A Joint Robust Header-Compression and Multi-layer RTP packet Multiplexing for Real-time IP services over DVB-S2 Networks

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Abstract: *Interactive and real-time IP service provisioning over DVB-S2 network is a difficult task that underlies several control and optimization mechanisms. This paper describes a joint header compression and packet encapsulation scheme that aims to maximize the resource efficiency in all-IP DVB-S2 satellite networks. The bandwidth gain is obtained by means of the integration of a robust headers compression (ROHC) and a novel low-overhead IP/RTP packet encapsulator & multiplexor called RTP4mux. RTP4mux uses several encapsulation and multiplexing profiles that are defined via a Satellite Resource Management System and service-level priority lists. The targeted IP real-time services over DVB-S2 are Voice over IP, IP Videoconferencing, Low resolution multicast DTV for mobile and embedded devices. Finally, the integration of RTP4mux in the satellite platform of IST IMOSAN project is also presented.*

1. Introduction

The wide coverage area and high bit rate capabilities make satellite links an ideal medium for the provision of a variety of services, to urban an rural areas. As a consequence, there is increasing commercial interest to use satellites in modern communications. However, this strong commercial interest is balanced, by the cost of the lease of satellite transponders and by the time-variation of satellite channels, especially near the bounds of the satellite footprint, which affects the performance of the whole system. The success of satellite communications is heavily dependent on new solutions that : (a) maximize the efficiency of satellite spectrum, which reduces the cost of satellite link per service or per user and (b) use adaptive channel coding techniques to compensate for the time variations of the satellite channel.

No complete and unified solution that maximizes the efficiency of the satellite transponder and compensates for time variations exists today. The problem has been –and is currently being– investigated in many layers (physical, network, service), but only partial solutions have been proposed so far; the solutions provide optimisation in special cases, where, for example, only one of the above layers is considered, or a static channel is assumed. With real-time IP-based services over satellite communications, encapsulation overhead is a major factor affecting efficiency, since about 50% (with low bitrate services) of overhead is introduced by the well known TCP/IP cross-layer encapsulation and possible DVB convergence layers [1][1].

The European BROADWAN [3] and IST SATLIFE projects [4], together with ETSI [5] normalization group, promote the use of buffer management and congestion control [6] mechanisms to enforce some sort of QoS strategies at the satellite network access point. While a common approach relies on simple RED-like (Random Early Discard) mechanisms, we argue that within the context of a variable coding and modulation the overall available bandwidth is likely to shift between several rates, which entail the use of more sophisticated scheduling mechanisms. The goal is to dynamically adjust the drop level in each queue so as to maintain bounded delays for real-time services at the time where the basic packet service time changes frequently. Thus the queues' policies should be coordinated through a traffic access control that interacts with both service and physical layers.

Taking the concept of bandwidth optimisation a step further, the EC-funded IST IMOSAN **Error! Reference source not found.** project focuses on developing and evaluating an innovative architecture for the provision of all-IP triple play services (TV, telephony, Internet) over an interactive DVB-S2/DVB-RCS network. The IMOSAN project aims to develop a multi-layer Satellite Resource Management System (SRMS), operating across the physical, network and applications layer which achieves the optimal allocation of resources within the satellite channel. Towards this aim, IMOSAN also employs a novel IP stream encapsulation and multiplexing scheme called RTP4mux.

RTP4mux operates at both RTP session level and RTP payload level. It is based on *a-priori* established priority list (e.g., pre-committed SLAs) and an overall services' characterization (max/mean bit rate, frame rate, etc.). RTP4mux is able to select several encapsulation/multiplexing/compression "*profiles*" depending on service priorities and uplink channel availability. One particular advantage of RTP4mux is that no additional delays are introduced by the multiplexing and compression process at the terrestrial satellite uplink gateway. Which is very appreciable in high bandwidth-delay product networks [7] such as satellite links where the latency is also a real issue for the support of interactive IP services.
The reminder of the article is as follows. Section 2 is devoted to the description of the RTP4mux modules. In section 3, the integration of RTP4mux in the IMOSAN service management and optimization architecture is discussed. Finally, we conclude in section 4.

2. RTP4mux

2.1 Robust Header Compression Module Description

The ROHC [1] family of header compression protocols is intended for wireless links which have moderate to high error rates and/or packet loss rates. The prime application of ROHC will probably be to 3G wireless mobile telephony to facilitate the efficient integration of voice (and other real-time traffic) with data. ROHC compresses only packet headers, not packet payloads. As depicted in Figure 1, the RTP compression protocol compresses the IP/UDP/RTP headers of voice and video RTP traffic typically from 40 bytes down to 4 bytes. The ROHC family also includes IP/UDP, IP/TCP and IP-only header compression. The word robust distinguishes ROHC from other header compression protocols which are *not* robust for links with moderate to high error rates.

It should be noted, however, that smaller packets offer a smaller target for bit errors. So the packet loss rate for any compression method should be lower than for uncompressed packets. Thus if RoHC has only a very small probability of loss of sync between compressor/decompressor state machines, there should be a small reduction in overall packet loss rate between applications. The main purpose of RoHC is to not increase the packet loss rate between applications.

In our context, all traffic will be transmitted through TS packet streams, which means that all streams are fragmented into constant-size (188 Bytes) TS packets.

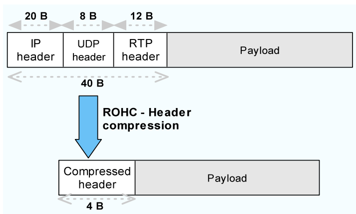

Figure 1: Robust header compression operation.

2.2 Multi-level RTP multiplexing module description

As highlighted above, the header compression doesn't achieve any optimization at service level (transported payload). Although many advanced audio/video formats (especially those intended for wireless clients) provide very low bit rates for fairly good perceptual quality (such as H.264 and AAC), the generated streams usually entail a small video/audio frames (payloads) that may severely affect the link utilization. The small payloads translates into small network packets, which significantly increase the overhead and thus confining the overall link exploitation in terms of the number of QoS-enabled services that could be carried in the FWL. It is quite clear that an optimal services encapsulation will widely contribute to increase the network operator revenues by further reducing the cost of satellite link per service [9].

We propose a novel real-time transport profile (RTP payload format) that basically provide media stream fragmentation, multiplexing and aggregation so as to maximize the link throughput while minimizing end-to-end transmission delays in DVB-S2 Gateways and Satellite Terminals (ST) by simultaneously packing/unpacking in the same RTP payload, several media frames (media fragment) originating from different streams. This technique is, in fact, a two-level stream multiplexing/demultiplexing process taking place in both RTP core level and RTP specific payload level.

RTP4mux is optimized for low bit rates audio and video services transmitted over wireless IP links (e.g. VoIP, IP Videoconferencing, Low resolution multicast DTV for mobile and embedded IP devices, …) supporting several voice/audio/video encoding standards such as MPEG4/H.264, MP3, CELP, ….

For short, the different streams are grouped together in a single RTP session based on their application-level intrinsic properties (codec format, bit rate) to maintain the overhead needed for their synchronization below a certain limit (in-band signaling of timestamps and sequence numbers are used – see Extra header in Figure 2). Moreover, the end-to-end transmission service delay constraint is considered during the configuration and operational phases of the mechanism permitting the customization of the transport process over DVB-S2 channels.

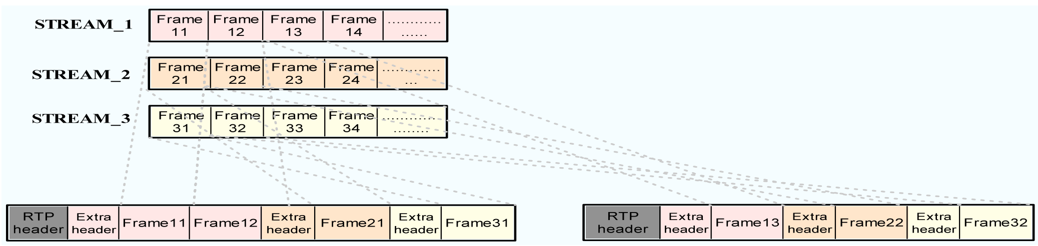

Figure 2: Multi-level RTP packet multiplexing and aggregation protocol.

[9] gives a performance comparison of RTP4mux with existing real-time transport protocols proposed in the frame of IETF AudioVisual Transport working group – AVT [10]). The results are promising and should be validated in an integrated DVB-S2 architecture such as IMOSAN.

2.3 Typical Transport Overhead and Expected Bandwidth savings

Table 1 gives the bit rate overhead introduced by a conventional content delivery over RTP/UDP/IP stack. The nominal bandwidth consumption on the forward Link is analyzed for some of the multimedia streams identified as the main services to be provided in IMOSAN's use cases, e.g., H.264 video, together with its AAC audio stream, as well as well the most popular VoIP codecs. The main cause of the extra bandwidth usage is RTP, UDP, IP Ethernet, and ULE/DVB [1] headers. Low-bitrate video streams and VoIP codecs sends small packets and so, many times, the headers are actually much larger than the data part of the packet.

Media Format	Coding profile	Frame Size	Overhead (bytes) (RTP/UDP/IP/802.3/ULE-DVB)	Overhead Ratio	Nominal DVB Bitrate (Kbps)	Estimated Gain
H.264 (MPEG-4 AVC)	**120 Kbps**, QCIF (144x176), QP=30	501 Bytes (33 ms ≈ 30 fps)	12+8+20+20+14 = 74	13%	**137.3**	**8%**
AAC (Advanced Audio Coding)	**64 Kbps**, 24 KHz	342 Bytes (42 ms)	12+8+20+20+14 = 74	18%	**77.7**	**13%**
ITU G.711	**64 Kbps**, 16 KHz	245 Bytes (30 ms)	12+8+20+20+14 = 74	23%	**83.3**	**18%**
iLBC (Internet Low bitrate Coding)	**15 Kbps**, 8 KHz	39 Bytes (20 ms)	12+8+20+20+14 = 74	66%	**43.9**	**61%**
GSM	**13 Kbps**, 8 KHz	34 Bytes (20 ms)	12+8+20+20+14 = 74	69%	**41.9**	**64%**
CELP (Speech Coding)	**6 Kbps**, 8 KHz	15 Bytes (20 ms)	12+8+20+20+14 = 74	83%	**34.9**	**79%**

Table 1: Estimated transport overhead and Bandwidth savings for typical IP multimedia services.

It is important to mention that in network with large delays (like DVB-S2) it is mostly an imperative to convey only one frame per packet so the delay may be kept at a minimum, though the poor bandwidth exploitation is the main side-effect in this case. At this point, the use of a dual-layer multiplexing protocol that combines different AV streams in a single IP payload may contribute in optimizing the network usage while minimizing the end-to-end delay.

For instance, when AAC (Advanced Audio Coding) is used for encoding of stereo audio signal at 64 Kbit/s (the format currently used within MPEG-2 and MPEG-4/AVC), AAC frames contain an average of approximately 342 bytes (with 1024 sample/frame). The transmission of a single AAC stream into a single RTP session will involve an overhead of 18% (bandwidth wasting); it gets worse when using DVB-S2 as transport medium since an additional ULE header [1] (14 Bytes of additional overhead for IPv6 encapsulation) is appended before finally conveying the stream through TS packets. At the end, the estimated DVB-S2 spectrum wasting may be about 21% (without accounting for the TS packets overhead) for audio streams transmission. Consequently, in our context with 1500 bytes MTU (Maximum Transfer Unit), a four AAC frames can be carried in one IP packet through using multiple streams multiplexing in a single RTP session, which divide the overhead rate by four.

In practice, since there is no path MTU restriction in the FWL (Forward Link), we can use an extended Ethernet frame size, which allow us to further increase the DVB-S2 spectrum utilization.

Since RTP4mux is operational only on the FWL where the streams are fragmented into TS packets stream, there are no constraints on maximum RTP4mux packet size to use. The only constraint is the service end-to-end delay to be respected by the communication system.

Obviously the encapsulation scheme is rather integrated to the service level (media-aware) in order to take care of the synchronization issues related to real-time services communication. As the services' multiplexing uses the common RTP header [11], it is not contradictory with robust header compression, and can clearly be deployed together with an appropriate ROHC strategy in order to combine both techniques advantages.

3 Integration of RTP4mux in the IMOSAN Architecture

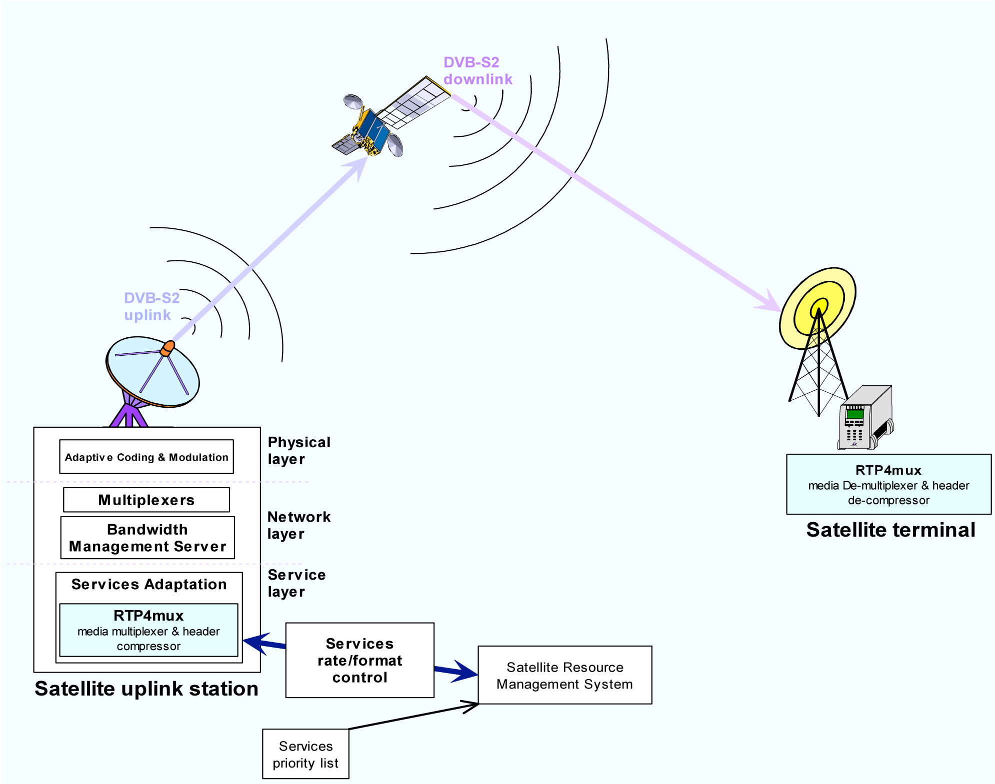

Figure 3: The overall IMOSAN service management architecture

As illustrated in Figure 1, RTP4mux requires interaction with the IMOSAN *Satellite Resource Management System* by the intermediary of the *Services rate/format control*. Also, as after the header compression, the priority identifier (DSCP) [12] is compressed it is the responsibility of RTP4mux to put the different multiplexed streams in the appropriate queue at the *Bandwidth Management Server*.

Within IMOSAN, three multiplexing profiles may be selected (along with an appropriate header compression) depending on the service characteristics (traffic pattern) after possible interactions with SRMS (services priority list). Preliminary analysis shows that IMOSAN require the use of, at least, three compression and multiplexing profiles (data, audio, and video) to accommodate the different type of services that are envisioned in the project.

4. Conclusion

In this paper we have proposed a new resource optimisation protocol for the provisioning of IP multimedia services over DVB-S2 networks with a return channel based on DVB-RCS. This protocol, named RTP4mux, achieves bandwidth gain by the association of a RTP/UDP/IP robust headers compression (ROHC) and a low-overhead RTP packet multiplexing and encapsulation over DVB. RTP4mux is currently under evaluation within the European project IST IMOSAN which studies a multi-layer integrated management solution that achieves optimum usage of the satellite spectrum.

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