications in real systems, control information, (e.g. Source Sensitivity Information (SSI) and Channel State Information (CSI) need to be transferred through the network and system layers. Unfortunately, the impact of the network and networking protocols are quite often discarded while presenting the joint source and channel coding systems and only minimal effort is put into finding solutions for providing efficient and backward compatible inter-layer and network signalling mechanisms. Still, some theoretical work has been carried out to provide cross-layer protection strategies for video streaming over wireless network, such as combining the adaptive selection of application-layer forward error correction (FEC) and medium access control (MAC) layer automatic repeat request (ARQ). There are already some mechanisms in use for reliable and timely generic information exchange between the different system layers, as the QoS features of relatively novel architectures, namely differentiated services (DiffServ) and integrated services (IntServ), which provide means for an application to have reserve transmission resources guarantees and specific service level from the interconnecting IP network by mapping the application requirement at network protocol level. Another example of the inter-layer signalling can be found from IEEE 802.11e standard where the QoS provisioning is achieved by coordination between the application and the medium access layers in WLANs. Furthermore, different research projects such as ENTHRONE [15] or more recently SEA [16] have introduced respectively system management solutions or adaptation procedure to deal with end-to-end QoS improvement and wireless adaptation. The OPTIMIX approach aiming at working transparently over existing systems, it could be used in conjunction with these solutions.

2. OPTIMIX SIMULATION ARCHITECTURE

The simulation chain set-up by the OPTIMIX project partners includes the different OSI layers, with the addition of controlling modules that will ensure the jointly optimised usage of the different protocols to maximise the perceived quality for the end-user. As shown by Figure 1, the protocol stack has been separated into three main groups, that are the application and session layers, the transport and network layers and finally the radio access (data link and physical layers).

2.1 Application and session

On the application layer, two video coding schemes – H.264 Advanced Video Coding (AVC) and Scalable Video Coding (SVC) – are included in the OPTIMIX context. H.264/AVC, standardized jointly by ISO/IEC MPEG and ITU-T VCEG, represents the state-of-the-art in single layer video coding, and is therefore a reference for our simulation, to which additional functionality compared to the original software package have been introduced: enhanced capabilities to operate in an error prone environment (i.e., packet loss and bit errors), soft-input decoding of variable length codes to deal with bit errors, and better error concealment techniques to handle unavailable information due to packet loss. H.264/SVC has been standardized only recently as an annex to H.264/AVC. It introduces enhancement layers to a base layer stream compatible with H.264/AVC. In addition, temporal scalability can be introduced by using prediction structures allowing for the removal of intermediate frames.

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Like for H.264/AVC, additional error resilience techniques have been introduced in the decoder.

In addition, audio codecs (i.e., AAC+ and AMR-WB+) will be introduced in the second version of the chain, in particular to study audio/video synchronization aspects. Other future extensions include encryption ("ciphering") and application-layer forward error correction (FEC) as it is applied in mobile broadcasting (e.g., MBMS, the Multimedia Broadcast Multicast Service in 3G networks). Those additions foreseen as plug-in modules to a standard codec (denoted application processing), to allow backward compatible usage but also to allow their usage as transcoding tools in middle of the transmission chain, for instance at the base station level.

The transmission is driven by a simplified client-server approach, relying on OMNeT++ messages relaying RTSP [7] commands to allow the end-user to signal its requests to the server, that will then provide the corresponding content.

For the simulations, the video input is read from a file that either contain raw YUV 4:2:0 video data, which is encoded during the simulation cycles or pre-coded video data. The first approach is used especially for H.264/AVC, since the data-rate can be adapted after the encoding step with more difficulties. The alternative approach with a pre-coded video file is considered valid for SVC, where the data-rate (and also the image size and frame rate) can be adapted in the compressed domain by simply removing packets (i.e., NAL units) from the bit-stream.

2.2 Transport, packetization and network

The packetization block of the simulation chain builds or removes the standard IPv6/transport headers. Various transport protocols are considered here: from the common TCP or UDP to the more recent ones like DCCP [8] and SCTP that will be included in the second version of the chain. ICMPv6 [10] is also implemented to provide ICMP style information delivery (e.g., in the case of feedback information messages). Advanced built-in features of IPv6 [11] (e.g. mobility support, multihoming) will also be used in future versions. Over UDP, the real-time transport protocol (RTP) [6] is considered, with optionally its specific secure profile (SRTP [9]) that provides ciphering and authentication in unicast and multicast modes.

Advanced mechanisms have been introduced and designed to allow for mobility of the users, as well as aggregation of signalling. Typically, Host Identity Protocol (HIP) [13] will be used to provide advanced mobility mechanisms and per-application mobility (every application can have different policies for handover management). Furthermore, future releases of the simulation chain will include an anycast addressing scheme to support feedback aggregation at the network layer.

The impact of an IPv6 wired network, corresponding to a LAN or Internet crossing is modelled with a network module. The purpose of this module is to resemble the wired trunk of the telecommunication infrastructure to generate the effects (mostly packet loss and delay) of crossing multiple IP routers on the data transmission. The corresponding loss and delay parameters are established via analytical modelling generating artificially packet losses and delays based on statistical distributions (mean value, variance, etc.) from measurements of the real world environment (i.e. the Public Internet). The point to multipoint transmission with multicast functionality is emulated by a routing function inside the IP Network Module that realises the replication of the data flow addressed to the different users when needed.

2.3 Radio access

The first module in the radio access part, that will be added in the second version of the chain, consists in reducing the inner header redundancy via the RoHC [12] (Robust Header Compression) protocol. The bandwidth economized in this way can be used, for example, on the application side to decrease the constraint on the video source bit rate, allowing then a better end to end final video quality.

Below is found the data link layer (DLL), whose implementation is technology-specific. In the chain are introduced basic DLL services specified in two standards of interest for OPTIMIX: WLAN (IEEE 802.11g) and Mobile WiMAX (IEEE 802.16e), with amendments to allow end-to-end optimisation. One of the key additional techniques is a partial checksum, using only DLL headers in the checksum calculation, at the DLL which allows passing a corrupted payload data to higher layers for further processing. In the future version of the simulator, a support for prioritization using buffers above the actual DLL will be added. The buffers are used to prioritize video packets (e.g. layers of SVC video) in the access point destined to the different users. As well, a support for IEEE 802.21 standard will be implemented to provide timely channel state information (CSI) between access points and mobile nodes.

The PHY layer module receives the data packets from the DLL and arranges them for the transmission over the radio channel. At the receiver side, it has the dual role of demodulating and decoding the received symbols before passing the correspondent packets to the upper layers. The PHY module is composed of several submodules, i.e. channel co-decoder, MIMO-OFDM mo-demodulator and physical frame de-assembler unit. Working in strict cooperation with the MAC layer and under the supervision of the Base Station Controller, the PHY layer module proposes different multiple access schemes for downlink, like TDMA, FDMA, SDMA and OFDMA. The possibility to choose between various channel coding and MIMO-OFDM modulation schemes permits to evaluate the results of the end-to-end optimization process addressed in the project in case of different radio communication scenarios, e.g. WiMAX or 802.11a/g/n. Currently, the channel co-decoder includes Rate Compatible Punctured convolutional (RCPC) codes and Rate Compatible Punctured IRA LDPC codes. OFDM modulation is supported, as well as multiple transmitting and receiving antennas. More complex space-time coding (STC) and linear beamforming techniques will be included in future versions.

Finally, the transmission is done over a frequency-selective channel sub-module which introduces the typical radio transmission impairments met in wideband mobile communications (e.g. different Rayleigh fading for the various subcarriers, log-normal slow fading and additive thermal noise).

2.4 Controllers and observers

Different entities are involved in the JSCC/D adaptation process proposed by OPTIMIX, namely the Master Application Controller, the Base Station (BS) Controller and the Mobile Observers. This approach comes from the splitting of the general joint adaptation problem into a number of subproblems, addressed by distinct controlling modules strictly cooperating in order to optimise the end-to-end perceived

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quality. This collaboration between the master application controller and the base station controller is realised through a cross-layer exchange of side information and control signals across the network (provided by the observers).

The application controller can be seen as an intelligent streaming pump implementing the controlling strategies ensuring that the compression and protection functions are decided jointly and efficiently from the end-user point of view. This intelligence in the streaming server is driven by a controller whose role is to improve the long term average received video quality, by controlling the compression and protection levels as well as the different modules in the transmission chain and adapt their parameters based on the feedback information it receives on transmission conditions. In that aspect, the application controller is the master of the whole system, as it transmits QoS targets to the different base station controllers, which will make their best to meet such targets, and inform the master of their success, or failures, to decide of new parameters based also on the observers feedbacks.

The Base Station (BS) Controller is designed to manage the large set of degrees of freedom that the radio resources typically provide in point-to-multipoint communication scenarios, in terms of frequency, time and space multi-user allocations. Best-effort adaptive algorithms for the BS Controller have been considered within the project, capable to intelligently (and fairly) serving the users according to their priorities and their radio channel state. One of the main BS Controller goals is to exploit the flexibility of different channel coding and modulation techniques in order to jointly adapt the wireless transmission scheme to both the source characteristics specified by the Master Application Controller and to the radio channel state experienced by the different users. In fact, a detailed set of CSI are supposed available at the base station, mainly through feedback channels from the receivers, and proper SSI information come from the upper layers, together with a set of requirements and constraints imposed by the Master Application Controller. In the first version of the simulator chain, the Application Controller has not been integrated and a simple fixed compression/protection ratio is applied at the higher layers. For the lower layers, the BS Controller sends configuration messages to the DLL and PHY layer modules, specifying fixed transmission parameters in terms of channel code type and rate, OFDM constellation size, transmission power, etc. Next versions of the simulation chain will include some of the opportunistic multi-user scheduling techniques and the adaptive radio resource allocation strategies addressed

The iterative exchange of information relevant to requirements and feedbacks between adaptive entities and the techniques capable to exploit them constitute the core of our preliminary design of the JSCC/D Controllers. In particular the Mobile Unit Observer is another key-element of our approach, since its main purpose is to provide to all the requiring entities the needed feedback information from the end-user mobile terminal.

The Observer runs functions of a triggering engine (TRG) which provides a unified service for cross-layer information collection, temporary storage, and dissemination within the network protocol stack. TRG is the central component of the cross-layer signalling architecture based on the triggering framework [4]. The role of TRG in the triggering architecture is to collect information (triggers) of several events occurring at different layers of the protocol stack, to process the information carried in the events into triggers, and to deliver the triggers to their consumers according to specific rules and policies defined for the trigger delivery. For example, a video quality estimation is performed after the decoding process. This event constitutes a Quality Information trigger collected by TRG, which, in turn, delivers the trigger to the Master Application Controller which has subscribed for this trigger.

Figure 2 details the implemented architecture for the simulator, that has been realised in the generic OMNeT++ [5] framework. The different modules detailed in this section can be found, with the same colour codes as in Figure 1.

3. INTEREST OF THE SIMULATION AND RESULTS

The main reason for creating such a complex simulator is simple: only a full scale simulator can allow to envisage all possible interactions of the advances proposed by the research topics pursued within the OPTIMIX project on all the different layers of the OSI protocol stack. Needing a common reference chain for our observations and results, we elected to use the already existing OMNeT++ framework, with its recognised flexibility.

Aiming at propose solutions suitable for the next generation of networked radio communications, and willing to take into account as many options as possible, it is clear that the number of parameters that can be used in the simulator can not be systematically tested. This is why, as with any complex simulation, the results will be of interest when a detailed (and possibly realistic) scenario is considered and compared with recent or legacy standard solutions, to obtain fair comparison and gains figures.

An example of results that can be extracted from the chain is presented in Figure 3, where the quality evolution with time (represented by the frame number in the figure) can be drawn for different conditions of use. Typically, we plotted here an H.264/AVC transmission with an RTP/UDP/IPv6 encapsulation and transmission over a 802.11g like radio access, with using or not the SRTP extension. It should be noted that the results presented in the figure are purely illustrative, and as a consequence we are not commenting them into details. Indeed, the interest of such a simulation chain is the opportunity it will offer, once all foreseen elements are integrated, to compare detailed scenarios and average simulations results for a representative number of realisations. Nevertheless, it is interesting to point out that the simulation chain can also be used as a tool to validate specific modules and algorithms.

4. CONCLUSIONS

The OPTIMIX project main goal is to effectively exploit the available bandwidth on wireless links such as WiFi or WiMAX ones. The complexity of the complete transmission chain, and the great numbers of parameters that can be jointly optimised explains the interest and necessity of establishing a reference simulation chain, shared by the project partners in order to allow the optimisation of the multimedia transmission. This simulation chain, developed over OMNeT++, will allow to prove the efficiency of the controlling elements introduced by OPTIMIX to drive the transmission and provide an enhanced perceived quality for the end-users.

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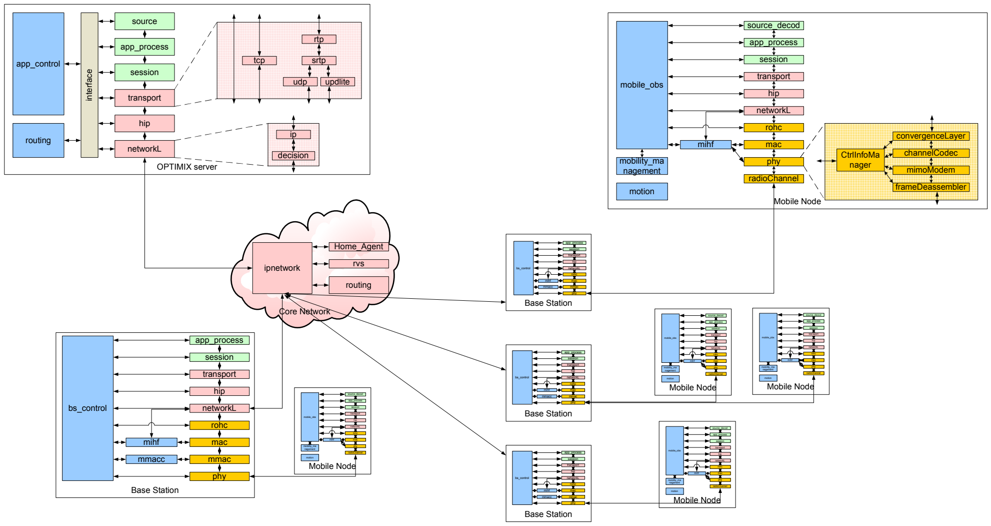

Figure 2: Overview of the OPTIMIX OMNeT++ complete simulator (realisation case with four base stations and five mobile nodes).

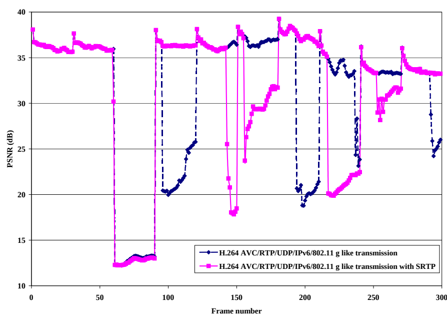

Figure 3: Example of obtained quality evolution with time for different simulation conditions: impact of using SRTP or not.

5. ACKNOWLEDGMENTS

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