Optimisation of Multimedia over wireless IP links via X-layer design*

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ABSTRACT

Following the path opened by FP6 IST PHOENIX project which was shown allowing an optimised allocation of resources for multimedia transmission over wired/wireless links in a joint source-channel coding approach for unicast transmissions, OPTIMIX project proposes to study innovative solutions enabling enhanced video 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.

Expected applications for such improvements are numerous in a time where users demand to have at their disposal on the move the same services they are already experiencing since a few years at home or in their offices. Typically, visiophony, video on demand on the move, access to streaming Internet websites, access to mobile personal television, location based services for educational, health, transport and environment purposes, virtual meetings or security professional applications are envisioned applications for the European end-user.

Keywords

end-to-end optimisation, joint source channel coding and decoding, multimedia transmission, point to multi-point video delivery, cross–layer design, IPv6 mobility, adaptive medium access control, QoS.

1. OVERVIEW ON TANDEM JOINT SOURCE AND CHANNEL CODING PRINCIPLE

In wireless communications over power- and band-limited channels, it is always of prime concern to maintain an optimum compromise in terms of the contradictory requirements of low bit rate, high robustness against channel errors, low delay, and low complexity for a target Quality of Service (QoS). The minimum bit rate at which distortionless 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 have 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. When using a block code of length N channel coded symbols in order to encode K source symbols with a coding rate of $R = K/N$, the symbol error rate can be rendered arbitrarily low, if N tends to infinity and the coding rate is below channel capacity. This condition also implies an infinite coding delay. Based on the above considerations and on the assumption of additive white Gaussian noise (AWGN) channels, 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-

*This work has been carried thanks to INFSO-ICT-214625 OPTIMIX project, which was partially funded by the European Commission within the EU 7th Framework Programme and Information Society Technologies.

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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.

A range of practical limitations must be observed when designing mobile radio speech or video links. Although it is often possible to further reduce the prevailing typical bit rate of state-of-art speech or video codecs, in practical terms this is possible only after a concomitant increase of the implementation complexity and encoding delay. A good example of these limitations is the half-rate GSM speech codec, which was required to approximately halve the encoding rate of the 13 kbps full-rate codec, while maintaining less than quadrupled complexity, similar robustness against channel errors, and less than doubled encoding delay. Naturally, the increased algorithmic complexity is typically associated with higher power consumption, while the reduced number of bits used to represent a certain speech segment intuitively implies that each bit will have an increased relative significance. Accordingly, their corruption may inflict increasingly objectionable speech degradations, unless special attention is devoted to this problem.

In a somewhat simplistic approach, one could argue that because of the reduced source rate we could accommodate an increased number of parity symbols using a more powerful, implementationally more complex and lower rate channel codec, while maintaining the same transmission bandwidth. However, the complexity, quality, and robustness trade-off that such a scheme implies may not always be attractive.

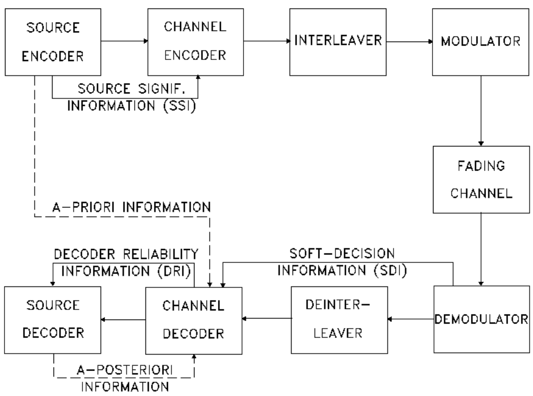

Figure 1: Intelligent transceiver schematic.

A more intelligent approach is required to design better speech or video transceivers for mobile radio channels, which is the main objective of the OPTIMIX project. Such an intelligent transceiver is portrayed in Figure 1. Perfect source encoders operating close to the information-theoretical limits of Shannon's predictions can only be designed for stationary source signals, a condition not satisfied by most source signals. Further previously mentioned limitations are the encoding complexity and delay. As a consequence of these limitations the source-coded stream will inherently contain residual redundancy, and the correlated source symbols will exhibit unequal error sensitivity, requiring unequal error protection. Following the open literature, we will refer to the additional knowledge as regards to the different importance or vulnerability of various multimedia source-coded bits as source significance information (SSI). Furthermore, we may term the confidence associated with the channel decoder's decisions as decoder reliability information (DRI). These additional links between the source and channel codecs are also indicated in Figure 1. A variety of such techniques have successfully been used in robust source-sensitivity-matched channel coding.

If for example, due to imperfect source coding, a source message has a high sensitivity, it needs to be protected by a stronger code than another having a lower sensitivity, since the impact of its corruption on the user perceived quality will be more dramatic.

The role of the channel interleaver and de-interleaver seen in Figure 1 is to rearrange the channel coded bits before transmission. The mobile radio channel typically inflicts bursts of errors during deep channel fades, which often overload the channel decoder's error correction capability in certain speech or video segments. In contrast other segments are not benefiting from the channel codec at all, because they may have been transmitted between fades and hence are error-free even without channel coding. This problem can be circumvented by dispersing the bursts of errors more randomly between fades so that the channel codec is always faced with an "average-quality" channel, rather than the bimodal faded/nonfaded condition. In other words, channel codecs are most efficient if the channel errors are near-uniformly dispersed over consecutive received segments.

Coming back to Figure 1, the soft-decision information or channel state information (CSI) link provides a measure of confidence with regard to the likelihood that a specific symbol was transmitted. Then the channel decoder often uses this information in order to invoke iterative detection. In interference-limited fading environments the channel quality fluctuates dramatically and hence it is unrealistic to expect that any fixed-mode system becomes capable of maintaining a constant quality of service. This is the case even in Multiple-Input Multiple-Output (MIMO) systems, since in the presence of shadow fading the received signal of all elements fades together. Hence even the MIMOs have to be reconfigured on a near-instantaneous basis. A simple manifestation of this is when the MIMO system's transmitter is reconfigured as a Spatial Division Multiplexing (SDM) scheme, provided that the receiver already has a sufficiently high diversity gain. This allows the system to increase the attainable throughput without reducing the integrity.

For the scenario where the mobile stations are unable to accommodate multiple transmit/receive antennas, the concept of distributed MIMO elements will be employed, which are also capable of mitigating the effects of shadow fading. Naturally, this requires the provision of extra resources for the associated side-information signalling, which ameliorates the achievable gains. The above-mentioned adaptive multi-functional MIMOs can be combined with sophisticated channel quality estimation and the radically new family of transmit pre-processing techniques, which are also often referred to as multi-user transmission arrangements. At this stage the network-layer has to support the operation of this refined physical layer. For the sake of avoiding the latency of automatic repeat request (ARQ) techniques, new Fountain-coding techniques can be designed.

Cross layer design is thus a further evolution of the concept of joint source and channel coding, targeting at jointly designing the classically separated OSI layers at any layer. Characteristics and limits of such approach are described, *e.g.*, in [3, 4].

In order to benefit from joint source channel coding (JSCC) and cross-layer communications in real systems, control information (e.g. SSI and CSI) needs 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 inter-layer and network signalling mechanisms. Some work has, however, been carried out in order to provide cross-layer protection strategies for video streaming over wireless network, such as combining the adaptive selection of application forward error correction (FEC) and medium access control (MAC) layer automatic repeat request (ARQ). There are already some mechanisms in use for generic information exchange between the different system layers, as the QoS features, namely differentiated services (DiffServ) and integrated services (IntServ), which provide means for an application to reserve transmission resources 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. However, QoS information such as the IP packet priority, to drop them selectively, is not alone sufficient as an optimization method for multimedia transmission. Extended information needs to be delivered in order to fully optimize the end-to-end transmission in a cross-layer manner. One possible solution for transferring the required controlling information is to leverage the current protocols such as Internet Protocol version 6 (IPv6) or Internet Control Message Protocol version 6 (ICMPv6) through the employment of header options and defining new message types. This concept of transmitting cross-layer information in a backward compatible manner can be referred as "network transparency" [5], which includes the abstract idea of making the underlying network infrastructure almost invisible to all the entities involved in joint optimization.

The presented transparency solutions are potential candidates for transferring the control information through both wired and wireless networks but they do not solve fully the problem of transferring control information through protocol stack from application to physical layers and vice-versa. Furthermore, they do not propose solutions to use this information for a thorough end-to-end optimization, which requires taking into account all protocol layers and particularly applications. In addition, QoS information consisting of IP packet priority alone is not sufficient for delivering optimization information between the layers of source and destination devices. Thus an enlarged information set needs to be delivered in order to fully optimize the end-to-end QoS of multimedia transmission system on different system layers.

Solutions for such an optimisation, relying on a cross-layer design for point-to-point link with feedback channel have been presented in [5], that show the noticable gain that can be obtained when optimizing the end-to-end transmission chain in a cross-layer manner.

Following this review of the main concepts on the basis of the project, let us now turn our attention to the specific OPTIMIX objectives.

2. OPTIMIX OVERVIEW

In light of the existing state-of-the-art and achievements done in the context of joint source and channel coding, and following the path opened by FP6 IST PHOENIX project for overall end-to-end adaptation of a multimedia transmission over wireless IP links, during which an optimised allocation of resources was demonstrated over wired/wireless (WiFi) links in a joint source-channel coding approach, OPTIMIX project proposes to study innovative solutions enabling enhanced video 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.

Cross-layer mechanisms proposed by OPTIMIX enable efficient joint approach between application world and transmission world. To achieve this goal, it is proposed to develop a scheme including all elements of major importance in a point to multi-point video streaming chain, in particular video coding, networking modules, MAC layer and physical layer, efficiently communicating together through the use of joint controllers (at the server side) and mobile unit observers (at the client side). This will be achieved by:

- considering innovative techniques to improve the efficiency of scalable video codecs when used in a wireless multi user environment with respect to robustness, efficient compression and intelligent use of scalability schemes. This will lead to the design of novel controlling strategies in the scope of point to multi-point scenarios, taking in consideration the aggregation of multiple feedbacks and the overall optimisation criteria in a multi user context.
- developing cross-layer mechanisms to enable the communication between application world and transmission world through the use of enhanced transport and network protocols.
- validating the overall system with respect to end to end quality optimization, and innovative techniques developed in all the fields of interest of the project will be evaluated. Efficient bandwidth use, real time constraints, robustness (both to errors, loss and network faults) and video quality will be amongst the major evaluation criteria

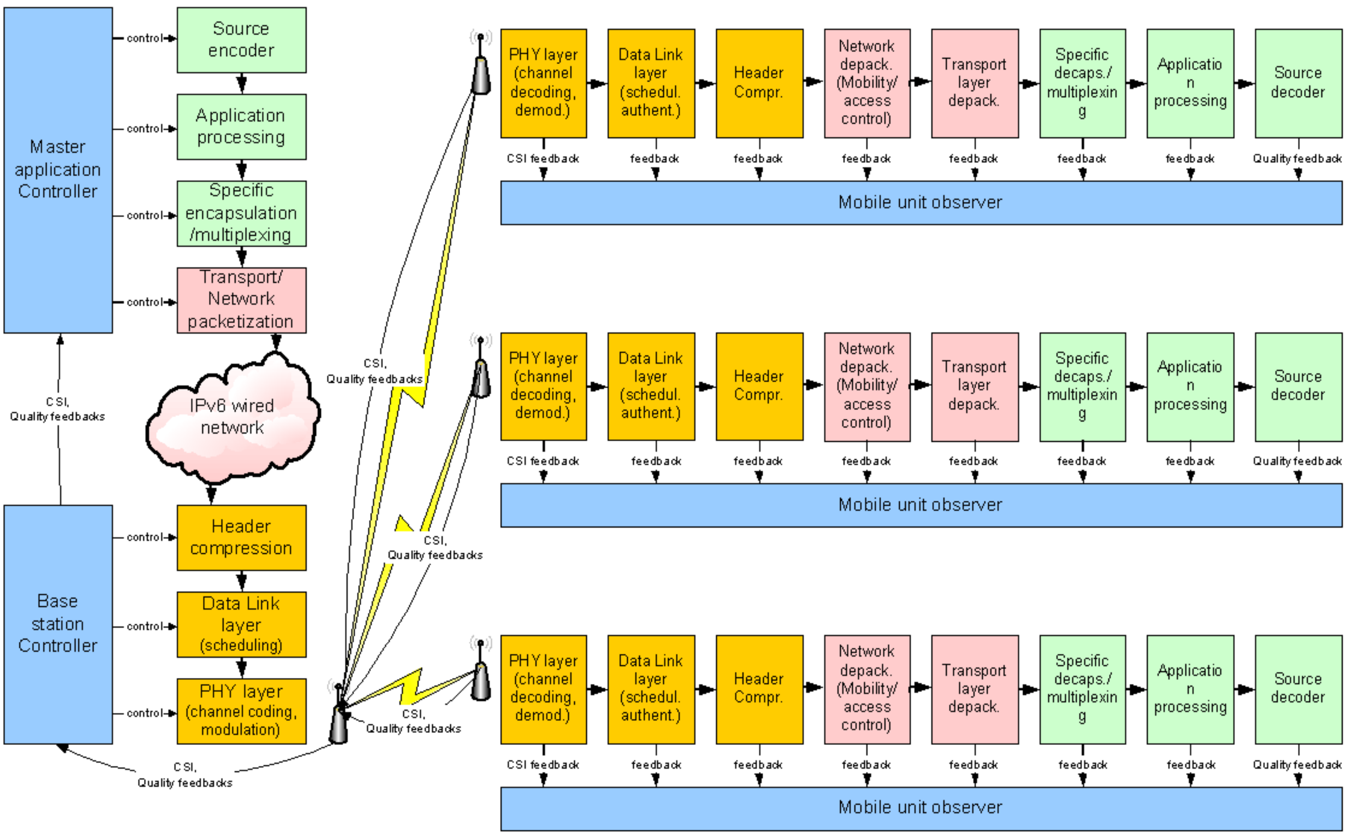

Figure 2: OPTIMIX transmission chain: introduction of observer and controller units.

The global architecture of the system proposal that will be considered within OPTIMIX project is depicted on Figure 2 for the particular case of one base station and three mobile units. In addition to the improvement of each respective block of the transmission chain (source coding, application processing, networking, header compression, . . .), the overall system optimisation will be done in the master application and base station controller (for server side) and in mobile units observers (for clients side).

With respect to existing solutions, a key innovation of OPTIMIX project is the introduction of the controller unit and observers, whose tasks are respectively to collect the side information on the system at the transmitter side (in terms of source characteristics, paquetization possibilities) and at the receiver side (in terms of channel state information, network conditions, source characteristics, . . .) at the receiver side to allow for the consequent tuning of parameters of the different system blocks at the transmission side. As illustrated in Figure 2, among the different elements that will be attuned or adapted for improvements are: application level through scalable content delivery, and corresponding packetization and protection against errors in streaming to multi-point context; transport protocols, for which new enhancements are targeted to efficiently adapt the network transmission to the overall system (including radio and infrastructure impairments); IP level mechanisms enhanced for a simple, but effective, network infrastructure supporting mobility and differentiated services; MAC layer adaptive schemes for bandwidth saving typically through improved network header compression protocols and queuing management. Resource allocation [6], prioritization and adaptive multi-user scheduling, as well as robustness to network impairments, can be performed at different layers according to the available information on the source, network and channel.

3. CONCLUSIONS

The OPTIMIX project main goal is to effectively exploit the available bandwidth on wireless links (WiFi, WiMAX, . . .) that is dynamic by nature, providing optimised solution for multimedia transmission over IP-based wireless networks in a point to multi-point context. Following first works realised in the context of IST FP6 PHOENIX project, units for collecting information on perceived quality at different layers of the system, and corresponding controllers to decide of the transmission and session parameters have been introduced at the emission and reception sides of the chain. Those units are communicating with each other through cross-layer mechanisms that are meant to be totally transparent for the rest of the chain, typically for routers in between, allowing for a deployment of such solutions over existing systems.

4. ACKNOWLEDGMENTS

The authors would like to thank their colleagues, who have participated in IST PHOENIX project and who will participate to ICT OPTIMIX project. Further information can be found on http://www.ict-optimix.eu.

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