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Rate Control Optimization for Bandwidth Provision over Satellite Independent Service Access Points

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***Abstract*—The *Quality of Service* (QoS) provision requires the cooperation of all network layers from bottom-to-top. More specifically, in the ETSI *Broadband Satellite Multimedia* architecture, the physical layers are isolated from the rest by a *Satellite Independent-Service Access Point* (SI-SAP). The structure of SI-SAP protocol model "opens the door" to the problem of mapping the performance requests of *Satellite Independent* (SI) layers over (SD) *Satellite Dependent* technology. In this perspective, we investigate here a novel control algorithm for QoS mapping at the SI-SAP interface. Through the adoption of *Infinitesimal Perturbation Analysis*, we are able to capture the "bandwidth need" of the different flows when they are conveyed to the SD core. Simulation results validate the proposed approach.**

Keywords—Satellite QoS Architectures, QoS Interworking, Measurement-based Control, Infinitesimal Perturbation Analysis.

I. INTRODUCTION

The definition of "heterogeneous network" may assume different aspects. Network portions may use different transmission means such as cable, satellite, radio, and may implement different protocols. Actually, from a user perspective, an end-to-end QoS in satellite/wireless/terrestrial network depends on the QoS achieved at each layer of the network and it is based on functions performed at the layer interfaces. ETSI-BSM (*Broadband Satellite Multimedia*) QoS architecture (defined in technical reports [1, 2]) is a good example and it is the reference of this paper, which concerns satellite networks.

The considered protocol stack separates the layers identified as *Satellite Dependent* (SD) and *Satellite Independent* (SI). SD layers strictly depend on the physical implementations and are often covered by industrial copyright. SI layers, on the other hand, are composed of IP and upper layers. The interface between SI and SD is defined by ETSI through SI-SAPs (*Satellite Independent – Service Access Points*). QoS requirements must flow through SI-SAPs and be implemented at SD layers.

In this view, the paper envisages the QoS management issues arising at the SI-SAP interface and proposes a novel control scheme for the optimization of the bandwidth provision at the SD layer. The aim is to "map" the QoS defined at the SI layer into the SD technology, as if the satellite portions were transparent to the rest of the network.

The remainder of the paper is organized as follows. In the next section we highlight the traffic management problem arising at the SI-SAP interface. In section III, we briefly summarize the role of the fading counteraction and, in section IV, we formulate the QoS mapping problem and detail our optimized approach. Simulation results are proposed in section V to validate our control methodology. Finally, we conclude in section VI by summarizing the obtained results and emphasizing the directions for future research.

II. QOS MAPPING OVER BSM TECHNOLOGY

In BSM communications, the reference is the support of services offered by the UDP-TCP/IP suite over a satellite bearer service. The protocol architecture is defined in references [1, 2] and is reproduced in Fig. 1.

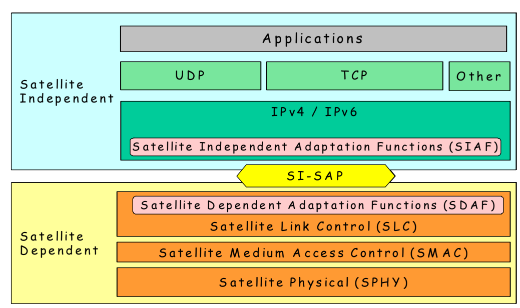

Figure 1. ETSI BSM Protocol Architecture. [1, 2].

The physical layers (i.e. satellite physical, MAC and Link Control, strictly satellite dependent) are isolated from the rest by a SI-SAP, which should offer specific QoS services to the upper layers. In such a context, the interworking between SI and SD layers reveals to be a hot topic of research. There are two main problems: **1)** the change of information unit (encapsulation) and **2)** the need to aggregate traffic.

A queue model describing the QoS mapping operation performed at the SI-SAP interface, similarly used in reference [5], is reported in Fig. 2. The first block (IP traffic classifier) assigns the arriving IP packet to a queue in dependence of the DSCP (*DiffServ Code Point*) value.

The example reports one *Expedite Forwarding* (EF) class, 4 *Assured Forwarding* (AF) classes and 1 *Best Effort* (BE) class. Below the DiffServ scheduler that picks the packets up from the queues, an AAL5 adapter block encapsulates the IP packet within the AAL5 frame and creates the ATM cells.

The problem is how much bandwidth must to be assigned to each SD queue so that the SI IP-based SLA (*Service Level Agreement*, i.e. the performance expected) is guaranteed.

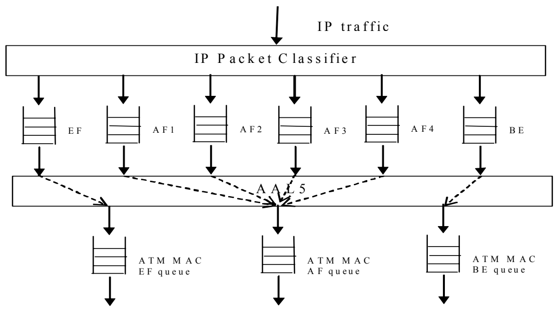

Figure 2. SI-SAP interface. SI layer (DiffServ) over SD layer (ATM). [3, 5].

III. THE ROLE OF FADING

Before detailing our operative proposal about QoS mapping, it is necessary to consider an additional problem, peculiar of satellite networks, represented by the fading degradation affecting the satellite channel and to specify the mathematical model used to describe it in this work.

Variable fading conditions over the channel can heavily affect the transmission quality in wireless networks. *Forward Error Correction* (FEC) mechanisms allow recovering erroneous packets despite the channel degradation, but, because of their overhead, cause bandwidth reduction with consequent congestion and additional packet loss in the network. Modelling this behaviour is very important.

Let $\theta^{SD}(t)$ be the service rate assigned to a traffic buffer at the SD layer at time t. The effect of fading can be modelled as a reduction of the bandwidth actually "seen" by the buffer. The reduction is represented by a stochastic process $\phi(t)$. At time t, the "real" service rate $\hat{\theta}^{SD}(t)$ (available for data transfer) is:

$$\hat{\theta}^{SD}(t) = \theta^{SD}(t)\cdot\phi(t);\quad \phi(t)\in[0,1] \tag{1}$$

The model may be associated with fading countermeasure mechanisms located at physical layer [10]. Whenever the fading causes bit errors and, as a consequence, erroneous packets, an adaptive control, monitoring the C/N (*Carrier/Noise Power*) factor, can increase the redundancy bits of the packets on the basis of this measure.

IV. THE SI-SAP QOS MAPPING PROBLEM

To summarize, the QoS management at the SI-SAP interface has to face three components: **1)** fading counteraction; **2)** change of encapsulation format; **3)** need to aggregate traffic.

In this perspective, the proposal of this work concerns the adoption of a novel control mechanism, suited for real time control of QoS mapping operations.

A. *System constraints and assumptions*

IP *Packet Loss Probability* (PLP) is chosen as performance metric. The proposed investigation can be generalized for other QoS constraints, such as delay and delay jitter.

Implementation details may be protected by industrial patents. Moreover, being the interface between SI and SD layers defined through SI-SAP primitives, SI and SD layers might also be located within remotely located tools. As a consequence, it is necessary to make use of a control mechanism that does not need information about traffic statistics and allocated resources, but only a priori reference values and on-line measures.

B. *Stochastic fluid model and optimization problem*

The chosen mathematical framework is based on a *Stochastic Fluid Model* [6] of the traffic buffers. We take Fig. 2 as a reference and we consider the presence of N SI queues and a single SD queue.

Let $\alpha_i^{SI}(t)$ be the *inflow rate* process entering the *i*-th traffic buffer at the SI layer at time *t*, $i=1,\dots,N$. After entering one single buffer (with service rate $\theta_i^{SI}(t)$) at the SI layer, each $\alpha_i^{SI}(t)$ process is conveyed to a single SD buffer (whose service rate is $\theta^{SD}(t)$) at the SD layer. We denote by ${}^{i}L_V^{SI}(\alpha_i^{SI}(t),\theta_i^{SI}(t))$ the *loss volume* of the *i*-th IP buffer according to the bandwidth allocation $\theta_i^{SI}(t)$.

Let $\alpha^{SD}(t)$ be the *inflow rate* process of the buffer at the SD layer at time *t*. The $\alpha^{SD}(t)$ process derives from the *outflow rate* processes of the SI buffers (or directly from the $\alpha_i^{SI}(t)$ processes, if no buffering is applied at the SI layer). In any case, a change in the encapsulation format is applied when the $\alpha^{SD}(t)$ is produced (for example, in case of IPoATM or IPoDVB traffic, by applying LLC-SNAP with AAL5). We denote by ${}^{i}L_V^{SD}(\alpha^{SD}(t),\theta^{SD}(t)\cdot\phi(t))$, the loss volume of the *i*-th traffic class within the SD buffer. It is a function of the following elements: the SD inflow process $\alpha^{SD}(t)$ (deriving from the aggregation of the SI inflow processes $\alpha_i^{SI}(t)$, $i=1,\dots,N$, and the transport technology change), the fading process $\phi(t)$ and the SD bandwidth allocation $\theta^{SD}(t)$. No analytical expressions for ${}^{i}L_V^{SD}(\cdot)$ is available, since there are no instruments for the mathematical description of the statistical behaviour of the packets belonging to a specific connection within an aggregated trunk.

We suppose that resource allocation of SI can satisfy the required QoS at the SI level. Here, the key problem is to "equalize" the QoS measured at the SD layer in dependence of the QoS imposed at the SI layer. To capture this concept, it is useful to think at a *penalty cost function*, whose values can be interpreted as an indication about the current inability of the SD layer to guarantee the required QoS. In practice, ${}^{i}L_V^{SD}(\cdot)$

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can chase both the performance measured at the SI layer and a threshold value (fixed or time variant) that flows from SI to SD layer and represents the SLA. In both cases, the performance reference of SI is described through ${}^{i}L_V^{SD}(\cdot)$: in the first case, the loss volume of SI layer is just the measure of the buffer loss over time; in the second case, it is the product of the required PLP, (denoted by PLP_i^*), and the sub-portion of SD inflow volume relative to class i over a given time horizon $[k,k+1]$, i.e., ${}^{i}L_V^{SD}(\cdot) = PLP_i^* \cdot \int_k^{k+1} \beta_i^{SI}(t)\cdot dt$, where $\beta_i^{SI}(t)$ is the outflow rate process of i-th SI buffer.

The optimization problem can now be stated. **QoS Mapping Optimization** (QoSMO) **Problem**: find the optimal bandwidth allocation ${}^{Opt}\theta^{SD}(t)$, so that the cost function $J(\cdot,\theta^{SD}(t))$ is minimized:

$$
\begin{gathered}
{}^{Opt}\theta^{SD}(t) = \arg\min_{\theta^{SD}(t)} J(\cdot,\theta^{SD}(t));\; J(\cdot,\theta^{SD}(t)) = \mathop{E}_{\omega\in\Theta} L_{\Delta V}(\cdot,\theta^{SD}(t)) \\
L_{\Delta V}(\cdot,\theta^{SD}(t)) = \sum_{i=1}^{N}\left[{}^{i}L_V^{SI}(\alpha_i^{SI}(t),\theta_i^{SI}(t)) - {}^{i}L_V^{SD}(\alpha^{SD}(t),\theta^{SD}(t)\cdot\phi(t))\right]^2
\end{gathered}
\tag{2}
$$

We denote by ω a *sample path* of the system, i.e., a realization of the stochastic processes involved in the problem ($\phi(t)$, $\alpha_i^{SI}(t)$, $i=1,\dots,N$, $\alpha^{SD}(t)$) according to the statistical behaviour of the IP sources and to the channel degradation and with $\mathop{E}_{\omega\in\Theta}(\cdot)$ the mean over the set Θ of all the possible sample paths.

It is worth noting that finding a solution of (2) through analytical tools is a very hard task and approaching the problem by numerical approximations is recommended.

C. The equivalent bandwidth approach

Before detailing our proposal, we point out the peculiarities of a possible *equivalent bandwidth* approach, by underlying its adoption for the solution of the QoSMO problem under investigation.

Traditionally, equivalent bandwidth techniques are based on the statistical characterization of the traffic generated by users' applications. Sophisticated mathematical descriptions are proposed to this aim, based on proper traffic descriptors (*peak rate*, *mean rate*, *maximum burst size*, etc.) [7]. Unfortunately, in our case, the inflow process α^{SD} is the result of the outflow processes of the SI buffers and the aforementioned user oriented traffic descriptors can hardly be applied. The only simply applicable statistics, useful for the SD rate provision, are the *mean* (m) and the *standard deviation* (σ) of the α^{SD} process. Hence, a popular equivalent bandwidth technique, actually applicable in this context, is introduced in [7] and is ruled by (3) below. It exploits on-line measures without assuming any a-priori knowledge of traffic statistics and buffer size. Being: $k=1,2,\dots$ the time instants of the SD rate reallocations, $m_{\alpha^{SD}}(k)$ and $\sigma_{\alpha^{SD}}(k)$ the mean and the standard deviation respectively of the SD inflow process measured over the time interval $[k,k+1]$; the bandwidth provision ($\theta^{SD}(k+1)$) at the SD layer, assigned for time interval $[k+1,k+2]$, may be computed as a function of the measured statistics $m_{\alpha^{SD}}$ and $\sigma_{\alpha^{SD}}$:

$$\theta^{SD}(k+1) = m_{\alpha^{SD}}(k) + a\cdot\sigma_{\alpha^{SD}}(k) \tag{3}$$

where $a=\sqrt{-2\ln(\varepsilon)-\ln(2\pi)}$ and ε represents the upper bound on the allowed PLP. Such operative proposal is identified as **Equivalent Bandwidth approach** (EqB) in the following.

Another important difference from traditional equivalent bandwidth techniques relies on the heterogeneity of the QoS levels required in the SD aggregated trunk. Usually, the equivalent bandwidth is defined as the minimum rate allocation necessary to guarantee a specific level of QoS to a given flow. In our case, this definition is generalized, since we are looking for the minimum θ^{SD} satisfying all the PLP levels defined for the different IP classes in the SD core. EqB approach may be adapted to this need, by choosing the parameter ε as the most stringent PLP required at the SI layer. This allows guaranteeing all the PLP thresholds in the SD trunk, but it may introduce a bandwidth waste [11].

D. The Sensitivity Estimation algorithm

To summarize, we are looking for a novel measurement-based equivalent bandwidth algorithm that can face: **1)** heterogeneity of the QoS requests in the aggregated trunk, **2)** change of encapsulation format, **3)** fading counteraction, **4)** no knowledge of SD inflow process's statistical properties and **5)** of SD buffer size. To match these requirements, we capture the temporal behaviour of each single performance level ${}^{i}L_V^{SD}(\cdot)$ through on-line measurements and perform the SD rate reallocations accordingly to them. To reach the aim, we exploit the cost function $L_{\Delta V}(\cdot)$ derivative that can be obtained as:

$$\frac{\partial L_{\Delta V}(\cdot,\theta^{SD})}{\partial\theta^{SD}} = 2\cdot\sum_{i=1}^{N}\frac{\partial\, {}^{i}L_V^{SD}(\theta^{SD})}{\partial\theta^{SD}}\left[{}^{i}L_V^{SD}(\theta^{SD}) - {}^{i}L_V^{SI}(\theta_i^{SI})\right] \tag{4}$$

Due to the application of *Infinitesimal Perturbation Analysis* (IPA), recently developed in the field of *Sensitivity Estimation* techniques for *Discrete Event Systems* (see, e.g., [6] and references therein), each $\frac{\partial\, {}^{i}L_V^{SD}(\theta^{SD})}{\partial\theta^{SD}}$ component can be obtained in real time only on the basis of some traffic samples acquired during the system evolution. Let $[k,k+1]$ be the time interval between two consecutive SD bandwidth reallocations. The periods of time in which the buffer is not empty are defined as *busy periods*. The derivative estimation is computed at the end of the decision epoch $[k,k+1]$ as:

$$\left.\frac{\partial\, {}^{i}L_V^{SD}(\theta^{SD})}{\partial\theta^{SD}}\right|_{\hat\theta^{SD}(k)} = \phi(k)\cdot\sum_{\zeta=1}^{N_k^i}\left.\frac{\partial\, {}^{i}L_{k,\zeta}^{SD}(\theta^{SD})}{\partial\theta^{SD}}\right|_{\hat\theta^{SD}(k)} \tag{5}$$

$$\left.\frac{\partial\, {}^{i}L^{SD}_{k,\zeta}(\theta^{SD})}{\partial \theta^{SD}}\right|_{\hat{\theta}^{SD}(k)} = -({}^{i}v^{k}_{\zeta}(\hat{\theta}^{SD}(k)) - {}^{i}\xi^{k}_{\zeta}(\hat{\theta}^{SD}(k))) \quad (6)$$

where ${}^{i}L^{SD}_{k,\zeta}(\theta^{SD})$ is the ζ-th contribution to the SD loss volume of the i-th traffic class for each busy period B^{ζ}_{k}, within the decision interval $[k, k+1]$; ξ^{ζ}_{k} is the starting point of B^{ζ}_{k}; v^{ζ}_{k} is the instant of time when the last loss occurs during B^{ζ}_{k}; N^{i}_{k} is the number of busy periods within the interval $[k, k+1]$ for service class i. As to (1), $\hat{\theta}^{SD}(k)$ represents the SD bandwidth reduction due to fading, as outlined in section III, within time interval $[k, k+1]$ (i.e., $\hat{\theta}^{SD}(k) = \theta^{SD}(k)\cdot\phi(k)$).

The IPA-based derivative estimator (4) is then used to optimally tune the SD rate provision. The proposed optimization algorithm is based on the gradient method, whose descent step is ruled by (7):

$$\theta^{SD}(k+1) = \theta^{SD}(k) - \eta_k \cdot \left.\frac{\partial L_{\Delta V}(\cdot, \theta^{SD})}{\partial \theta^{SD}}\right|_{\hat{\theta}^{SD}(k)} \quad ; k = 1, 2, \ldots \quad (7)$$

We denote by η_k the gradient stepsize and by k the reallocation time instant. The algorithm tracks the optimal solution ${}^{Opt}\theta^{SD}(t)$ of the QoSMO problem (2) over time through the on-line gradient descent (7). It is called **Reference Chaser Bandwidth Controller** (RCBC).

V. PERFORMANCE ANALYSIS

In this section, we investigate the performance of our rate control mechanism through simulations. An ad-hoc C++ simulator has been developed for the SI-SAP environment shown in Fig. 2. In the following, for the sake of simplicity, we face the traffic aggregation problem by assuming no channel degradation over the satellite channel.

We consider the case of two SI traffic buffers. The first one conveys the traffic of 30 *Voice over IP* (VoIP) sources. Each VoIP source is modeled as an exponentially modulated on-off process, with mean on and off times (as for the ITU P.59 recommendation) equal to 1.008 s and 1.587 s, respectively. All VoIP connections are modeled as 64.0 kbps voice flows over RTP/UDP/IP. IP packet size is 80 bytes. SI service rate for VoIP (θ^{SI}_{VoIP}=232 kbps, measured through simulation) assures a SLA target PLP below 10^{-2} (SI VoIP buffer size is 30 IP packets). The second buffer is dedicated to a video service. "Jurassic Park I" video trace, taken from the web site referenced in [8], is used. Data are H.263 encoded and have an average bit rate of 260 kbps as well as a peak bit rate of 1.3 Mbps. SI rate allocation for video, θ^{SI}_{video} (also measured through simulations), is 350 kbps. It assures a PLP<10^{-3}, which is the target SLA for video (SI video buffer size is 10500 bytes). Both the outputs of the SI buffers are conveyed towards a single queue at the SD layer. DVB encapsulation (header 4 bytes, payload 184 bytes) of the IP packets through the LLC/SNAP (overhead 8 bytes) ([5]) is implemented in this case. The SD buffer size is 300 DVB cells.

In Fig. 3, we compare the SD bandwidth provision produced by RCBC and EqB. The loss probability bound ε for EqB is set to 10^{-3}, being the most stringent PLP constraint imposed at the SI level. The time interval between two consecutive SD bandwidth reallocations is denoted by T_{RCBC} and T_{EqB}, respectively for RCBC and EqB.

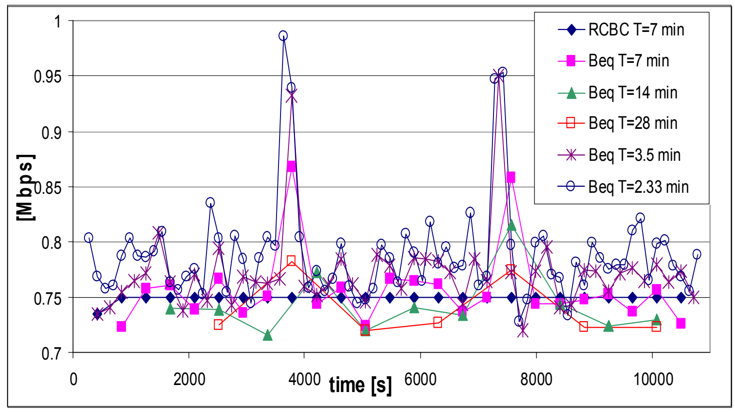

Figure 3. Aggregation of VoIP and Video. SD allocations. RCBC vs EqB.

T_{RCBC} is fixed to 7 minutes, while T_{EqB} is set to the following values:

$$\left\{T_{RCBC}\cdot 1/3,\ T_{RCBC}\cdot 1/2,\ T_{RCBC},\ T_{RCBC}\cdot 2,\ T_{RCBC}\cdot 3\right\}$$

in different tests. EqB allocation is parametrized by the dimension of the time horizons in which $m_{\alpha^{SD}}$ and $\sigma_{\alpha^{SD}}$ are computed. It allows highlighting the possible inaccuracy introduced by the real time computation of the EqB statistics.

The RCBC captures the bandwidth need of the SD layer in a single reallocation step. The EqB produces strong oscillations in the SD rate assignment. The instability arising in the EqB allocations is evident and it is inversely proportional to T_{EqB}. On the other hand, RCBC finds the ${}^{Opt}\theta^{SD}$ after the first reallocation step and no changes are produced for the rest of the simulation. The gradient descent (7) is initialized through $\theta^{SD}(0) = m_{\alpha^{SD}}(0) + a\cdot\sigma_{\alpha^{SD}}(0)$, where $m_{\alpha^{SD}}(0)$ and $\sigma_{\alpha^{SD}}(0)$ are the EqB statistics computed during the first 7 minutes of simulation. The gradient stepsize η_k is ruled by $\eta_k = \eta_k,\ \eta_k = \left.\frac{\partial L_{\Delta V}(\cdot, \theta^{SD})}{\partial \theta^{SD}}\right|_{\hat{\theta}^{SD}(k)} \cdot \upsilon;\ \upsilon = 1.22\cdot 10^{-9}\ \forall k$ to assure a decreasing behaviour of η_k, which is a regular condition of gradient stochastic algorithms to guarantee convergence. We must note that $\eta_k \xrightarrow{k\to+\infty} 0$, since the sequence $\theta^{SD}(k),\ k = 1, 2, \ldots$, driven by descent (7), approximates the solution of (2). The value of parameter υ has been obtained by a simulation inspection, aimed at maximizing the convergence speed, which, actually, is evident in Fig. 3. It is also clear from Fig. 3 that the IPA-based estimation (4) is more robust than the on-line estimation of $m_{\alpha^{SD}}$ and $\sigma_{\alpha^{SD}}$. The performed sensitivity estimation can drive RCBC toward the optimal solution of the QoSMO problem, with a higher degree of precision than EqB approach.

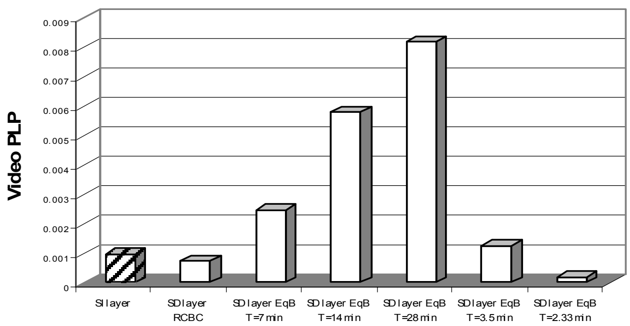

Figure 4. Aggregation of VoIP and Video. Video PLP.

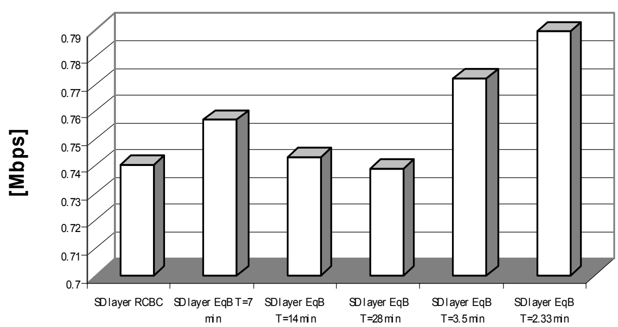

Fig. 5. Aggregation of VoIP and Video. Average SD layer provision.

SD buffer video PLP (whose performance threshold is 10^{-3}), averaged over the entire simulation horizon, is shown in Fig. 4. The performance of RCBC, referenced as "SD layer RCBC" is very satisfying: PLP is below the SI threshold, reported in the first column. Actually, the RCBC video PLP is $7.56 \cdot 10^{-4}$. A result "below threshold" has been measured for EqB only for frequent reallocations ($T_{EqB} = \frac{T_{RCBC}}{3}$=2.33 minutes). Similar comments may be done for VoIP PLP (not shown here) whose average value for RCBC is $1.01 \cdot 10^{-4}$, below 10^{-2} threshold. The corresponding bandwidth allocations, averaged over the simulation duration, are shown in Fig. 5. RCBC not only allows saving bandwidth compared to the "SD layer EqB T=2.33 min" strategy, which also guarantees performance requirement, but offers a performance comparable to EqB strategies, whose offered PLP is far from the SI threshold. On the other hand, when EqB assigns more bandwidth than necessary, SD PLP (both video and VoIP) is much lower than the thresholds. The behaviour is exactly the contrary when SD bandwidth need is underestimated. This outlines the difficulty to find the optimal measurement window T_{EqB} for EqB statistics. RCBC, on the other hand, effectively finds the optimal operation point of the system, namely, the minimum SD bandwidth provision needed to track the SI QoS thresholds.

VI. Conclusion and Future Work

The challenging problem of *Broadband Satellite Multimedia* architecture is the communication between *Satellite Independent* (SI) and *Satellite Dependent* (SD) layers. SD stack should offer QoS guarantees to upper layers in dependence of the *Service Level Agreements* fixed at the SI layer. A control scheme has been studied to allow bandwidth adaptation at SD layer and consequent tracking of the loss performance metric. The results have shown a good efficiency.

Further simulation scenarios, including other transport technologies (as MPLS, IPv6), are currently the subject of ongoing research, as well as a deep analysis of the traffic aggregation problem in dependence of different traffic categories.

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