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About QoS in DVB-S2/RCS Systems

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1. Introduction

The standardization of a Return Channel via Satellite and the satellite community efforts in term of interoperability over the last few years stands for major milestones in the development of reliable, efficient and low cost satellite equipments. It leads to quite a positive outcome: geostationary satellite networks are expected to play a decisive role in bridging the existing digital divide through providing broadband access to multimedia services in low terrestrial infrastructure areas.
However, unlike cable or 3GPP access networks, a lot of work on IP over satellite has been needed, especially about Quality of Service (QoS). The QoS architecture takes benefits from DVB-RCS dynamic allocation schemes and IP QoS architecture to cope with the satellite delay and the scarce uplink resources. This chapter deals with design and evaluation of Quality of Service Architecture to be implemented in DVB-S2/RCS systems.

The first section of this chapter aims at introducing DVB-S2/RCS Systems.
The long term efforts to optimize the DVB-S standard to lower the price of satellite access networks led to a new evolution of the standard: DVB-S2. A better encapsulation mechanism of IP packets and a new adaptative transmission scheme are the main concerns for the QoS architecture.
The encapsulation of IP packets in DVB-S has always been a complex problem. This section presents the evolution of the standard from the Multiprotocol Encapsulation (MPE) and the Ultra Lightweight Protocol (ULE) to the Generic Stream Encapsulation (GSE).
The Adaptive Coding and Modulation (ACM) technique that increase the network efficiency according to the weather conditions is a major evolution. The variable transmission rate impacts the QoS management and offers new perspectives for future system evolution.

DVB-S Satellite Terminals can only receive frames from the satellite. The need for a return link rapidly becomes essential so as to support emerging Internet services via satellite.
The return link access scheme in DVB-S/RCS systems is MF-TDMA. The return link is segmented into portions of time and frequency ("superframes. A Network Control Center (NCC) performs the entire satellite system control, especially Satellite Terminals synchronization and resource allocation. It periodically broadcasts a signaling frame, the TBTP (Terminal Burst Time Plan), which updates the timeslot allocation within a

superframe between every competing ST. This allocation can be dynamically modified on STs demand thanks to a bandwidth on demand protocol called Demand Assignment Multiple Access (DAMA). This system is presented here.

The next section rapidly overviews the concepts and mechanisms of Quality of Service management in basic architectures such as IETF Intserv and IETF / Diffserv. Others mechanisms such as Traffic shaping / conditioning, SLA, Scheduling and Admission control that have a main impact on the QoS are also described.

The next part aims at describing what means QoS in satellite networks thanks to the DVB-S2/RCS QoS Architecture example.
From the very first system only based on MPLS, a first architecture based on Diffserv was proposed. It was then enhanced to better fit to the DVB-RCS system in the IST project Satsix.

The next part will answer a main question related to the satellite networking systems that is: How to develop new services with Satellite Systems?
Based on our research work and results in the field, we'll explain how to use Simulation (using NS-2 or NS-3), Basic Emulation (using Linux TC/Simnet) and Advanced Emulation testbed like the one that was developed in various projects we were involved in. And we'll conclude that part with our skill on Real Deployed Systems.

The last part deals with Performance Evaluation of the described proposals. We first evaluated DVB-S/RCS NS-2 emulation model with QoS. The next way used to evaluate the proposed architecture was done through the PLATINE emulation testbed coming as the main result of the Satsix project. Our last experiment was done in the OURSES project, labeled in the Aerospace Valley research center with the following main devices from Astra (satellite), Thales Alenia Space (Gateway) and Advantech (Satellite Terminal).

To conclude this chapter, results summary and lessons learned will be given and future work will be described.

2. DVB-S2/RCS main features

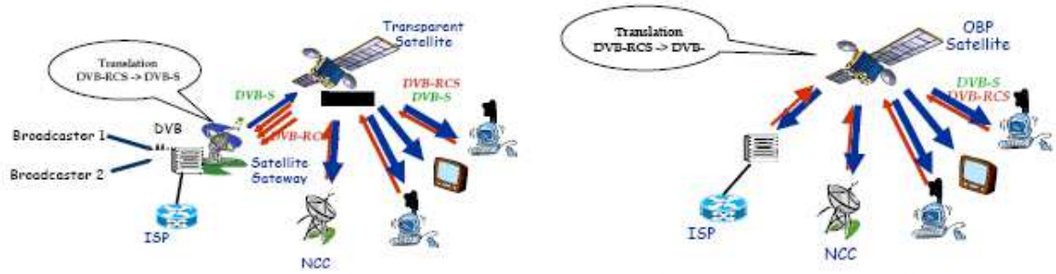
2.1 First overview of DVB-S/RCS systems

Started in 1993, the international European DVB Project published, in the end-nineties, a family of digital transmission specifications, based upon MPEG-2 (Motion Picture Expert Group) video compression and transmission techniques. Data are transported within MPEG-2 transport streams (MPEG2-TS) which are identified through DVB Service Information Tables. Adapted for satellite systems, DVB-S defines one of the most widespread formats used for Digital TV over the last years and still nowadays. However, DVB-S Satellite Terminals can only receive frames from the satellite. The need for a return link rapidly becomes essential so as to support emerging Internet services via satellite, leading to 3 solutions:

- UDLR (UniDirectional Link Routing) which emulates a cheap bidirectional solution through a terrestrial return link,
- DVB-S system with low speed terrestrial return link,

– DVB-RCS, which provides a full bidirectional satellite architecture [Fig. 1].

A good overview of DVB-S/RCS satellite networks architecture is given in Fig. 1, compliant with the architecture adopted within the ETSI BSM [3] group and the DVB-RCS standards. It consists in a geostationary satellite network with Ka MF-TDMA (Multiple Frequency Time Division Multiple Access) uplinks and Ku TDM (Time Division Multiplexed) downlinks.



a) transparent satellite b) regenerative satellite

Fig. 1. DVB-S/RCS architecture

Satellite Terminals (RCST) provide single PC or LANs with the access to the network, while Gateways (GWs) allow the connection with Internet core networks. The uplink access from each RCST is managed through DVB-RCS interfaces.

On the 2 topologies, the end-user side of the platform is on the right. On the left is shown the provider/enterprise/Internet side of the platform. It can be distinguished also between the satellite network side (in the middle) and the IP network sides (on left and right ends), interconnected by RCSTs. So, the 3 main components in the satellite network side (middle) are the Satellite, the Return Channel by Satellite Terminals (RCST) and the Network Control Center (NCC).

In Fig. 1–a, the architecture relies on a transparent satellite offering a star topology. All the forward links from GW to the RCSTs are DVB-S links and all the return links are DVB-RCS links. This allows the satellite payload to work in a simple transparent way without any computation to be made on the received frames before resending. Such a payload is easier to design and was the first implemented in such GEO satellites. But the main constraints of such architecture are due to the mandatory double hop to be done to go from one RCST to another one as it is needed to go through the GW to access to an RCST.

On the opposite, this problem is solved in the second kind of architecture shown in Fig. 1–b. In this topology, the uplinks (to the satellite) are DVB-RCS links only and the downlinks (to the RCST) are DVB-S. The complexity of this solution is located in the satellite where the payload has to be regenerative to translate incoming frames in DVB-RCS in outgoing frames in DVB-S. More complex to implement, the regenerative payload was designed later than the transparent one.

It has to be noticed that it is now time to implement hybrid payload including two parts one transparent and the other one regenerative inducing more complexity of the payload, but nothing new in the architecture components where the two kinds of network components coexist, but in separated configurations.

2.2 Specific DVB-RCS features

DVB-RCS systems involve lots of specific techniques, but only a few of them impact the QoS of such a satellite network. So this section is dedicated to the 2 main ones that are DAMA and Encapsulation.

2.2.1 DVB-RCS Demand Assignment Multiple Access (DAMA)

Furthermore, DVB-RCS requires a Medium Access Control (MAC) protocol because Satellite Terminals (ST) is able to simultaneously access the return channel capacity. The standard method relies on a Multi-Frequency Time Division Multiple Access (MF-TDMA). It basically relies on the availability of several TDMA channels (corresponding to different carrier frequencies), each subdivided into frames and further into timeslots of fixed length (bursts) during which the STs are able to transmit data through MPEG2-TS or ATM traffic bursts.

The entity responsible for this timeslot allocation within the Superframe shared by competing STs is the **NCC (Network Control Center)** that centralizes the satellite resources management. Thus it periodically broadcasts a signaling frame, the TBTP (Terminal Burst Time Plan) that contains the information on which STs relies to know when to transmit their bursts. This allocation can be dynamically modified by STs requests so as to prevent from wasting satellite resources that would be otherwise statically allocated. The implementation of such a mechanism is generally known as bandwidth on demand (BoD) algorithm.

In order to dynamically manage the bandwidth allocation, a bandwidth on demand protocol called Demand Assignment Multiple Access (DAMA) is defined. It relies on the STs ability to request frequently "capacities" to the NCC which enables a regular update of the TBTP to fit to the STs respective traffic load [Fig. 2]. The latter provides signaling schemes as well as MAC QoS Classes and their mapping on capacity types.

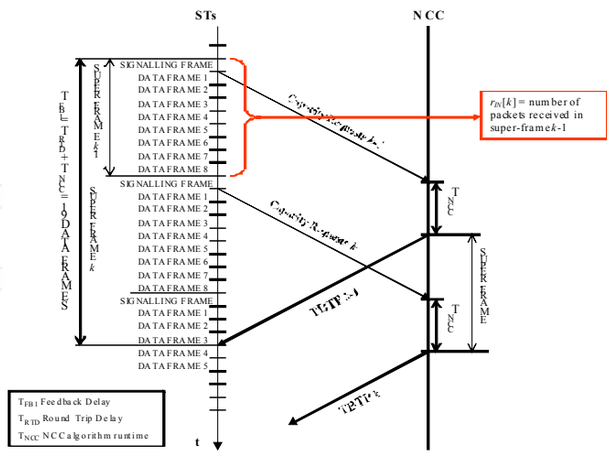


Fig. 2. DAMA algorithm: TBTP computed from RCST requests

Thus, the norm defines 5 Capacity Categories to fit the applications needs that will be detailed further in this paper. Capacity types are vital to return path QoS support at MAC layer; therefore they will be described in more details in the following. Any given ST can be

assigned one or a mix of the four capacity types. Generally, higher priority classes of service are associated with guaranteed capacity (CRA, RBDC), while lower priority classes are predominantly given best effort capacity (VBDC, FCA).
The DVB-RCS standard has left many issues open, e.g. how capacity requests are triggered, how and when certain parameters are negotiated (CRA), and if they can be re-negotiated, etc. It defines that when the NCC assigns timeslots to a certain RCST through the TBTP table, it can indicate a "channel" to which the timeslots are assigned.
It is obvious that this DAMA mechanism has great impact on what we will discuss later in this paper.

2.2.2 Encapsulation: from MPE to ULE

The multiprotocol encapsulation (MPE) provides a mechanism for transporting data networks on top of the MPEG2 TS in DVB networks. It has been adapted for carriage of IP packets, both IPv4 and IPv6. The encapsulation shall be done in accordance with the "Multiprotocol Encapsulation" technique described in the ETSI/DVB standard EN 302 192 and TR 101 202]. MPE includes methods for addressing the receivers of the data in the MPE header, which is necessary when many users have access to the same data channel. This feature allows several logical networks to be established without assigning PID values to each service.
IP datagrams are encapsulated in "datagram_sections" as defined in ISO 13818-6. The section_number and last_section _number must be "0" when carrying the IP protocol. The section format provides a format for mapping the datagram to the MPEG2 TS, and support filtering of datagram based on the MAC address using hardware or software demultiplexers.
The mapping of the datagram_section into MPEG2-TS is defined in ISO 13818-1. The sections are inserted into the payload of the MPEG2 packets (only first packet). The MPEG2 header is added to each packet. The resulting stream is the output of the data encapsulator/multiplexer, which is fed to the DVB modulator and satellite uplink equipment.
Many network operators and manufacturers of electronic equipment have adapted the MPE standard. That means that the standard is already in use and working well. Even though it is not the most efficient scheme for IP data transport.

An alternative encapsulation method has been defined by the IETF, RFC 4259. This directly places packet data into a Stream. This is called the Uni-Directional Link Encapsulation (ULE) defined in RFC 4326. The design of ULE simplifies processing, by reducing the number of header bytes and by significantly reducing the number of protocol fields that a receiver needs to parse. ULE also uses a Type field that resembles that adopted by the IEEE Ethernet standard, permitting easy interfacing to a wide range of network service (including IEEE 802.1pQ; MPLS; IPv4; IPv6).

ULE allows transmission of SNDUs up to 32 KB (compared to a maximum of 4KB in MPE). ULE also provides an extension header format (with an associated IANA registry), which allows future addition of new protocol fields to an encapsulated PDU, while providing backwards compatibility with existing implementations. This method is used to provide an efficient bridging method, but in future could also be used for encryption, compression, etc.

ULE is still old fashioned and solutions better fitted to Internet communications for instance have led to other proposal. The most promising called GSE will be presented later in this paper.

2.3 DVB-S enhancement: DVB-S2 standard and its new mechanisms

This section deals with the presentation of the new standard DVB-S2 and will be dedicated to the presentation of the main new features of such a satelite network, that are the DRA/ACM and GSE encapsulation scheme.

2.3.1 DRA/ACM

2.3.1.1 Return link

For the return link, different physical layers for individual terminals and for collective terminals have to be considered. The combination of adaptive coding, adaptive modulation and variable symbol rate can lead to different trade-offs depending on the type of scenario and of terminal.

The return link physical layer is based on the DVB-RCS standard with an adaptive waveform. DRA is considered, this means it is possible to change the coding rate, the modulation scheme and/or the transmission symbol rate. DRA is not included in the current DVB-RCS standard, but can be implemented using the standard capabilities without any additional information or changes (except for adaptive modulation using 8PSK and BPSK, which are not currently available in the DVB-RCS standard).

A DRA scheme is defined as an association of coding rate, modulation scheme and carrier symbol rate. The coding rate may be chosen among the set of DVB-RCS coding rates. Modulation can be either QPSK or 8PSK (with BPSK if required by link budgets). There is no specific constraint on the transmission symbol rate. This must be adapted to the terminal requirements in terms of the peak data rate range. However, due to equipment constraints (demodulation in particular), frequency plan constraints (in particular if the frequency plan needs to be reconfigured dynamically), the only transmitted symbol rates that are considered are multiples of each other.
A total of 70 combinations are thus possible.

The choice of DRA schemes to be retained is a trade-off between:

- the spectral efficiency (and thus the system capacity, the total number of users in the system and the average bit rate per user),
- the useful peak data rate (maximum data rate seen by a user)
- and the required SNR for each DRA scheme (the increase in required SNR should be reasonable with respect to the increase in peak rate or spectral efficiency).

When going from lowest DRA schemes to higher ones (i.e. from low SNR to a better SNR), we should:

- increase the required C/N0 with at least 2 or 3 dB steps between schemes to ensure the stability of the control loop,

- increase the spectral efficiency or the symbol peak data rate or both (that is increase the useful data rate).

When considering all possible combinations, we have 14 possible combinations of coding rate and modulation scheme, and 70 possible DRA schemes (combination of coding rate, modulation scheme and symbol rate). To select the set of DRA schemes required for the system, we consider some additional constraints resulting from implementation issues, as well as some specificity from the scenarios. We target a residential scenario with individual terminals, where we will try to optimise the spectral efficiency (to get a higher system capacity and a large number of users) rather than the peak data rate (that will be more a priority for collective terminals). However a reasonable peak data rate should still be offered to remain competitive with terrestrial solutions.
At the end, the selection process leads operational real systems to focus on around ten schemes only.

2.3.1.2 Forward link

The forward link physical layer is based on the DVB-S2 standard that supports an adaptive physical layer thanks to ACM (Adaptive Coding and Modulation). It addresses different kinds of terminals and different link conditions by allowing a large set of possible MODCOD schemes. With ACM, the coding rate and Modulation scheme can be chosen depending on the link quality within the set of available MODCOD (MODulation and CODing) schemes. The link can also be adapted dynamically by the implementation of a control loop for each station and dynamic measurement of the channel quality based on the SNIR estimation (with pilot symbols transmitted within the DVB-S2 frame).

For the forward link, there is no specific trade-off in the physical layer definition for the individual versus the collective terminals. They will all use a subset of the overall DVB-S2 possible MODCOD schemes. The MODCOD schemes retained for such systems only depend on the performances of the MODCODs proposed by the DVB-S2 standard. We select the best schemes in terms of spectral efficiency versus required Es/N0. Then the MODCODs that are actually used by individual or collective terminals will only depend on the Es/N0 range that they can reach.

This means the same DVB-S2 downlink carrier could be used to address both individual and collective terminals, the only constraint being that all the MODCOD schemes shall be supported at the same time. However, for the computations hereafter we will separate the MODCOD schemes distribution computation for individual and collective terminals.
And, as before on the return link, the selection process leads operational real systems to focus on around ten schemes only.

2.3.2 Encapsulation in DVB-S2: GSE

S2 introduces a new physical layer that supports a set of transmission waveforms that use a combination of higher-order modulation and powerful FEC coding. Although backwards-compatibility with existing DVB-S is supported, the main advantage arises when S2 is used with a range of terminal capabilities, particularly when the waveform is dynamically chosen

to match the prevailing conditions of the receivers. A transmission frame consists of a 90-bit physical layer header providing a preamble and identifying the ModCod used. The payload of a physical layer frame is known as a baseband frame (BBframe) and includes a 10 byte signalling header, which is followed by the BBframe payload. The size of this payload depends on the ModCod that was selected and can be up to 8 KB, significantly larger than an MPEG-2 TS Packet. The BBframe payload may carry a sequence of TS Packets, as in other MPEG-2 networks. Therefore, PDUs can be sent by encapsulating them using MPE or ULE. In S2, the transmission is optimised by omitting any Null TS Packets from the BBframe, and re-inserting these at the Receiver (preserving end-to-end timing).

The requirements for GSE differ to those of MPEG-2 TS, in that the GS uses a strong FEC code, and a much larger frame compared to that of a TS Packet. The requirement to check the integrity of a received SNDU therefore differs from that for the MPEG-2 TS. The current proposal is not to check every SNDU, but instead to place a 32-bit CRC at the end of each BBframe. Fig. 3 shows a general picture of the different encapsulation mechanisms involved in IP communication over DVB-S2 through GSE.

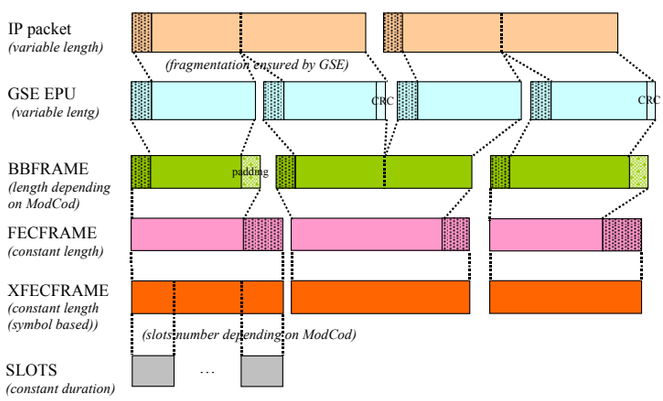


Fig. 3. GSE encapsulation in DVB-S2

The BBFRAME is the container that adapts GSE to fit the DVB-S2 properties to a data stream. DVB-S2 allows the use of a different ModCod (Modulation and Coding scheme) for each BBFRAME. This permits to take into account (while sending DVB-S2 data) the reception conditions, if available (e.g. through DVB-RCS).
A scheduling algorithm needs to take such variations into account.

In the DVB-S2 architecture, both modulation and coding can change from one BBFRAME to another, leading to variable length and variable duration BBFRAMEs. These two parameters are, however, tackled independently with the help of FECFRAMEs and XFECFRAMEs.
However, once the ModCod (more precisely the coding scheme) has been chosen, the BBFRAME length is implicitly fixed so that a padding mechanism is needed.

The FECFRAME is a constant size container that is used to encapsulate the BBFRAME with its associated coding. There are two possible values for a FECFRAME length (16200 bits or 64 800 bits) only one of which is implemented in a given system.

We will not describe more detail about FECFRAMEs, as the only pertinent information regarding a scheduler implementation is the constant length of the FECFRAME and the "variable" length of its payload.

The GSE protocol uses the BBframe header to determine the frame size (and presence of padding). The encapsulated PDUs also known as EPUs (Encapsulated Payload Units) are carried in DVB-S2 BBFRAMEs.
Not all receivers necessarily support the same set of ModCods (e.g. implementation of different sets of ModCods, different locations within the satellite footprint, local propagation conditions, or other reasons). Care therefore needs to be taken to ensure that all SNDUs or SNDU fragments are sent using GSUs with a ModCod that can be received by the intended recipient(s). Sending a SNDU or a SNDU fragment with a higher ModCod than required consumes unnecessary satellite capacity, while sending with a lower ModCod could result in loss.
At the time of writing, issues remain open such as flow security, support for signalling/control, the effect of small packets (concatenation) and these are the overall performance of the different features when used in an environment where traffic and terminal reception conditions are both varying.

3. QoS in terrestrial networks

The concept of quality of service (QoS) has gradually emerged to meet the various requirements demanded by certain types of applications, primarily in the world of interactive. Indeed, initially, the majority of Internet traffic consisted of textual data having no specific requirements but progressively, tools involving simultaneously files transfer, instant messaging, audio and video appeared. Consequently, guarantees on bandwidth, delay or jitter should be provided to users to ensure proper functioning.
However, the architecture of the Internet, based on TCP/IP, is not designed to differentiate the types of traffic and is currently dominated by a single model of service: best effort. This architecture can only guarantee correct functioning of all types of applications by offering the oversizing of the network that consists in configuring a network with a capacity that far exceeds the requirements. But this approach only postpones the problem and can hardly be applied to wireless access technologies with limited bandwidth.
Thus, effective management of resources is necessary to provide to Internet users QoS guarantees adapted to their needs. In this part, we will therefore firstly introduce the fundamentals of QoS. Then, the precursor models of QoS, IntServ and DiffServ will be detailed and finally we will show how the two signaling protocols, COPS and SIP, can participate in the implementation of QoS.

3.1 The basis of QoS

According to the standard ISO 8402 (ISO, 2000), quality of service is defined as "the totality of characteristics of an entity that bear on its ability to satisfy stated and implied needs". The E800 recommendation ITU-T (ITU-T, 1993) defines it as "the collective effect of service performance which determines the degree of satisfaction of a user of the service".
Thus, different types of QoS can be defined from these definitions and can be separated into three categories as defined in (Hardy, 2001):

- The intrinsic quality of service that is provided directly by the network itself and described by objective parameters such as, for example, the delay or the losses. It is the primary IETF focus.
- The perceived quality of service that matches the quality perceived by the user (also called QoE, Quality of Exeprience). It depends heavily on network performance but is measured by an average of users' opinions. The most common method is the MOS (Mean Opinion Score) in which a group of users separately evaluate perceived quality of an application between 1 and 5, an average score being then performed. The MOS is typically used for audio or video quality of an application, but may also involve perceived QoS connection time, the perceived security of user, service availability, etc… In addition, there is not necessarily a correspondence between intrinsic QoS and perceived QoS, the latter being very subjective. ETSI and ITU primarily use the term QoS as perceived QoS and prefer the term network performance to describe the technical part of QoS.
- The assessed quality of service that refers to the willingness of a user to continue using a particular service. It depends on the QoS perceived but also the price, service support offered by the supplier and other commercial aspects.

The main parameters that describe the intrinsic QoS in IP networks are:

- The transfer delay of packets, in milliseconds. It is usually measured from end to end but may be calculated on a portion of the network.
- The jitter or delay variation of packet forwarding, expressed in milliseconds.
- The bit rate, expressed in bits per second (bit/s or bps) or bytes per second.
- The rate of packet loss, defined as the percentage of packets lost in relation to the total number of packets transmitted.

3.2 The main existing models to ensure QoS

Two major proposals were made by the IETF to ensure QoS and proper functioning of IP services, real time or not: IntServ (Integrated Services) and DiffServ (Differenciated Services) each associated with a working group of the same name.

3.2.1 The IntServ model

The work of the IntServ working group led in 1994 to the definition of an integrated services architecture (Braden et al., 1994) composed essentially of two parts: an extended service model, called the IS model and a reference implementation framework for the development of this model.

This architecture, allowing to support QoS without changing the IP, is based on a per flow resource reservation. Each router must maintain the state of the flows going through it, which fundamentally modifies the principle of the Internet, which, on the contrary, was till now based on a conservation of the state of flows at the user terminals. That's why routers are then equipped with four additional features:

- The packet scheduler, responsible for the delivery of packets streams, using a set of queues as well as other mechanisms such as timers.
- The classifier that realizes the correspondence between an incoming packet and the class of service to which it is associated. The level of QoS provided by each service class is programmable for each stream.

- The admission control that implements the decision algorithm used by the router to determine whether a new flow may or may not obtain the requested QoS without degrading the guarantees offered previously.
- The reservation establishment protocol, needed to create and maintain the state of flow on routers. The protocol chosen for this function is RSVP (ReSerVation Protocol), defined later as Resource reSerVation Protocol in (Braden et al., 1997).

Two new classes of service are then defined in addition to best-effort that receives no special treatment at routers:

- The Guaranteed Service (GS) (Shenker et al., 1997) provides guarantees in terms of bandwidth and maximum transfer delay of packets, expressible quantitatively. If the stream respects the reserved parameters, this service ensures that all packets will arrive with a maximum delay and that they will not be lost in the queues in case of congestion. This service is suitable for real-time applications with strong time constraints such as videoconferencing or VoIP. However, no average delay is guaranteed, so it's the application itself which has to manage the delay variations at the receiver side by using buffering mechanisms.
- The Controlled-Load service (LC) (Wroclawski, 1997) is a service expressible qualitatively in terms of bandwidth, which ensures the user that its data stream will be transmitted with a QoS close to the one obtained in a network not overloaded (not congested).

The guarantees are obtained from end-to-end by the concatenation of such assurances, offered separately by each router crossed along the path. Furthermore, as stated previously, the protocol used to configure these routers is RSVP and we will now explain its functioning.

Initially, a PATH message is propagated from the transmitter (sending application) to the receiver. This message contains the traffic specification (TSPEC) that will be generated by the application. This specification can not be modified along the path but other information can be added through additional specification (AdSpecs) to precise specific resources constraints. Once the message arrived, the receiver responds with a RESV message that contains the description of traffic flow to which the resource reservation should apply (TSPEC Receiver) and the parameters demanded to implement the required service (RSpec). These descriptions may also change along the path.

RESV messages must follow the reverse path of PATH messages and trigger an effective reservation of resources (state have to be stored at each router) if the admission control mechanisms of each router validate the request. If a router validates the request, it creates and maintains a state corresponding to this flow. However, the reservations lifespan is limited and PATH/RESV messages must be exchanged periodically so that the reservation remains valid. It allows the system to be robust to changes in routing for exemple.

Once QoS is configured, when a reserved flow is going through a router, the classifier identifies it by its 5-tuple (Source IP address, destination IP address, Protocol, TCP/UDP source port, TCP/UDP destination port). The scheduler then handles the queue management.

RSVP is so a signaling protocol that allows to reserve dynamically bandwidth and ensure a maximum delay from end-to-end. This reservation, initiated by the receiver, can prevent that some applications monopolize resources unnecessarily and allows in the case of multicast communication to differentiate the reservation (and billing) for each receiver.

Moreover, its dynamic functioning can adapt to changing communications (changes in the number of participants, route changes, etc…). Finally, this protocol also has the advantages of being adaptable to both IPv4 and IPv6 and seamlessly passing non-RSVP routers.
However, RSVP requires maintaining state information for each flow at each node or router along a path connecting a transmitter to its receiver. So, when the number of users and flows increases, the number of states becomes significant and the traffic is all the more saturated as refreshments between RSVP routers become important and create overhead. The main shortcoming of the IntServ model and the associated RSVP protocol remains their lack of adaptation to the scale factor, especially as the reservation in RSVP is unidirectional. So, for a bidirectional application requiring QoS in both directions, the amount of messages is twice as high. The IntServ/RSVP model is therefore more adapted to small networks such as LAN.
It is one of the reasons that explain that another model of architecture has been proposed: the DiffServ architecture.

3.2.2 The DiffServ model

To solve the problem of scalability posed by the per flow management of QoS in the core network routers, the DiffServ working group has therefore proposed ((Blake et al., 1998) and (Grossman, 2002)) to separate traffic by classes of service. Thus, per flow QoS treatment is realized only at edge routers that aggregate flows by traffic class. The number of state maintained by core routers is so reduced to the number of classes and not anymore to the number of flows, which greatly reduces the complexity.
Each service class is identified by a value encoded in an existing field of IP header, redefined by the DiffServ group and named DSCP (DiffServ Code Point), which presents the advantage of not requiring the use of an additional signaling protocol. This is the TOS (Type Of Service) for IPv4 and TC (Traffic Class) for IPv6.

The DiffServ architecture is primarily based on the concept of DiffServ domain that consists in the grouping of one or more networks under a single administrative authority. This domain is composed of nodes or core routers that are only connected to nodes within the same DiffServ domain and edge routers that interconnect the DiffServ domain to other domains, DiffServ or not. The edge routers can also acts as well an incoming router when the traffic enters into a DiffServ domain as an outgoing router when the traffic leaves the domain.
A client of a DiffServ domain (which can be either a user or another domain DiffServ) must negotiate a contract with the service provider responsible for this area that specifies the terms and conditions of use of the concerned services: this contract is called an SLA (Service Level Agreement) and the technical part is specified by different SLS (Service Level Specification). An SLA contains the following information:

- The traffic that the customer is likely to generate in terms of data volume, rate, number of users, etc...
- The QoS that the service provider has to provide the customer in terms of availability, security, reliability or performance (delay, bandwidth, etc…).
- The policy followed by the service provider in case of overflow traffic (rejected, accepted but surcharge, etc...).

Finally, (Nichols, 1999) defines an entity called bandwidth broker with knowledge of the resources availability and policies of the associated domain. One of its main tasks is the admission control. In addition, to allow an end-to-end allocation of resources across all the domains taking into account the different SLAs negotiated between them, this entity must communicate with the bandwidth broker of the neighbouring domains.
To manage a DiffServ domain, the service provider in charge of it first starts by sizing its network according to the contracted SLAs with all of its customers.
The processing of packets entering the DiffServ domain is then realized at the edge routers in charge of flows classification by class of service, and traffic conditioning. To do this, they are composed of a classifier, a meter, a marker, a regulator and finally a dropper of non-conforming traffic. The classifier identifies a flow from either the DSCP field only, or a combination of one or more fields such as source IP address, destination IP address, DSCP, protocol ID, source and destination ports or other information as the input interface. Then, mechanisms for profiling and traffic measurement allow on the one hand marking or re-marking the packets and, on the other hand, to shape flows or drop them totally or partially to meet the negotiated traffic profiles.
During the marking, the DSCP field is updated using one of the different classes of service or PHB (Per Hop Behaviour). In addition to the Best Effort class which is not subject to any special treatment, two PHBs have been defined by IETF:

- Expedited Forwarding (EF) (Jacobson et al., 1999) which corresponds to the highest priority and ensures the transfer of flows with high temporal constraints (eg VoIP, videoconferencing) guaranteeing a certain bandwidth and low delays, jitter and loss rate.
- Assured Forwarding (AF) (Heinanen et al., 1999) which defines four levels of priority (AF1, AF2, AF3, AF4) on the delivery of certain packets in case of congestion.

Once packets have been marked by the edge routers, they are treated within the DiffServ domain by the core routers depending on the PHB deducted from the DSCP field which describes the forwarding behavior of these routers. The priorities are then performed by scheduling algorithms such as PQ (Priority Queuing), WFQ (Weighted Fair Queuing) or CBQ (Class Based Queuing).
The DiffServ model therefore provides a QoS management more adapted and more realistic than the IntServ one. The per class management (or per aggregate) can indeed be much more resistant to the scale factor. Moreover, Diffserv does not require signaling protocol like RSVP by adapting the header of IP packets, which saves bandwidth.
However, the sizing of a DiffServ domain taking into account the SLAs contracted with neighboring domains and users is a heavy and complex task to implement that does not dynamically adapt to rapid changes in traffic. Similarly, not using a signaling protocol at the application level implies that the user is unable to dynamically change the resources according to its needs.

3.3 QoS Signaling protocol

The previous paragraphs show that, although IntServ is unsuitable for scaling up, the use of a resource reservation protocol (RSVP) allows a more precise (adapted to each stream) and dynamic QoS management. The idea of using one or more signaling protocols to negotiate and establish end-to-end QoS has consequently been the subject of a number of research among the scientific community.

The objective of this part is then to present some of these protocols that enable the implementation of QoS, the other concepts (security, supervision, etc.) being not addressed.

3.3.1 COPS and the notion of QoS management by policy

The RAP (Resource Allocation Protocol) working group of the IETF defined in 2000 an architecture based on the notion of policy to improve the admission control mechanisms of a network. A policy is defined as a set of rules for the administration, management and access control to network resources. Each rule is then associated with a set of conditions which corresponds to a series of actions to do in case of these conditions are met. To enable the exchange of such policies on a client/server model, a protocol is also defined by that working group, the protocol COPS (Common Open Policy Service) (Durham et al., 2000).

This model of by policy management is composed of two core elements: PEP (Policy Enforcement Point), responsible for implementing policy decisions and the PDP (Policy Decision Point), in charge of making decisions based on defined policy. These two components communicate via the COPS protocol. To take its decisions, the PDP communicates with a database of policies through the LDAP protocol (Leightweight Directory Access Protocol) and can query other entities such as an authentication server or a bandwidth broker using SNMP (Simple Network Management Protocol) for example.

This generic model of network control by policy allows for two distinct modes of control: outsourcing and configuration (also known as provisioning).

In the outsourcing model, a router (including PEP) which has to make a decision on acceptance of a reservation sends a COPS request to the PDP and the latter responds with the decision taken in accordance with the policy rules. It can especially be used in relation with the IntServ model when a router receives a RESV message and must decide to accept or decline the resource reservation. Indeed, that decision may require other information than the resources locally available at the router and the use of a PDP can thus be judicious.

In the configuration model, when external events require a change in the configuration of routers (and thus PEPs), the PDP may communicate to them new rules to apply through the COPS protocol. They no longer have to seek the PDP before making a decision. This model can overcome the main weakness of DiffServ model in which classes of service configuration is static. Indeed, a network administrator can define, for example, two types of policies, one appropriate to the day where many VoIP calls are held and one for the night which suits rather backup's servers or data downloads. In this case, edge routers act as PEPs, while the bandwidth broker plays the role of PDP.

3.3.2 SIP: the session control and the QoS

SIP (Session Initiation Protocol) is a signaling protocol standardized by the IETF (Rosenberg et al., 2002) designed to establish, modify and terminate sessions with one or more participants. Its main use is now the session management of voice over IP (VoIP), for which it is currently the most common open standard, but the fact that it is independent of the type of data transmitted and the type of protocol used also allows him to develop many other applications such as instant messaging, video conferencing, distance learning, video games or virtual reality. The parameters of these sessions are described by the Session Description Protocol or SDP (Handley et al., 2006) and, in this part, we will see how these parameters can be helpful for the dynamic reservation and release of resources.

These parameters are negotiated during the establishment of the session but also during the session modifications. This negotiation can be conducted directly by the SIP clients located on users' terminals but, although they are aware of some important parameters, such as codecs they can use, the latter have no real means to know the status of available resources along the path their communications will cross. Moreover, it would imply the integration of QoS mechanisms such as RSVP or COPS on SIP clients, which has the double disadvantage of making the SIP clients more complex and not allowing SIP clients which are unaware of QoS to use this option.

That's why these new features must be rather implemented at the SIP proxy: the intermediate entities where SIP clients are registered and that intercept SIP messages. Indeed, they may allow operators to know the duration of the session, the number of involved media, the codecs used and their associated characteristics (bandwidth, etc..). From this information, automatic management of the resources reservation/release and of admission control according to the policy adopted by the operator becomes feasible and no changes are required at the SIP clients.

Two modes of session establishment with QoS reservation can then be distinguished:

- The "enabled" mode in which the session establishment and resource reservation are performed in parallel but do not depend on each other. The session will start regardless of the outcome of the reservation. For example in the case of DiffServ networks, QoS will be BE if the reservation has failed and EF if it worked.
- The "assured" mode in which the establishment of the session is done only if the QoS reservation has been realized.

A standard (Camarillo et al., 2002) detailing the operation of these two modes has been proposed. It is based on new messages such as Session Progress, UPDATE or PRACK and a set of preconditions added to the session descriptors. The SIP clients can then specify whether the implementation of QoS is "mandatory" ("assured" mode) or "optional" ("enabled" mode) for each direction of communication (reception, transmission) and if the QoS must be e2e (end-to-end) or local (at the access network). However, RFC 3312 (Camarillo et al., 2002) considers that QoS is implemented by the participants of the SIP session while we will consider that it is implemented by the SIP proxy. The Fig. 4 illustrates an example of a SIP session integrating resource reservation/release in "assured" mode.

IntechOpen

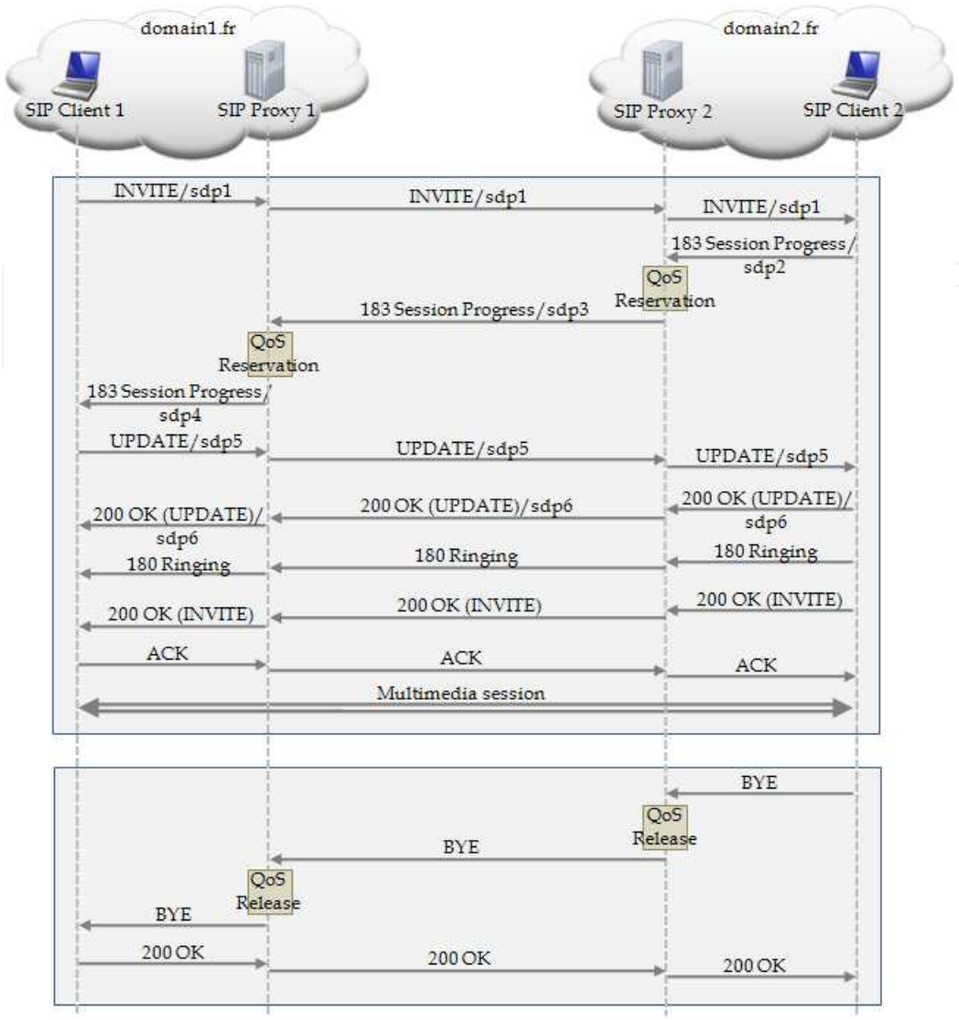


Fig. 4. SIP session with resources reservation/release

PRACK and 200 OK (PRACK) messages are not presented to make the figure more comprehensible but they are normally exchanged at the reception of a 1xx message to make it reliable. It can also be noticed that, in this example, all messages are exchanged via all the SIP proxies to manage the QoS along the whole path. Finally, QoS reservation/release in Fig. 4 may for example represent the exchange of messages (COPS, RSVP, etc..) allowing SIP proxies to communicate with the entities responsible for resources management within the concerned domains. For example, if we consider a DiffServ domain between domain1.fr and domaine.fr, the SIP proxies will be able to exchange COPS messages with the Bandwidth Broker in charge of the DiffServ domain. The latter can then configure the edge routers of this domain to prioritize the flows of the future SIP session.

Moreover, whatever the session establishment mode is, if a change of session is triggered by a re-INVITE message, SIP proxies can analyze the new parameters and automatically warn the entities responsible for resource management. The SIP signaling thus allows much more dynamic management of QoS for the applications it controls.

SIP, although it was designed originally to allow the session control, can be used very efficiently in the dynamic management of QoS.

4. QoS in satellite networks: DVB-S2/RCS QoS Architecture

The quality of service in DVB-S2/RCS networks is basically managed at layer 2. The first section gives an overview of classical layer 2 QoS architecture. The QoS offer should satisfy the application QoS requirements then this relationship is covered by the upper layers. Three QoS management strategies that can be considered as three different architecture maturity degrees are presented in this section. The first one, corresponding to a short term technique to introduce QoS in satellite system is based on MPLS over satellite systems. Then a fully IP based solution is presented as a medium term solution and the complete integration of satellite networks in Next Generation Networks (NGN) consists in the long term evolution of theses architectures.

4.1 Layer 2 QoS management

- In star architectures, the gateway (GW) centralizes the traffic and the signaling path and uses all of the offered bandwidth. Then, the quality of service management could be simply ensured by a correct scheduling algorithm, according to the packets destination and their service classes.
  But, when considering meshed topologies, the return link is shared among multiples satellite terminals. As one of the first things that impact the quality of service in the satellite network is the network access, packets should be sent on the air interface as soon as possible or even compliantly with the delay or the bandwidth required by the application. Then, in theses architectures, the quality of service management is essentially assured for the return link, i.e the DVB-RCS part.
  As said previously, the DAMA request/assignment cycle exhibits a non negligible latency and additional delays that cannot always match interactivity requirements of multimedia services. In order to both maximize satellite resource use and meet multimedia requirements, the DVB-RCS norm discriminates RCST capacity requests into 5 categories:
- Continuous Rate Assignment (CRA): Fixed slots are assigned in each MF-TDMA frame for the whole duration of a RCST connection
- Rate-Based Dynamic Capacity (RBDC): a dynamic rate capacity (in slots/frame) granted in response to explicit RCST requests
- Volume-Based Dynamic Capacity (VBDC & AVBDC): a dynamic cumulative volume capacity (in slots), granted in response to explicit RCST requests
- Free Capacity Allocation (FCA), which is assigned to STs on an “as available” basis from unused capacity

The standard defines separate MAC traffic priority queues and suggests a requesting strategy for each of them, that is to say a relevant mapping between traffic and request categories. Any given RCST can be assigned one or a mix of the four capacity types. In general, higher priority classes of service are associated with guaranteed capacity (CRA, RBDC), while lower priority classes are predominantly given best effort capacity (VBDC, FCA).
The generally recommended MAC queues are the following:

- Real Time (RT) queue used by temporally constrained applications as VoIP or visioconference. A CRA allocation mechanism is often used to feed it.
- Critical Data (CD) queue used by critical applications. It could be file transfer applications as well as low importance visioconference. RBDC requests are generally used.
- Best Effort (BE) queue used by non critical applications, as emailing or web browsing. RBDC, VBDC, or AVBDC are recommended.
- Network Management (NM) queue is recommended to offer a guaranteed path to signaling protocols as those presented in the following sections.

The CRA capacity allocated to the RCST should be consumed by the Real Time queue, but could be shared with other classes if unused.
However those suggestions are not yet sufficient to seamlessly integrate satellite networks into an end-to-end NGN QoS Architecture. The next sections propose three solutions, from the simple one to the NGN compatible one.

4.2 QoS with MPLS based architectures

One of the challenges the Internet is facing since the middle of 1990's is the exponential increased level of the IP traffic, sometime beyond what routers are able to handle. One solution to overcome the router bottlenecks it to delegate more of the IP forwarding to the layer 2. This allows the router to shortcut the heavy layer 3 processing based on routing table and implemented in software, while using instead a lightweight layer 2 switching table implemented in dedicated ASICs in the forwarding process.
MPLS is being defined by the Internet Engineering Task Force (IETF) as a standards-based approach to apply label-switching technology to large-scale networks. Although MPLS, as its name suggests, can conceptually support multiple protocols, the current work is focused on the integration of IPv4 and IPv6 with ATM, Ethernet, Wavelength Division Multiplexing (WDM) and Frame Relay. Thus, MPLS could be seen as the simplest solution for the seamless IPv4 to IPv6 migration of a heterogeneous network.
MPLS forwards data using labels that are attached to each data packet. Intermediate MPLS nodes do not need to look at the content of the data in each packet. In particular the destination IP addresses in the packets are not examined, which enables MPLS to offer an efficient encapsulation mechanism for private data traffic traversing the SP backbone. Service providers are therefore offering MPLS as a VPN technology.

4.2.1 Intserv / Diffserv QoS support in MPLS

MPLS acts as a level 2/3 protocol, its role is to transport a native network level traffic. IP includes in its header ToS field in IPv4 or Traffic Class field in IPv6, which provides information on the kind of traffic to transport. The traffic level quality information that should be borne by MPLS is fully dependent on its upper layer protocol or the resources reservation provided by RSVP. Both IntServ and DiffServ classes are supported over the MPLS architecture. IntServ and DiffServ classes are mapped onto DVB-S/RCS classes. Bandwidth allocations and flow isolation is efficiently supported by means of the MPLS VPN mechanisms.
The **IntServ** service architecture incorporates one main component to support the IntServ QoS model: RSVP-TE (Awduche, 2001). This IntServ Signalling protocol for MPLS is an

extension of RSVP for MPLS. It allows traffic engineering via label distribution. RSVP-TE works on the Downstream-on-Demand principle, i.e. the upstream router requests a label from the downstream router. This allows an ingress router to specify a route for a given flow. Resources reservation is optional with RSVP-TE; this allows the label distribution even if the paths do not need reserved network resources. Unlike RSVP, RSVP-TE works only between ingress and egress routers, and not between the sources and the destinations of the flows. RSVP-TE also allows the traffic policing working with the LSP instead of the destination IP address.

Unlike IntServ, **DiffServ** does not require the reservation of resources end-to-end. It actually runs on a Per-Hop-Behaviour (PHB) principle. This means that each router filters the incoming packets. In MPLS, LSRs do not read the IP headers, thus they do not have any direct information regarding the PHB for arriving packets. MPLS encapsulates IP packets with a shim header that includes the label and the Exp field. This last one is composed of 3 bits intended to support marking of packets for DiffServ, which on its side defines its classes of service with the 6-bit DSCP field. The problem is that DiffServ can have 64 possible DSCPs whereas MPLS/Exp bits can only address up to 8 possible types of Per-Hob Behaviors (PHBs). In order to support for instance 1 EF class, 4 AF classes with 3 dropping priorities and a default Best Effort class larger field containing QoS control information is needed. Two possible labeling configurations exists (Le Faucheur, 2002):

- A configuration called E-LSP (Exp LSP) is able to treat most of the QoS requirements. E-LSP allows one label, thus 8 different PHBs. This is enough to support a minimal configuration: EF, BE, and four AF classes with two drops precedence level.
- If a network has more than 8 PHBs, then 3 Exp bits will not be able to convey all the PHBs to LSPs. One way is to use the label itself to convey PHBs, therefore, an LSR uses the label to determine the PHB. For L-LSP PHB is determined from label and AF drop precedence is determined from Exp bits.

Both E-LSP and L-LSP can be used, but it depends on which kind of traffic is going to be forwarded. E-LSP is simpler to implement and is close to DiffServ. However, L-LSP is more sophisticated and could offer more possibilities when the QoS over MPLS is more mature. It depends on the requirements.

4.2.2 IntServ/Diffserv Mapping over DVB-S/RCS

Performances of MPLS are also determined by the capabilities of the satellite architecture lower layers, based on DVB-S/RCS technologies, especially, in terms of bandwidth allocation capabilities. MAC/DAMA concerns the satellite segment procedures for bandwidth assignments, which, on one hand, aim at an efficient use of the valuable satellite capacity while, on the other hand, should allow the respect of QoS requirements for IP traffic streams.

Two algorithms of bandwidth allocation exist:

- A Connection Admission Control (CAC) provides the RCST with a statically assigned capacity, which is always available to the ST.
- While a Bandwidth-on-Demand (BoD) mechanism provides the RCST with an additional amount of capacity which dynamically adapts to the current RCST traffic conditions.

Due to the long propagation delay associated with the satellite links, the BoD schemes implicitly introduce an additional delay. This additional delay, which is given by the

queuing delay in the RCST buffers, is in the range of 0.5-1 sec., depending on the BoD control policy.
Taking into consideration the **CoS contract** characterizing the MPLS traffic classes, the crucial point is to supply a proper mapping; possibly, the traffic classes might be supported by a mix of the capacity request categories.
The **CoS requirements** can be classified in bandwidth, delay and jitter constraints:

- Bandwidth constraints might impose that a defined amount of capacity is granted to the connection for the connection lifetime;
- Delay constraints enforce an upper limit on the queuing delay in the buffers;
- Jitter constraints require a limited delay variation in the transmission of the packets.

Due to the additional queuing delay introduced by BoD mechanisms, MPLS connections with stringent delay and jitter requirements – i.e., real time (RT) connections – cannot rely on dynamic access to the network: the proper amount of capacity, computed on the basis of the bandwidth requirements, must be statically reserved via Connection Admission Control (CAC) procedures.
On the other hand, Best Effort (BE) connections can avail of dynamic capacity assignments (via the BoD mechanism), since no particularly stringent delay requirements have to be satisfied. Furthermore, since no bandwidth requirements are defined, also the CAC procedure is not triggered; this means that BE connections are the most subject to congestions.
Finally, the connections with no jitter requirements, no (or loose) delay requirements but with bandwidth constraints – i.e., Critical Data (CD)– must be subject to CAC admission decisions but they may access a portion of reserved capacity via the BoD mechanism. In this way, these connections are not subject to congestions but, when the reserved capacity is not fully requested, the leftover part can be distributed among the best effort connections.
For each identified MPLS traffic category, the following table shows the CoS requirements, the suitable DVB priority class, the QoS guarantees and the functional modules, which allow the QoS requirements to be met.

IP classes	*CoS requirements*	*DVB priority class*	*QoS guarantees*	*CAC*	*BoD*
GS - IntServ, EF – DiffServ	Bandwidth Delay Jitter	RT	Bandwidth Delay Jitter	yes	no
CS-IntServ, AF-DiffServ	Bandwidth	CD	Bandwidth	yes	optional
Best Effort	-	BE	-	no	yes

Table 1. IntServ/DiffServ QoS classes mapping onto DVB-S/RCS

4.3 IP oriented architecture

Motivated by the need of more efficient and simplier to manage architectures, the next evolution of DVB-S2/RCS satellite network resides in IP over DVB, or at least as closed as it can. In fact, several encapsulation layers are still needed (MPE, ULE, GSE...) according to the referred architectures, but MPLS could be avoided to save its encapsulation overhead.
The satellite terminal becomes an **IP/Diffserv edge router**, and embeds all the mechanisms required to manage several PHB. The QoS is managed in three steps (Fig. 5).

- The first step take place at Medium Access Control level where the DAMA allocates the bandwidth on a fixed basis for the real time queue and on an on-demand basis for other queues (critical data and best efforts queues).
- The second step is at IP layer as described above. A dedicated IP level module implements a queue management system aiming at providing a differentiated service with regards to several service classes.
- The third step is at user layer level. The user classifies its own flows and provides the classification to a QoS server, located in the RCST, that configures its QoS architecture. The goal is to exploit the capabilities offered by the IP-level QoS.

In other words each step involves a service offering to the above.

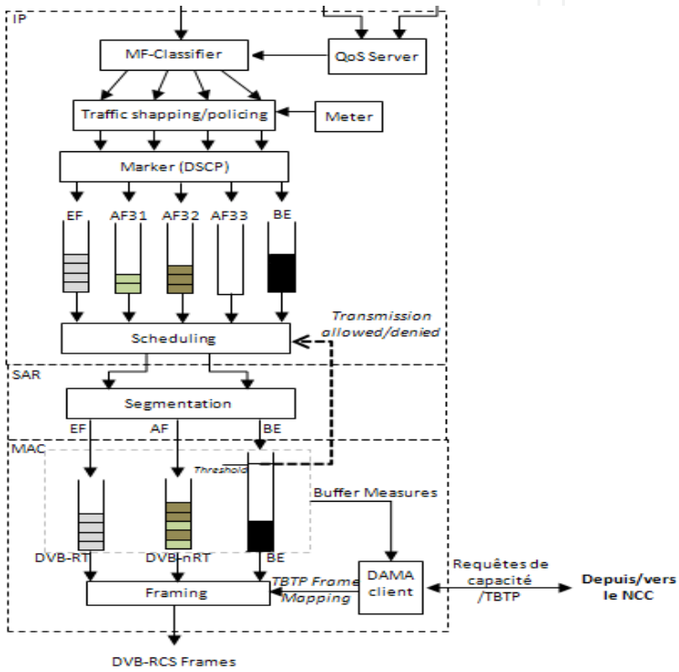


Fig. 5. The QoS Architecture

4.3.1 RCST QoS architecture components

A *Service Class* (SC) is given by the underlying applications behaviour that uses the service. In our example there exist five Service Classes for IP layer and three for MAC layer. Service classes for MAC layer are: Real-Time, Critical Data and Best Effort. Service classes for IP layer are: Real-Time (EF), three Assured Forwarding level (AF1, AF2, AF3) and one Best-Effort (BE).

- Real time (RT) service class: this is the service class for non-jitter, low delay flows typically filled with small packets. The underlying satellite bandwidth allocation mechanism (the DAMA at MAC level) should have reserved sufficient static bandwidth. This could be done at the terminal logon.
- AF service classes: these service classes deserve a special treatment with regards to best effort but with more priority. For exemple, signalling could use the AF1 queue, streaming the AF2, and web browsing the AF3.

- Best effort (BE) service Class: all the traffic that does not belong to real-time or critical data service class belongs to best effort service class.

A *scheduler* is used to differentiate IP queues and to map IP queues over MAC queues. This is a crucial component of the architecture. The EF class is generally mapped with the DVB-RT class and the AF queues are mapped over the CD queue that could use RBDC and VBDC allocation schemes. For this purpose, the traffic categories could be served by a scheduler using a simple priority queuing (PQ) discipline or a Weight Fair Queuing (WFQ). In fact, MAC queues should be as small as possible (only the necessary space to save the fragment of several IP packets) to reduce the end-to-end delay. To avoid MAC queues overflooding, interaction mechanisms (cross layering) between IP and MAC layer should be setup. For instance, when the BE queue (at MAC level) exceed a given threshold, a "backpressure" algorithm stops the scheduler to feed the MAC queue. This threshold could be set up according to the DAMA algorithm.

A *Traffic Flow* (TF) is a given data flow generated by protocol entities of a given application. A TF is uniquely recognized by a TF ID designing the originating network address, the destination network address, the originating transport service access point and the destination transport service access point. Classically in (TCP, UDP)/IP networks, this is given by the 4-tuple <IP-source, IP-destination, port source, port destination>. The *MF-Classifier* sets up the DSCP packet field according to the QoS policy.

A *shaper* is a specific algorithmic that enforces the emission profile of a flow according to its specifications (rate, delay, etc). A dual leaky bucket (DLB) is a simple solution to implement it but a Hierarchical Dual Leacky Bucket (HDLB) offers better flexibility in the queue management.

A *dropper* is a more stringent version of a shaper. In place of attempting to shape the traffic, it simply drops the non-conforming traffic. It does this by dropping each packet of the flow until the traffic goes back to a normal state. This can be implemented by reducing the input queue size of a dual leaky bucket to one.

4.3.2 Network components

The *Connection Admission Control* (CAC) algorithm in the NCC performs preventive congestion control by deciding the admission of new connections in the network. CAC decisions are taken on the basis of the QoS requirements and guarantee the availability of bandwidth and/or the fulfillment of delay requirements.

The elements in this QoS architecture that perform or can perform CAC are the following: the QoS server on the RCST side, which is informed of the user QoS requests; the ARC, Access Resource Controller (in IP-oriented architecture), which is a separate component located at the NCC side that takes the final decision according to the SLA, the network capacity, etc. To ensure the CAC procedure, a dedicated protocol has to be set up between RCST, NCC and ARC.

These components are mandatory to set up a Diffserv compliant QoS architecture in a DVB-S2/RCS system. However, this is not sufficient to have a perfect integration of satellite systems in NGN networks. The section 0 reviews the lack components.

4.3.3 Satlabs

To reach a better interoperability of DVB-RCS products, after several year of extensive work and discussions, both "Quality of Service" and "Management and Control Harmonization" Satlabs Working Groups have proposed recommendations (Combes & Pirio, 2008). With the incorporation of the recommendations into the SatLabs certification program, DVB-RCS certified that terminals will be able to interoperate in different systems requiring only changes to their configuration and to provide real-time services efficiently.
In line with the latest revision of the DVB-RCS standard, these recommendations detail how the following enhanced features operate in a DVB-RCS system:

- Harmonized support for IP Classes of Service according to the IETF DiffServ recommendations, and their mapping on DVB-RCS specific MAC layer Capacity Categories and Dynamic Requests,
- Standardized management procedures based on the IETF SNMP recommendations,
- A Common SNMP Management Information Base (MIB) for all DVB-RCS terminals,
- Standardized procedures and protocols for terminal's Configuration file and log files management,
- A Common protocol for terminal's software updates based on Multicast distribution.

As part of the short term QoS harmonization, SatLabs defines that the RCST shall support at least the following PHBs:

- Expedited Forwarding (EF)
- At least one Assured Forwarding PHB Class (AF3), as defined in RFC2597 , with at least two drop precedences
- Best effost PHB, as defined in RFC 2474.

4.4 Next Generation Satellite Networks

The following definition is an excerpt of ETSI's NGN-Starter Group conclusions: « The term NGN is commonly used to give a name to the changes to the service provision infrastructures that have already started in the telecom and IT industry. As such it is not a term that can be precisely defined but is rather an umbrella term to describe developments following PSTN/ISDN/GSM phase 2+ era.
One of the main characteristics of NGN is the *uncoupling of services and networks*, allowing them to be offered separately and to evolve independently. Therefore, in the NGN architectures proposals, there is a clear separation between the services functions and the transport functions. An *open interface* is provided between both. NGN allow the provisioning of both existing and new services independently of the network and the access type used. In NGN the *functional entities* controlling policy, sessions, media, resources, service delivery, security, etc. may be distributed over the infrastructure, including both existing and new networks. When they are physically distributed, they communicate over open interfaces. That is why the NGN architectures proposed in standards bodies and fora consist of layers and planes and show a lot of reference points. New protocols are being standardised to provide the communication between those functional entities. *Interworking* between NGN and existing networks such as PSTN, ISDN and GSM is provided by means of Gateways.
The following deals with the IP Multimedia Subsystem (IMS) architecture that could be considered as the most NGN mature architecture. The integration of DVB-RCS system in an IMS global network is presented.

4.4.1 IMS architecture

The QoS management in IMS architecture relies on several functional components:

- The Call Session Control Function (CSCF) is the standardized access point for an IMS user. It is in charge of the service policy management. According to the network architecture it could be a proxy (P-CSCF), a server (S-CSCF) or an edge entity (I-CSCF).
- The Policy Decision Function (PDF) is in charge of the service policy decisions.
- The Policy Enforcement Function (PEF) is an IP packet conditioner (traffic shaping) and a classifier.
- The Application Functions (AF) offers the services to the user.

IMS doesn't implement directly the QoS management but relies on the underlying networks. IMS standardizes the interface for the QoS management of these networks.
When considering access networks, the interface between PDF and the access link is ensured by COPS and the interface between the user terminal and the P-CSCF is based on SIP protocol.

4.4.2 Resource provisioning using SIP in DVB-RCS networks

In the IP oriented satellite architecture, the QoS functions are triggered in the RCST by the QoS Server as explain before, but this could directly be done by application servers to avoid user configuration. A SIP Proxy (Alphand, 2005) with extended functionalities has been developed to automate QoS reservations with multimedia applications. SIP (Session Initiation Protocol) should be used by multimedia applications for the session management and codec negotiation. The main functionality of a SIP proxy is to route SIP messages (session establishment for instance) from a SIP entity (an application) to another SIP entity. To remain the QoS reservation transparent for the application, the enhanced SIP proxy has to extract SDP (Session Description Protocol) descriptions from SIP messages exchanged between caller and callee, translate the parameters of each media in the session in QoS characteristics and send QoS reservations with the QoS Server on behalf of the application. The QoS-Aware SIP proxy could be centralized in the operator network part, but it could be also implemented in each RCST to reduce the connection setup time.

4.4.3 Integration of DVB-S2/RCS networks

The IP oriented architecture presented before is compatible with the implementation of the QoS in the IMS architecture (Baudoin, 2009) according to several modifications.
Compared to this medium term architecture, the P-CSCF functions are managed by a QoS-Aware SIP proxy collocated with the Gateway. The ARC should act as a PDF. It has "enforcement" functionalities when it configures at the same time the DAMA. The QoS Server has the role of QoS enforcement function (PEF) and could be triggered by the ARC using the COPS protocol. The SIP protocol is used continuously during the QoS configuration.
However, the IMS concept is well suited to a deployment in mobile networks, where end-users are directly linked across the wireless link. Conversely, in the DVB-RCS model, end users and satellite networks are separate entities, which is the major obstacle for IMS integration in DVB-based networks. DNS or DHCP could partially provide solutions. Moreover, accounting in satellite systems is generally private and centralized. Efforts have to be done to adapt it to a larger and compatible accounting architecture.

5. Evaluation of a DVB-S/RCS satellite system

5.1 Methodologies

In this section, we will describe the different methodologies allowing evaluating the performance of a DVB-S/RCS satellite system.

5.1.1 Simulation: NS-2

Development and implementation of new services on satellite networks take time and money, so before being implemented on a real system, an implementation on a test-bed has to be done. The downside of emulation is the time needed to build and configure it, moreover it needs consequent resources and it is not easily scalable. Simulations allow overcoming some of these issues by providing a scalable and effortlessly reconfigurable network at lower cost, the major disadvantage being the realness of the results obtained since they are ruled by the model accuracy.

Network Simulator 2 (NS-2) is a discrete event simulator for networks research. Supported by DARPA and NSF, its modularity made it one of the most famous network simulators: researchers use it all around the world to implement new components or to check theoretical works. NS-2 provides protocols and communication mediums that can be settled depending on simulations needs. Updates from the development team and patches from research groups allow NS-2 to simulate current networks and latest protocols. NS-2 handles simulations of satellite networks but several components are missing, dynamic bandwidth allocation is provided by patch and there is neither QoS architecture nor encapsulation scheme. The TDMA-DAMA mechanism has been implemented and tested in (Gotta et al., 2006); the MAC layer introduced is used by our model to represent the dynamic bandwidth allocation (see section 0).

5.1.2 Advanced Emulation testbed: PLATINE

Simulation and emulation both provide the opportunity to evaluate performances, at low cost, on more or less realistic systems. When simulation needs a complete modeling of the systems from applications to physical network and operates in virtual time, emulation is more demonstrative since real applications can be deployed over the model describing transfer characteristics, delay and error behavior for instance.

The (IST SATSIX, 2004) project relies on these different methods of evaluation and validation, with the design and development of simulations, emulations and trials. The emulation testbed relies on the satellite emulation platform (PLATINE) (Baudoin et al., 2007) formerly developed within the frame of the (IST SATIP6, 2001) project. It aims to demonstrate the network and application services integration on next generation satellite systems and the possibility to interoperate with terrestrial networks. With regards to the previous release, PLATINE includes a DVB-RCS and DVB-S2 emulation, with ULE/MPEG2-TS and AAL5/ATM stacks together with the adaptive physical layer simulation and the associated radio resources management (RRM). A complete QoS architecture mixes ehanced SIP proxies (as described in 0), IP/MAC scheduling and cross layer techniques is available.

5.1.3 Real experiments: OURSES

During the (OURSES, 2006) project, we had he opportunity to use a DVB-S2/RCS system. A platform compliant with the IP oriented architecture was setup during the project. The gateway and the terminals are compliant with the Satlabs recommendations. The four (VoIP, ViC, Critical Data, Best Effort) Diffserv class of service are offered on the STM satlink 1000 terminals we used. A Service Level Agreement (SLA) is setup on the gateway side (Thales A9780 model) for each customer. It fixes the limits in terms of bandwidth with each MAC service classes. The tests were done using a satellite channel emulator and the Ka band.

5.2 Performance evaluations

5.2.1 DVB-S/RCS NS-2 simulation model with QoS

This section briefly describes the DVB-S/RCS NS-2 simulation model with QoS architecture that have been developed at LAAS/CNRS, further details on implementation and simulations can be found in (Gayraud et al., 2009). Such model can be used to simulate new protocols or to compare results with measurements obtained through emulation or experimentations done on a real link. To be efficient, architecture and behaviour have to be as closed as possible from the chosen satellite network (the one from OURSES project in our case). The model is using a TDMA-DAMA MAC layer above the physical layer defined by NS-2. The simulation made on the model without our contribution shows a really efficient behaviour with a dynamic bandwidth allocation and a fast establishment of connections. However, to be closer from the real system and to improve performances, some features need to be added:

- Dynamic encapsulation of IP packets,
- Substitution of the single queue at MAC layer by two distinct queues,
- Addition of queues at IP layer (inspired from DiffServ architecture).

Since packets fragmentation is not possible with Network Simulator 2, the MAC layer adjusts the sending time to the available bandwidth based on the assigned slots. The dynamic encapsulation (from IP to ATM frames) doesn't fragment the packets either, but their size is settled according to AAL5 protocol, resulting in a consistent overhead (around 10 percent).

QoS architecture is implemented by duplicating the queue at MAC layer and by adding buffers at IP layer: flows are aggregated, differentiated and stocked according to the DIFFSERV architecture from terrestrial network; management is done by a packet scheduler below the queues. To study the model behaviour, two kind of traffic with specific constraints were generated:

- Constant Bit Rate (CBR) needing low delay and jitter, associated to Real Time flows (RT),
- File Transfer Protocol (FTP) needing large bandwidth regardless to delay, associated to non Real Time flows (n-RT).

The chosen transport protocols are respectively UDP and TCP, the most commons for such flows. During experiments, the available bandwidth is settled to 128kbps per slot (this ratio depending on weather conditions and chosen coding scheme), since one satellite terminal can have at most two slots (256kbps), CBR rate has been settled to 128 kbps (without encapsulation overhead). Flows will compete with each other, the main point of simulations being to show the efficiency of the QoS architecture added to the model: CBR flows should

get the lowest delay possible while FTP flows would still be able to establish communication and transfer data.
The rest of this section will focus on the QoS architecture by taking a look at the model's behavior. To illustrate the competition between the two types of flows, delay suffered by the communications and the throughput they can achieved are shown on Fig. 6.

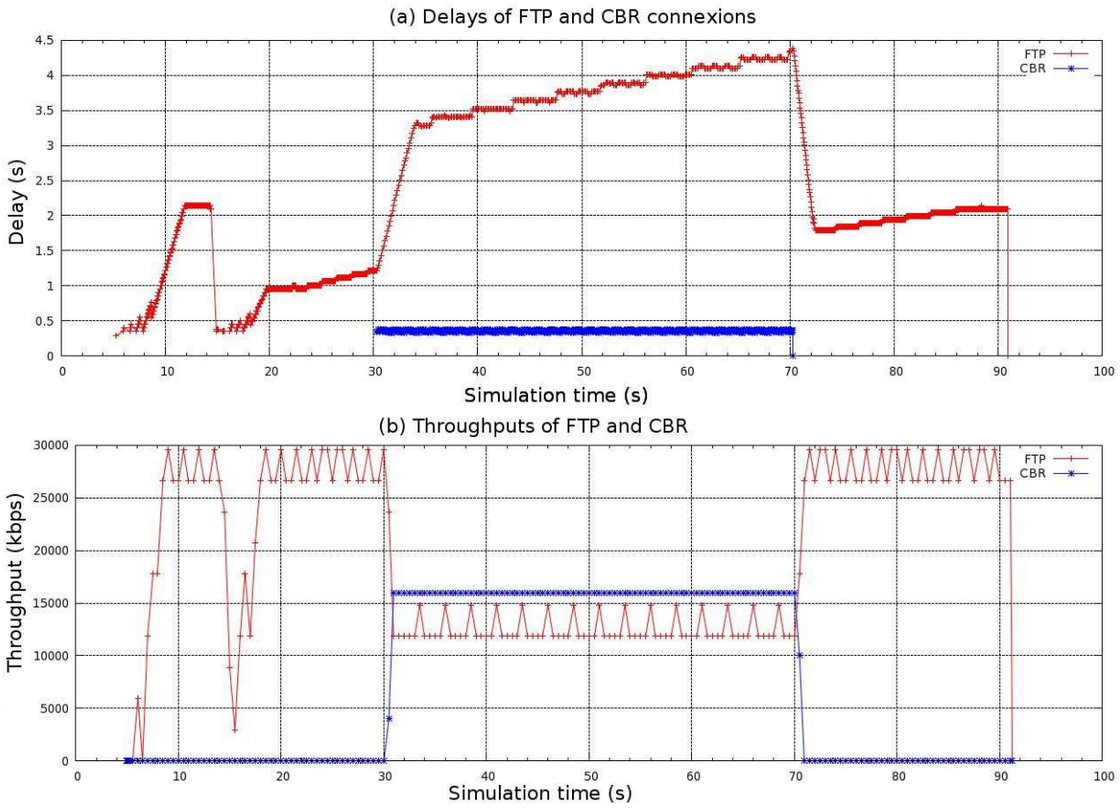


Fig. 6. a) Delay suffered by connections. b) Throughput of connections.

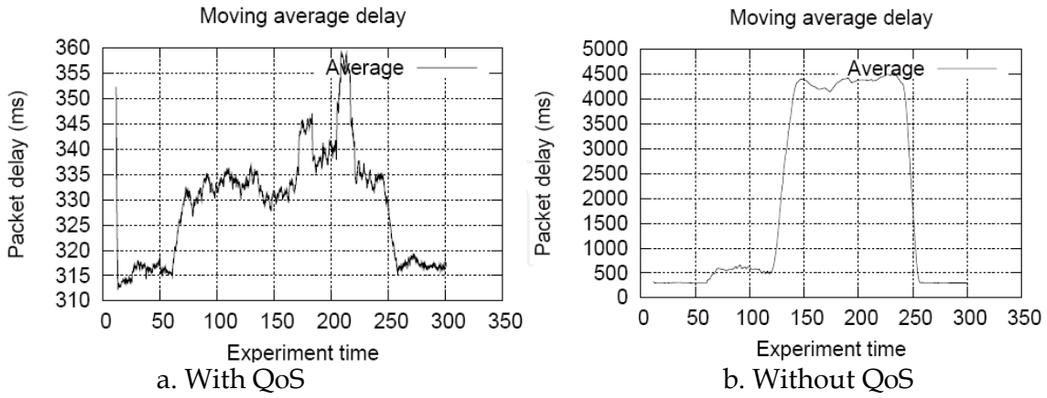
Differences between RT and n-RT flows are clearly visible: delay suffered by CBR is stable and below 500ms while FTP delay fluctuates and is above 3s (Fig. 6a). On Fig. 6.b, it is noticeable that FTP throughput is restricted by bandwidth taken by CBR. These two results illustrate the model's behavior by showing the differentiation done on those flows; the one with more constraints is getting the lower delay possible and enough bandwidth so no loss occurs. For n-RT flow, the throughput and the delay are fluctuating depending on network load and bandwidth allocated to the satellite terminal.
The model behaves properly and reacts as we expect: indeed; it provides an efficient QoS architecture to the basic satellite network from NS-2. But some improvements can still be done on the model: using a more efficient manager below the MAC buffers and providing a thinner encapsulation mechanism. There are also some features needing to be tested: using RED instead of DropTail policy in satellite terminal buffers or using a more realistic error model (already implemented but not used during simulations). The latest experiments were done to study SCTP (Stream Control Transport Protocol) behavior on a QoS satellite network and compare it with TCP; results can be found in (Bertaux et al., 2010).

5.2.2 PLATINE performances evaluations

In the following parts, we will show two exemples of QoS management in DVB-S2/RCS satellite systems, using the VisioSIP client (a SIP videoconferencing tool) and a QoS-aware SIP proxy (located behind each ST and based on the NIST-SIP Proxy) that send reservation or release messages to a QoS Server, located on RCSTs and able to reconfigure DiffServ queues to prioritize flows with strong time-constraints (VoIP, videoconferencing, etc...). Moreover, we consider that each ST has a total bandwidth of 1000kbps.

5.2.2.1 Impact of the queue management: BE vs EF

We consider here that a SIP videoconferencing session is initiated between two SIP clients located behind two separate STs. The SIP session starts at t=t0 + 10s and then 3 concurrent UDP flows (500 kbps) start respectively at t=t0+60s, t=t0+120s and t=t0+180s and terminate at t=t0+240s. Finally the SIP session ends at t=t0+300s. Moreover, 150 kbps of CRA is allocated to the studied ST to support, in terms of bandwidth, the video and audio flows.
We will make the analysis on the audio delays graphs presented on Fig. 7, but the same analysis will apply to the video delays graphs that are similar.
These two series of delays' graphs show a real benefit of the IPv6 QoS usage and a fair separation of the classes of service can be observed on the first graphs. Detail analysis of those graphs is now provided.
First, concerning the comparison of the graphs with and without QoS, it can be observed a real improvement when the QoS architecture is running especially when background traffic is high: The "moving average delay" graphs show that when two or three concurrent flows are running (between 120 and 240 ms) a very high increase of average delay is experienced by the audio flow when the QoS is not set (above 4 seconds delay) while the average delay remains below 360 ms when the QoS is set, which is compatible with audio conference requirements. In the case of high load on the satellite return link, the impact of the QoS architecture is clearly shown here.



a. With QoS b. Without QoS

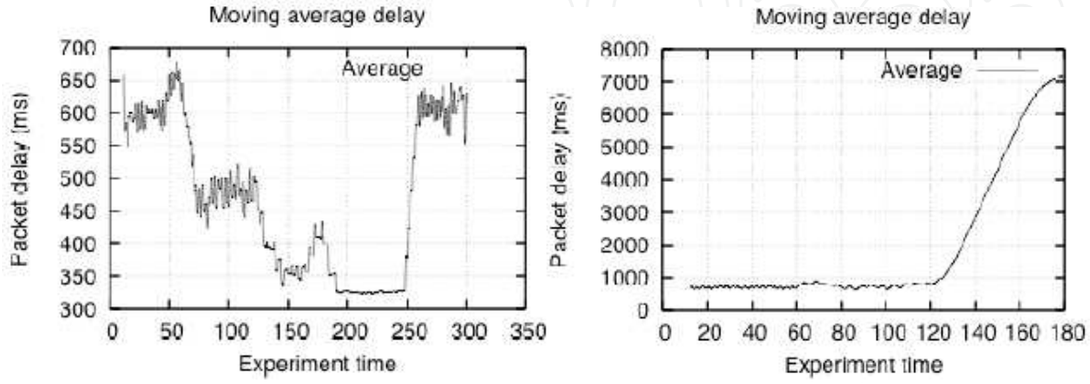
Fig. 7. Moving average delay for the audio flow

When no concurrent flows are running, delay for the audio flow is around 300ms in both cases (with and without QoS), cf. graphs between 0 and 60 seconds. This can be explained by the fact that all CRA resources, in this case, are used by the multimedia flows and no on-demand capacity is needed. When just one concurrent UDP flow of 500kbps is running the delay of

VoIP application is increasing in both cases but very slightly when QoS is set (from 315 ms up to 330 ms on average) while it's increasing up to 500 ms on average in the case no QoS is set. The capacity of the channel should be enough for both flows but the CRA capacity is not enough and on-demand RBDC bandwidth is required. So the audio flow experiences more delay when QoS is not set; this is due to the fact that all flows (audio, video and best-effort flows) are using the same MAC buffer and PVC, and so the same delay is experienced by all packets in this buffer, implied by the capacity allocation scheme. When the QoS is set, a different MAC buffer and PVC is used for high priority traffic (audio and video packets) and is served first compared to the low priority MAC buffer. Consequently, the audio flow is protected and the delay is increasing very slightly: it's experiencing an end-to-end delay compatible with audio conference application requirements (under 400ms).
Secondly, concerning the classes of service separation, we can notice on the first graph (a) with QoS that the impact on high priority classes of service of concurrent flows is rather low, and does not degrade the overall quality for end-to-end users: the delay remains below 400ms which is acceptable for interactive audio conference applications. The delay increases from 315 ms up to 360 ms, which can be explained by the sending time for large low priority packets.

5.2.2.2 Impact of the RBDC mechanism on interactive applications

The following experiments show the impact of the DAMA algorithm on interactive applications. The teleconferencing application takes the EF service class and the background traffic the Best Effort service class, but, unlike the previous experiment, there is no CRA allocated to this Satellite Terminal, all the capacity is given with RBDC requests. The teleconferencing application is first started, then 3 concurrent UDP flows of 400 kbps start and terminate at the same time than the previous experiment.
On Fig. 8.a., the delay experienced by the audio stream is less than 700 ms (this is the same values for video stream) and decreases to very low delay. The first noticeable thing is that the DAMA algorithm works fine with audio and video streams. The delay stays stable, around 650 ms, even with the throughput variation. The second noticeable thing is the delay diminution that occurs during the experiments. This can be explained by the fact that the teleconferencing application takes benefit from the RBDC requests made for background traffic as this traffic has a better priority.



a. Audio stream delay b. First UDP flow delay

Fig. 8. Moving average delay for the audio flow

On Fig. 8.b, as the link capacity is not reached, the packet delay is stable, below one second but, of course, when there is no more capacity, the delay increase, but only for the Best Effort Class.
The main problem, in this case, is that the delay of audio and video flows is often higher than what is advised in ITU-T recommandations (ITU-T, 2001), namely a value inferior to 400 ms. Consequently, to provide a solution to lower the delay given with the RBDC mechanism when no CRA (or no sufficient CRA) are allocated to a specific ST, a new extension of the SIP Proxy has been proposed to allow it to communicate with an entity located at the NCC side: the Access Resource Controller (ARC). When a SIP session is initiated, the SIP proxy can intercept the SDP, deduct the codec bitrate and ask to the ARC to increase the quantity of CRA allocated to the concerned ST corresponding to the sum of codec bitrates. The ARC checks if the SIP clients are authorized to use this service and decides to accept or reject the resource reservation.

6. Conclusion

This chapter has explained the way that can be used to provide such a satellite network client with the QoS he requested. It was proven that these QoS archtectures are feasible, that their performances are good enough by several actions like simulation, emulation, and real systems.
The work on QoS architecture is still ongoing and heterogeneous access networks mixing satellite and other radio techniques such as Wimax, and wireless systems in general. This work will lead in the very next future to the implementation of some of ours. It seems that the first network ensuring QoS may be the satellite systems that were described, designed and evaluated in the work as described in this paper.

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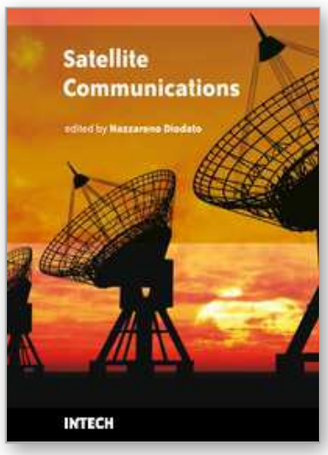
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This study is motivated by the need to give the reader a broad view of the developments, key concepts, and technologies related to information society evolution, with a focus on the wireless communications and geoinformation technologies and their role in the environment. Giving perspective, it aims at assisting people active in the industry, the public sector, and Earth science fields as well, by providing a base for their continued work and thinking.

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