Scalable QoS Management in Next Generation GEO-Satellite Networks

Floriano De Rango[1], MauroTropea[2], Peppino Fazio[1], Valeria Loscrì[1], Salvatore Marano[1]
D.E.I.S. Dept., University of Calabria
Via P.Bucci, 87036 Rende (CS), ITALY
Email[1]:{derango, pfazio, vloscri, marano}@deis.unical.it
Email[2]: mauro.tropea@tin.it

Abstract– **This paper focuses on the Architectural changes and adaptive QoS mechanisms that should be supported by next generation Satellite networks. Scalability is an important issue in the view of heavy traffic load support on the Satellite links and in the integration of Terrestrial wired and wireless backbone and the Satellite segment. QoS Adaptivity, on the other hand, is an important requirement that QoS architectures, MAC, network and application layer protocols should offer in order to support multimedia traffic (e.g. MPEG-2) and to meet QoS requirements in time changing wireless channel conditions (e.g. long and short term fading). After a brief overview of IP-QoS Scalable architectures in GEO-Satellite networks, in order to offer a scalable IP end-to-end QoS among satellite terminals, QoS mechanisms of the Scalable CORE architecture have been considered in a DVB-RCS Satellite system and an adaptive resource management has been applied. EF and AF have been considered in the SCORE Satellite network and a Traffic Resource Management with EF and AF services has been evaluated. Performance evaluations have been considered in terms of Burst/GOP loss ratio, satellite utilization and admitted calls.**

Index Terms-Scalable Satellite Network, SCORE, DiffServ, IP-QoS, CAC, TRM,

I. INTRODUCTION

Since the launch of the first satellite in orbit these systems have been considered an important element of telecommunications networks serving, in particular, long distance telephony and television broadcasting. The cooperation between satellite and terrestrial networks is a new trends in the world of telecommunications. The satellite capability of covering large geographical region places the satellite in a privileged position in communications systems. In a global telecommunication system, conceived with a terrestrial and a space segment, the satellite constellations, can play an important role [1-4]. Hybrid satellite and terrestrial solutions will provide an interconnectivity with distant/isolated nodes of the terrestrial network. Moreover these new hybrid platforms resolve the problem due to the increasing worldwide demand for more bandwidth.

Today it is clear that satellite networks will be a significant player in the digital revolution, and will specially benefit from on-board processing (OBP) and switching, as well as other such technological advances as emerging digital compression, narrow spot beams for frequency reuse, digital intersatellite links, advanced link access methods and multicast technologies.

A key design issue for satellite networks include efficient resource management schemes and QoS architectures [5-6].

In the last years the IETF has provided two different QoS architecture to resolve the problem of the currently Internet Protocol (IP) that only has minimal traffic management capabilities and provides best effort services.

They are Integrated Services (*IntServ*) and Differentiated Services (*DiffServ*) [7-9]. The first one manages the traffic and the QoS on flow basis, introducing high overhead in terms of signalling and state information. The second architecture, instead, adopts a different paradigm, offering more scalability to the network through the QoS management on a class basis. However the *DiffServ* model does not offer strict guarantees for single flow.

The most scalability of the second architecture carried an increased interest in developing *DiffServ* architectures for provisioning IP QoS. *DiffServ* aims to provide scalable service differentiation in the Internet that can be used to permit differentiated pricing of Internet service. It performs traffic classification and conditioning only at network boundary nodes. In order to achieve the advantage of the *IntServ* architecture in *DiffServ* one, recently a new type of network architecture called Scalable CORE (SCORE) has been proposed because it performs guaranteed services on an aggregated basis without maintaining state info in the core routers by the use of Dynamic Packet State (DPS) technique which allows the state info on a flow basis in the core routers to be eliminated [10-11]. This work applies the SCORE architecture on a Geo-Satellite segment and proposes a scalable way to manage traffic resources through a distributed local admission control and a scheduling, based on the state info carried by the data packet such as suggested by the DPS approach. The DVB-RCS platform has been considered as the reference architecture [12-15].

This paper is organised as follows: section II gives an overview of two scalable approaches applied in the satellite networks; call admission control and traffic resource management on Satellite network are introduced in section III; the reference scenario is addressed in section

IV; section V presents the simulation results; finally the conclusions are summarised in the last section.

II. Scalable Satellite Networks

In order to apply the satellite segment as a network element able to interconnect heterogeneous networks such as depicted in fig.1, a scalable architecture should be considered [16,23]. The satellite network, with the increasing OBP capability, is becoming an element able to make routing/switch decision and able to serve a lot of traffic connections. As well as the core routers in a IP backbone should offer high scalability because of the high number of connections that they serve, so the Satellite node should offer IP QoS avoiding to maintain heavy per-flow state info. Two IP QoS architectures with good scalability properties are introduced in the following and our attention will be focused on the SCORE architecture [10,11].

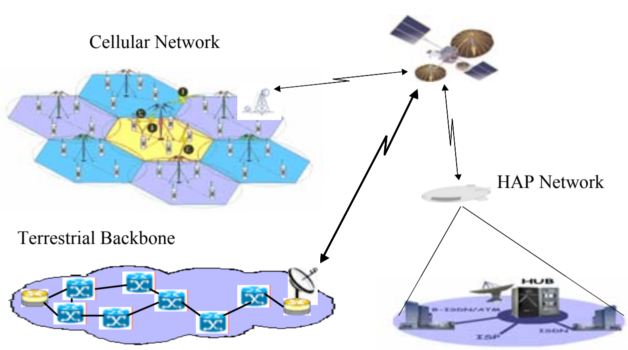

Fig.1 Satellite segment: a key network element to connect heterogeneous systems .

A. Differentiated Service Architecture

The most scalable approach proposed in the IETF is the *DiffServ* architecture [9]. It applies an IP packet classification at the boundary of the network and an appropriate mark, called DS codepoint (DSCP), is inserted in the packet header. This marking mechanism is necessary to the management of data packet in the core network though some QoS policy mechanism called Per Hop Behaviour (PHB).

The *DiffServ* architecture can be applied to the Satellite scenario for the following reasons:

I. High state info scalability due to the per class traffic management.

II. Easy integration with the terrestrial backbone than can make use of *DiffServ* paradigm.

III. High protocol scalability with reduced signalling overhead, due to the marking mechanism and QoS policy distributed among the RCST terminal and GTW.

The PHBs are associated to the QoS traffic class and they are divided in two main categories:

- Expedited Forwarding: (EF) [29]: it is adopted for real time traffic that need to receive QoS guarantees in terms of bandwidth, maximum delay and/or jitter delay.
- Assured Forwarding (AF) [18,19,30]: it is suggested for non real-time media streams and for application delay tolerant. This class assures timely delivery of data packet when the network is not overloaded and a degraded services when the congestion is reached. It is divided in four independent forwarding sub-classes with different discarding probabilities.

The *DiffServ* architecture is a good candidate for terrestrial backbone and some recent studies exist that applied *DiffServ* approach to the GEO satellite network [22]. However the pure *DiffServ* paradigm is not able to guarantee single flow mechanism due to the per-class traffic management and to avoid carring state info about the network or flow [24]. In order to obtain fine QoS granularity preserving the state scalability of *DiffServ* paradigm, a novel architecture called SCORE can be considered. It is a *DiffServ*-like architecture with some differences such as explained in the following.

B. SCORE Architecture over Satellite

SCORE architecture has the advantage of presenting a deterministic control on an aggregated basis. It permits to reach an high scalability on the Satellite node where a lot of connections can be served. In a similar way to the approach followed in [4], the Aggregate RSVP protocol [28] is applied in the SCORE network in order to reduce the control messages on the satellite network while preserving network utilization. The considered SCORE architecture uses a differentiated management of the routers which compose the network. In particular, it makes a distinction between border nodes and inner nodes. The DPS is the technique applied in the SCORE architecture to avoid the storage of state info on flow-basis on the core router. An example of SCORE architecture applied to Satellite network is depicted in fig.2

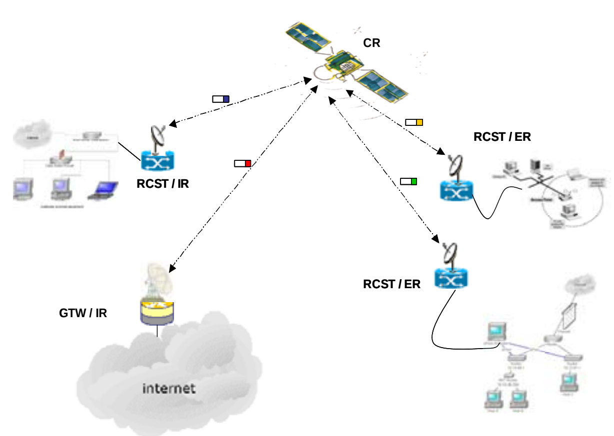

Fig.2 DPS technique and SCORE Network over Satellite segment.

With DPS, each packet carries in its header some state that is initialised by the ingress router (IRs) (fig.2). Core routers (CRs) process each incoming packet based on the state carried in the packet's header, updating both its internal state and the state in the packet's header before forwarding it to the next hop. At the end, the egress node (ER) removes states from the packet's header.

This technique is used in two algorithms, one for the data plane to schedule packets and the other one for the control plane to perform admission control.

The schedule algorithm is called Core Jitter Virtual Clock (CJVC) [10] which is implemented like the Jitter Virtual Clock.

This choice is made principally because in the Jitter Virtual Clock packet's deadline depends only on the state variables of the flow it belongs to.

CJVC uses the DPS technique. The key idea is to have the ingress node to encode scheduling parameters in each packet's header. The core router can then make scheduling decisions based on the parameters encoded in packet headers, thus eliminating the need for maintaining flow state at the core nodes. For details see [10,11].

The control plane that performs the CAC algorithm is based on the idea to overestimate the aggregated bandwidth of the calls accepted on the path. This policy permits to be conservative and to avoid loss of burst or gop.

However, in order to increase the link utilization, especially for high burstiness traffic, a *periodical recalibration* is applied on the RCST terminal and Satellite node. This recalibration permits to reduce the waste of bandwidth produced by the R_{bound} overestimation. R_{bound} represents the sum of single rates of any accepted call. To perform recalibration, a new variable R_{DPS} is periodically recomputed by the Egress Router (IR) and by any router on the forward path from IR to ER to account for the actual traffic flowing in the network.

Time is divided into intervals of dimension T_W: $(t_k, t_{k+1}]$ with k>0. The IR monitors the amount of bits transmitted by any flow *i* in the interval T_W. $b_i(t_k, t_{k+1})$ is denominated as the sum of the bits received for flow *i* in the interval $(t_k, t_{k+1}]$. This information is included in the IP packet header by the IR and it is used to update the variable $B(t_k, t_{k+1})$ by all crossed *CRs*. $B(t_k, t_{k+1})$ is the sum of bits $b_i(t_k, t_{k+1})$ of all flows *i* belonging to the same *DSCP* crossing the router.

So at the end of any interval T_W, each router computes the actual rate R_{DPS} and then, based on R_{DPS}, it computes a new variable R_{cal} defined as follows:

$$[R_{DPS}(t_{k+1})] = \frac{[B(t_k, t_{k+1})]}{t_{k+1} - t_k} = \frac{[B(t_k, t_{k+1})]}{T_W} \tag{1}$$

$$[R_{Cal}(t_{k+1})] = \frac{[R_{DPS}(t_{k+1})]}{1 - f} + R_{new}(t_{k+1}). \tag{2}$$

where *f* represents the frequency of rate recalibration, and R_{new} represents a variable locally initialised by each router at the beginning of any new interval T_W and accounting for the contributes in terms of rate R of any new flow accepted $R_{new} = R_{new} + R$. So it is possible to recalculate R_{bound} this way:

$$R_{Bound}(t_{k+1}) = \min\left(R_{Bound}(t_{k+1}), R_{Cal}(t_{k+1})\right) \tag{3}$$

R_{bound} is calculated by each core router and RCST/ER. The local admission control verifies that for each CR or ER, the condition 4 is verified:

$$R_{bound} < C \tag{4}$$

where C is the capacity associated with the link.

In order to apply this mechanism in a multi-beam satellite network, a R_{bound} variable need to be stored for each spot beam on the Satellite node and each satellite terminal. The scalability of the system is guaranteed because the number of spot-beams is low in comparison with the number of connections served by the satellite network. Thus the SCORE architecture applied on the Satellite network offers more scalability than Integrated Service over Geo-Satellite preserving the single flow QoS guarantee.

The aggregate bandwidth estimation procedure and the recalibration mechanism can be applied indifferently for EF and AF services and the different resource management is applied on the Traffic Resource Manager on the Satellite payload. In the following, the differences in the admission control phase and the satellite resource management are presented.

III. Call Admission Control and Traffic Resource Management

Call Admission Control (CAC) on Network Control Centre (NCC) and Traffic Resource Manager (TRM) on Satellite node need to be harmonized to respect QoS constraints such as burst or GOP loss ratio, end-to-end maximum delay and so on [25,26]. CAC and TRM module applied in this work for AF and EF services are briefly explained in the next sub-section.

A. EF Services

The EF services refer to application that are delay sensitive such as audio and video traffic. Two main classes can be considered belonging to the EF PHB: maximum delay assured and MPEG traffic.

The first category needs a specific bandwidth to respect a maximum bandwidth target in the worst situation and the second one presents a highly variable rate nature due to the standard MPEG encoding [27]. Both of them can be associated to the EF class, but a dynamic management of *Service Level Agreement* (SLA) at the RCST/ER should be applied. Taking advantage of the next generation of Satellite system that can use the return channel via satellite (RCS), it is possible to request the bandwidth on burst basis. Thus a semi-permanent channel assignment such as introduced in [6,25,26] can be applied. The multimedia MPEG traffic can also request a bandwidth on GOP basis and the satellite can release the bandwidth when the GOP has been transmitted permitting to other terminal to request bandwidth resources. A dynamic permanent and semi-permanent management is based on the traffic characteristics and it depends on the burstiness of data traffic. For example, as explained in [6], if the burstiness and token bucket parameters of a traffic source fall in the range expressed in table I an ideal mapping can be applied. For details to see [6].

TABLE I
MAPPING ON THE SATELLITE :LINK

ß \ b/r	<1.8s	>1.8s
<1.5	Permanent	Permanent
>1.5	Permanent	Semipermanent

Depending on the application layer, it is possible to update the aggregate bandwidth of EF class following the R_{bound} approach for maximum delay tolerant application and a criteria is presented below for the multimedia application.

Statistical Multiplexing based on the Normal Distribution algorithm (SMND) is a good candidate for multimedia traffic because it takes advantage of some properties of multimedia MPEG traffic stream [20,32]. In particular, a bandwidth expansion factor (BEF) is defined for VBR traffic (e.g. MPEG) so that the aggregate instantaneous rate exceeding the fraction of the capacity of the VBR traffic will not be greater than a pre-specified threshold value γ:

$$P_r\{R_{bound} > B_T\} = \int_{x=B_T}^{\infty} f_x(x)dx \le \gamma \tag{5}$$

with $R_{bound} = \sum_{i=1}^{M} R_i$ representing the aggregate instantaneous bandwidth rate and $B_T = BEF \cdot \sum_{i=1}^{M} \mu_i$.

Previous studies [21] have assumed that a generic multimedia MPEG traffic stream can be modelled by a statistical Normal Distribution of Group of Pictures (GOP) which is characterized by a mean data rate value (μ) and a standard deviation value (σ); thus, considering independent multimedia traffic streams, the aggregate of *n* multimedia streams can be considered like a flow characterized by a Normal Distribution with mean data rate as sum of the single mean data rate (μ_{TOT}) and standard deviation (σ_{TOT}) as the square root of the sum of single variance σ_i^2 :

$$\mu_{TOT} = \sum_{i=1}^{M} \mu_i \text{ and } \sigma_{TOT} = \sqrt{\sum_{i=1}^{M} \sigma_i^2} \tag{6}$$

Thus the outage probability p_o of the system can be determined as follows:

$$p_o = P_r\{R_{bound} \ge B_T\} = \int_{B_t}^{\infty} \frac{1}{\sqrt{2\pi\sigma_{TOT}^2}} \cdot e^{\frac{-(x-\mu_{TOT})^2}{2\sigma_{TOT}^2}} \tag{7}$$

And the *(k+1)*-th MPEG call is admitted if the eq. 8 is verified [20]:

$$R_{bound}(k+1) = R_{bound}(k) + \mu_{k+1} + BEF \cdot \left(\sqrt{\sigma_{TOT}^2(k) + \sigma_{k+1}^2}\right) < B_T \tag{8}$$

where $R_{bound}(k+1)$ is the aggregate bandwidth accounting the *(k+1)-th* admitted call, $R_{bound}(k)$ is the aggregate bandwidth at the previous step and the BEF value is obtained by the table in [17] that guarantees that the p_o does not overcome the threshold γ ($p_o \le \gamma$).The parameters adopted by SMND algorithm are presented in table II.

TABLE II
PARAMETERS OF THE SMND CAC ALGORITHM

	SMND Parameters
M	Number of MPEG traffic sources
R_i	bandwidth associated with i-th source
B_T	capacity assigned to VBR traffic
α	bandwidht expansion factor
μ_i	average rate of the i-th source
$f_x(x)$	probability density function (pdf) of the aggregate rate

For details about an SMND-like approach to see [20,32].

In order to unify the admission of multimedia traffic (e.g MPEG) and maximum delay bounded traffic, a change in the computation R_{new} variable needs to be made.

$$R_{new} = \sum_{i=1}^{M} \left(\mu_i + BEF \cdot \sqrt{\sigma_i}\right) + \sum_{i=1}^{N} r_i(t_k, t_{k+1}) \tag{9}$$

where M is the number of MPEG active calls and N is the number of *maximum delay-bounded* accepted calls in the interval $[t_k,t_{k+1}]$. r_i represents the rate requested by the *maximum delay-bounded* calls to guarantee a fixed delay bound DB_i. r_i can be calculated such as expressed in eq.10 and eq.11 and according with *IntServ* paradigm [8]. For details to see previous work [6].

$$r_i = \frac{p(b-M)+(p-r)(M+C_{tot})}{(p-r)(DB-RTT_{SAT})+(b-M)} \qquad \text{if } r \le R < p \qquad (10)$$

$$r_i = \frac{M + C_{tot}}{DB - RTT_{SAT}} \qquad \text{if } r \le p \le R \qquad (11)$$

In (10) e (11) C_{tot} and D_{tot} are two error terms, which represent how the network elements' implementation of single data flow deviates from the fluid model [6]. While p, r, b, M are the peak rate, average rate, depth size of the token bucket at source terminals and M is the IP datagram size. Thus, it is possible to have two kinds of EF traffic sources: MPEG and maximum delay bound ON/OFF source. The first one is characterised by an average GOP rate μ and a standard deviation σ. The second traffic source is characterised by a maximum delay bound DB that permits to calculate the rate such as expressed in eq.11. a new EF call is admitted if the condition expressed in eq.8 is verified.

B. *AF Services*

The AF class is characterised by a delay tolerance and it is suitable for web browsing, e-mail access, FTP services and so on. In order to increase the total satellite utilisation, it is possible to take advantage of this characteristics and to make use of smoothing factor such as presented in [31], to reduce the bandwidth of call belonging to AF class and freeing the bandwidth for other calls such as EF services or BE services.

Previous studies on smoothing factor showed a dependence of resource allocation for non real-time traffic by traffic burstiness β and traffic load ρ. In particular, when the satellite system has low EF traffic load and the traffic burstiness is high, a lot of non real-time calls can be accepted by the system to increase the satellite link utilization. This is due to the delay tolerance of calls belonging to AF class. Thus if a new AF call arrives, checking the traffic load on the satellite segment and the estimated burstiness of the new call, it is possible to select the best allocation for the AF services. In section V an example of this approach is given.

The CAC module requests two parameters for admitting an AF call: burstiness value for regulating the statistical multiplexing of traffic sources and a bandwidth requests for assigning a certain number of satellite channels. For AF class, the bandwidth request in the admission phase or during the request on burst basis is expressed as follows:

$$r_{AF} = k * p \qquad (12)$$

where k is a reduction factor called smoothing factor and p is the token bucket peak value. k can vary between 0 and 1. Reducing k, less satellite channels are given to AF services and this means that it will be more time consumed to serve AF traffic but more traffic managed on satellite connections. AF requests are mapped on lower priority queue (e.g. AF can mapped on nrt-VBR in a ATM satellite system or on VBDC in a DVB-RCS platform) and presents a timeout n_{VBDC}>12 (564 ms). The k value can affect the overall satellite system perfomance because it can increase the low priority AF requests in the system and increase the channel holding time of the AF calls. This can produce burst/GOP loss of the EF services. Thus it is important to choose the optimal k^* value that can increase or decrease the bandwidth reservation of the AF request if the system conditions change. In section V a table that maximises the system utilization preserving the outage satellite system probability is presented for some parameters of the system.

C. *Traffic Resource Management*

The Traffic Resource Manager (TRM) is the entity that manages transmission resources. It is equipped with different databases and with a calculator for an optimum resource management. The calculator, for each frame time (47 ms), memorizes the arrived requests. In the next frame time, it analyses the requests and checks the possibility of satisfying the requests through the consulting of the occupation resources table. If it is possible to satisfy the requests, TRM sends a resource assignment message to the terminal requesting satellite channels. The definition and the management of the priority of the requests to be satisfied play a key rule in the resource allocation of the satellite system. It is necessary to define an adequate criterion for assigning the satellite resources to the service requests, distinguishing the low delay tolerant requests from the more delay resilient ones . For this reason, the TRM have different priority queues that allow guaranteeing of the fairness for the real-time connections. In particular the CRA queue is used to store requests of application with a maximum delay bound of 47ms, the RBDC for applications with a delay greater than 47 and lower than 564ms, VBDC with timeout greater than 564ms and lower than 1,034s and FCA for application greater than 1,034

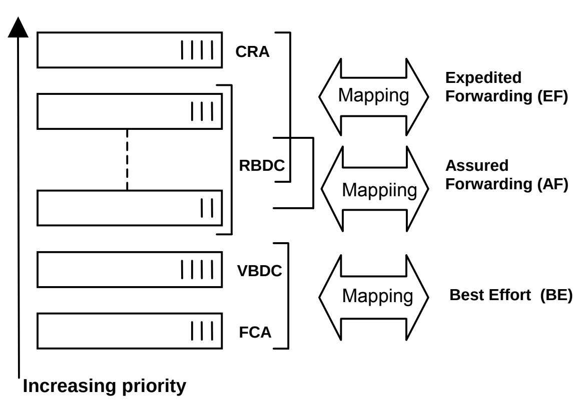

Fig.3 Requests' Queuing system in TRM module

IV. Reference Scenario

A DVB-RCS platform has been considered [12-15]. The DVB-RCS is an open standard which leaves space to different employment solutions. In fact the standard previews two different types of architecture.

One that uses an earth station, called *hub*, as control centre and monitor of all system. Such solution reduces the complexity on satellite allowing to have an economic and simple to manage system. Another one, instead, previews on OBP that excludes the presence of the hub.

The DVB-RCS standard have two communication channels, one that goes from user to satellite, called Return channel and one that goes from satellite to user called Forward channel. Each spot beam (coverage area) has one return channel and, inside each spot, there are certain particular terminals, called RCST (Return Channel Satellite Terminal) divided in four different types, indicated with letter A (144kbits/s), B (384kbits/s), C (1024kbits/s), D (2048kbits/s), and that provide different transmission capacity.

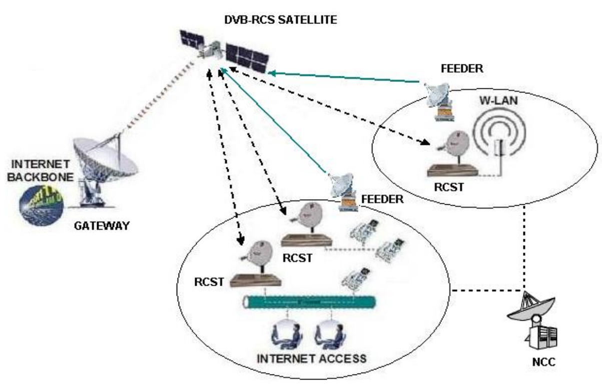

Fig.4- DVB-RCS Reference Architecture

The ETSI standard provides also a reference model of the satellite Interactive Network which consist of:

- Network Control Centre (NCC): it provides monitoring & control functions;
- Traffic Gateway (TG): it receives the RCST return signals, provides accounting functions, interactive services and/or connections to external public, proprietary and private service providers and networks;
- Feeder: it transmits the forward link signal, which is a standard satellite digital video broadcast (DVB-S) uplink, onto which are multiplexed users' data and/or the control and timing signals needed for the operation of the Satellite Interactive Network.

The satellite terminal uses a frame structure of the duration of 47 ms. The packet format consists of 188 bytes as "digital data containers" in an MPEG2 transport stream (MPEG2-TS), 4 of which (bytes) are reserved for the packet header and the rest for the payload.

Moreover the standard previews some class of services:

- Continuous Rate Assignment (CRA): CRA should be used for traffic which requires a fixed guaranteed rate.
- Rate Based Dynamic Capacity (RBDC): RBDC should be used for variable rate traffic which can tolerate some delay.
- Volume Based Dynamic Capacity (VBDC): VBDC should be used only for traffic that can tolerate delay jitter.
- Free Capacity Assignment (FCA) is volume capacity which shall be assigned to RCSTs from capacity which would be otherwise unused. Such capacity assignment shall be automatic and shall not involve any signalling from the RCST to the NCC. It shall be possible for the NCC to inhibit FCA for any RCST or RCSTs. FCA should not be mapped to any traffic category, since availability is highly variable.

In this work the DVB-RCS RBDC class has been considered and a mapping of EF and AF diffserv classes on RBDC class has been applied. In particular MPEG calls or low delay tolerant calls that represent EF traffic are mapped on RBDC with low value of timeout (47ms) and delay tolerant traffic that represents the AF class is mapped on the RBDC queue with higher timeout value (611ms)

V. Simulation Results

In order to evaluate the performance of the satellite system with EF and AF services in a SCORE Satellite network, simulation campaigns have been deployed. Only semi-permanent satellite connections have been evaluated.

The parameters considered for performance evaluation are:

- Total Satellite Utilization: r/R where r is the average token bucket data rate and R represents the bandwidth requested by the satellite receiver for EF services requesting a fixed delay bound (DB).

- Total Accepted Calls: it is the number of EF and AF services accepted by the satellite respecting the system outage probability

- GOP Loss: ratio between the number of lost GOPs and number of total GOP transmitted by RCST terminal during a call.

In Table-III are summarized the most important considered and fixed parameters in the simulation scenario.

TABLE III
SIMULATION PARAMETERS

General Source parameters	Value
Traffic Sources	Real time variable bit rate
Number of Sources	256
Load Factor (ρ)	0.1, 0.2, 0.3, 0.4, 0.5, 0.6, 0.7, 0.8, 0.9
Delay Bound (DB) requested (ms)	600
On/Off Source parameters	**Value**
Burstiness β	2, 3, 4, 5
Bucket size b (kbit)	1024
Peak rate p (kbps)	$\beta * r$
Rate r (kbps)	128
Mpeg Source parameters	**Value**
μ (kbps)	144
σ (kbps)	16
Receiver parameters	**Value**
Number of satellite receivers	256
Delay Bound (DB) requested (ms)	500, 600, 700, 800, 900, 1000
Smoothing Factor k	1, 0.9, 0.8, 0.7, 0.6
Satellite parameters	**Value**
Round Trip Time (ms)	540
Medium Access Protocol	MF-TDMA
Timeout n_{EF} for EF requests in OBP	1 (47ms)
Timeout n_{AF} for AF requests in OBP	13 (611ms)
Target burst loss probability (γ)	0.01
Return Channel's slots	1400
Forward Channel's slots	4000
Atomic satellite channel (slot)	32 Kbit/s
Return and Forward Channel's trama	47 ms
RCST type	D
Max number of sources for RCST	16

The simulation campaign has been conducted considering increasing traffic load of AF services. The traffic load ρ is defined as follows:

$$\rho = \frac{n_{AF}}{n_{AF} + n_{EF}} \tag{12}$$

where n_{AF} and n_{EF} represent respectively the number of AF and EF arrived calls. The traffic load is evaluated for each trama interval (47ms) in order to consider the working condition and to select the optimal k value (k^*). A lot of simulations have been assessed to find the k^* value that maximise the total satellite utilisation preserving the GOP or burst loss ratio specified for EF and AF services. In particular, in table IV some optimal values for different traffic condition and burstiness value of AF services are reported.

TABLE IV
K^* VALUES UNDER MANY TRAFFI CONDITIONS

ß \ ρ	10%	20%	30%	40%	50%	60%	70%	80%	90%
2	1	1	0.9	0.9	0.9	0.9	0.8	0.7	0.6
3	0.8	0.8	0.7	0.7	0.6	0.6	0.6	0.6	0.6
4	0.8	0.7	0.7	0.6	0.6	0.6	0.6	0.6	0.6
5	0.6	0.6	0.6	0.6	0.6	0.6	0.6	0.6	0.6

Through a current traffic load estimation ρ and a burstiness traffic knowledge, it is possible to select the smoothing factor to apply in the admission phase or in the bandwidth request of traffic source on burst basis. It is possible to see in table IV the decrease of smoothing factor k for increasing EF traffic load. This is due to the low number of AF calls that can be served with lower bandwidth preserving the GOP/loss ratio threshold γ (1%). The k^* values are obtained through simulation campaign where the burstiness value has been changed between 2 and 5 and the traffic load in the range of [10-90]%. If other burstiness values are considered (>5) it is possible to find further k values (<0.6) that guarantee an high system utilization respecting the threshold γ.
In fig.5, fig.6 and fig.7 the satellite utilisation, number of accepted calls and GOP loss ratio are reported. It is possible to observe as the system utilisation increase for high AF traffic load. In particular, in fig.6 the simulation results for β=2 and 5 are showed.

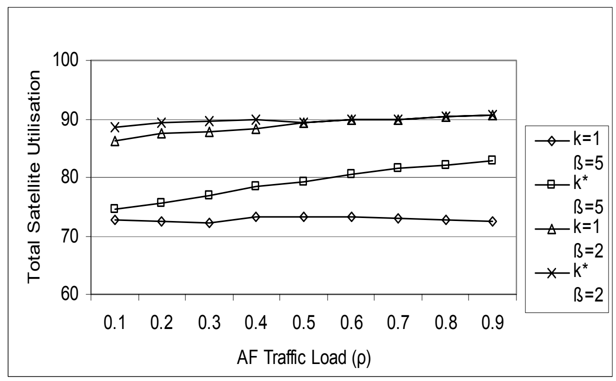

Fig.5- Total Satellite Utilisation vs increasing AF traffic load ρ.

The improvement are more evident when the burstiness is high. This is due to the high multiplexing gain that can be obtained by the AF services. For low burstiness values (β=2) few AF calls can be admitted in the system when AF traffic load increases otherwise the GOP loss ratio is not respected. When the burstiness increases (β=5) some further improving margin exists and it is possible to reduce the bandwidth request of AF services through lower k values (<6). In this paper only k values in the range of [0.6-1] have been considered. In this scenario it is possible to observe an improvement of 25% in the system utilisation for β=5, an increase of 1500 calls and a GOP loss ratio <1%. For k values lower than 0.6, the system utilization can further increase and the GOP loss ratio continue to be under the threshold (1%).

VI. CONCLUSIONS

A scalable framework for Geo-Satellite network has been proposed. A DVB-RCS architecture has been considered and end-to-end IP QoS guarantees have been provisioned over the satellite system through the introduction of SCORE network. The DPS technique, the CAC for AF and EF services and the TRM module permit to respect the QOS constraints and to maintain the state scalability in the Satellite module. Bandwidth aggregation and statistical multiplexing are applied for AF and EF services. In order to increase the system utilisation and the number of accepted calls, the smoothing factor is applied to AF services. Great improvements are obtained in terms of total satellite utilisation (> 20%) and number of admitted calls (>1200 calls). The outage probability is maintained under the pre-specified threshold (<1%).

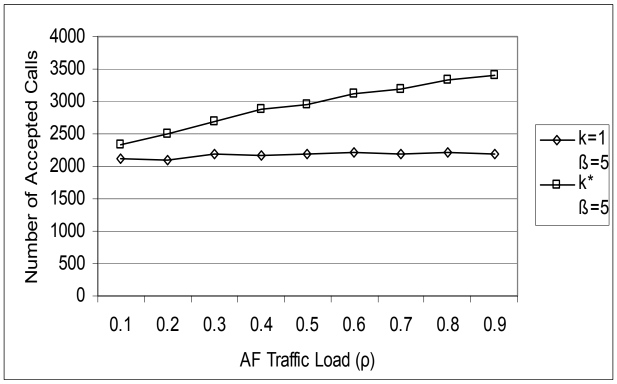

Fig.6- Number of accepted calls vs increasing AF traffic load ρ.

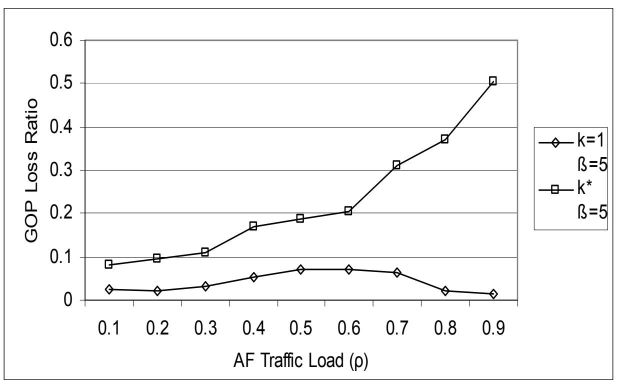

Fig.7- GOP Loss Ratio vs Increasing AF traffic load ρ.

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