Call Admission Control with Statistical Multiplexing for Aggregate MPEG Traffic in a DVB-RCS Satellite Network

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***Abstract*—This paper presents a novel Call Admission Control (CAC) algorithm based on the statistical multiplexing of VBR traffic. The proposed algorithm is called Statistical Multiplexing based on Discrete Bandwidth levels of GOP rate (SMDB) because the solution is based on the discretisation of the GOP rate in a set of bandwidth levels and on the time characteristics of discrete bandwidth levels of MPEG sources. SMDB is compared with another statistical CAC based on the Normal/Lognormal distribution of the gop rate (SMND). SMDB overcomes SMND in many situations such as high variance around the average video traffic gop rate of sources and heavy traffic load. The new CAC has been tested in the new generation satellite platform called Digital Video Broadcasting via Return Channel Satellite (DVB-RCS). A high system utilization and multiplexing gain respecting QoS constraints is obtained.**

***Index Terms*—DVB-RCS, Call Admission Control, Statistical Multiplexing, MPEG traffic sources.**

I. INTRODUCTION

The majority of new and existing service providers are working toward ubiquitous and low cost broadband access. Current broadband access is typically provided through cable infrastructure, but an alternative exists in broadband wireless, like satellite access, that has the advantages of broad and continental coverage, and quick installation [1].

The success of DTH (direct-to-home) television has proven that satellite services can compete with terrestrial alternatives in certain markets. The lack of a widely supported access standard has limited the introduction of satellite access system.

Equipment expenses have been mainly driven by low volume production, thereby eliminating the economies of scale prevalent in other access technologies (cable/DSL).

The development of the DVB Return Channel via Satellite (RCS) standard, recently accepted by ETSI, is the first attempt at introducing a wide-scale satellite access standard. The RCS standard is built around volume equipment delivery that can reach the price points required for market acceptance [2-4].

The RCS standard is tailored to the specific nuances of Internet access over a Geosynchronous Satellite.

In the quick deployment of the DVB-RCS platform that interoperates with the terrestrial Internet, a good Call Admission Control (CAC) should be performed. DVB-RCS supports many classes of service and MPEG transport stream. Thus, an efficient management and admission control for Variable Bit Rate (VBR) sources and for MPEG traffic is an important issue. There are many studies in the literature that try to take advantage of the statistical characteristics of MPEG traffic [5-8]. However, these approaches do not account for the variation of MPEG traffic in the time; thus, a high multiplexing gain for video sources with high standard deviation around the average GOP rate and heavy traffic load cannot be obtained. A novel CAC is proposed and it is based on the multiplexing of discrete bandwidth levels of MPEG sources. The video traffic is discretised in a finite number of bandwidth levels and it is characterised by a Finite State Markov Chain (FSMC) that tries to model the time characteristics of traffic sources . The proposed algorithm called Statistical Multiplexing based on Discrete Bandwidth levels of GOP rate (SMDB) is compared with a call admission control used for MPEG traffic and based on the Normal distribution of the aggregate GOP rate.

This paper is organized as follows: section II presents the DVB-RCS architecture; MPEG traffic sources and their characterization are addressed in section III; the CAC algorithms with Normal GOP approximation and with discrete GOP rate approach are formulated in section IV; finally, performance evaluation and conclusions are respectively summarised in sections V and VI.

II. DVB-RCS ARCHITECTURE

A DVB-RCS system with On Board Processing (OBP) on the Satellite payload has been considered as reference architecture in this work such as depicted in fig.1.

Most of such satellites in operation today or planned for deployment in the near future are characterised by the support of two interesting technologies: On-Board Switching (OBS) and Spot-beams. The first can be used to actively control signal routing and it copes with high network transfer rate, the switching of packets from an uplink to a different downlink spot beam needs to be done at a very high speed. The Spot-beam technology, instead, permits the division of the beam into a number of spot-beams rather covering the whole footprint of a satellite with a global beam. Dual benefits are offered by spot beam technology: smaller antennas due to the reduction of user terminals power requirements, increased capacity due to frequency beams re-use.

The DVB Return Channel System via Satellite (DVB-RCS) was specified by an ad-hoc ETSI technical group founded in 1999 [2-4]. This specifies a typical network that consists of the following elements (see fig.1).

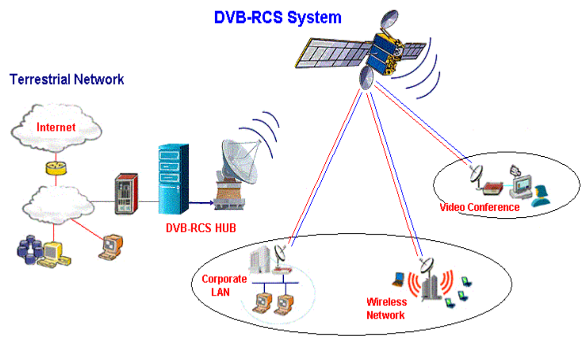

Fig.1- DVB-RCS reference architecture

- Satellite: it uses typical frequency bands in Ku (12-18 GHz) for the forward link and/or Ka (18-30 GHz) for the return link.
- User Terminals: (sometimes known as a Satellite Interactive Terminal (SIT) or Return Channel Satellite Terminal (RCST)) they support a two/way DVB satellite system. They can be of four different types on the basis of different bandwidth capacity: RCST A (144 Kbps), B (384 Kbps), C (1024 Kbps), D (2048 Kbps).
  Each RCST is assigned to a specific bit rate, depending on its capabilities and local conditions.
- Hub Station: it implements the Forward Link via a conventional DVB-S channel whereby the IP information is encapsulated into DVB-streams.

The Return Link is implemented using the DVB-RCS standard. The transmission capability uses a Multi-Frequency Time Division Multiple Access (MF-TDMA) scheme to share the capacity available for transmission by user terminals.

The mapping between source traffic type and capacity category depends on the types of service provided, on the transmission protocols used and on constraints imposed by the satellite orbit. Capacity categories supported by standard are listed below:

- Continuous Rate Assignment (CRA): it should be used for traffic that requires a fixed guaranteed rate;
- Rate Based Dynamic Capacity (RBDC): it should be used for variable rate traffic that can tolerate some delay;
- Volume Based Dynamic Capacity (VBDC): it should be used only for traffic that can tolerate delay jitter;
- Absolute Volume Based Dynamic Capacity (AVBDC): AVBDC is similar to VBDC and should be used instead of VBDC when the RCST senses that a VBDC request might be lost. This might happen when requests are sent on contention bursts or when the channel conditions (PER, Eb/N0) are degraded. Traffic supported by AVBDC is similar to the VBDC one.

The standard previews a frame structure of the duration of 47 ms and a possible use of IP packets carried via DVB/MPEG2-TS. A frame consists of a number of time slots on a certain number of carriers. The number and composition of time slots per frame is determined by the information bit rate to be supported by the frame.

The frame structure consists of 188 bytes, 4 of which are of header and 184 of payload. For details about the frame composition to see [3,4].

The sum of allocated or requested capacity for any given RCST shall not exceed the maximum transmission capability of that RCST.

III. MPEG Traffic Sources

MPEG (Moving Picture Experts Group) is a video compression standard for multimedia applications [9]. This standard previews two types of correlation: intra-frame (space type) and inter-frame (temporal type), where a frame is a single photogram of the video sequence. MPEG subdivides the video in GOPs (Group of Picture) constituted from a consecutive frames sequence. Each GOP contains three types of different frames: I frame (Intra-coded frame), P frame (Predictive frame), B frame (Bidirectional frame).

MPEG compression has a variable bit rate, and VBR traffic requires a variable bandwidth, which goes from some Kbps to various Mbps.

MPEG traffic sources are characterised by a variable GOP rate $r(t)$ that can change during time, so that $r(t) \in [r_{\min}, r_{\max}]$, where r_{min} and r_{max} are the minimum and maximum GOP rate requests. They are characterised by a Finite State Markov Chain (FSMC) that approximates the time behavior of the GOP rate. In particular, the source's GOP rate is divided in to a fixed number l of discrete bandwidth levels $[r_1, r_2, \ldots, r_l]$. If a GOP's source requests a bandwidth $r(t)$ at a particular instant time t, and $r_{\min} \le r(t) \le r_1$, the bandwith request is associated with the state r_1. In the general case, if $r_{i-1} \le r(t) \le r_i$ with i=1,2,…,l, ($r_0 = r_{\min}$ and $r_l = r_{\max}$), $r(t)$ is associetd with the state i. It is possible to calculate the sojourn time t_i in the l states and the transition probabilities p_{r_i, r_j} with i,j=1,2,…l in order to know the steady state probabilities π_i with i=1,2,…,l.

Through the approximation of the GOP rate's time dynamic, it is possible to get full advantage of statistical multiplexing. This can be useful to project a Call Admission Control, such as explained in section IV, where the probability of requesting a discrete bandwidth level i, the duration of the GOP rate and the GOP loss ratio associated with the state i can be considered in the admission phase. In fig.2 the aggregate GOP rate with and without bandwidth level discretisation is shown. It has been observed that the sojourn time in a particular discrete bandwidth is exponentially distributed for a single source; the stationary probability has been calculated such as shown in section IV. When the aggregate GOP rate is considered, it is possible to multiplex the traffic sources considering the property of the discrete bandwidth levels. In the following a clearer idea is given.

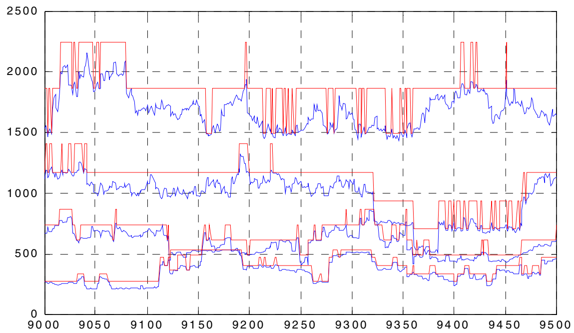

Fig.2- Aggregate GOP rate with 1, 2, 3 and 4 traffic sources with 4 discrete badwidth levels.

IV. CALL ADMISSION CONTROL FOR MPEG TRAFFIC SOURCES

Call Admission Control (CAC) represents a module of the Network Control Centre (NCC) disposed on a terrestrial station. Its task is to regulate the access to the satellite segment. It permits a flexible handling of the bandwidth and avoids the *a priori* partitioning of resources among different types of service. The resource allocation algorithm is performed by DVB satellite to accept the source channel requests. The kind of service category that we have considered is Rate Based Dynamic Capacity (RBDC).

In this work, we want to compare the novel approach based on the discretised GOP rate with previous work based on the Normal distribution of aggregate GOP rate [8].

A Statistical Multiplexing based on Normal GOP Distribution (SMND)

Statistical Multiplexing based on the Normal Distribution algorithm (SMND) takes advantage of some properties of multimedia MPEG traffic stream, but it does not account the time dependence of aggregated MPEG traffic stream. 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:

$$P_r\{B > B_T\} = \int_{x=B_T}^{\infty} f_x(x)dx \le \gamma \qquad (1)$$

with $B = \sum_{i=1}^{N} R_i$ representing the aggregate instantaneous bandwidth rate and $B_T = BEF \cdot \sum_{i=1}^{N} \mu_i$. The parameters adopted by SMND algorithm are presented in table I.

TABLE I
PARAMETERS OF THE SMND CAC ALGORITHM

	SMGD Parameters
N	MPEG traffic source
R_i	bandwidth of i-th source
B_T	capacity assigned to VBR traffic
BEF	bandwidht expansion factor
μ_i	average rate of the i-th source
$f_x(x)$	probability density function (pdf) of the aggregate rate

Previous studies [8] 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 :

$$\mu_{TOT} = \sum_{i=1}^{n} \mu_i \text{ and } \sigma_{TOT} = \sqrt{\sum_{i=1}^{n} \sigma_i^2} \qquad (2)$$

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

$$p_o = P_r\{B \ge B_T\} = \int_{B_t}^{\infty} \frac{1}{\sqrt{2\pi\sigma_{TOT}^2}} \cdot e^{\frac{-(x-\mu_{TOT})^2}{2\sigma_{TOT}^2}} \qquad (3)$$

And the $(k+1)$-th call is admitted if the eq. 4 is verified:

$$B(k+1) = B(k) + \mu_{k+1} + BEF \cdot \left(\sqrt{\sigma_{TOT}^2(k) + \sigma_{k+1}^2}\right) < B_T \qquad (4)$$

where $R(k+1)$ is the aggregate bandwidth accounting the $(k+1)$-*th* admitted call, $R(k)$ is the aggregate bandwidth at the previous step and the BEF value is obtained by the table in [10] that guarantees that the p_o does not overcome the threshold γ ($p_o \le \gamma$). For details about the SMND approach to see [8].

B Statistical Multiplexing based on Discrete Bandwidth Levels of GOP Rate (SMDB)

SMND over-estimates the bandwidth for the aggregated traffic because it does not account for the time behaviour of the MPEG source. It degrades its performance when high variance for each item of video traffic is considered, as explained in section V. In order to better use the time variation of MPEG traffic, a novel way to characterise the MPEG source is considered, as explained in section III. A CAC, called SMDB, able to better manage the time characteristics of each video source is proposed.

SMDB is based on the characterization of MPEG sources, as presented in section III. Thus, a call presents the l discrete bandwidth levels $\{r_1, r_2, ..., r_l\}$, the average bandwidth sojourn times $\{t_1, t_2, ..., t_l\}$ and the steady state probability $\{\pi_1, \pi_2, ..., \pi_l\}$ associated with the bandwidth levels of traffic sources. The algorithm, through this knowledge, can calculate the GOP loss ratio of the aggregate MPEG traffic for each level and can perform statistical multiplexing as specified below.

It is clear that, to guarantee users' QoS guaranteed service, certain admission control should be enforced to limit the number of users in the system. In this case, a greater multiplexing gain can be obtained, accounting for the time property of MPEG traffic streams associated with discrete bandwidth levels. Given an FSMC's transition probabilities matrix A that approximates the MPEG source behaviour, the steady state probability $\Pi = [\pi_1, \pi_2, ..., \pi_l]$ can be calculated by solving the following vector equation:

$$\Pi = \Pi A \tag{5}$$

if state i's average sojourn time is t_i, then, at a particular instant time the probability of the GOP rate being around the discrete bandwidth level i is:

$$p_i = \frac{\pi_i \cdot t_i}{\sum_{i=1}^{l} \pi_i \cdot t_i} \tag{6}$$

The steady state probabilities associated with the discrete bandwidth levels of a call, together with other variables introduced below, permit to project an algorithm that is able to maintain aggregate states associated to the aggregate GOP rate and calculate new aggregate states when a new call is admitted. An idea of the proposed approach is depicted in fig.3 and fig.4.

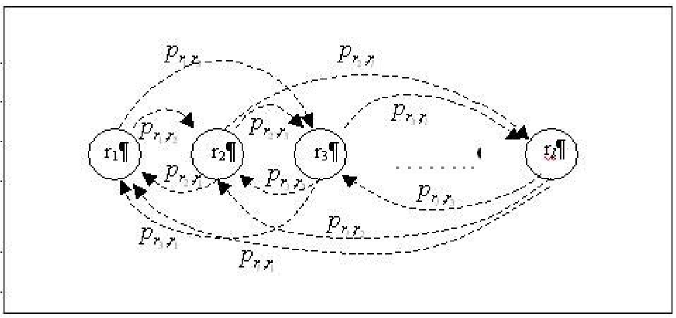

Fig.3- FSMC of discrete bandwidth levels associated with the single traffic source

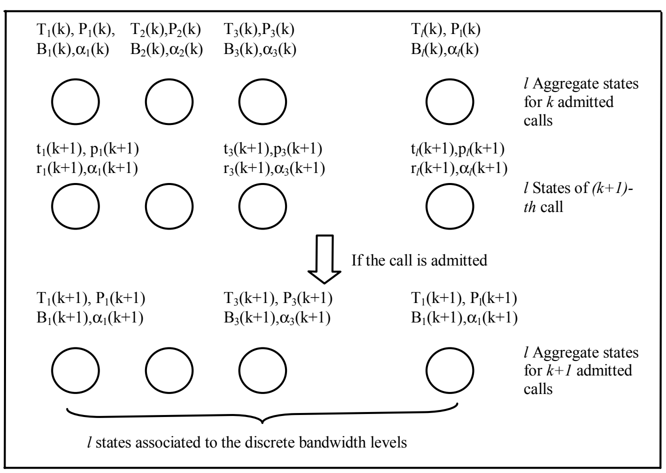

Fig.4- Aggregate states of bandwidth levels of Aggregate GOP rate.

The way to calculate bandwidth levels associated to the aggregate GOP rate is now explained.

Recall that when users' bandwidth levels are on a particular state i in the same time for an amount of time t, the GOP loss percentage associated with the state i (if it exists) is evaluated as follows. Define the bandwidth gap $\Delta_{i,j}$ associated with the combination of the aggregate state i of the k admitted calls and the new *(k+1)-th* arrived call as the amount extra requested bandwidth:

$$\Delta_{i,j}(k+1) = B_i(k) + r_j(k+1) - B_T \tag{7}$$

where $B_i(k)$ is the aggregate bandwidth reservation associated to the i-th state, $r_j(k+1)$ is the gop rate of the *(k+1)-th* call associated to the state j and B_T is the total bandwidth assigned for the MPEG sources management. This *excess demand* bandwidth is accounted if $\Delta_{i,j}>0$ otherwise a $\Delta_{i,j}=0$ is considered. To calculate the excess bandwidth request associated with the state i, all possible combinations between new call j and the previous aggregated bandwidth reservation in the state i need to be performed. The GOP loss ratio associated with the aggregated traffic in the state i and the novel call in the state j can be represented as

$$\alpha_{i,j}(k+1) = \frac{\Delta_{i,j} \cdot \min(T_i(k), t_j(k+1))}{B_i(k) \cdot T_i(k) + r_j(k+1) \cdot t_j(k+1)} \tag{8}$$

where $r_j(k+1)$ is the j-th discrete level of the $(k+1)$-th call, $t_j(k+1)$ is the average sojourn time of the $(k+1)$-th call in the state j and $T_i(k)$ is the time in which all k sources will transmit in the same time. $T_i(k)$ can be calculated as follows:

$$T_i(k) = \max\left(\min\{T_i(k-1), t_j(k)\}_{j=1}^{l}\right) \tag{9}$$

with $t_i(k)$ representing the average sojourn time of the k-th call associated to the state j. This calculus guarantees an overestimation of the $T_i(k)$ time.

To calculate the GOP loss ratio associated with the state i of the aggregated traffic when the *(k+1)-th* call is admitted the following term should be calculated:

$$\alpha_i(k+1) = \alpha_{i,1}(k+1) \cdot P_i(k) \cdot p_1(k+1) + \alpha_{i,2}(k+1) \cdot P_i(k) \cdot p_2(k+1) + ... + \alpha_{i,l}(k+1) \cdot P_i(k) \cdot p_l(k+1) + \Theta_i(k) \tag{10}$$

where $P_i(k)$ is the probability of the aggregated traffic to be in the state i; $\alpha_i(k+1)$ represents the GOP loss percentage associated with the aggregate state i when $k+1$ calls are managed in the satellite system, and $p_j(k+1)$ is the probability of the *(k+1)*-th call to be in the state j. $\Theta_i(k)$ is expressed as:

$$\Theta_i(k) = \alpha_i(k) \cdot P_i(k) \tag{11}$$

where $\Theta_i(k)$ is the contribution given by the GOP loss ratio associated with the aggregate state i without accounting the further GOP loss contribution given by the *(k+1)-th* call. The GOP loss percentage of the aggregate state i when $k+1$ calls are admitted is summarised as follows:

$$\alpha_i(k+1) = P_i(k) \cdot \left[\sum_{j=1}^{l} \alpha_{i,j}(k+1) \cdot p_j(k+1) + \alpha_i(k)\right] \tag{12}$$

However, a further observation needs to be done. Because the call can stay in a particular state i for a fraction of the overall transmission time, the outage probability p_o of the system can be calculated as follows.

The system is considered to be in outage and QoS constraints are not respected if the probability of losing GOP frames overcomes a fixed threshold γ. In particular, imaging to manage k active calls in the system, if the *(k+1)-th* call arrives, it can be admitted if:

$$p_o = \sum_{i=1}^{l} \alpha_i(k+1) \cdot P_i(k+1) \leq \gamma \tag{13}$$

At this point, it is important to calculate the probability to be in the state i of the aggregated traffic when a new call is admitted. The following equation can be applied:

$$P_i(k+1) = \frac{\sum_{j=1}^{l} P_i(k) \cdot p_j(k+1) \cdot T_{ij}(k)}{\sum_{i=1}^{l}\sum_{j=1}^{l} P_i(k) \cdot p_j(k+1) \cdot T_{ij}(k)} \tag{14}$$

where:

$$T_{ij}(k) = \min\left(T_i(k), t_j(k+1)\right) \tag{15}$$

If a source terminates its transmission, the index associated with the aggregated traffic needs to be recomputed. In particular, the aggregated probability $P_i(k)$, the GOP loss percentage $\alpha_i(k)$ associated to the state i and the minimum time $T_i(k)$ during which the sources can simultaneously transmit can be calculated as follows.

The minimum time period $T_i(k)$ when the *(k+1)-th* call leaves the system is easily calculated, re-computing the minimum average interval time in which remaining calls can simultaneously transmit such as expressed in eq.9.

The GOP loss percentage $\alpha_i(k)$ associated with the state i can be calculated as follows:

$$\alpha_i(k) = \frac{\alpha_i(k+1) - P_i(k)\sum_{j=1}^{l} \alpha_{i,j}(k+1) \cdot p_j(k+1)}{P_i(k)} \tag{16}$$

The SMDB algorithm needs to store only the aggregate GOP loss percentage $\alpha_i(k)$, the aggregate Probability $P_i(k)$ of being in the aggregate state i, the aggregate bandwidth $B_i(k)$ associated with the bandwidth level i, parameters of single sources such as steady state probability π_i,, discrete bandwidth sojourn times $t_{i,}$ and bandwidth levels b_i.

V. Performance Evaluation

Many simulation campaigns have been assessed to test the performance of SMND algorithm.

Evaluated simulation indexes are:

- GOP Loss Ratio: it is the percentage of lost GOPs of a traffic source. A GOP is considered lost if its bandwidth request on the TRM module cannot be satisfied in a frame period (47ms);
- Admitted Calls: number of accepted calls in the overall satellite system;
- Satellite Utilization: it is the ratio between real transmitted traffic over satellite channel and potentially transmittable traffic.

In table III the adopted simulation parameters in the DVB-RCS platform have been considered.

TABLE II
DVB-RCS Simulation Parameters

Parameter	Value
Round Trip Time	540 ms
Return Channel's slots	1000
Forward Channel's slots	4000
Satellite Atomic Channel	32 Kbps
Return and Forward Channel's trama	47 ms
GOP Loss Probability (RBDC sources)	0.01
RCST typology	D (2048 Kbps)
Number of RCST	2, 4, 6, 8
Max Number of Sources for RCST	16
Number of Sources	32, 64, 96, 128
Normalized System Load	0.6, 4
Bandwidth Expansion Factor	2.33

A Simulation Scenario

In order to analyse the SMND limitations and the improvements of SMDB when standard deviation increases in the traffic video the scenario represented in table IV is applied. Two different MPEG sources with increasing average GOP rate are considered and for each source an encoding with increasing standard deviation σ in the range of [10%,30%] around its average μ is considered. The frame rate used during the coding phase is 25 fps and the GOP structure is IBBPBBPBBPBBPBB with a resolution of 224x168.

For these traffic sources a light (0,6 calls/min) and heavy traffic (4 calls/min.) conditions are applied in order to verify the benefits of SMDB in many situations. Four number of sources have been evaluated on the RCS channel and a bandwidth corresponding to the number of slot necessary to admit these sources according with the SMND algorithm have been considered. For example on RCS channel 334 slot with 10% of standard deviation around an average gop rate of 300kbps have been considered to admit 32 sources.

TABLE III
Simulated Scenario

	Traffic Profile			
	Class	media	sigma	%sigma
Video 1	RBDC	144	14.4	10%
		144	28.8	20%
		144	43.2	30%
Video 2	RBDC	300	30	10%
		300	60	20%
		300	90	30%

A number of slot equal to the slots given to the SMND algorithm has been applied for the SMDB algorithm. This permitted to evaluate the benefits offered by the SMDB algorithm in comparison with SMND.

B Heavy Traffic Load

If the heavy traffic scenario with a call arrival rate of 4calls/min is considered, the improvements of SMND are also perceptible for variance of 20% and 30% and they can be around the 15-20% for the Satellite utilization such as depicted in table V. This scenario permits to verify the reduction of call block probability of SMND through the admission of a greater number of calls than SMDB CAC. If the standard deviation of traffic sources is little, the improvements of SMND in comparison with SMDB are low perceptible. This is due to the reduced multiplexing capability of the traffic sources. The GOP Loss Ratio is below the threshold value of 1% as depicted in table VI.

TABLE V
SATELLITE UTILIZATION

μ Kbps	σ Kbps	Number of Sources 32	64	96	128	CAC
300	30	86.012	87.138	87.715	87.961	SMND
300	30	88.648	89.547	90.387	91.677	SMDB
300	60	80.928	82.859	83.686	84.207	SMND
300	60	85.962	87.882	89.803	92.684	SMDB
300	90	70.661	73.900	75.433	76.532	SMND
300	90	83.534	85.992	88.449	92.134	SMDB

TABLE VII
AVERAGE NUMBER OF ACCEPTED CALLS

μ Kbps	σ Kbps	Number of Sources 32	64	96	128	CAC
300	30	175.500	349.750	525.750	702.750	SMND
300	30	182.750	365.875	549.000	823.688	SMDB
300	60	175.000	349.750	523.750	697.750	SMND
300	60	185.500	374.625	563.750	847.438	SMDB
300	90	162.250	323.750	488.250	653.250	SMND
300	90	193.750	386.875	580.000	869.688	SMDB

TABLE VI
GOP LOSS RATIO

μ Kbps	σ Kbps	Number of Sources 32	64	96	128	CAC
300	30	0.001	0.000	0.000	0.000	SMND
300	30	0.375	0.401	0.427	0.465	SMDB
300	60	0.000	0.000	0.000	0.000	SMND
300	60	0.327	0.465	0.603	0.810	SMDB
300	90	0.001	0.000	0.000	0.000	SMND
300	90	0.742	0.826	0.909	0,998	SMDB

C Light Traffic Load

In tables IV and V the number of admitted calls and the GOP loss percentage of SMND and SMDB algorithms are presented. SMND outperforms SMDB in light traffic condition because it can admit more calls preserving the prefixed GOP loss percentage of 1%.

VI. CONCLUSIONS

A novel CAC algorithm that accounts for the time characteristics of the discrete bandwidth levels associated with the aggregate GOP rate has been proposed. A high multiplexing gain is obtained through the FSMC characterization of discrete bandwidth levels associated with MPEG traffic sources. The proposed solution is compared to a well-known standard mechanism based on the Normal distribution of the GOP rate(SMND). SMDB outperforms SMND in all situations where the standard deviation of video is high and in heavy system traffic load conditions. A higher system utilization in the DVB-RCS platform is obtained, respecting the QoS constraints applied on the system.

TABLE III
AVERAGE NUMBER OF ACCEPTED CALLS

μ Kbps	σ Kbps	Number of Sources 32	64	96	128	CAC
144	14.4	116.250	234.750	346.500	463.250	SMND
144	14.4	128.000	246.500	375.750	491.500	SMDB
144	28.8	118.000	234.250	353.750	474.500	SMND
144	28.8	126.750	247.000	374.250	498.000	SMDB
144	43.2	121.750	242.250	362.750	479.500	SMND
144	43.2	142.250	278.500	417.500	562.250	SMDB
300	30	141.750	284.250	424.000	567.000	SMND
300	30	164.250	325.500	495.750	664.250	SMDB
300	60	140.250	285.500	422.250	566.750	SMND
300	60	173.750	341.500	514.500	675.750	SMDB
300	90	146.250	287.500	435.500	572.000	SMND
300	90	183.000	355.500	541.750	731.250	SMDB

TABLE IV
SATELLITE UTILIZATION

μ Kbps	σ Kbps	Number of Sources 32	64	96	128	CAC
144	14.4	74.856	78.226	78.288	78.604	SMND
144	14.4	83.596	84.489	85.578	86.803	SMDB
144	28.8	77.513	79.675	80.818	81.277	SMND
144	28.8	83.344	83.791	84.934	86.374	SMDB
144	43.2	72.768	75.734	77.403	78.403	SMND
144	43.2	83.417	86.213	87.788	89.255	SMDB
300	30	68.922	69.978	70.174	70.761	SMND
300	30	79.145	79.553	81.438	82.232	SMDB
300	60	64.775	66.823	67.414	67.775	SMND
300	60	78.909	78.417	82.240	81.176	SMDB
300	90	63.897	65.602	66.593	67.273	SMND
300	90	79.785	82.031	84.157	84.114	SMDB

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