Effective Admission Policy for Multimedia Traffic Connections over Satellite DVB-RCS Network

Pasquale Pace and Gianluca Aloi

Thanks to the great possibilities of providing different types of telecommunication traffic to a large geographical area, satellite networks are expected to be an essential component of the next-generation internet. As a result, issues concerning the designing and testing of efficient connection-admission-control (CAC) strategies in order to increase the quality of service (QoS) for multimedia traffic sources, are attractive and at the cutting edge of research. This paper investigates the potential strengths of a generic digital-video-broadcasting return-channel-via-satellite (DVB-RCS) system architecture, proposing a new CAC algorithm with the aim of efficiently managing real-time multimedia video sources, both with constant and high variable data rate transmission; moreover, the proposed admission strategy is compared with a well-known iterative CAC mainly designed for the managing of real-time bursty traffic sources in order to demonstrate that the new algorithm is also well suited for those traffic sources. Performance analysis shows that, both algorithms guarantee the agreed QoS to real-time bursty connections that are more sensitive to delay jitter; however, our proposed algorithm can also manage interactive real-time multimedia traffic sources in high load and mixed traffic conditions.

Keywords: Satellite, connection admission control, real-time, multimedia, QoS, MPEG, multiplexing, onboard processing.

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I. Introduction

Telecommunication satellites have proved their worth since the beginning of the space age, mainly due to their ability to set up long-haul links and cover extensive geographic areas which remain under-equipped with traditional telecommunication networks. After a pioneering phase in which civil and military institutions were largely responsible for developing and advancing space technology, satellite telecommunication has achieved a high degree of reliability and industrial and commercial maturity. As a result, satellites now have a major role in telecommunication systems.

In the last 20 years, satellites have become 1000 times more powerful and at the same time more cost effective. There does not seem to be anything in the laws of physics to prevent this trend continuing. There will be no long term need for gap fillers as satellite power will be sufficient to provide building and urban coverage. Adaptive air-interface technologies and reconfiguration will remove the current limitations of the satellite channel; the rapid advance in memory devices and their reduction in cost means that personal terminals can store and play back large content volumes. Judging by past trends, broadband satellite multimedia systems will be an integral part of the global information infrastructure as one of the major technologies providing both broadband access and broadcast services for users who are mostly exchanging real-time applications based on several data types (such as text, voice, images, and video); furthermore, the advent of Ka-band satellites has made small and low-cost user terminals feasible. This has hastened the development of multimedia satellite networks such as AmerHis (*Satellite Broadband Interactive Services for Europe and America*) which is an advanced communications system, supported and co-funded by the

European Space Agency (ESA) and the industry. This network is based on a regenerative payload onboard the *Hispasat Amazonas* satellite and could provide real-time mesh communications between any two low-cost terminal users (DVB-RCS).

The use of open-standard air interfaces is a clear trend. Satellite communications systems have mainly been based on proprietary air interfaces up to now. Over the last few years, the DVB-format has become the preferred format for digital satellite broadcasting in Europe, and also on other continents. There are several European satellite communications projects such as SATLIFE [2], SatNEx [3], HealthWare [4], and EDEN [5] based on the DVB-RCS standard [1].

Today, the internet has become the *de facto* global networking infrastructure. IP-based services have increasingly been used for delivering multimedia applications, including various voice, video, and data applications. Such applications generate traffic with different characteristics and consequently require various levels of services. Service differentiation and end-to-end quality of service (QoS) provisioning in IP networks have thus become major preoccupations. Since IP services and applications are dominating terrestrial networks, the space segment is challenged to be "QoS-aware" to seamlessly integrate with IP terrestrial networks to efficiently use resources and serve a maximum number of connections. The meaning of QoS-aware is twofold: the space segment must be able to interpret QoS parameters and settings engineered by the terrestrial segment, and the space segment must be able to actively participate in QoS operations.

It is well known that, to be cost effective, a GEO satellite system has to reach a large customer population. To fulfil this objective, further conditions should be fulfilled:

- The bandwidth utilization needs to be optimised to reduce service costs.
- Small and low-cost terminals need to be implemented.
- Network delays have to be mitigated in order to support multimedia real time services.

Originally developed to bring digital television home through satellites, DVB-S and DVB-RCS standards empower interactive satellite communications with economical standardized satellite terminals. DVB-RCS has great potential, mostly due to the many solutions aimed at increasing the efficiency of bandwidth usage. It supports both transparent bent-pipe satellites and satellites with onboard switching and processing, and different QoS classes and QoS-aware resource management mechanisms.

The onboard processing (OBP) and switching technology is the most attractive due to optimised bandwidth usage, fully-meshed network topology through one satellite hop, and reduced delay by accomplishing end-to-end terminal communication within one satellite hop. Furthermore, the OBP technology can work together with a suitable connection-admission-control (CAC) strategy in order to guarantee both a reasonable QoS level for DVB-RCS service classes and the maximum usage of the system capacity.

DVB-RCS standard is based on the use of the already existing DVB-S standard in the forward link. DVB-RCS and DVB-S are used in return (ground-to-satellite) and forward (satellite-to-ground) links, respectively. Both of them adopt the 188-byte Moving Picture Expert Group 2 Transport Stream (MPEG2-TS) packet format for IP data transmission, but DVB-RCS can also alternatively use the 53-byte ATM cell format [1].

Combining the DVB-S2 adaptive coding and modulation (ACM) scheme with multi-spot Ka-band satellites and a DVB-RCS return link, current satellite capacity can be increased by a factor of 10. The next step is to introduce mobility into the standard which will then enable it to be used for broadband mobile multimedia connections [6].

The return link in DVB-RCS is based on a multiple-frequency time-division multiple-access (MF-TDMA) scheme, where terminals allocate capacity in slots within a certain time and frequency frame. The allocation mechanism is based on the statistical multiplexing principle, expecting overbooking of the return link radio resources in order to exploit the satellite bandwidth optimally.

The three following user profiles have been identified based on commercial requirements:

- the *corporate* user represents a large group of users behind a single terminal, typically with a LAN connected to the RCST (return channel system terminal);
- the *prosumer* (the "professional consumer") is a user who requires broadband, high-quality services (typical prosumers are home offices, small media or graphic design offices, medical centers, and so on);
- the *consumer* will probably be the last profile to feel the need for, or to be able to afford, these kinds of services but the increased need for higher capacity and enhanced services make this profile the most realistic future market.

The DVB-RCS standard supports different service classes: continuous rate assignment (CRA), rate-based dynamic capacity (RBDC), volume based dynamic capacity (VBDC), and free capacity assignment (FCA). The CRA class is used for traffic which requires a fixed guaranteed rate, with minimum delay and minimum delay jitter like the CBR class of ATM; the RBDC class is used for traffic that supports a small delay jitter of a few frames and does not demand a fixed guaranteed rate; the VBDC class is used only for traffic that can tolerate delay jitter, such as the UBR class of the ATM traffic or the standard IP traffic; while the FCA class has a volume capacity which is assigned to RCSTs from capacity which would be otherwise unused. The FCA service class should not be mapped to any traffic category, since availability is highly variable. Capacity assigned in this category is intended as bonus capacity which can be used to reduce delays in any traffic which can tolerate delay jitter. VBDC and FCA connections do not have particular constraints in terms of QoS so we have designed a centralized CAC procedure implemented in the network control center (NCC) of the DVB-RCS system with the aim of preserving the QoS of the CRA and RBDC classes.

Using a centralized approach, all connections are managed by the NCC ground station; obviously the CAC procedure situated in the NCC has all the needed knowledge in order to allocate the transmit resources over all connection paths. The centralized solution is easy to implement and it is also a very logical solution according to the simulated system architecture in which only one GEO satellite is used for testing the CAC algorithms.

This paper is organized as follows: section II summarizes the reference scenario and the system architecture for supporting the DVB-RCS standard; section III illustrates how the CAC algorithm and the scheduling strategy work together in order to guarantee a good QoS for each traffic source; section IV reports simulations assumptions, traffic scenarios, and numerical results; and finally, conclusions are given in section V.

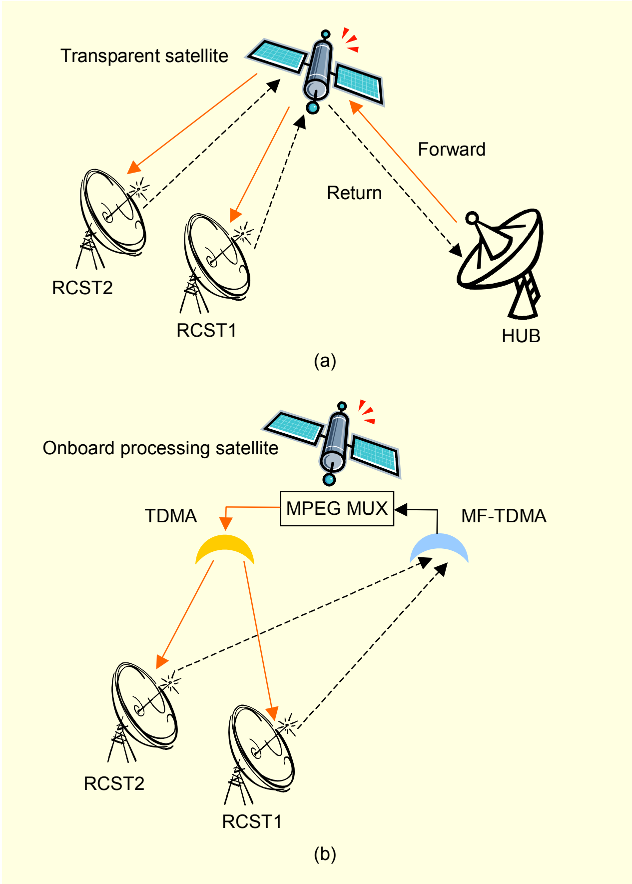

Fig. 2. Bent pipe satellite system vs. satellite system with OBP.

II. The Reference Scenario and Architectures

A generic DVB-RCS system proposed by ETSI [1] is illustrated in Fig. 1. It consists of a ground gateway station for each satellite spot beam that receives data by using the return link; a feeder station for each satellite spot beam that sends data by using the forward link; one or more satellites in the forward direction; a population of *return channel system terminals* (RCSTs) at the user's location; one or more satellites in the return direction; and a *master control station* (MCS), responsible for the overall system supervision and control, located in the NCC.

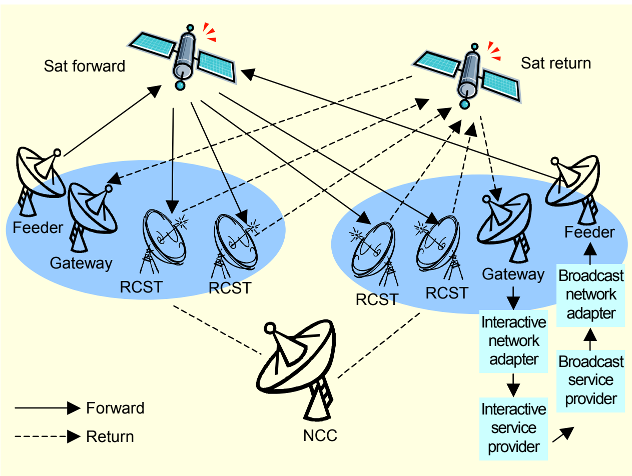

Fig. 1. A generic DVB-RCS system.

The DVB-RCS standard allows the use of regenerative satellite multimedia systems (RSMS) architectures in which the communication between the NCC, gateways, feeders and terminals transit through a satellite with OBP functions (as opposed to a conventional, bent-pipe satellite). This enables mesh connectivity to be established in the most efficient way. This is depicted in Fig. 2.

The onboard processors are classified as follows.

- *Regenerative with onboard switching*: This class of RSMS can provide full traffic re-arrangement for point-to-point connections between terminals in a mesh network. The onboard processor can also be configured to support point-to-multipoint, multi-point-to-point connections and/or concentration/multicasting/broadcasting through flexible routing/switching between input and output ports.
- *Regenerative without onboard switching*: This class of RSMS is particularly attractive when the number of uplink

and/or downlink beams is relatively small and the requirements for an onboard traffic arrangement are moderate. In such cases the requirements for concentration and/or a multicasting/multiplexing type of connectivity prevail.

- *Regenerative in conjunction with transparent repeater*: This class of RSMS systems assumes a hybrid payload including both transparent and regenerative onboard switching repeaters. Terminals are connected to the RSMS network through the transparent repeater. Point-to-point connectivity between terminals is provided by the OBP processor.

The functional requirements of the OBP processor are the following:

- to receive all traffic and control data sent by the terminals,
- to receive all traffic and control data sent by the NCC,
- to extract the traffic data to be sent on the downlink within DVB-S format and route them to the appropriate output(s) towards the receiving terminals,
- to generate/extract the control data to be sent to the NCC and route them to the appropriate output(s), and
- to format downlink streams including all of the necessary downlink signaling messages in DVB-RCS/DVB-S compatible format and route/switch them to the appropriate output(s).

In this paper, we investigate a DVB-RCS system architecture that uses a geostationary satellite with regenerative payload. We chose to use a regenerative onboard processor with switching capability. According to the DVB specification, data traffic (that is, IP for internet services) is encapsulated in MPEG-2 cells in order to allow the onboard satellite processor to switch received data segments to the appropriate spot beams based, for example, on a switching table containing the list of outgoing spot beams for each active session.

Due to the different user profiles identified in the introduction section, a distinction between different types of RCSTs based on the transmission rate is necessary [7]. The following four different typologies of RCSTs, having different bandwidth capacity and belonging to different user profiles (consumer, prosumer, and corporate), are considered in this paper and tested in the return link. They are illustrated in Fig. 3 as follows:

- RCST A: destined to a consumer market, it allows a return link with a 160 kbps rate;
- RCST B: supposed to cover the prosumer/consumer market, it has a return channel with a rate up to 384 kbps;
- RCST C: thought for to be a prosumer market, it has a return channel ranging up to 1024 kbps;
- RCST D: corporate oriented, it has a return channel with a rate that can reach 2048 kbps.

Each user terminal can manage up to 16 different connections. This way, by fixing the maximum number of active terminals for each spot, the maximum number of active connections that can be managed is also automatically fixed.

In the proposed scheme, the terminals can transmit data over the return and forward links by using an MPEG2-*traffic stream* (TS) format; the forward link capacity is fixed to 128 Mbps and an MF–TDMA scheme is chosen in the return direction.

According to the spot dimensions and the four different RSCT typologies, a possible MF-TDMA access scheme is shown in Table 1. Table 1 specifies the number of slots and carriers that each terminal can exploit supposing that approximately a quarter of the total available bandwidth is assigned to each different RCST typology. The scheme is obtained from a previous one [7] proposed by ETSI for an MF-TDMA system based on an ATM transport stream with an atomic channel dimension of 16 kbps.

A transmission frame consists of a number of time slots within some carriers. In our case, traffic slots accommodate one MPEG packet (that is, 188 bytes, 752 modulation symbols plus preamble, guard-time and an FEC redundancy part with rate 3/4) thus, assuming an atomic channel user rate of 32 kbps, the frame length needs to be 47 ms. The number and

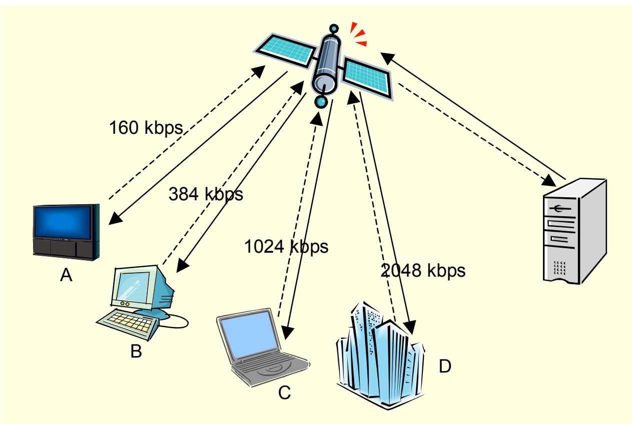

Fig. 3. Different RCST typologies.

Table 1. Features of the MF-TDMA return link.

Peak information data rate	Slots per carrier and per frame				Carriers per frame
	Traffic slots	CSC/ acquisition slots	SYNC/ synchroni-zation slots	Total slots	
160 kbps	5	1	2	6	102
384 kbps	12	2	4	14	42
1024 kbps	32	4	8	36	16
2048 kbps	64	8	16	72	8

composition of the time slots per frame is determined by the information bit rate to be supported by the frame. For example, a class D terminal can transmit over the return link using 64 traffic slots of 188 bytes each in order to make use of its own transmission bandwidth; furthermore, 8 carriers of 2048 kbps each are reserved over the return link for the traffic coming from the D terminals. The same considerations are valid for the traffic data coming from terminals A, B, and C.

The overhead slots, namely the clock and scheduler card (CSC) and synchronization (SYNC) slots, are aligned across the carriers and at the beginning of each frame. CSC slots have the same duration as traffic slots, although the actual bursts may be shorter than the slots. The duration of synchronization slots is half the duration of traffic slots. The number and composition of the time slots per frame (TRF, CSC, and SYNC) is determined by the information bit rate to be supported by the frame.

Traffic capacity is assigned on a frame basis. This means that the repetition rate of the terminal burst time plan (TBTP) is equal to the frame period, thus the TBTP shall be distributed every 47 ms. The number of traffic slots for each information bit rate allows the generation of bit rates that are multiples of 32 kbps. With a CRA or RBDC assignment of n time slots in every consecutive frame, the RCST is assigned the bit rate equal to n times and 32 kbps.

Since DVB-RCS standard defines two types of traffic bursts carrying either ATM cells or MPEG2-TS packets, we propose carrying IP packets via DVB/MPEG2-TS (188 byte) in the return and forward directions; furthermore, we propose meshing the functions of the gateway and the feeder stations into one hub station that receives the transmission requests from the return link format, and sends the requested services over the forward link [8].

III. CAC Algorithms and Requests Scheduling

Obviously, in order to guarantee a reasonable QoS level for the DVB-RCS service classes, a good CAC strategy is needed and also represents one of the main research features of the satellite research community.

A traditional admission control scheme based on Poisson process models may provide sufficient precision for 2G cellular networks [9], and the admission control schemes for IP-based networks may provide guaranteed QoS for differentiated services networks [10], [11]; however, there are stringent requirements for multimedia satellite networks. An adaptive call management system for real-time (low-interactive) VBR traffic over GEO satellite links was proposed in [12], but the authors assumed the use of on-off traffic sources. Due to the traffic characteristics of multimedia real-time streams, the last assumption is quite stringent; therefore, we want to find a good CAC strategy for managing multimedia interactive connections. Such a strategy should allow the offering of good quality of service at critical traffic sources, minimizing the signaling exchange between the onboard and on-Earth segments of the system, and reducing delays due to the processing of the call requests onboard. Towards achieving this goal, we have designed a new Gaussian CAC and we have evaluated its performance, comparing it with the reference iterative CAC proposed in [12]. Moreover, we want to demonstrate that the proposed admission strategy can also be used for managing on-off traffic connections.

1. Gaussian Connection Admission Control

As explained in the introduction section, only CRA and RDBC connections need to be preserved by the CAC, thus a simple policy peak data rate (PDR) base is utilized for CRA connections; resources are permanently assigned during the connection lifetime. The admission of RDBC connections is more complex because the resource utilization of this kind of traffic source is time-variant. The policy adopted for CRA sources may cause an inevitable waste of bandwidth if it is adopted for RDBC ones. To allow efficient management, a statistical multiplexing admission policy, harmonized with a semi-permanent approach is the adopted solution for RDBC connections [12].

The admission condition of the proposed CAC takes into account the previous consideration so a generic connection will be admitted if, and only if, the following expression is strictly verified:

$$\text{If } B_{CRA}(k)+B_{RBDC}(k)\le B_T \text{ then} \quad (1)$$

Admit(call$_i$)

else

Block(call$_i$)

The $B_{CRA}(k)$ term corresponds to the total allocated bandwidth of CRA connections, the $B_{RBDC}(k)$ term represents the allocated bandwidth of all multiplexed RBDC connections at the k-th iteration of the algorithm and B_T represents the total link bandwidth.

The bandwidth contribution of CRA connections is represented by

$$B_{CRA}(k)=B_{CRAi}+B_{CRA}(k-1), \quad (2)$$

where B_{CRAi} is the bandwidth required by the i-th call and $B_{CRA}(k\text{-}1)$ indicates the total amount of bandwidth allocated to the admitted connections during the previous iteration:

$$B_{CRA}(k-1)=\sum_{n=1}^{i-1} PDR_n, \quad (3)$$

where PDR_n is the peak data rate of the n-th call.

The admission procedure for the RBDC connections is more complex because is more difficult to evaluate the total bandwidth contribution ($B_{RDBC}(k)$) of a multiplexed stream traffic sources [13], [14]. A statistical analysis must be introduced.

A generic multimedia MPEG traffic stream can be modelled by a statistical normal distribution of "group of pictures" (GOP) with mean data rate (μ) and standard deviation (σ) as shown in previous studies [15], [16].

In [17] it was demonstrated that, if there is a large set of multiplexed MPEG streams then the autocorrelation effect will be negligible and, as a result, all traffic streams can be considered independent. To validate this assumption, we need to have at least six and eighteen MPEG streams respectively multiplexed over the return and forward link. These conditions are always respected in our scenario; consequently, the behaviour of a multiplexed set of (i–1) RBDC calls can be represented as an aggregated flow with a statistical normal distribution of mean data rate μ_{TOT} and standard deviation σ_{TOT}.

$$\mu_{TOT} = \sum_{n=1}^{i-1} \mu_n \quad \text{and} \quad \sigma_{TOT} = \sum_{n=1}^{i-1} \sigma_n \tag{4}$$

In general, it is easy to know in advance the mean GOP rate and the standard deviation of MPEG streams. For example, these parameters can be calculated during the MPEG video coding phase according to the desired video quality.

The RBDC traffic sources are statistically multiplexed so, in certain cases, the total bandwidth request can exceed the bandwidth available. The excess demand probability (EDP) is defined as the probability that this event will occur. The CAC algorithm tries to keep the EDP value below a fixed threshold (ε). Obviously, the fixed ε parameter also coincides with the maximum tolerated cell-loss ratio. Since the total contribution $B_{RDBC}(k)$ is related to parameters μ_{TOT}, σ_{TOT}, and ε, it is obtainable using the inverse function of the normal cumulative distribution F:

$$\begin{aligned} B_{RBDC}(k) &= F^{-1}(P \mid \mu_{TOT}, \sigma_{TOT}) \\ &= \{B_{RBDC}(k) : F(B_{RBDC}(k) \mid \mu_{TOT}, \sigma_{TOT}) = (1-\varepsilon)\}. \end{aligned} \tag{5}$$

The integral equation of the cumulative normal distribution

$$\begin{aligned} &F(B_{RBDC}(k) \mid \mu_{TOT}, \sigma_{TOT}) \\ &= \frac{1}{\sigma_{TOT}\sqrt{2\pi}} \int_{-\infty}^{B_{RBDC}(k)} e^{\frac{-(t-\mu_{TOT})^2}{2\sigma_{TOT}^2}} dt \end{aligned} \tag{6}$$

does not have a closed solution then the computation of the bandwidth value $B_{RBDC}(k)$ should be carried out using printout values.

It is practically impossible to compute printout values for each of the parameters μ and σ, so it is convenient to make them independent. By executing a change of variables, it is possible to evaluate a probability enclosed inside the interval between a and b values:

$$\int_a^b \frac{1}{\sqrt{2\pi}\sigma_{TOT}} e^{-\frac{(t-\mu_{TOT})^2}{2\sigma_{TOT}^2}} dt$$

$$\int_a^b f(t)\, dt = t\, d \int_{\frac{a-\mu_{TOT}}{\sigma_{TOT}}}^{\frac{b-\mu_{TOT}}{\sigma_{TOT}}} \frac{1}{\sqrt{2\pi}} e^{-\frac{Z^2}{2}} dz = \int_{Z_a}^{Z_b} \frac{1}{\sqrt{2\pi}} e^{-\frac{Z^2}{2}} dz$$

$$= P\ (Z_a \le Z \le Z_b) = P\left(\frac{a-\mu}{\sigma} \le Z \le \frac{b-\mu}{\sigma}\right). \tag{7}$$

The Z term is called the standard normal variable and the probability function Z ≈ N(0,1) is called the standard normal distribution. It is easy to observe that the standard normal distribution is a particular case of the normal distribution with null mean value and standard unitary deviation.

$$(Z) = \frac{2}{\sqrt{\pi}} \int_0^Z e^{-t^2} dt \tag{8}$$

Finally, when a new RBDC call with μ_i and σ_i parameters asks for admission, the new total multiplexed bandwidth contribution can be evaluated as

$$B_{RBDC}(k) = B_{RBDC}(k-1) + \mu_i + BEF \cdot (\sigma_i), \tag{9}$$

where

$$B_{RBDC}(k-1) = \mu_{TOT} + BEF \cdot (\sigma_{TOT}). \tag{10}$$

The term BEF (band-wide expansion factor) represents the Z value on the x axis (Fig. 4), corresponding to the area's value of the standard normal distribution equal to (1-ε). The BEF is a constant term directly obtainable by fixing the desired ε value; choosing ε equal to 1%, the correspondent value of BEF is 2.33.

Once the new value of $B_{RBDC}(k)$ is obtained, the admission condition (1) must be recomputed. The dismiss procedure of a generic i-th CRA call is

$$B_{CRA}(k) = B_{CRA}(k-1) - PDR_i, \tag{11}$$

while, for a generic i-th RBDC call, the rule is

$$B_{RBDC}(k) = B_{RBDC}(k-1) - \mu_i - BEF \cdot (\sigma_i). \tag{12}$$

The proposed CAC algorithm has a very low linear computational complexity because it is founded on a few elementary sums, subtractions, and product operations.

Moreover, it is very flexible to any changes in the service management policy because it is based on a small set of configurable parameters. Due to these considerations, the choice whether to admit or to refuse a new connection is easier and faster.

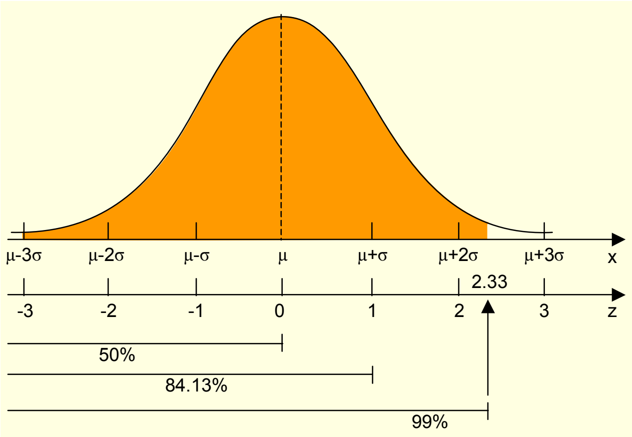

Fig. 4. Standard normal distribution.

2. Iterative Connection Admission Control

The iterative CAC is based on the reservation of bandwidth resources on the satellite links for each call, as long as data are sent. This algorithm was designed mainly for managing bursty on-off traffic sources. The decision on call acceptance is taken following evaluation of the EDP, which is the probability that the accepted calls will ask for more atomic channels than actually available in a satellite spot beam. This means that the CAC algorithm accepts a given number of calls if, at a random instant, the EDP is such that a target QoS can be guaranteed anyway. In the following, the way of computing the EDP is reported.

A two-state Markov model describes each source: in the on state the source is active and transmits at its peak data rate, in the off state it is silent and does not transmit.

Let L be the number of atomic channels in a spot beam and 32 kbps be the capacity of an atomic channel, then the total spot beam capacity is (L×32) kbps. The number of atomic channels B_i requested by the i-th call, during its on period, is given by

$$B_i = \left\lceil \frac{PBR_i}{32} \right\rceil \text{kbps},$$

where PBR_i is the peak bit rate of the i-th call (i = 1,⋯,N). Let x_i be the random variable representing the number of atomic channels requested by a call at a given instant. The following probabilities are defined:

$$\Pr(x_i = B_i) = \frac{SBR_i}{PBR_i}, \qquad \Pr\ (x_i = 0) = 1 - \frac{SBR_i}{PBR_i},$$

where $SBRi$ is the sustained bit rate of the i-th call.

Let us define the random variable

$$X = \sum_{i=1}^{N} x_i$$

that is the total demand for atomic channels at a given instant. The distribution probability for X is found by means of a generating function, as shown in [18]. Let

$$p_i = \frac{SBR_i}{PBR_i}$$

be the source utilization factor (or activation probability),

$$\overline{p}_i = 1 - \frac{SBR_i}{PBR_i}$$

and the generating function for the i-th call is

$$G_{x_i}(z) = \left(\overline{p}_i + p_i z^{B_i}\right). \tag{13}$$

Being the x_i i.i.d. random variables, the generating function of their sum, X, is

$$fx(z) = \prod G_{x_i}(z)$$

$$\begin{aligned} fx(z) &= \prod_{i=1}^{N}\left(\overline{p}_i + p_i z^{B_i}\right) \\ &= \left(\overline{p}_1 + p_1 z^{B_1}\right)\left(\overline{p}_2 + p_2 z^{B_2}\right)\left(\overline{p}_3 + p_3 z^{B_3}\right)\cdots \\ &= \prod_{i=1}^{N} \overline{p}_i + tz^{B_1} + vz^{B_2} + \cdots + qz^{B_1+B_2} + \cdots \\ &= C_0 + C_1 z + C_2 z^2 + \cdots + C_k z^k, \end{aligned} \tag{14}$$

where constant terms t, v, and q are products of activation probabilities and

$$k = \sum_{i=1}^{N} B_i\ .$$

In each term $C_j\, z^j$, C_j represents the probability that the total request of channels is equal to j

$$\Pr(X = j) = C_j\ , \quad j = 0, \cdots, k.$$

The excess demand probability is determined by computing C_{L+1}+ C_{L+2} +⋯. The C_j coefficients are computed in an incremental fashion each time a new call is accepted into the system, according to the incremental formula

$$C'_j = C_j \overline{p}_{N+1} + C_{j-B_{N+1}} p_{N+1}\ , \tag{15}$$

where $C_k = 0$ for $k < 0$.

When the (N+1)th call is released, the C_j coefficients are re-computed according to the following decremental formulas:

for variable rate calls,

$$C_j = \left(\frac{C'_j}{\overline{p}_{N+1}}\right) - \left(\frac{C_{j-B_{N+1}} p_{N+1}}{\overline{p}_{N+1}}\right); \qquad (16)$$

and for constant rate calls,

$$C_{j-B_{N-1}} = C_j, \qquad (17)$$

starting the computation from C_0, then C_1, and so on. A set of calls can be safely multiplexed if the excess demand probability is below a given bound ε:

$$C_0 + C_1 + \cdots + C_L \geq 1 - \varepsilon. \qquad (18)$$

A check is performed on both the uplink and downlink of the processed call. More details on this algorithm and its performance can be found in [12].

The connection acceptance decision is taken in real-time by the iterative CAC, because it is based on the iterative computation of the C_j coefficients each time a connection is activated or released. Computations are performed by the MCS on Earth and they do not burden the onboard processor. Anyway, the two critical points of this CAC could be: (i) the frequency of update operations of the C_j coefficients each time a new call arrives an active call finishes, and (ii) the memory requirements in the MCS for storing the old C_j coefficients. These two weaknesses need to be take into account in comparing the CAC algorithm with the proposed Gaussian admission algorithm.

3. Scheduling and Queue Management

Accurate scheduling and queue management techniques on board the satellite become vital for the provision of variable rate profile and bandwidth-on-demand services.

For this reasons capacity requests are queued in the traffic resources manager (TRM) module which will manage the available resources over the forward link.

There is a queue for each service class in the system and the requests are sorted based on their relevant timeout values.

With a view to avoiding the possible starvation of the VBDC requests, a scheduling policy consisting of priority-based management of the first request of each queue that presents the smallest timeout value, is implemented. For this reason, if a VBDC request has a lower timeout value, then it will be served even if other traffic classes of higher priority are present in the scheduler; however, requests with the same timeout value will be managed according to the class priority value. In this way, the scheduler algorithm guarantees an efficient "aging" mechanism of the requests based on the timeout values and on the traffic class categories as shown in Fig. 5.

IV. Performance Evaluation Results

In this section, we present some simulation results that show how the proposed Gaussian CAC scheme guarantees a high QoS, and also provides an elevated level of connectivity and network throughput. In addition, we compare our algorithm with a previous one designed for managing only on-off traffic sources and we demonstrate that our scheme can also be adopted for those connections and can guarantee the same performance.

1. Simulation Approach

The batch means [19] method is used for interval estimation in a steady-state simulation. This method is based on one long run (versus numerous shorter ones) in which data needs to be deleted only once. The raw output data are placed in a few large batches, and the analyst works with these few batch means as if they were independent. For our simulations, we chose one long run of 8000 minutes and 8 batches according with Law [20] that demonstrated how, for a fixed total sample size, it was best to use a very small number of batches of the longest possible length in order to strongly reduce the autocorrelation of the observed process.

Statistical error associated with the final result of any statistical experiment, or, in other words, the degree of confidence in the accuracy of a given final point estimate, is commonly measured by the corresponding confidence interval (CI) at a given confidence level (CL), that is, the interval CI expected to contain an unknown value with the probability CL. In any correctly implemented simulation, the width of a CI will

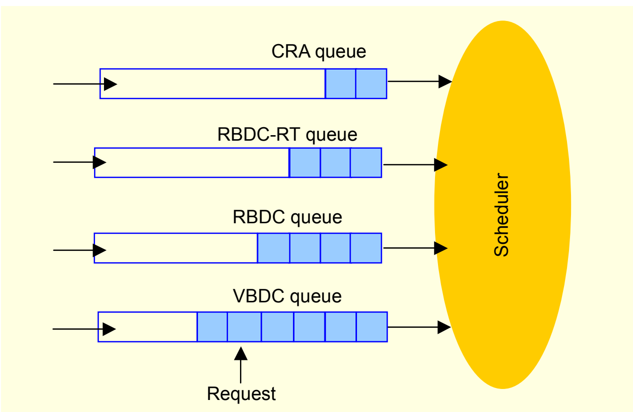

Fig. 5. Queues in the traffic resource manager.

tend to shrink with the number of collected simulation output data, that is, with the duration of the simulation. In our simulations, the confidence interval level is fixed at 0.95 and the first batch is excluded by the statistical error computation in order to reduce the effects of the system transitory.

2. First Campaign: Testing the Gaussian CAC Algorithm

We chose a set of real-time multimedia video streaming traffic sources, as indicated in Table 2 (videos A, B, C, and D). The four video traffic sources, taken from movies such as *The Gladiator* and *Mission Impossible 2*, were chosen in order to cover a wide range of possible applications such as video streaming or video conference with a variable bit rate; moreover, these video traffic sources have different mean data rate and standard deviation values in order to verify the quality of the CAC behaviour under various traffic conditions. The two constant traffic sources represent a classical high quality digital voice call (64 kbps) and a video conference application (384 kbps). The compression technique adopted is the MPEG-2 [21] with a high interactivity level and different mean GOP data rate (μ) values.

Each frame in a video sequence is coded in one of three modes as follows:

- *Intra (I) Frame*: Image data is transmitted using variable-length codes representing the coefficients derived by breaking the picture into 8×8 blocks, applying the discrete cosine transform (DCT) to these blocks, and quantizing the DCT coefficients to a user-defined level.
- *Predictive (P) Frame*: The motion compensated prediction from the previous I or P frame is used, with the residual difference data being coded in much the same way as for an I frame.
- *Bidirectional (B) Frame*: The motion-compensated prediction is achieved by one forward prediction from the previous I or P frame, backward prediction from the next (later) I or P frame or an interpolation between these two I or P frames.

Table 2. Traffic sources parameters - first campaign.

Traffic sources	Traffic profile			
	Class	PDR	μ	σ
Constant 1	CRA	64 kbps	-	-
Constant 2	CRA	384 kbps	-	-
Video A	RBDC	128 kbps	96 kbps	32 kbps
Video B	RBDC	480 kbps	320 kbps	64 kbps
Video C	RBDC	576 kbps	544 kbps	32 kbps
Video D	RBDC	1152 kbps	928 kbps	128 kbps

The prediction mode used can change for different parts of the picture.

MPEG-2 provides supreme picture quality at 720 × 480 resolution and 29.97 fps frame rate for NTSC standard, which is 720 × 576 pixel resolution and 25fps for phase alternating line (PAL) with a maximum bit rate of 15 Mbps. According to the RCST bandwidths, we have chosen to use the maximum pixel resolution of 352×288 pixels for traffic sources having a mean data rate equal to 900 kbps.

This resolution is particularly recommended for MPEG-2 compression on DVB networks carrying standard definition television (SDTV) applications [21]. For the traffic sources with lower mean data rates (80 kbps, 300 kbps, and 500 kbps) we have chosen a pixel resolution that is proportionally reduced (176×144, 240×192, and 264×216).

More detailed information on the MPEG-2 video compression standard can be found in [21].

We used the Tsunami MPEG Encoder (TMPGEnc) [22] during the encoding phase with a frame rate of 25fps and with a GOP structure composed of 15 frames in the following order IBBPBBPBBPBBPBB according to the PAL standard. It is notable that the histogram representation of the MPEG stream in terms of GOP amount, has a normal distribution trend as shown in Fig. 6.

The maximum cell loss probability tolerated by real-time connections is fixed at a value equal to 10^{-2} in the whole paper according to [23], [24]. This value is much higher than the value considered in video error resilience work [24], but it is potentially very realistic and important in wireless applications.

System performance is evaluated according to the traffic composition defined in Table 3. The traffic composition in terms of traffic source percentage has been chosen in order to test the system behaviour in a very heterogeneous and exhaustive way. For this reason, since we have no realistic traffic composition for a generic DVB-RCS satellite system, we decided to split the whole traffic load between the distinct traffic classes focusing our attention on the video traffic sources that are more difficult to manage during the admission phase.

Others key parameters used for all simulation campaigns are summarized in Table 4. We simulated a system with only two spots beams and a population of 300 satellite terminals for each spot. RCSTs are equally distributed between the four distinct typologies (A-B-C-D), and each traffic source belonging to a certain terminal has an average mean call holding time exponentially distributed over 10 minutes. In order to accommodate a whole MPEG-2 packet in each slot, the needed atomic channel user rate is fixed at 32 kbps. The BEF is a constant term directly obtainable by fixing the desired cell-loss ratio for the RBDC connections as explained in section III.

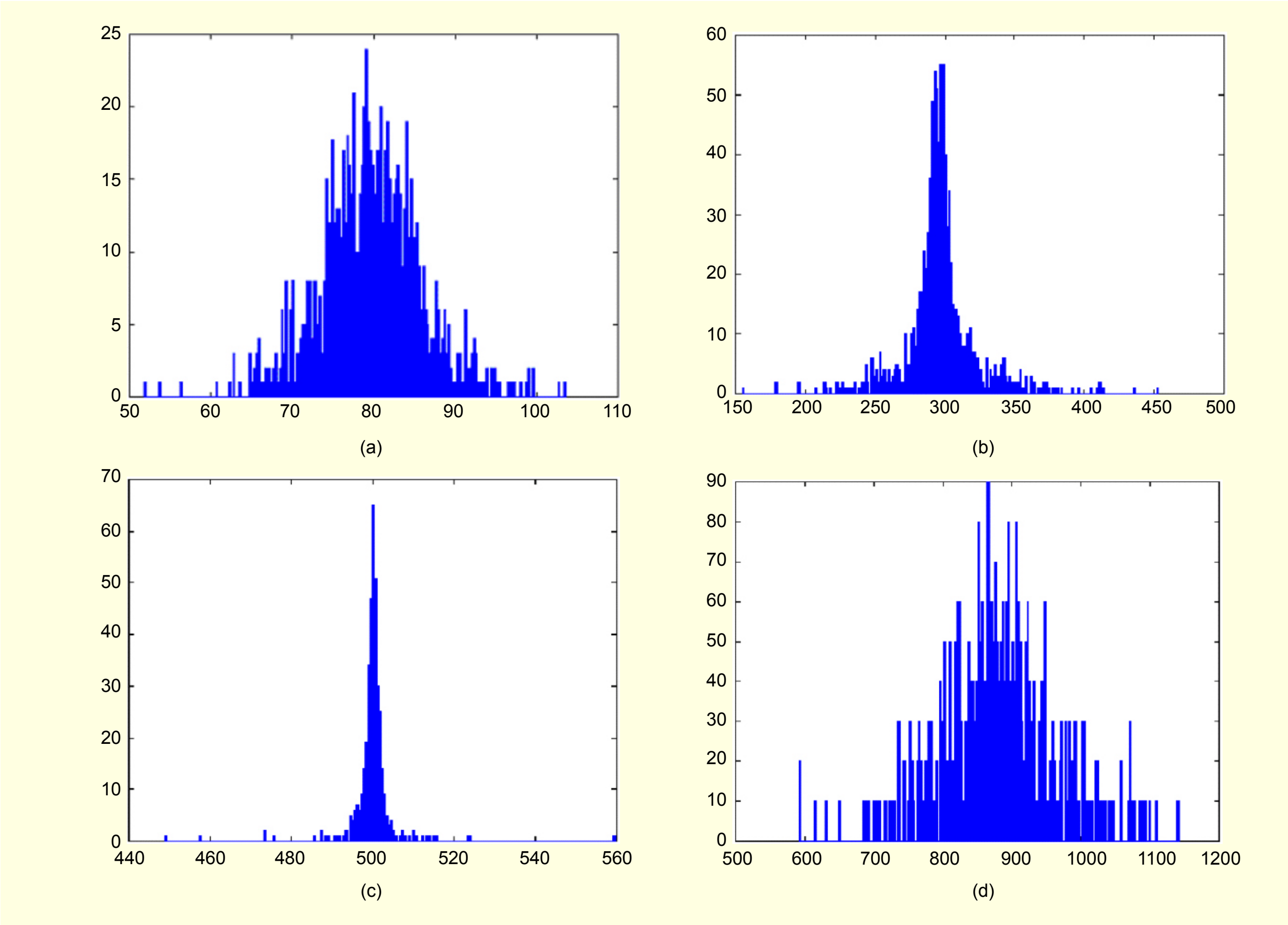

Fig. 6. Histogram of GOP-rate distribution: (a) 80 kbps, (b) 300 kbps, (c) 500 kbps, and (d) 900 kbps.

The maximum peak-to-peak cell delay variation value has been chosen equal to the lowest value (1 frame) in order to make a worst-case analysis. Finally, the entire simulation campaign has been conducted under very high traffic-load conditions (that is, a normalized system load value greater than 1) in order to stress the system verifying the robustness and quality of the admission control module.

The obtained results show the good behaviour of the proposed CAC algorithm in terms of multiplexed and accepted traffic sources, particularly under heavy traffic-load conditions. It is notable (Fig. 7) that there is minimal bandwidth waste; the throughput of the system is always greater than 85% in all the mixed traffic scenarios (Fig. 8).

The statistical multiplexing effect is more evident when the return link is loaded by sources having a small mean peak data rate such as in the scenario S1. This is due to the lower number of slots needed for transmission. Clearly, in low load conditions (0.5), the CLR value is equal to zero because there are always enough resources to satisfy the traffic source requests. The same considerations are still valid in all of the simulated traffic scenarios from S1 to S8 even if, due to the lack of space, only the results related to the scenario S1 have been shown.

Table 3. Heterogeneous traffic scenarios – first campaign.

Scenarios	Traffic sources					
	Video A	Video B	Video C	Video D	CRA 64 kbps	CRA 384 kbps
S1	50%	50%	-	-	-	-
S2	50%	-	50%	-	-	-
S3	50%	-	-	50%	-	-
S4	25%	25%	25%	25%	-	-
S5	-	50%	50%	-	-	-
S6	-	50%	-	50%	-	-
S7	12.5%	12.5%	12.5%	12.5%	25%	25%
S8	-	-	50%	50%	-	-

The quality of service offered to the video traffic sources in terms of cell loss ratio is shown in Fig. 8 where it is clear that the fixed threshold value equal to 0.01 (ε = 1%) is always guaranteed to the traffic sources.

Table 4. Commons simulation parameters.

Parameter	Value
RCST typology	A-B-C-D
Number of RCST	300 per spot
Mean connections duration	10 minutes (exponentially distributed)
Return channel	1000 slots (32 Mbps)
Forward channel	4000 slots (128 Mbps)
Slot dimension (rate)	32 kbps
Cell loss ratio (RBDC sources)	1%
Bandwide expansion factor	2.33
Frame duration over return and forward channel	47 ms
Delay propagation	135 ms
Max peak-to-peak CDV (real-time sources)	47 ms (1 frame)
Normalized system load	0.5 - 1 - 1.25 - 2 - 4

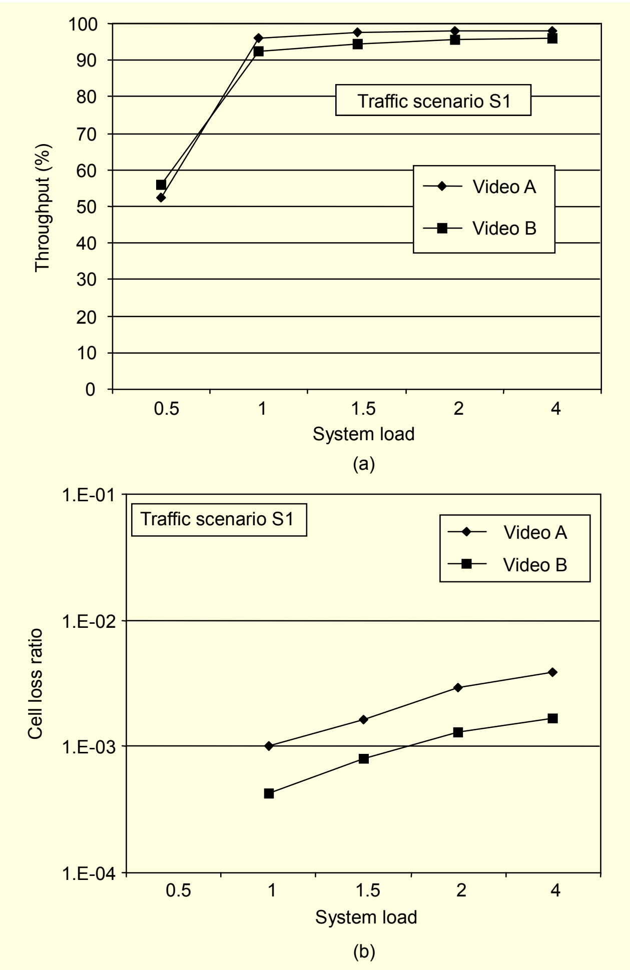

Fig. 7. (a) Return link throughput and (b) cell loss ratio for video traffic sources.

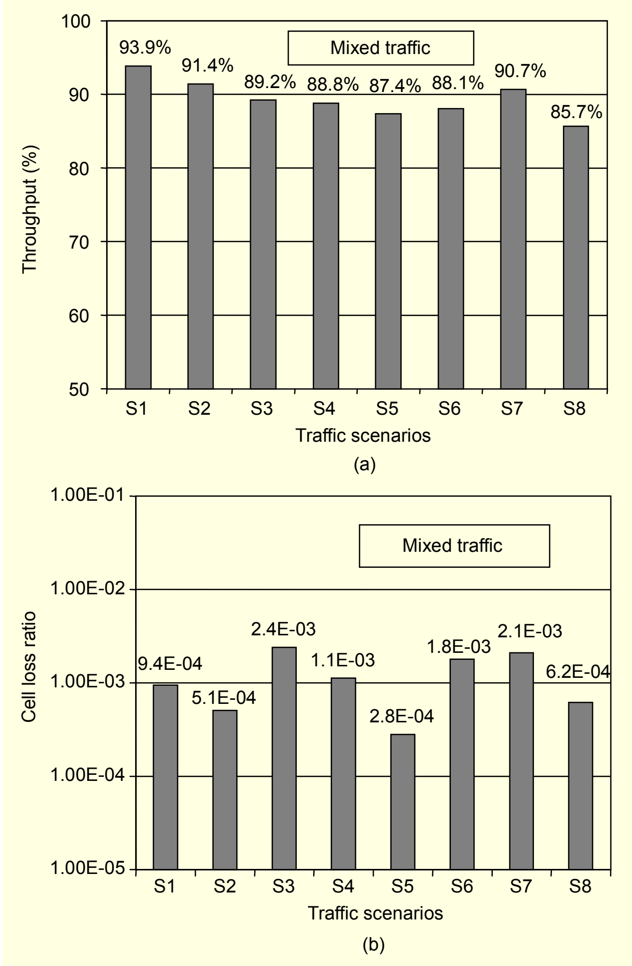

Fig. 8. (a) Return link throughput and (b) cell loss ratio in mixed condition.

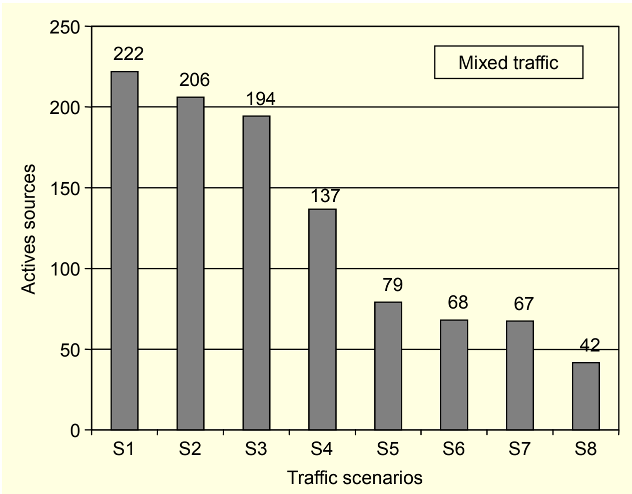

Fig. 9. Heterogeneous traffic sources multiplexing.

The average number of admitted sources that are simultaneously active over the return link is illustrated in Fig. 9. Obviously the augment of the percentage of the sources with higher PDR causes a reduction in the number of handled calls. The high number of active sources is compatible with the supposition of independent traffic streams made in section III.1.

In conclusion, all of the presented results demonstrate the quality and robustness of the proposed CAC algorithm which assures the agreed quality of service in any conditions.

3. Second Campaign: Gaussian CAC vs. Iterative CAC

In this section we show the comparison between the Gaussian CAC and the iterative CAC. We want to demonstrate that the Gaussian CAC is also well suited for managing bursty traffic sources. Some considerations are needed to allow the Gaussian algorithm to admit on-off traffic sources. Considering formula (10) it is clear that the Gaussian algorithm admission condition is mainly based on two characteristic traffic parameters: the mean data rate and the standard deviation of each connection. On the other hand, the iterative algorithm is based on the PDR and burstiness values for each connection. For this reason, a matching of the different traffic parameters of both algorithms is really needed. Due to the exponential distribution of the on-off traffic sources it is feasible to use the mathematical properties of this kind of distribution for evaluating the mean data rate and the standard deviation parameters in terms of PDR and burstiness as shown in the next equations,

$$\mu_i = \frac{PDR_i}{B_i} \quad \text{and} \quad \sigma_i = \frac{1}{\sqrt{B_i}},$$

where PDR_i and B_i are, respectively, the peak data rate and the burstiness of the *i*-th on-off traffic source.

During the simulation campaign we chose to test the system under homogeneous traffic conditions using sources with different PDR, T_{on} and burstiness values; moreover, the normalized system load is always equal to 1 in order to make the system work in realistic load conditions.

Other key parameters used during simulation campaigns for both algorithms are summarized in Table 5.

The obtained results have been compared with the previous ones using the iterative CAC under the same traffic load conditions.

Figure 10 shows that a good quality of service, in terms of cell loss ratio, is always offered to the real-time traffic sources. The CLR value is always under the agreed threshold value (10^{-2}) during the admission phase, and also varies the traffic source profiles; moreover, the system behaviour using the two different CAC algorithms is very similar even under worst case conditions with high T_{on} and burstiness values.

Figure 11 shows the number of admitted sources with varying burstiness values. Obviously, when the burstiness value increases, a greater number of traffic sources can be admitted due to the statistical multiplexing; however, the differences between the two algorithms in terms of admitted calls is always very slight, so the CAC algorithm's behaviour is extremely analogous at all times.

To further validate the previous considerations concerning the similarity of the two CAC algorithms, the results regarding the throughput of the satellite system over the return link (Fig. 12) are shown.

Table 5. Traffic sources parameters - second campaign.

Parameter	Value
Traffic composition	100% on-off real-time sources
Peak data rate	64 kbps - 1024 kbps
Burstiness	2 – 4 – 8 – 16
Avg. T_{on} duration	200 – 500 – 1200 – 5000

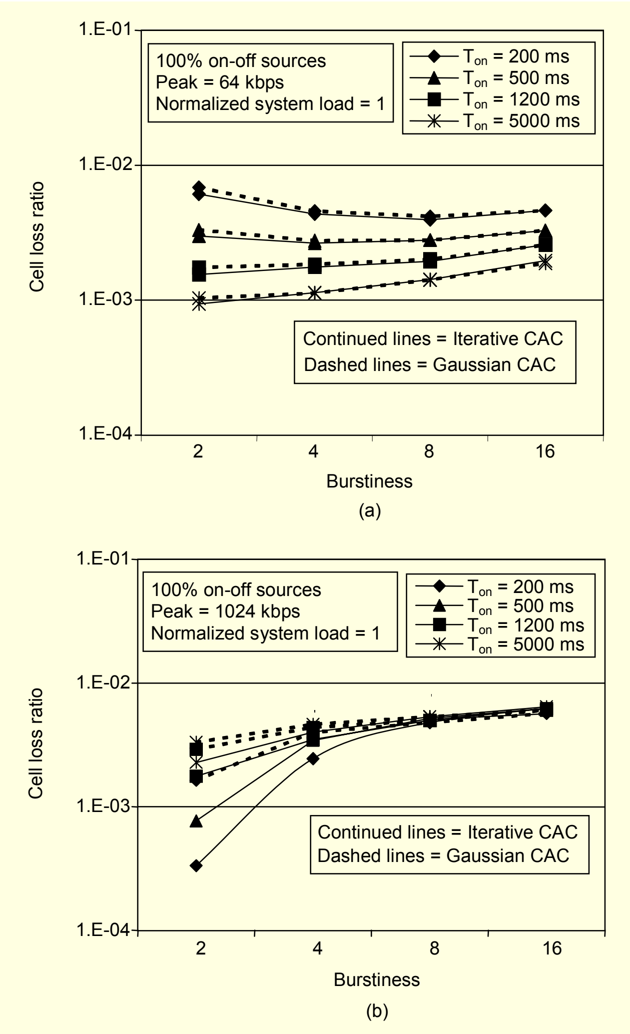

Fig. 10. CLR over the return link: (a) PDR=64 kbps and (b) PDR=1024 kbps.

It is clear that using traffic sources with a small PDR, like 64 kbps, the throughput value over the return link is always very high (about 92%); at an increased PDR, the throughput value is lower because it is more difficult to allocate a large number of slots for the same traffic source; moreover, in this particular condition, the Gaussian algorithm guarantees a slight throughput increment. In light of all of the previous considerations, it is reasonable to conclude that the proposed Gaussian CAC algorithm presents the same performance and robustness compared to the iterative CAC; moreover, it presents a very low computation complexity and it can be used to manage multimedia and bursty (on-off) traffic sources as well.

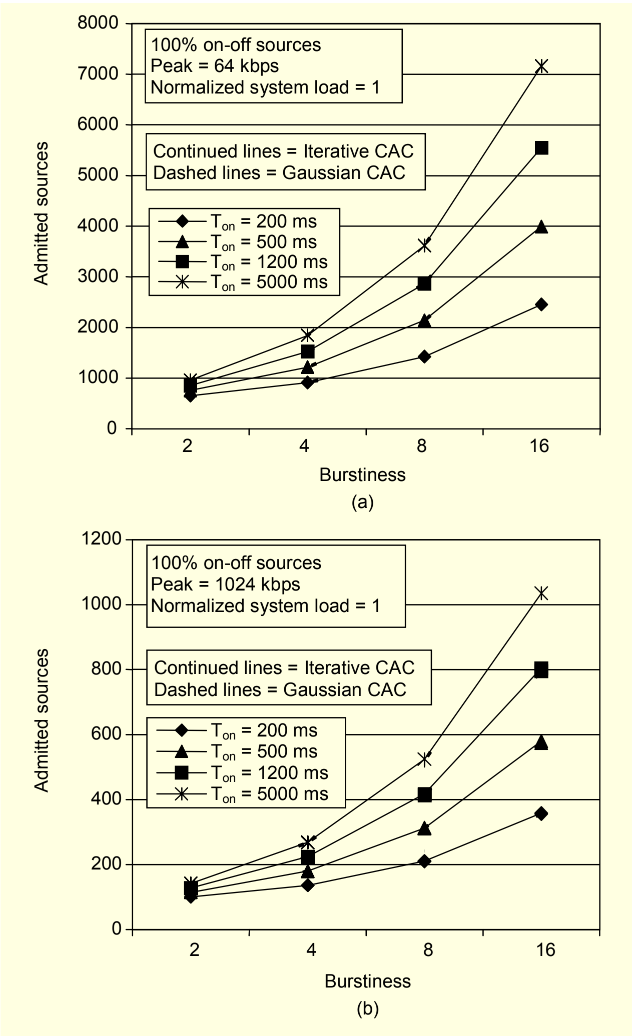

Fig. 11. Admitted sources over the return link: (a) PDR = 64 kbps and (b) PDR=1024 kbps.

V. Conclusions

In this paper a generic DVB-RCS satellite system consisting of a multi-spot beam GEO satellite with regenerative payload was explored highlighting the strengths of this architecture; moreover, a new CAC algorithm was proposed with the aim of efficiently managing real-time multimedia video sources both with constant and high variable data rate transmission. Through an extensive set of simulations, the robustness and the efficiency of the proposed admission strategy has also been verified in high load and mixed traffic conditions, demonstrating that the Gaussian algorithm can be used for managing on-off sources as well. In comparison with the iterative CAC, the Gaussian CAC presents very low computational complexity, offering a very fast and suitable admission phase.

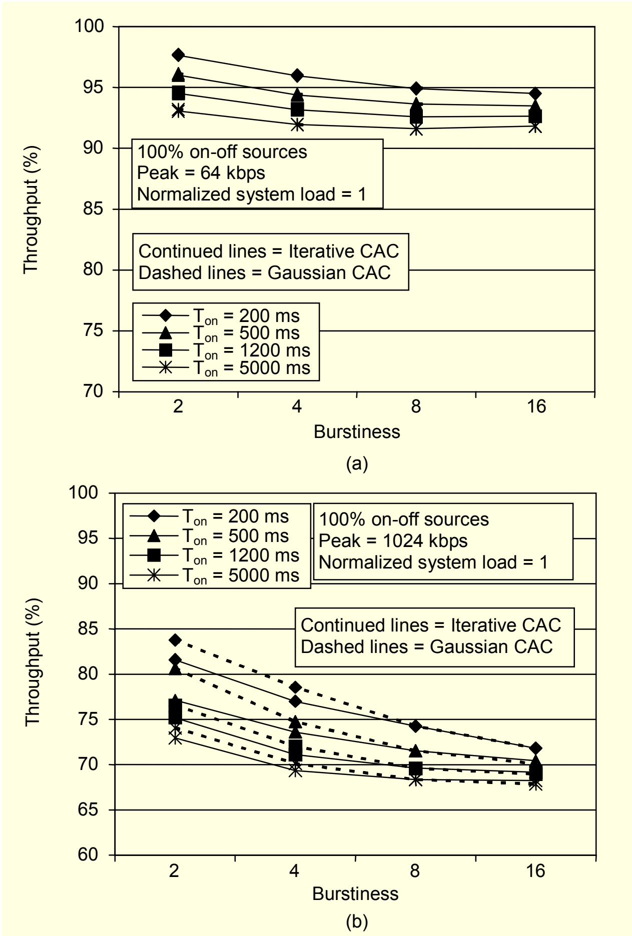

Fig. 12. System throughput over the return link: (a) PDR = 64 kbps and (b) PDR=1024 kbps.

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