Bandwidth Allocation in Geostationary Satellite Faded Channels for Internet Traffic

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Abstract - **Resource allocation schemes dedicated to satellite environment often considers Internet traffic as the superposition of traffic sources without distinguishing between TCP and UDP flows, even if TCP and UDP imply very different traffic characteristics. The basic idea of this work is that a resource allocation algorithm where the allocation function is conscious of the difference may result more efficient and fair. To reach the aim, the paper introduces: a system control architecture with three types of flows entering the network (Constant Bit Rate (CBR), UDP and TCP) and a cost function including an analytical measure of the packet loss for TCP flows The work joins the novelties introduced with a bandwidth allocation control already in the literature (called CAP-ABASC) and derives a new allocation scheme called E-CAP-ABASC (Extended-CAP-ABASC).**

Performance evaluation includes the comparison of the two allocation strategies mentioned above.

I. Introduction

Matching the applications that use TCP/IP with the advantages offered by satellites, it is natural to think of TCP/IP-based applications over satellite networks but the general characteristics (e.g. fading) of the links heavily affect the performance of the communication. Resource allocation is an issue of particular importance in this environment.

Within this environment, the paper takes the adaptive bandwidth allocation system (called CAP-ABASC, Constrained Average Probability - Adaptive Bandwidth Allocation in Satellite Channels) proposed in [1] as reference. The satellite network is composed of earth stations connected through a geostationary satellite. An earth station (or the satellite itself, if switching on board is allowed) represents the master, which manages the resources and provides the other stations with a portion of the overall bandwidth; each station shares the assigned portion between its traffic flows. Three types of traffic are considered: a QoS guaranteed traffic, operating at a fixed speed (measured in Kbps) and two non-guaranteed best-effort traffic: UDP, modeled by a self-similar Pareto distribution [3, 4], and TCP, whose packet loss model is introduced in [2] and briefly summarized in this paper. The fading is considered by assigning a probability of channel degradation to each link, along with a weighting coefficient to 'measure' the degradation itself. The TCP packet loss formulation, together with the CBR and UDP models, is used to get a new cost function and a bandwidth allocation scheme (called Extended-CAP-ABASC), which represents the novelty of this work. E-CAP-ABASC is compared with CAP-ABASC.

II. Network Topology and Channel Model

The satellite network is composed of I earth stations. One station is the "master" and controls the allocation of the satellite resources. Each station gathers traffic from the users directly connected or through local area networks. Ka-band (20-30 GHz) may be the technological reference (rain may cause satellite link deterioration essentially due to fading) even if the study is not linked to a particular bandwidth choice. The real availability of the channel bandwidth is strictly dependent on rain fade compensations; due to its main role, it is very important to get a method to describe the effect of fading: a simple way is bandwidth reduction [1]. Mathematically, it means that the nominal bandwidth C_{tot} (assigned to a station) is reduced of a factor β, which is a stochastic parameter distributed in the real numbers interval [0,1].

$$C_{real} = \beta \cdot C_{tot} \tag{1}$$

A specific value β corresponds to a fixed fading level. A technical interpretation of the factor β may be the bandwidth reduction due to the presence of a FEC (Forward Error Correction) scheme. Each fading level, happening with an associated probability p_f, deserves a particular FEC. In satellite environments, link layer corruption due to noise is typical and in general the packet loss is due to it. Nevertheless if FEC schemes are used, the link layer corruption problem may be seen as congestion problem. In other words, the FEC strategies make negligible the channel errors but reduce the available service capacity so creating possible bottleneck. In this view, this work explicitly consider reduction through the factor βand assumes that all the packet losses happening during communication may be considered due to congestion event. On the other hand the conditions described, with the use of FEC, the loss due to link layer corruption may be supposed tending to zero and the simple channel model proposed in (1) seems to be a reasonable approximation of satellite channel behavior.

III. Control System Architecture

The traffic considered is divided into: CBR flow, which is privileged since a QoS threshold in term of call blocking probability is assured through a call admission control (CAC) scheme; TCP and UDP flows that, in this work, are considered as two separated components. The traffic is conveyed in each earth station and, virtually, enters the system bandwidth allocation.

IV. Source Models for CBR, UDP and TCP

The models reported in the following are taken from the literature. The models for CBR and UDP have been already used in CAP-ABASC [1] and are quickly summarized here for the sake of completeness. The analytical expression of the TCP packet loss probability over geostationary satellite channels has been proposed by the authors in [2], together with the performance evaluation. The expression obtained is function of the bandwidth available and is suited to be used in control mechanisms, as applied in this work. The method to get it is briefly reported in the following.

CBR. The relevant metric, which will be used in the allocation strategy, is call blocking probability that is modeled with the Erlang B formula (M/M/m/m queuing system, in formula (2)).

$$P_B(k_{max}) = B\left(\frac{\lambda}{\mu}; k_{max}\right) \qquad (2)$$

k_{max} is the maximum number of servers; λ and μ are, respectively, the call arrival rate and the service rate, both of them exponentially distributed. Given $k(t) \leq k_{max}$ active CBR sources at the instant t (i.e., having k(t) busy servers), emitting data at R_{CBR} [Kbps], the bandwidth used by real-time CBR traffic is, at time t:

$$C_{CBR} = k(t) \cdot R_{CBR} \qquad (3)$$

UDP. The metric used in the allocation scheme is the packet loss probability. The model applied is Y/D/Cs/Q and has been proposed in reference [4]. It has been used in the CAP-ABASC strategy to model all the IP-based traffic with no distinction between UDP and TCP. When the model was proposed, there was not any reference to congestion control algorithm and to the transport layer used. Nevertheless, the particular type of statistical ON-OFF fractal traffic nature suggests its use for UDP traffic, where no acknowledgement-based mechanism is provided for congestion control. The analytical approximation is reported in (4):

$$P_{loss}^{UDP}(C_s) = min\left\{\frac{c\lambda_s\left(\alpha(\alpha-1)\right)^{-1}}{\left(C_s - \lambda_s\bar{\tau}\right)} Q_{UDP}^{-(\alpha+1)}, 1\right\} \qquad (4)$$

C_s is the number of servers busy in the Y/D/Cs/Q system and Q_{UDP} is the length (measured in packet of 1500 bytes) of the IP buffer dedicated to UDP source, which is supposed to tend to infinite. α is the Pareto parameter (1<α<2), c is a normalization constant and λ_s is the arrival rate of UDP sources equal to λ_{asy}·T, where T is the source packetization time and λ_{asy} the burst of packets generated by the UDP sources. The approximation reported in (4) is valid if $C_s > \lambda_s\bar{\tau}$, otherwise the packet loss probability is 1. It is important to specify the relation between C_s and the transmission bandwidth available for this kind of sources: if the peak bandwidth for each UDP source is B_p, the average value of IP packets length is L and T, defined as L/ B_p like in [4].

$$C_s = \frac{C_{UDP}}{L} T = \frac{C_{UDP}}{L} \cdot \frac{L}{B_p} = \frac{C_{UDP}}{B_p} \qquad (5)$$

Where, C_{UDP} represents transmission bandwidth expressed in Kbps dedicated, in this work, to UDP traffic.

TCP. The model used within the control for the TCP traffic is directly taken from [2] and it is specifically studied for the geostationary satellite environments.

T_n is the round trip time at the TCP layer for the n-th connection. It is supposed constant for each packet of the n-th connection. W^{pipe} is the maximum volume of information that can be transmitted to the system composed of a channel server of capacity C_{TCP} [Kbps] and of the IP buffer of size Q_{TCP} (measured in packets of 1500 bytes and dedicated to TCP traffic sources).

Defining C_{TCP}^n and Q_{TCP}^n, constant over time, respectively, the maximum portion of the capacity C_{TCP} and of the buffer Q_{TCP}, "seen" by the n-th connection, and W_n^{pipe} the maximum volume of information that can be transmitted to the system by the n-th connection, it is true that:

$$W^{pipe} = \sum_{j=1}^{N} W_j^{pipe} = \sum_{j=1}^{N} (C_{TCP}^j \cdot T_j + Q_{TCP}^j) \qquad (6)$$

Being the satellite geostationary, the round trip time may be supposed fixed and equal for all the sources. This condition, written mathematically in (7), together with the hypothesis of synchronization, gives origin to the fairness condition.

$$T_j = T_n = RTT, \forall j, n \in [1, N] \qquad (7)$$

Remembering that $\sum_{j=1}^{N} Q_{TCP}^j = Q_{TCP}$ and $\sum_{j=1}^{N} C_{TCP}^j = C_{TCP}$, from equation (6), it is true that:

$$W^{pipe} = RTT \cdot C_{TCP} + Q_{TCP} \qquad (8)$$

Taking TCP Reno as reference, the dimension of the congestion window W_n of a generic source *n* varies between a minimum and a maximum value as introduced in [6] (TCP-Reno simplified model). Its size grows up to saturate the channel; if a packet is lost, the window decreases its maximum size in dependence of a factor *m* [8] that varies between 0 and 1 (typically *m*=1/2, as indicated in [7]). The packet loss probability p_n of n-th TCP connection, fixed the parameters defined above, is reported in (9). b_n is the number of packets covered by one acknowledgement for connection n-th. The detailed computations needed to get it may be found in reference [2], where (9) has been originally proposed.

$$p_n = P_{loss}^{TCP} = c \cdot \frac{8N^2}{b_n \cdot (C_{TCP} \cdot RTT + Q_{TCP})^2} \qquad (9)$$

c is a numerical constant equal to 16/27. If $b_n = b$, for all *n*, as supposed in this work, the packet loss probability in (9) correspond to the entire TCP aggregate. The packet loss probability is used within the E-CAP-ABASC allocation scheme as proposed in the next section.

V. E-CAP-ABASC

The TCP packet loss introduced in the previous section is used within the control mechanism called E-CAP-ABASC (Extended - Constrained Average Probability - Adaptive Bandwidth Allocation in Satellite Channels) which is the novelty of the paper. The new control algorithm uses a global packet loss probability for best-effort traffic but separates the buffer dedicated to UDP and the TCP buffer and to uses an appropriate traffic model for TCP. Best-effort packet loss probability is defined as the probability that either one UDP packet or one TCP packet is lost or both of them are lost simultaneously. Formally, it may be written as a function of the bandwidth available and of the number of active TCP and UDP sources:

$$P_{loss}\left(C^{be},N,M\right)=P_{loss}^{TCP}+P_{loss}^{UDP}-P_{loss}^{TCP}P_{loss}^{UDP} \quad (10)$$

P_{loss}^{TCP} has been defined in (9). P_{loss}^{UDP} is modeled as in formula (4), which, in CAP-ABASC, was used to describe generic best-effort traffic. C^{be} is the overall bandwidth for best-effort traffic (C_{TCP} plus C_{UDP}). The bandwidth assigned to UDP and TCP flows is proportional to the number of active sources ((11) and (12)). M is the number of UDP sources, N of TCP sources.

$$C_{UDP}=\frac{M}{N+M}C^{be} \quad (11)$$

$$C_{TCP}=\frac{N}{N+M}C^{be} \quad (12)$$

The control scheme is composed of a high control layer called Centralized Bandwidth Allocator (CBA) that distributes the bandwidth capacity among the earth stations and of a low layer, called Local Controller (LC), which splits the capacity allocated to each station to two contributions: C^{CBR}, for CBR real time traffic and C^{be}, for the Internet traffic including UDP and TCP. Each i-th earth station solves a local optimization problem to share the capacity assigned, so finding a threshold $k_{max}^{(i)}$, which is the maximum number of acceptable CBR calls that allows guaranteeing a specific QoS level in terms of call blocking probability. The following parameters are defined for the stations.

- CBR: call arrival rate $\lambda^{(i)}$ [calls/s], call duration mean value $1/\mu^{(i)}$ [s], bit rate of the i-th call $R_{CBR}{}^{(i)}$ [Kbps].
- UDP: packet arrival rate $\lambda_{asy}^{(i)}$ [burst/s], UDP buffer size $Q_{UDP}^{(i)}$ [packets of 1500 bytes], Pareto parameter $\alpha^{(i)}$.
- TCP: RTT (round trip time for all the TCP flows), TCP buffer size $Q_{TCP}^{(i)}$ [packets of 1500 bytes].

For the sake of completeness, the optimization algorithm, defined in [1] and used in this work for bandwidth allocation is quickly revised in the following. $C_{min}^{(i)}$(Kbps) is defined as follows:

$$C_{min}^{(i)}=\arg\min_{X^{(i)}}\left\{X^{(i)}\in\Re : P_B^{(i)}\left(\left\lfloor\frac{X^{(i)}}{R_{CBR}^{(i)}}\right\rfloor\right)\le\gamma^{(i)}\right\} \quad (13)$$

$P_B^{(i)}(\cdot)$ is the call blocking probability for the i-th call, generically defined in (2). The variable $X^{(i)}$ ranges between 0 and C_{tot}, which is the overall available channel bandwidth. $K_{max}^{(i)}$ is the maximum number of calls guaranteed by $C_{min}^{(i)}$. Formula (14) establishes the relation among the two quantities.

$$C_{min}^{(i)}=\left\lceil K_{max}^{(i)}R_{CBR}^{(i)}\right\rceil \quad (14)$$

For a given bandwidth assigned to the i-th station $X^{(i)}$, the bandwidth allocation fixes the following maximum number of real time (CBR) calls:

$$\hat{K}_{max}^{(i)}\left(X^{(i)}\right)=\begin{cases}\left\lfloor\dfrac{C_{min}^{(i)}}{R_{CBR}{}^{(i)}}\right\rfloor & if\ \ X^{(i)}>C_{min}^{(i)}\\ \left\lfloor\dfrac{X^{(i)}}{R_{CBR}{}^{(i)}}\right\rfloor & if\ \ X^{(i)}\le C_{min}^{(i)}\end{cases} \quad (15)$$

Fixed the maximum number of CBR calls acceptable in the system (i.e. the number of servers available within the system at station i), $k^{(i)}(t)\le K_{max}^{(i)}$ is the number of CBR calls in progress at time t, at station i. The bandwidth dedicated to CBR traffic is defined in (3) ($R_{CBR}{}^{(i)}\cdot k^{(i)}(t)$). The residual capacity ($C_{be}^{(i)}=X^{(i)}-R_{CBR}{}^{(i)}\cdot k^{(i)}(t)$) is available for the Internet traffic at the i-th station. The penalty function is defined as:

$$F_{CAP}^{(i)}\left(X^{(i)}\right)=\begin{cases}0 & if\ \ \bar{P}_B^{(i)}\left(\hat{K}_{max}^{(i)}\left(X^{(i)}\right)\right)\le\gamma^{(i)}\\ H & if\ \ \bar{P}_B^{(i)}\left(\hat{K}_{max}^{(i)}\left(X^{(i)}\right)\right)>\gamma^{(i)}\end{cases} \quad (16)$$

where H is a very large constant, and

$$\bar{P}_B^{(i)}\left(\hat{K}_{max}^{(i)}\left(X^{(i)}\right)\right)=\sum_{f=1}^{F}p_f^{(i)}\cdot P_B^{(i)}\left(\hat{K}_{max}^{(i)}\left(X^{(i)}\right)\right) \quad (17)$$

F is the maximum number of fading level and p_f their associated probability. The Centralized Bandwidth Allocator (CBA) assigns the bandwidth portions to the earth stations by minimizing the function cost, defined as

$$J_{CAP}\left(X^{(1)},...,X^{(I)}\right)=\sum_{i=1}^{I}\sum_{f=1}^{F}p_f^{(i)}J_{CAP}^{(i)}\left(\beta_f^{(i)}X^{(i)}\right) \quad (18)$$

where

$$J_{CAP}^{(i)}\left(X^{(i)}\right)=P_{loss}^{(i)}\left(X^{(i)},\hat{K}_{max}^{(i)}\left(X^{(i)}\right)\right)+F_{CAP}^{(i)}\left(X^{(i)}\right) \quad (19)$$

whose solution is:

$$\left\{C^{(1)},...,C^{(I)}\right\}=\arg\min_{X^{(1)},...,X^{(I)}}\left\{J_{CAP}\left(X^{(1)},...,X^{(I)}\right)\right\} \quad (20)$$

$$if\ C_{tot}<\sum_{i=1}^{I}C_{min}^{(i)}\Rightarrow C^{(i)}=\frac{C_{tot}}{\sum_{i=1}^{I}C_{min}^{(i)}}\cdot C_{min}^{(i)} \quad (21)$$

P_{loss}^{TCP} is computed by using the new formula proposed in (10).

VI. PERFORMANCE COMPARISON

The aim is to compare the performance in terms of bandwidth saving of the E-CAP-ABASC (presented in (10)) with the results obtained with the CAP-ABASC. When Internet traffic is also considered other QoS metrics such as throughput, end to end delay and delay jitter may be important but the allocation strategies are compared in terms of packet loss probability to make a fair comparison with CAP-ABASC that used that metric. It is worth noting that without any distinction between UDP and TCP, performance measured with CAP-ABASC does not change when best-effort traffic balance is varied and the capacity assigned to the station 3 is more than the bandwidth provided by the other algorithms. It means that, without any distinction between TCP and UDP traffic, the bandwidth allocation is too rough. E-CAP-ABASC (identified as E-CAP in the following) is compared with a slight modified version of the CAP-ABASC strategy sensible to traffic variations. It uses two separated buffers for TCP and UDP traffics but describes the packet loss of both traffics through formula (4). That It allows stating the additional value of the new allocation scheme regarding both buffer separation and use a dedicated TCP packet loss model. The comparison is performed by varying the percentage of TCP and UDP traffic loading the system. The tests have been performed with 4 earth stations (numbered from 0 to 3) and using the following parameters for each i-th station.

CBR: $\lambda^{(i)} = 0.006$ [calls/s], $1/\mu^{(i)} = 600$ [s], $R_{CBR}{}^{(i)} = 128$ [Kbps], threshold $\gamma^{(i)} = 0.05$.

UDP: $\lambda_{asy}^{(i)} = 10$ [burst/s], $Q_{UDP}^{(i)} = 8000$ [packets], $\alpha^{(i)} = 1.5$, length L=1500 [bytes] for the packets, peak rate $B_p = 64$ [Kbps] and variable number M of UDP sources.

TCP: variable number N of TCP sources, a RTT=520 [ms] and $Q_{TCP}^{(i)} = 8000$ [packets], coherently with the UDP buffer.

The parameter choices are directly imposed by the requirements of the model applied for UDP traffic, where the buffer dimension would be supposed infinite. The overall number of best-effort active source is 100 (M+N=100), in all cases. The fade levels ($\beta^{(i)}$) are considered variable only for Station 3. Stations 0, 1 and 2 are in clear sky conditions. The probability p_f related to the fading level is fixed always equal to 1. The overall bandwidth available C_{tot} is set to 8 [Mbps]. The constraint over the call blocking probability for the CBR traffic is set to 0.05 and it is kept for all the following tests. No performance evaluation concerning CBR traffic is reported because it is out the scope of this paper. E-CAP-ABASC algorithm uses this TCP feature so improving the bandwidth utilization and the efficiency of the overall system. Fig. 1, Fig. 2 and Fig. 3 contain a comparison of the overall bandwidth necessary for all the four stations (the sum of the bandwidth needed to each earth station) to get a packet loss probability of station 3 lower than 0.001 (0.1%). The fade value ranges from 0 to 1. Three unbalance cases are considered: "10% TCP 90% UDP", "50% TCP 50% UDP", and "90% TCP 10% UDP". Concerning the "90% TCP 10% UDP" case (in Fig. 1), the bandwidth needed is similar for the two mechanisms compared because the advantage of E-CAP-ABASC differentiation is limited by P_{loss}^{UDP} that receives a low quantity of capacity and provides high values of itself, thus of the global packet loss probability. The 10% of TCP sources assures the slight bandwidth gain appearing in Fig. 1. Fig. 2 reports the "50% TCP 50% UDP" case. The bandwidth gain is more evident because half of the aggregated sources uses the rate-limiting TCP congestion control mechanism. The "10% TCP 90% UDP" case is contained in Fig. 3.

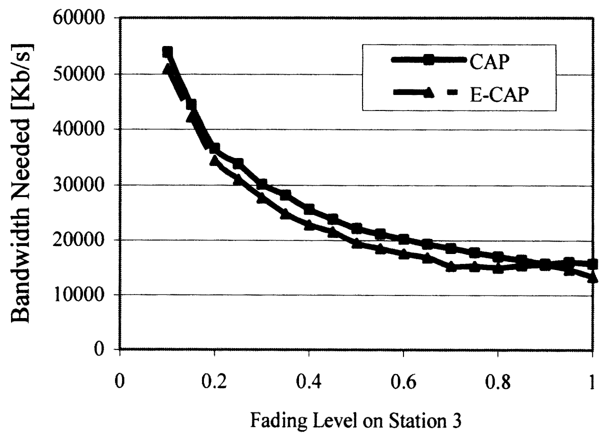

Fig. 1. Bandwidth necessary [Kbps] versus fading level of station 3 (90% TCP 10% UDP).

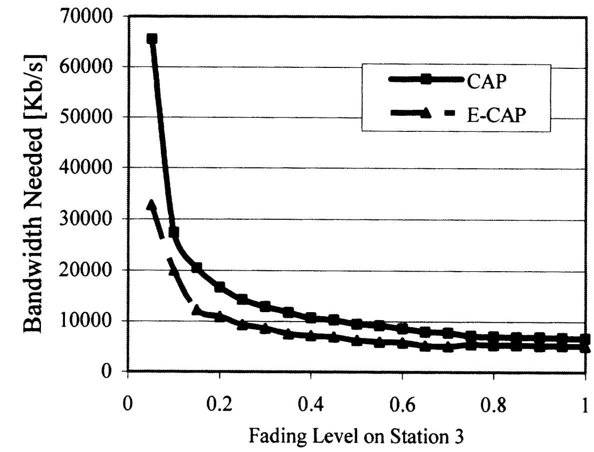

Fig. 2. Bandwidth necessary [Kbps] versus fading level of station 3 (50% TCP 50% UDP).

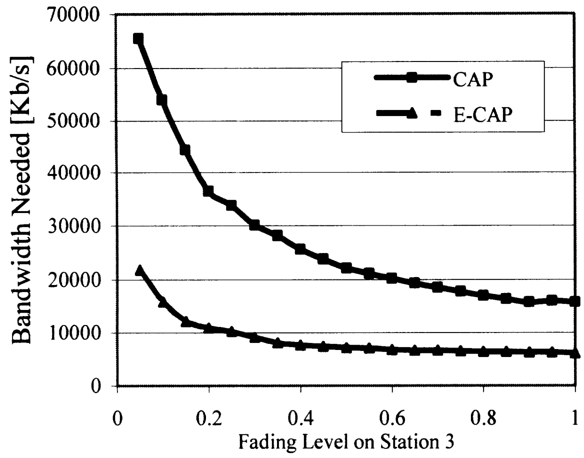

Fig. 3. Bandwidth needed [Kbps] versus Fade level over station 3 (10% TCP 90% UDP).

The bandwidth gain is really relevant in this case. The average bandwidth gain of the E-CAP-ABASC with respect to the strategy CAP-ABASC (CAP-2) is shown in Fig. 4, for each earth station. The gain is defined as the ratio between the bandwidth allocated by CAP-ABASC and the same quantity provided by E-CAP-ABASC, to keep the packet loss probability under the threshold, set to 0.001. The average value indicated is computed over all the β values considered in this study and it is shown for the unbalanced traffic conditions "90% TCP 10% UDP", "50% TCP 50% UDP", and "10% TCP 90% UDP". The gain is evident for all the earth stations when TCP traffic is dominant but it is noticeable also for the "50% TCP 50% UDP" case, in particular concerning the faded station.

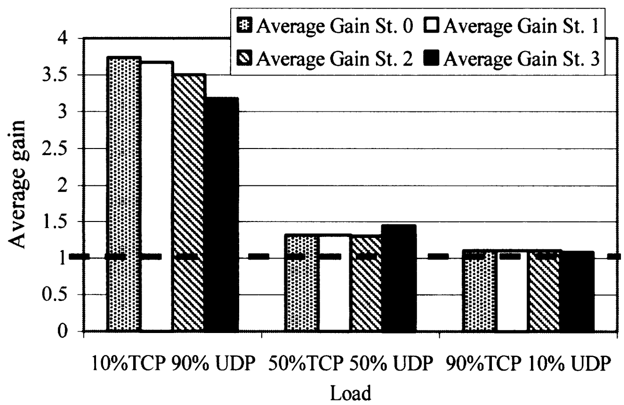

Fig. 4. Average bandwidth gain of the E-CAP-ABASC with respect to the strategy CAP-ABASC.

VII. Conclusions

The paper has introduced a novel control architecture (E-CAP-ABASC) for satellite systems with three types of flows entering the network and a measure of the packet loss for TCP. The satellite network is composed of earth stations connected through a geostationary satellite. The TCP packet loss formulation, together with the CBR and UDP models, are used to derive a new cost function and bandwidth allocation scheme. Performance evaluation contains the results of the new allocation strategy concerning the overall bandwidth gain and the comparison with an allocation mechanism already in the literature. E-CAP-ABASC, distinguishing TCP and UDP traffic, uses the TCP congestion control feature, which reduces the bit rate entering the network in case of congestion. It allows improving the bandwidth utilization and guaranteeing an efficient behavior of the overall system.

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