Call Admission Control for Aggregate MPEG-2 Traffic Over Multimedia Geo-Satellite Networks

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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 discretization 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 outperforms 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 with respect to QoS constraints is obtained.**

***Index Terms*—Call admission control, DVB-RCS, MPEG traffic sources, statistical multiplexing.**

I. Introduction

MULTIMEDIA services are becoming very popular due to development of advanced applications for video and audio streaming and to the advent of broadband wireless access infrastructure that is able to provide ubiquitous communication. Satellite access represents a reality that has the advantages of broad and continental coverage, and quick installation [1]–[3].

The success of DTH (direct-to-home) television has proven that satellite services can compete with terrestrial alternatives in certain markets or can be integrated with terrestrial backbone for providing scalable end-to-end QoS services [4].

The development of the Digital Video Broadcasting-Return Channel via Satellite (DVB-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], [3], [5], [6].

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 [7]–[10]. However, these approaches do not account for the variation of MPEG traffic in the time; some MPEG traffic management techniques request an explicit feedback at receiver in order to improve the perceived end-user quality such as proposed in [11], [12]. In this work, a statistical admission control is applied in the context of Geostationary Satellites (GEO) communications where there exists a propagation delay (around 540 ms of round trip time). Different from the approach followed in [10] where a low multiplexing gain is obtained for video sources with high standard deviation around the average GOP rate and heavy traffic load, a novel CAC is proposed that is based on the multiplexing of discrete bandwidth levels of MPEG sources. The video traffic is discretized in a finite number of bandwidth levels and it is characterized 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 (SMND).

This paper is organized as follows: Section II presents some previous works related to the MPEG traffic modeling and Call Admission Control algorithms over Satellite platform; the DVB-RCS architecture is briefly introduced in Section III; MPEG traffic sources and their characterization are addressed in Section IV; two CAC algorithms under examination are formulated with Normal GOP approximation (i.e., SMND) and with GOP rate discretization (i.e., SMDB) approaches, respectively, in Section V; finally, performance evaluation and conclusions are, respectively, summarized in Sections VI and VII.

II. Related Work

In the CAC literature, a number of publications such as [11], [13] are focused on wireless video delivery through the investigation of characteristics of existing mobile networks. However, these publications are focused on local fixed/mobile wireless networks where the wireless channel is affected by errors with low propagation delays. To the best of our knowledge, the wireless video delivery over high propagation delay link such as geostationary satellite network is considered in a few papers. Presented, in the following, is the state-of-the-art about video traffic modeling (MPEG traffic) and management over GEO satellite networks.

A. MPEG Statistical Modeling

MPEG (Moving Picture Experts Group) is a working group which has the role of developing video and audio encoding standards [14].

MPEG video traffic is characterized, typically, by constant transmission rate of two groups of picture (GOP) per second and 15 frames per GOP. Since the number of bytes in a frame is dependent upon the content of the video, the actual bit rate is variable over time. However, the MPEG video supports also the constant bit rate (CBR) mode. There are three types of frames [9], [14]:

- I-Frames (intraframes)—encoded independently of all other frames;
- P-Frames (predictive frames)—encoded based on immediately previous I or P frames;
- B-Frames (bidirectionally predictive)—encoded based on previous and subsequent frames.

Many works in literature faced the problem of analyzing and describing the structure of MPEG-2 traffic streams, trying to find a way to emulate them through statistic and stochastic streams generators, preserving the proper and intrinsic nature of the original stream. In [15] the input process model is viewed as a compromise between the Long Range Dependent (LRD) and short range dependence (SRD) models. Simulation results were found to be better than those of a self-similar process when the switch buffer is relatively small. The MPEG video model presented in [16] is a Markov chain model based on the 'Group of Pictures' (GOP) level process rather than the frame level process. This has the advantage of eliminating the cyclical variation in the MPEG video pattern, but at the expense of decreasing the resolution of the time scale. Typically, a GOP has duration of a half second, which is considered long for high speed networks. Of particular interests in video traffic modeling are the frame-size distribution and the traffic correlation. The frame size distribution has been studied in many existing works. Krunz proposed a model for MPEG video in [17] where a scene related component is introduced in the modeling of I frames, whilst ignoring scene effects in P and B frames. The scene length is independent and identically distributed (i.i.d.) with common geometric distribution. I frames are characterized by a modulated process in which the local variations are modulated by an Auto-Regressive (AR) process that varies around a scene related random process with log-normal distribution over different scenes; i.e., two random processes were needed to characterize I frames. The sizes of P and B frames were modulated by two i.i.d. random processes with log-normal marginal. This model uses several random process and needs to detect scene changes, thus complicating the modeling process. In [18] and [19], the adaptive source is modeled by means of a discrete-time queuing system representing a virtual buffer, loaded by the video source when its quantizer scale parameter is changed according to the feedback law implemented in the encoder system. The whole paper is based on the Switched Batch Bernoulli Process (SBBP) that has been demonstrated to be suitable to model an MPEG video source; in fact, being a Markov modulated model, an SBBP is able to capture not only the first-order statistics but also the second-order ones which characterize the evolution of the movie scene.

In this paper we introduce a new concept of GOP-rate modeling, based on the discretization method originally proposed in [20] for the wireless channel study. After the GOP-rate trend has been analyzed for the whole duration of the stream, it has been discretized in a certain number of states. Then the associated Markov Chain parameters have been evaluated.

B. Multimedia Call Admission Control

Efficient radio resource management and CAC strategies are key components in heterogeneous wireless system supporting multiple types of applications with different QoS requirements. CAC tries to provide QoS to multiple types of applications with different requirements considering both call level and packet level performance measures. A CAC scheme aims at maintaining the delivered QoS to different calls (or users) at the target level by limiting the number of ongoing calls in the system. Call admission control (CAC) schemes have been investigated extensively in each type of network. Different approaches of CAC exist in literature, centralized, distributed, Traffic-Descriptor-Based, Measurement-Based and so on [21], [22].

In satellite networks, different types of admission control have been studied. In [23], the authors present a novel strategy for handling ATM connections of different natures, traffic profile, and QoS requirements in enhanced satellite systems. CAC represents a module of Network Operation Center (NOC) disposed on a terrestrial station. Its task is to regulate the access to satellite segment. It permits a flexible handling of the bandwidth and avoids the a priori partitioning of the resources among different types of service. The CAC algorithm has been designed also to fulfill the objectives of minimizing the signaling exchange between the on-board and on-earth segments of the system. In order to reduce delays due to the processing of the call requests on board, the relevant parameters of the processed calls are stored and elaborated within the ground segment. The method is based on the concept of reserving buffer resources to each virtual circuit as long as data are sent. The decision on the call acceptance is taken following the evaluation of the excess demand probability, i.e., the probability that the accepted calls during their activation periods request more buffer resources than those available.

In [24] and [25], the authors propose an adaptive admission control strategy, which is aimed at facing link congestion and compromised channel conditions inherent in multimedia satellite networks. They present the performance comparisons of a traditional (fixed) admission control strategy versus the new adaptive admission control strategy for a Direct Broadcast Satellite (DBS) network with Return Channel System (DBS-RCS). Fixed admission control uses the same algorithm independent of the past traffic characteristics. The Bandwidth Expansion Factor (BEF) for VBR traffic is determined such that the probability of the aggregate instantaneous rate exceeding the fraction of the capacity assigned to the admitted VBR services will not be greater than a pre-specified probability value (ε). The dynamic approach recognizes that the admission control can only approximately estimate the statistical multiplexing and attempts to use the characteristics of past traffic streams to better estimate the gain that can be achieved. Unlike the fixed admission control, the adaptive admission control adjusts the BEF such that the actual value of the aggregate instantaneous rate is close to the desired value ε that is restricted by the acceptable QoS limits.

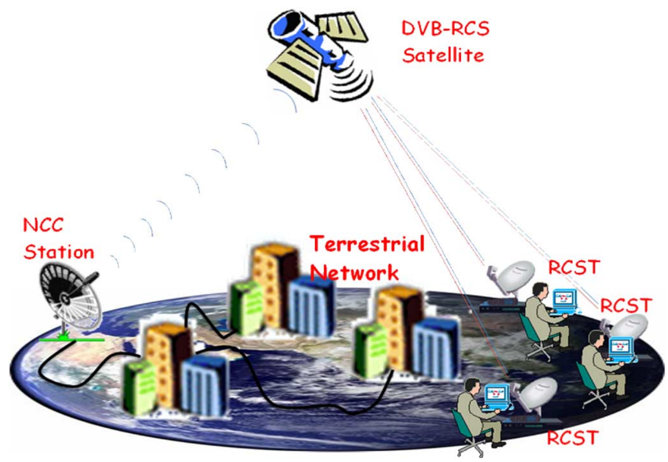

Fig. 1. DVB-RCS reference architecture.

With regard to the video broadcasting delivery and scalability properties to be offered for large scale and heterogeneous networks, the authors in [28] adopted a Video on Demand (VoD) scheme where VBR videos are mapped over CBR channels and a traffic smoothing scheme with a buffering delay control is proposed. The same authors proposed, in [29], novel broadcasting and proxy caching techniques in order to offer more scalability to the video delivery and to increase the overall performance of the system. In [30], the authors proposed a scheme to reduce the waiting time of the video application client side. The video traffic considered by authors was MPEG-2.

In [31] the author performs a comparison between Quality Oriented Adaptation Scheme (QOAS) against other adaptive schemes such a TCP Friendly Rate Control Protocol (TFRCP), Loss-Delay-based Adaptation Algorithm (LDA+) and a non adaptive (NoAd) solution when streaming multiple multimedia clips with various characteristics over broadband networks.

The purpose of the study in [32] is to propose a quality metric of video encoded with variable frame rate and quantization parameters suitable for mobile video broadcasting applications. In [33], the authors presented the results of a study that examined the user's perception of multimedia quality when impacted by varying network-level parameters (e.g., delay and jitter).

In this contribution, we consider the GOP loss ratio as a QoS parameter to be respected and VBR traffic has been considered in a DVB-RCS architecture.

III. DVB-RCS Architecture

Digital Video Broadcast with Return Channel via Satellite (DVB-RCS) is a technology that permits to have return channel and forward channel over the same medium where users can take advantages of the satellite communications. Moreover, it achieves interactive communications between the end-user and the service source.

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

DVB-RCS system was specified by an ad-hoc ETSI technical group founded in 1999 [2], [3], [5], [6]. It uses typical frequency bands in Ku (12–18 GHz) for the forward link and/or Ka (18–30 GHz) for the return link, and comprises of Return Channel Satellite Terminals (RCSTs), a Network Control Center (NCC), the satellite and a Feeder/Gateway (F/G). The core of the system is the NCC that has the task to manage the system, in particular it manages the connection, the system synchronization and informs all the RCST about the system. RCST consists of four different types on the basis of different bandwidth capacity: RCST A (144 kbps), B (384 kbps), C (1024 kbps), D (2048 kbps).

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 provided service, the used transmission protocols and constraints imposed by the satellite orbit. Capacity categories supported by the 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 that by the 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/N_0) 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 (Transport Stream). 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 (a 4-byte header and a 184-byte payload). More details about the frame composition can be found in [5].

The sum of allocated or requested capacity for any given RCST shall not exceed the maximum transmission capability of that RCST. We used the MPEG2-TS as channel to support our multimedia traffic.

IV. MPEG Traffic Modeling

MPEG compression has a variable bit rate, and VBR traffic requires a variable bandwidth, which ranges from a few kbps to several Mbps. For this reason we mapped MPEG traffic onto RBDC category. In previous work it has used a Gaussian distribution in order to characterize an MPEG traffic without considering the time dependence.

MPEG traffic sources are characterized 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 the maximum GOP rate requests. They are characterized 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 into 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 $r_{\min} \leq r(t) \leq r_1$, the

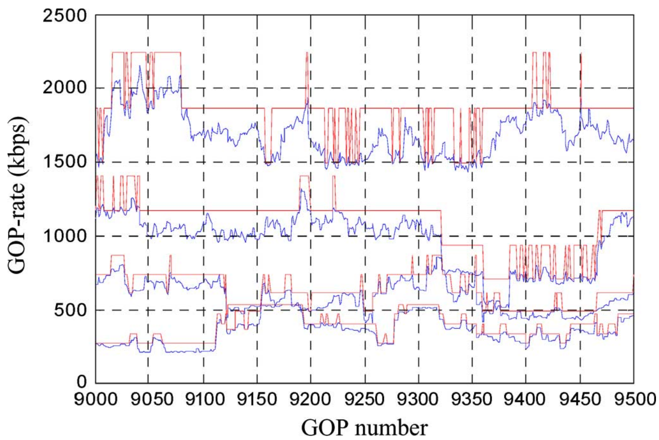

Fig. 2. Aggregate GOP rate with x $= 1, 2, 3, 4$ and 4 discrete bandwidth levels (x represents the number of aggregated traffic sources).

bandwith request is associated with the state r_1. In the general case, if $r_{i-1} \leq r(t) \leq r_i$ with $i = 1, 2, \ldots, l$, ($r_0 = r_{\min}$ and $r_l = r_{\max}$), $r(t)$ is associated 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, $j = 1, 2, \ldots l$ in order to know the steady state probabilities π_i with $i = 1, 2, \ldots, l$.

Through the approximation of the GOP rate's time dynamic, it is possible to take the full advantage of statistical multiplexing. This can be useful to design a CAC, such as explained in Section V, 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 rates with and without bandwidth level discretization are shown, respectively, for a different number of sources in the aggregation.

In Fig. 3, the Markovian approximation is shown: as it can be seen if a higher number of states is employed in the model the granularity increases, leading to a better approximation of the GOP-rate trend at expense of an increase in the computational complexity for the CAC scheme as will be shown in the next section.

It has been observed that for all the states of the considered streams, the sojourn time in a particular discrete bandwidth level is exponentially distributed for a single source as depicted in Fig. 4.

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.

V. CAC Algorithms for MPEG Traffic Sources

In the following, a clearer CAC represents a module of the NCC at a terrestrial station [26]. Its task is to regulate the access to the satellite segment. It permits a flexible handling of the bandwidth and it avoids the *a priori* partitioning of resources among different types of service. The resource allocation algorithm is performed by DVB satellite to manage the source channel requests. The kind of service category that we have considered to map MPEG traffic into a DVB-RCS class of service is RBDC.

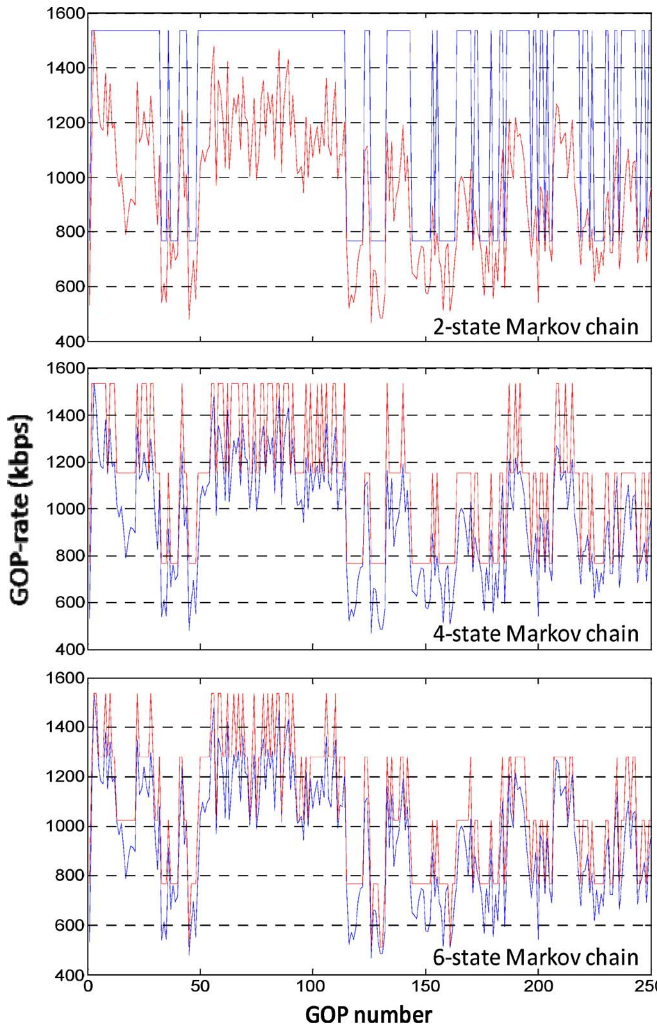

Fig. 3. GOP rate discretization of "The Gladiator" movie with different bandwidth levels (2,4,6).

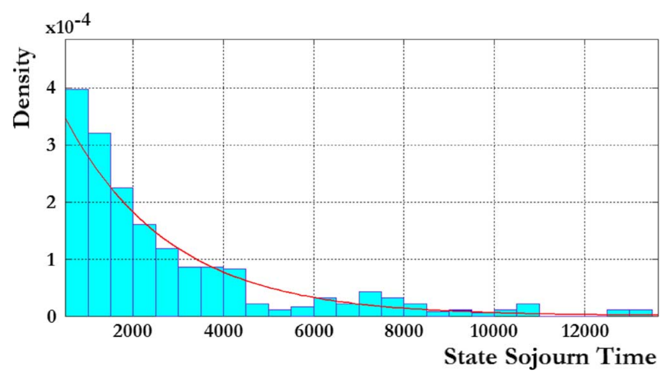

Fig. 4. Exponential approximation of the state sojourn time.

In this work, the novel approach based on the discretized GOP rate is compared with an existing method that is based on the Normal distribution of aggregate GOP rate [10].

Before discussing the work in [10] and our proposal on the novel admission of multimedia MPEG traffic, a list of parameters applied in the math formulation of the problem is defined:

— B: aggregate instantaneous bandwidth;

— N: number of MPEG traffic sources;

— B_T: capacity assigned to VBR traffic;
— r_i: bandwidth assigned to $i-th$ MPEG traffic source;
— μ_i: average rate of the $i-th$ traffic source;
— σ_i: standard deviation around the average rate μ_i of the $i-th$ traffic source.
— $f_x(x)$: probability density function of the GOP rate associated to a specific MPEG traffic.
— BEF: Bandwidth Expansion Factor. It is a multiplicative factor associated to GOP rate variance. More details about this term are given in the following.
— μ_{TOT}: aggregate average rate associated to the aggregate traffic.
— σ_{TOT}: aggregate standard deviation of the aggregate traffic. It is given by square root of the sum of single variances σ_i^2.
— $B(k)$: aggregate bandwidth when k calls are admitted.
— p_0: system outage probability. It expresses the probability to overcome the burst loss (GOP loss) ratio.
— γ: GOP loss ratio. It expresses the percentage of allowable lost GOPs.
— t_i: *average sojourn time* in the discretized $i-th$ GOP rate state.
— π_i: stationary probability associated to the discretized $i-th$ GOP rate state.
— p_i: probability to be in the discretized $i-th$ GOP rate.
— $\Delta_{i,j}(k)$: bandwidth gap (extra amount of bandwidth) associated with the combination of the aggregate state i of the k admitted calls and the new $(k+1)-th$ call.
— $\alpha_{i,j}$: GOP loss ratio of the aggregate traffic in the $i-th$ state with the novel call in the $j-th$ state.
— $t_i(k)$: average sojourn time of the $k-th$ call in the $i-th$ state.
— α_i: GOP loss percentage associated with the aggregate $i-th$ state.
— $P_i(k)$: probability of the aggregate traffic to be in the $i-th$ state.
— $T_{i,j}(k)$: simultaneous transmission of k calls in the aggregate state $i-th$ when the $(k+1)-th$ candidate call is in the $j-th$ state.

In the next two sections, SMND and SMDB CAC algorithms will be discussed.

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 take into account of the time dependence of aggregated MPEG traffic stream. In particular, a BEF is defined for VBR (e.g., MPEG) traffic 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:

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

where $B = \sum_{i=1}^{N} R_i$ represents the aggregate instantaneous bandwidth rate and $B_T = BEF \cdot \sum_{i=1}^{N} \mu_i$.

Previous studies [10] have assumed that a generic multimedia MPEG traffic stream can be modeled by a statistical Normal Distribution of 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 as a flow characterized by a GOP rate Normal Distribution with mean data rate as sum of the single mean data rates (μ_{TOT}) and standard deviation (σ_{TOT}) as the square root of the sum of single variances σ_i^2 [10], [24], [25]:

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

In general, it is simple enough 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 MPEG traffic sources are statistically multiplexed and, therefore, it is possible that total bandwidth request can exceed the available bandwidth. The excess demand probability (EDP) is defined as the probability that this event occurs. The objective of the CAC is to maintain the EDP below a fixed threshold (γ) value. Obviously, the fixed γ parameter coincides also with the maximum tolerated GLR (GOP Loss Ratio). The total contribution $B(k)$ is related to parameters μ_{TOT}, σ_{TOT} and γ thus it is available using the inverse function of the Normal cumulative distribution F:

$$\begin{aligned} B(k) &= F^{-1}(P|\mu_{TOT}, \sigma_{TOT}) \\ &= \{B(k) : F\left(B(k)|\mu_{TOT}, \sigma_{TOT}\right) = (1-\gamma)\} \end{aligned} \tag{3}$$

The integral equation of the cumulative Normal distribution (Fig. 5)

$$F\left(B(k)|\mu_{TOT}, \sigma_{TOT}\right) = \frac{1}{\sigma_{TOT}\sqrt{2\pi}} \int_{-\infty}^{B(k)} e^{\frac{-(x-\mu_{TOT})^2}{2\sigma_{TOT}^2}} dx \tag{4}$$

does not have a closed form solution and, therefore, the computation of the bandwidth value $B(k)$ need be carried out using printout values.

This estimate is an approximation of the aggregate rate, nevertheless it provides reasonably good results for moderate to large number of multiplexed streams [24], [25]. The accuracy of the approximation strongly depends on the value of predefined probability parameter γ, since this value determines the number of admitted requests. Executing a change of variable, it is possible to evaluate a probability enclosed inside the interval between a and b values:

$$\int_a^b \frac{1}{\sqrt{2\pi}\sigma_{TOT}} e^{-\frac{(t-\mu_{TOT})^2}{2\sigma_{TOT}^2}} dt \tag{5}$$

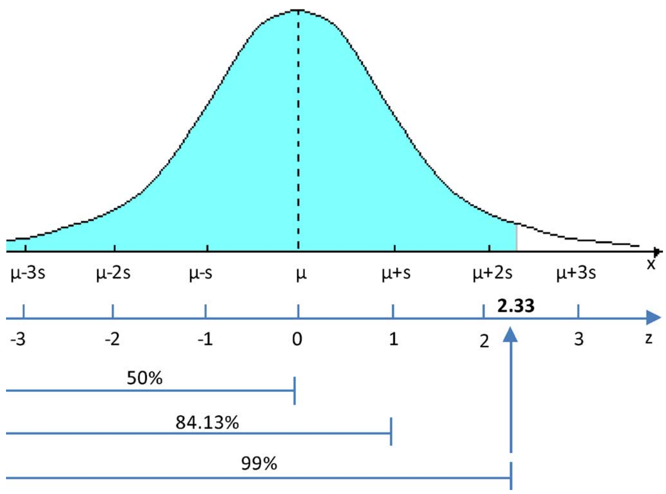

Fig. 5. Standard normal distribution.

$$\int_a^b f(t)dt = t\, d \int_{\frac{a-\mu_{TOT}}{\sigma_{TOT}}}^{\frac{b-\mu_{TOT}}{\sigma_{TOT}}} \frac{1}{\sqrt{2\pi}} e^{-\frac{Z^2}{2}} dz$$
$$= \int_{Z_a}^{Z_b} \frac{1}{\sqrt{2\pi}} e^{-\frac{Z^2}{2}} dz$$
$$= P(Z_a \leq Z \leq Z_b)$$
$$= P\left(\frac{a-\mu}{\sigma} \leq Z \leq \frac{b-\mu}{\sigma}\right) \quad (6)$$

The Z term is called *Standard Normal Variable* and the probability function $Z \approx N(0,1)$ is called *Standard Normal Distribution* (Fig. 5).

It is noted that the standard normal distribution is a special case of the normal distribution with zero mean value and unitary standard deviation.

$$Z = \frac{2}{\sqrt{\pi}} \int_0^Z e^{-t^2} dt \quad (7)$$

Finally, when a new MPEG call with μ_i and σ_i parameters wants to be accepted, the new total multiplexed bandwidth contribution is given by:

$$B(k) = B(k-1) + \mu_i + BEF \cdot (\sigma_i) \quad (8)$$

where

$$B(k-1) = \mu_{TOT} + BEF \cdot (\sigma_{TOT}) \quad (9)$$

The term BEF represents the Z value on the x axis, corresponding to the area's value of the standard normal distribution equal to $(1-\gamma)$.

The BEF is a constant term directly obtainable by fixing the desired γ value; choosing γ equal to 1%, the correspondent value of BEF is 2.33.

Once the new value of $B(k)$ is obtained, the admission condition (1) must be recomputed. The dismiss procedure of a generic i-th MPEG call is governed by the rule:

$$B(k) = B(k-1) - \mu_i + BEF \cdot (\sigma_i) \quad (10)$$

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

$$p_o = P_r\{B \geq B_T\} = \int_{B_t}^{\infty} \frac{1}{\sqrt{2\pi\sigma_{TOT}^2}} \cdot e^{\frac{-(x-\mu_{TOT})^2}{2\sigma_{TOT}^2}} dx \quad (11)$$

and the (k + 1)-th call is admitted if (12) is satisfied:

$$B(k+1) = B(k) + \mu_{k+1} + BEF \cdot \left(\sqrt{\sigma_{TOT}^2(k) + \sigma_{k+1}^2}\right) < B_T \quad (12)$$

where $B(k+1)$ is the aggregate bandwidth including the $(k+1)-th$ admitted call, $B(k)$ is the aggregate bandwidth at the previous step and the BEF value is obtained by the table in [27] that guarantees that the p_o does not exceed the threshold $\gamma(p_o \leq \gamma)$. For details about the SMND approach, see [10].

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 behavior of the MPEG source. It degrades its performance when high variance for each item of video traffic is considered, as explained in Section VI. In order to better use the time variation of MPEG traffic, a novel way to characterize the MPEG source is considered, as explained in Section IV. A CAC, called SMDB, is proposed, which is able to better manage the time characteristics of each video source.

SMDB is based on the characterization of MPEG sources, as presented in Section IV. Thus, a call presents the l discrete bandwidth levels $\{r_1, r_2, \ldots, r_l\}$, the average bandwidth sojourn times $\{t_1, t_2, \ldots, t_l\}$ and the steady state probabilities $\{\pi_1, \pi_2, \ldots, \pi_l\}$ 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 must 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 behavior, the steady state probability $\Pi = [\pi_1, \pi_2, \ldots, \pi_l]$ can be calculated by solving the following vector equation:

$$A\Pi = \Pi \quad (13)$$

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:

$$p_i = \frac{\pi_i \cdot t_i}{\sum_{i=1}^{l} \pi_i \cdot t_i} \quad (14)$$

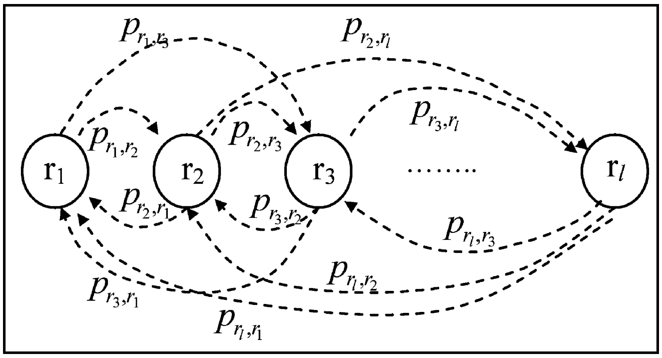

Fig. 6. FSMC of discrete bandwidth levels associated with the single traffic source.

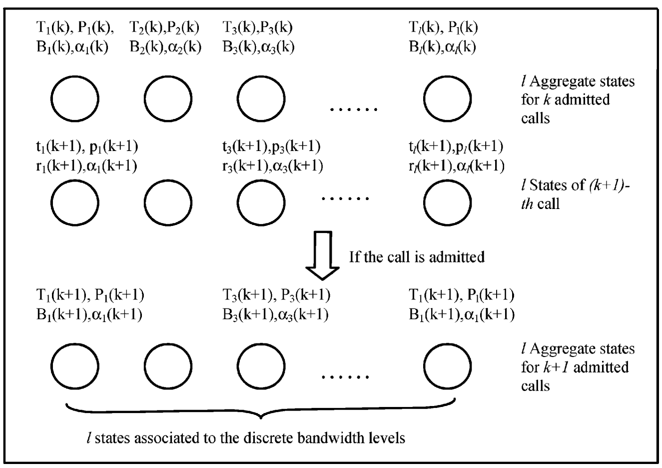

Fig. 7. Aggregate states of bandwidth levels of aggregate GOP rate.

The steady state probabilities associated with the discrete bandwidth levels of a call, together with other variables introduced below, permit design of an algorithm that is able to maintain aggregate states associated with the aggregate GOP rate and calculate new aggregate states when a new call is admitted. An idea of the proposed approach is depicted in Figs. 6 and 7. 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 at 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 $\Delta_{\mathrm{i,j}}$ associated with the combination of the aggregate state i of the k admitted calls and the new $(k+1)-th$ arrived call as the extra amount requested bandwidth:

$$\Delta_{i,j}(k+1) = B_i(k) + r_j(k+1) - B_T \qquad (15)$$

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 $\Delta_{\mathrm{i,j}} > 0$, otherwise, a $\Delta_{\mathrm{i,j}} = 0$ 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

$$\alpha_{i,j}(k+1) = \frac{\Delta_{i,j} \cdot \min\left(T_i(k), t_j(k+1)\right)}{B_i(k) \cdot T_i(k) + r_j(k+1) \cdot t_j(k+1)} \qquad (16)$$

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:

$$T_i(k) = \max\left(\min\left\{T_i(k-1), t_j(k)\right\}_{j=1}^{l}\right) \qquad (17)$$

where $t_j(k)$ represents the average sojourn time of the $k-th$ call associated with the state j. This formula 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:

$$\begin{aligned}\alpha_i(k+1) = {} & \alpha_{i,1}(k+1) \cdot P_i(k) \cdot p_1(k+1) \\ & + \alpha_{i,2}(k+1) \cdot P_i(k) \cdot p_2(k+1) + \cdots \\ & + \alpha_{i,l}(k+1) \cdot P_i(k) \cdot p_l(k+1) + \Theta_i(k) \end{aligned} \qquad (18)$$

where $\mathrm{P_i(k)}$ 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:

$$\Theta_i(k) = \alpha_i(k) \cdot P_i(k) \qquad (19)$$

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 summarized as follows:

$$\alpha_i(k+1) = P_i(k) \cdot \left[\sum_{j=1}^{l} \alpha_{i,j}(k+1) \cdot p_j(k+1) + \alpha_i(k)\right] \qquad (20)$$

where $\alpha_i(k)$ is the GOP loss percentage associated to the state i when the call $(\mathrm{k}+1)$-th in the state j-th is admitted.

Furthermore, the system outage probability p_0 needs to be estimated. Because the call can stay in a particular state i for a fraction of the overall transmission time.

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

$$p_o = \sum_{i=1}^{l} \alpha_i(k+1) \cdot P_i(k+1) \leq \gamma \qquad (21)$$

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:

$$P_i(k+1) = \frac{\sum_{j=1}^{l} P_i(k) \cdot p_j(k+1) \cdot T_{ij}(k)}{\sum_{i=1}^{l}\sum_{j=1}^{l} P_i(k) \cdot p_j(k+1) \cdot T_{ij}(k)} \quad (22)$$

where:

$$T_{ij}(k) = \min\left(T_i(k), t_j(k+1)\right) \quad (23)$$

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 be simultaneously transmitted as shown in (17).

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

$$\alpha_i(k) = \frac{\alpha_i(k+1) - P_i(k)\sum_{j=1}^{l} \alpha_{i,j}(k+1) \cdot p_j(k+1)}{P_i(k)} \quad (24)$$

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.

VI. Performance Evaluation

Many simulation experiments have been carried out to evaluate the performance of SMNB algorithm.

Table I resumes the main characteristics of the considered streams; in particular we encoded "THE GLADIATOR", "KING KONG", "THE FANTASTIC FOUR" AND "MADAGASCAR" (Fig. 8) from the original DVDs at a resolution of 224 × 168 pixels with different bitrates, as explained later; the second column of the table illustrates the transition probabilities matrices of the 3-state Markovian models associated to each stream; in the third column the means of the exponential distributed states sojourn times are illustrated, while in the last column the steady states probabilities are evaluated.

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 (47 ms);
- 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.

TABLE I
Markov Chain Parameters for 4 Movies (3 State Approximation)

	S	Transition probabilities matrix			State sojourn times (ms)	Steady State Probabilities
The Gladiator	1	0.0938	0.9062	0.0000	615.2	0.0178835
	2	0.0182	0.8522	0.1296	3691.9	0.890439
	3	0.0000	0.2803	0.7197	1756.3	0.0916779
King Kong	1	0.0000	1.0000	0.0000	600	0.0029016
	2	0.0037	0.6777	0.3186	1552	0.78421
	3	0.0000	0.1773	0.8227	2973.1	0.212889
Fantastic Four	1	0.0000	0.9523	0.0477	600	0.0405763
	2	0.0617	0.0000	0.9383	1601.9	0.640585
	3	0.0033	0.9967	0.0000	2186.8	0.318839
Madagascar	1	0.0000	1.0000	0.0000	600	0.0019103
	2	0.0025	0.6350	0.3625	1419.9	0.764123
	3	0.0000	0.1864	0.8133	2809.7	0.233967

Fig. 8. Four frames from the test video streams, with the mean and the standard deviation of the GOP-rate.

In Table II the adopted simulation parameters in the DVB-RCS platform have been considered.

A. Simulation Scenario

In order to analyze the SMND limitations and the improvements of SMDB when standard deviation increases in the traffic video the scenario represented in Table III is applied. Two different MPEG sources with different average GOP rates 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 30 fps and the GOP structure is IBBPBBPBBPBBPBB with a resolution of 224 × 168.

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 sources have been

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 frame period	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
Bandwidth Expansion Factor	2.33

TABLE III
SIMULATED SCENARIO

		Traffic Profile		
	Class	μ	σ	$\%\sigma$
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%

TABLE IV
SATELLITE UTILIZATION (%)

μ	σ	Number of Sources				CAC
kbps	kbps	32	64	96	128	
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

evaluated on the RCS channel and bandwidths corresponding to the number of slots necessary to admit these sources according with the SMND algorithm have been considered. For example, on RCS channel, 334 slots with 10% of standard deviation around an average GOP rate of 300 kbps have been considered to admit 32 sources.

A number of slots 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. Light Traffic Load

In Table IV, the satellite utilization for a traffic source of 300 kbps is presented with the number of sources varying between 32 and 128.

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

TABLE V
AVERAGE NUMBER OF ADMITTED CALLS

μ	σ	Number of Sources				CAC
kbps	kbps	32	64	96	128	
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	σ	Number of Sources				CAC
kbps	kbps	32	64	96	128	
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	1,034	SMDB

TABLE VII
SATELLITE UTILIZATION (%)

μ	σ	Number of Sources				CAC
kbps	kbps	32	64	96	128	
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

C. Heavy Traffic Load

If the heavy traffic scenario with a call arrival rate of 4 calls/min is considered, the improvements of SMDB are also perceptible for variance of 20% and 30% and they can be around the 15–20% for the satellite utilization such as shown in Table VII. This scenario permits to verify the reduction of call block probability of SMDB through the admission of a greater number of calls than SMND CAC such as illustrated in Table VIII. If the standard deviation of traffic sources is small, the improvements of SMDB in comparison with SMND are less 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 IX.

TABLE VIII
AVERAGE NUMBER OF ADMITTED 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 IX
GOP LOSS RATIO (%)

μ kbps	σ kbps	Number of Sources 32	64	96	128	CAC
144	14,4	0,000	0,000	0,000	0,000	SMND
144	14,4	0,920	0,695	0,553	0,888	SMDB
144	28,8	0,000	0,000	0,000	0,000	SMND
144	28,8	0,172	0,025	0,056	0,076	SMDB
144	43,2	0,000	0,000	0,000	0,000	SMND
144	43,2	0,149	0,163	0,136	0,632	SMDB
300	30	0,000	0,000	0,000	0,000	SMND
300	30	0,404	0,916	0,016	0,649	SMDB
300	60	0,000	0,000	0,000	0,000	SMND
300	60	0,115	1,015	0,824	0,003	SMDB
300	90	0,000	0,000	0,000	0,000	SMND
300	90	0,520	0,633	0,238	0,257	SMDB

VII. 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 Finite State Markov Chain characterization of discrete bandwidth levels associated with MPEG traffic sources. The proposed solution is compared to a well-known 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 for the DVB-RCS platform is obtained, respecting the QoS constraints applied on the system.

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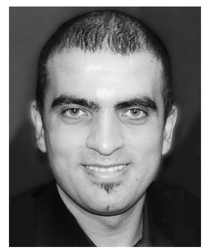

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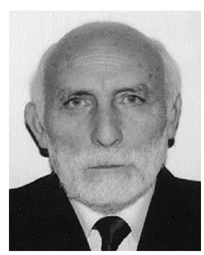

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