Suitability of DAMA and Contention-Based Satellite Access Schemes for TCP Traffic in Mobile DVB-RCS

Nedo Celandroni and Raffaello Secchi

Abstract—The problem of optimizing access and bandwidth sharing among Transmission Control Protocol (TCP) connections in the mobile digital video broadcasting return channel via satellite (DVB-RCS) is tackled in this paper. After sketching the general system architecture, we explicitly deal with the dynamic assignment of bandwidth to TCP connections on the return link, which is accomplished by a network control center (NCC) placed onboard the satellite. Mobile users access the satellite in multifrequency time-division multiple access (MF-TDMA), whereas they receive data from the NCC in time-division multiplexing (TDM). Two different techniques, based on deterministic and random access, are compared in terms of bandwidth usage and average completion time per connection, when the mobile user acts as both server and client. In the server case, to increase the TCP throughput, both packet-level forward error correction (FEC) on data sent by mobile users and a duplicated and delayed acknowledgment technique for TCP acknowledgment traffic from the NCC to the mobile users are applied. An analysis of the packet losses and a simulation campaign of file transfers by employing a realistic channel model has been carried out. The results of the analysis show the convenience of adopting a technique, in addition to the optimal data redundancy in different cases, such as the server or client role of users, their willingness to pay, the file size, and the environment type.

Index Terms—Demand assignment multiple access (DAMA), digital video broadcasting return channel via satellite (DVB-RCS), diversity slotted ALOHA (DSA), satellite mobile environment, Transmission Control Protocol (TCP) throughput.

I. INTRODUCTION

The convergence of video, audio, and Internet access with mobile services in a satellite network, which is currently referred to as quadruple play, is considered to be one of the key factors for the future success of digital video broadcasting (DVB) technology. Future releases of the current DVB return channel via satellite (DVB-RCS) standard [1] are indeed expected to support both Internet mobile access and interactive services across a satellite network. The development of mobile services in DVB-RCS is ongoing, and a substantial research effort is required to enable the medium access control (MAC) layer to mitigate negative effects induced by mobility. The goal of this paper is to propose improvements to the standard by presenting a comprehensive analysis of the suitability of various bandwidth allocation schemes to elastic traffic services in a vehicular environment. More specifically, we study the behavior of Transmission Control Protocol (TCP) flows originating from or directed to individual mobile sources (e.g., cars with radio equipment). Different from large mobile traffic terminals (e.g., trains, ships, and aircraft), which may convey traffic from several individual sources, these terminals tend to produce small and bursty flows and require careful management.

The design of a satellite network architecture that is suitable for TCP connections can begin by looking at various proposals presented in the literature (see, for instance, [2]) to mitigate impairments that are mainly due to the large latency. Solutions are generally categorized as end to end and link layer. End-to-end approaches focus on modifications of the transport layer to make TCP aware of the presence of the wireless link; such schemes may require the active participation of the network [3], only TCP sender modifications [4], or specific feedback in the TCP/Internet Protocol (IP) header [5], [6]. On the other hand, link-layer schemes employ forward error correction, automatic repeat request (ARQ), or a combination of both to make the link reliable. These solutions are typically embedded in a more general framework, which is sometimes referred to as the performance-enhancing proxy [7], [8], which may take advantage of dedicated entities, such as spoofing or splitting agents, traffic classification and conditioning, optimized transport protocols (e.g., [9]), and reservation schemes.

We propose a new satellite network architecture for vehicular users oriented to the link-layer paradigm. Starting from the discussions in [10] and [11], which are related to the TCP performance over fixed satellite links adopting DVB-RCS demand assignment multiple access (DAMA) schemes, this paper aims at presenting performance evaluations in terms of bandwidth cost and transfer time for mobile links. The original contribution of this paper consists of evaluating the suitability of the access scheme for TCP over a channel with impairments caused by mobility. Other studies consider the

1TCP flows are also called elastic in that the congestion-avoidance algorithm of TCP causes an elastic variation of the throughput, in reaction to network state variations.
TCP performance optimization over a discontinuous channel (such as [12]) but without taking into account the dynamics of the allocation scheme.

The rest of this paper is organized as follows: In Section II, we provide a description of the system architecture and the details of the analyzed algorithms. System design issues are discussed in Section III, and simulation results are presented in Section IV. Finally, in Section V, we conclude this paper, summarizing main contributions and outlining future work.

II. System Architecture

The multimedia mobile bandwidth allocation (MUMOBAL) architecture is the outcome of a joint research action undertaken in the framework of the SatNEx II European Network of Excellence [13]. The objective of MUMOBAL is to respond to design issues related to the user mobility in a satellite network. In this section, we summarize the main elements of this architecture.

The RCS terminal (RCST) acts as an access point for users in a multiservice transmission environment. The system is centrally controlled by a network control center (NCC), which may be placed in an Earth station or onboard the satellite (onboard processing). The results presented in this paper refer to a system with the NCC onboard the satellite. However, the architecture is also consistent, although with poorer performance, when the NCC is placed on ground.

MUMOBAL considers various classes of users: fixed, collective mobile (e.g., trains, ships, and aircraft, which collect traffic from several sources), and vehicular users. The last class poses the most challenging constraints in that, while each fixed and collective mobile user offers a moderately variable traffic (being the multiplexing of several sessions), vehicular users are characterized by connections that are highly variable in duration and bandwidth requirements. In fact, we refer to particular vehicles, such as rescue and civil protection vehicles, in addition to the cars of business and government people; the number of such vehicles may be large over a wide coverage area, but each of them uses the network with a very small duty cycle. In addition, the radio channel quality of mobile users is highly variable and subjected to intermittent link failures due to multipath and shadowing; this implies the adoption of mechanisms for compensating channel blockage periods.

MUMOBAL operates a class-based dynamic bandwidth allocation that guarantees a degree of independence among traffic classes by isolating each class of traffic in a different superframe portion. At each superframe, the boundaries between classes may be repositioned, taking into account the traffic load, channel conditions (e.g., users’ environmental positions), and quality-of-service requirements.

The allocation for each RCST is done in the time–frequency space, i.e., in multifrequency time-division multiple access (TDMA), following either a collision-free or a contention-based access scheme. The two alternatives dynamically share the available bandwidth as detailed here; moreover, the collision-free bandwidth portion is subdivided into two parts, which are devoted to carrying streaming (multimedia) and elastic traffic. More specifically, for each of the three bandwidth partitions, we adopt the following: 1) an independent bandwidth allocation scheme for multimedia traffic [14]; 2) a DAMA scheme (Section II-A) with dynamic or constant request of bandwidth [15], [16]; and 3) a contention-based access (Section II-B). After the NCC has allocated the amount of bandwidth required for partitions 1 and 3, the rest of the bandwidth is assigned in best-effort mode to the second partition.

The task of selecting the most suitable partition between the second and third partitions to serve a single TCP connection is demanded to the RCST. The choice is performed by considering the benefits for TCP in terms of bandwidth cost and transfer delay, as detailed in Section III-B. One of the factors that determine this choice is the length of the file to transfer; since this parameter has to be known at the MAC layer, cross-layer actions are performed [17].

Our approach is based on an infinite source asymptotic, i.e., we suppose that a single user is not able to significantly affect the load of the whole system. This hypothesis is justified by the fact that the population of vehicular users is generally vast.

A. DAMA Scheme

Since our objective is to define an allocation scheme that is suitable for TCP connections over a mobile satellite link, we carried out a set of simulations considering the request categories proposed in DVB-RCS [1, Sec. 6.8], i.e., continuous rate assignment (CRA), rate-based dynamic capacity (RBDC), and volume-based dynamic capacity (VBDC). The results of these experiments indicated that none of these categories dominates over the others in all cases of practical interest. A different candidate solution was indeed selected, depending on the environment type, connection length, and loading conditions. Moreover, the indications provided by different performance metrics (i.e., bandwidth consumption or connection completion time) led us to different choices. Thus, we design a flexible DAMA scheme that generalizes the DVB-RCS request categories and exhibits higher performance in many cases. Our approach consists of determining over a wide range of cases the optimal DAMA parameters, storing them in lookup tables, and using them at the run time.

In our system, the RCST has two options for requesting bandwidth: 1) fixed request (FR), which consists of asking for a fixed amount of bandwidth, and 2) variable request (VR). The VR option that we consider has some similarities with the First-in–first-out Ordered Demand Assignment (FODA) system described in [16]. When VR is enabled, the RCST computes the request, taking into account instantaneous queue size \( V \) and incoming traffic rate \( R \), which is the amount of data entering the RCST queue in each superframe. Requests for bandwidth are sent in each burst using the satellite access control (SAC) field, and new requests coming from RCSTs override the previous

\footnote{FR may be categorized as a CRA scheme. However, whereas CRA may block the assignment when there is not enough capacity available, the FR-based system is allowed, reducing the allocation during congested periods by making it lower than the request.}

\footnote{DVB-RCS v1.4.1 allows us to insert a SAC request both in an ATM burst after the preamble [1, Sec. 6.2.1.1] or in an MPEG-2 burst by using the MPEG-2 adaption field [1, Sec. 6.6.1.5].}
requests. Introducing a coefficient \(0 \leq \alpha \leq 1\) that weights the rate and volume components, request \(r\) is computed as

\[
r = \alpha R + (1 - \alpha)V
\]

where \(R\) is expressed in hundredths of slot (i.e., 184 B of payload) per superframe, and \(V\) is expressed in hundredths of slot. The RBDC and VBDC request categories can be obtained by setting \(\alpha = 1\) and \(\alpha = 0\), respectively. The results reported in Section IV show the benefits of a proper combination of volume and rate components, which, in our view, might justify the introduction of a new request category.

The NCC performs the bandwidth allocation by visiting in a round-robin fashion all active RCSTs and assigning to each of them a number of slots \(\alpha R = \beta r\) proportional to the request. An allocation cycle is the time required to complete a round of allocation. During an allocation cycle, our DAMA scheme guarantees at least one slot to each active RCST. An allocation cycle may be smaller than a superframe or larger in case a single superframe does not have enough slots to satisfy all requests. When the allocation cycle is smaller than the superframe, extra capacity is allocated to VR users, proportionally to their requests (free capacity assignment) to make the allocation cycle equal to the superframe. An RCST switches from idle to active when the NCC receives its bandwidth request for the first time. Conversely, it switches from active to idle upon reception of an explicit closure message at the NCC or when no data are received from the RCST for a certain time. The duration of such a time must be chosen in such a way as not to interfere with TCP timeouts.

Experimental studies [15] showed that the preceding allocation method requires a modest computational power and can be performed within an allocation cycle. When an RCST is going to access the DAMA system, it sends a request in contention mode by using the contention method provided by the standard [1, Sec. 6.6.1.4]. Section IV reports the TCP performance in terms of allocation cycle \(c\) expressed in superframe units.

### B. Contention-Based Scheme

In DVB-RCS, contention-based methods are used only to perform login operations and transmit signaling information. In this paper, we investigate the suitability of a contention-based scheme for user data delivery as well. We consider a variant of the contention-based access method slotted ALOHA (SA), which is called diversity SA (DSA) [18]. In this technique, a mobile RCST transmits multiple copies of the same packet to a pool of slots dedicated to contention-based access. The receiver retains only one of the copies correctly received. The method has been shown to provide better throughput up to a certain load level (in Erlangs) with respect to the basic SA. In our system, the NCC allocates the number of DSA slots necessary to maintain the throughput to a target threshold. Such an allocation corresponds to the average number of DSA slots successfully transmitted, which can easily be measured by the NCC, divided by the target throughput itself. In [18], throughput and collision probability formulas are provided in the case of Poisson traffic. Here, we follow a different approach by making the most generic assumption that a given number of RCSTs have a packet to send.

Let us consider random variable \(X(s)\), which represents the number of slots occupied when \(s\) packets are independently transmitted from a pool of \(n\) slots. The distribution of \(X(s)\) is given by

\[
\Pr\{X(s) = m\} = \frac{\binom{n}{m} S(s, m)}{n^s}
\]

\[
= \frac{1}{n^s} \binom{n}{m} \sum_{i=0}^{m} (-1)^i \binom{m}{i} (m-i)^s
\]

\[
m = 1, 2, \ldots, n
\]

where \((s, m)\) are the Stirling numbers of the second kind [19], which, by definition, represent the number of ways of partitioning a set of \(s\) elements into \(m\) nonempty sets, regardless of their order. The numerator in (2) is indeed the number of ways we can arrange \(s\) elements in a vector of \(n\) positions, so that exactly \(m\) positions are occupied, whereas the denominator expresses the total number of combinations. By using the generating function representation of Stirling numbers, we can derive the first- and second-order statistics of (2) as

\[
E\{X(s)/n\} = 1 - (1 - 1/n)^s
\]

\[
\text{var}\{X(s)/n\} = (1 - 2/n)^s - (1 - 1/n)^{2s} - 1/n \left[(1 - 2/n)^s - (1 - 1/n)^s\right].
\]

The second expression of (3) states that, when the number of slots \(n\) tends to infinity and \(s\) is made proportional to \(n\), the variance of the distribution of \(X(s)/n\) tends to zero. This means that, according to Chebyshev’s inequality [20], \(X(s)/n\) tends to a deterministic value that is equal to its mean as \(n\) tends to infinity.

In DSA, \(u\) RCSTs make \(k\) independent draws of slots in the set \(\{1, \ldots, n\}\) and experience a collision event when all the copies they sent collide with the packets sent by other RCSTs. To simplify the analysis, we assume that the same slot can be drawn more than once. Thus, the distribution of superframe occupancy seen by each RCST is given by (2), with \(s = (u-1)k\), and the corresponding collision probability is

\[
P_{\text{coll}}^{(k)} = \frac{\min(n, k(u-1))}{n^{k(u-1)}} \binom{n}{m} \sum_{i=0}^{m} (-1)^i \binom{m}{i} (m-i)^k (m-i)^k (\frac{m}{n})^k.
\]

However, we are interested in the system behavior when a large number of RCSTs access the system. Thus, we consider the first expression of (3), from which we are able to calculate a good asymptotic approximation of the collision probability

\[
P_{\text{coll}}^{(k)} \approx \left( E\left\{X(k(u-1))/n\right\} \right)^k
\]

\[
= (1 - (1 - 1/n)^{k(nG_e-1)})^k
\]
where $G_e = u/n$ is the equivalent system load, i.e., the traffic load that is not blocked by the channel and actually contributes in collisions. From this expression, the throughput can be immediately evaluated as

$$\lambda_n^{(k)} \simeq G_e \left(1 - \left(1 - (1 - 1/n)^{-k(nG_e-1)}\right)^k\right). \quad (6)$$

Finally, taking the limit of (5) and (6) for $n \to \infty$, we obtain expressions for the collision probability for the case of an infinite number of RCSTs

$$P_{coll\_lim}^{(k)} = (1 - e^{-kG_e})^k \quad (7)$$

and the relative throughput

$$\lambda_{lim}^{(k)} = G_e \left(1 - (1 - e^{-kG_e})^k\right). \quad (8)$$

Equations (7) and (8), for $k = 1$, are the classical results of SA with a system load of $G_e$ Erlangs. In Fig. 1, the throughput and collision probability, which are derived from (7) and (8), are reported as functions of the equivalent offered load for values of $k$ ranging from 1 to 4. The curve intersection points are for load values of 0.481, 0.281, and 0.199, for curves with $k = 1, 2, k = 2, 3, 4$, respectively.

III. SYSTEM DESIGN ISSUES

We choose a short TDMA superframe of 20 ms to track frequent variations of the mobile channel. Each slot is able to transport a single MAC packet, i.e., a 53-B asynchronous transfer mode (ATM) cell or a 188-B MPEG-2 packet. Since the bandwidth cost of DSA is directly dependent on the slot size, a smaller slot should be used when DSA is not used to carry large TCP segments. Even if small IP packets could be multiplexed in the same MPEG-2 slot, its size is excessive in transmitting individual acknowledgments (ACKs). Short-lived TCP connections generate few relatively spaced ACKs that tend to be encapsulated in separate slots. The inefficiency of using MPEG-2 can be obviated by employing an ATM stream or the mini-slot feature of the DVB standard [1, Sec. 6.2] with a mini-slot dimensioned to accommodate a full-sized TCP/IP header.

Reference [21] indicates that the general-purpose multiprotocol encapsulation (MPE) header adopted by the DVB standard carries redundant information when used in conjunction with an MPEG-2 transport stream and can be replaced by a 4-B header. This smaller header is, in fact, sufficient to convey per-packet information; the other MAC functionalities of MPE can be provided by extension headers or MAC tables. Assuming a MAC slot of $TS\_size$ bytes and one-half convolutional coding rate $EncRate$, the mini-slot size at the physical (PHY) layer is

$$\text{MiniSlotSize} = EncRate^{-1} \cdot TS\_size + PhyOH \quad (9)$$

where $PhyOH$ is the PHY layer overhead associated to each burst (typically some tens of bytes). Equation (9) tells us that the mini-slot size needed for the TCP/IP header in the worst case is about one half the size that would be required to carry an MPEG-2 (184-B payload) packet. However, simulation results reported in Section IV show that the actual bandwidth saving, by employing mini-slots with respect to MPEG-2 slots, is lower than the expected one half. This is because, in many cases, the system that uses MPEG-2 has the ability to pack multiple ACKs in the same slot.

In our reference system, RCSTs receive a terminal burst time plan (TBTP) in TDM mode at each superframe from the NCC. Aside from the TBTP, each RCST receives all other data coming from the hub and the fixed and mobile stations. Upon the correct reception of the TBTP, the RCST is able to evaluate the channel state over the forward link and infer, from this event, the probable state of the return link. Recent studies [22] demonstrated that, indeed, the link quality is strongly influenced by the terminal being able to see the satellite, that is when the terminal is along the line of sight (LOS) of the satellite; this is a symmetrical property of the channel. According to this property, when we do receive the TBTP in TDM mode, we can consider it to be along the LOS. The channel model assumes a negative exponential distribution of the time spent by the RCST in LOS state (and, thus, in good state). Due to the memoryless property of the negative exponential distribution, the sojourn time in a state has the same statistics as the residual sojourn time; thus, we assume that the channel remains in good state for a negative-exponential-distributed time, provided that the TBTP has been received. We base the segment loss estimation in Section III-A on this assumption.
Thus, assuming that the RCST is able to detect the LOS condition, it is able to predict the periods in which a TCP segment has high chances to be successfully received by the NCC. This avoids many segment losses due to return link failures and ultimately limits the performance degradation of TCP due to the channel intermittence. Clearly, every estimation is error prone. To compensate such estimation errors, a packet-level coding can be used. The coding type adopted in this paper is the Reed–Solomon erasure (RSE) [23], [24]. An RSE level coding can be used. The coding type adopted in this paper is derived from the three-state model described in [22], which applies to the Ku band. According to this model, each packet is correctly received out of the \( h \) sent packets, including redundant packets.

### A. Segment Loss Estimation on the Return Channel

The mobile channel is very hostile to TCP: Both forward and return channel impairments can greatly degrade TCP performance. However, the loss rate can sensibly be reduced by making the RCST aware of the return link channel conditions and enabling packet transmissions in nonblocked periods only. Supposing that one channel estimation is done at every superframe, for instance, at the TBTP reception time, a packet loss occurs when the RCST attempts to transmit after a transition between the nonblocked and blocked states before the next channel state estimation. Choosing a frequent sampling of the channel state (e.g., with a superframe duration of 20 ms), the channel state change during the transmission of a segment turns out to be rather unlikely.

1) **Channel Model**: The channel model assumed in this paper is derived from the three-state model described in [22], which applies to the Ku band. According to this model, each state is characterized by a different distribution of the received signal strength and average loss with respect to the LOS condition. We adopted a simplified version of the model by merging the two non-LOS states into a single blocked state and considering the LOS state as loss free. Such an approach is often used in the literature to describe the land mobile satellite channel (see, for instance, [25]). Table I reports the average duration of the nonblocked and blocked states, i.e., \( t_g \) and \( t_b \), respectively, together with the resulting steady-state probability of channel blocking for the two scenarios used in simulations, i.e., *highway* and *rural*.

<table>
<thead>
<tr>
<th>Environment</th>
<th>( h )</th>
<th>( b )</th>
<th>Blocking prob.</th>
</tr>
</thead>
<tbody>
<tr>
<td>highway</td>
<td>3.02</td>
<td>0.36</td>
<td>0.107</td>
</tr>
<tr>
<td>rural</td>
<td>2.03</td>
<td>0.55</td>
<td>0.215</td>
</tr>
</tbody>
</table>

The segment loss process can be assumed to be Bernoulli distributed for an \( n_a \) assignment and the RSE(\( h, i \)) code adopted. The estimation is made for each case. Considering that \( t_r \) is dimensioned for the maximum difference in the distance from the satellite and various stations (assumed to be 3 ms in the succeeding calculations) and taking a superframe length of 20 ms, we have \( p_{nl} < 1.5 t_r / t_g \). This allows us to neglect the events relative to packet losses in two consecutive superframes, because \((\max p_{nl})^2 < \max p_{nl}\).

Equations (11) and (12) can be used to select suitable redundancy levels for a certain allocation in FR mode, i.e., when the allocation is always the same. As an example, let us consider the case of \( n_a = 8 \) and RSE(13, 8) and express all the possible combinations in which a TCP segment of eight MPEG-2 packets can be fragmented into consecutive superframes. Indicating in parentheses the number of lost packets that can cause a segment loss (nl means that, even if all the packets are lost, no segment loss occurs), we have the following description of the 13-superframe period: \( 8/6, 5 + 3(nl), 8/6, 2 + 6/6, 7 + 1/7, 8/6, 4 + 4(nl), 8/6, 1 + 7/6, 6 + 2/8, 8/6, 3 + 5(nl), \) and \( 8/6 \).

Thus, the segment loss rate (SLR) is \( \text{SLR}(8/13, 8) = (8p_{nl} + (p_{nl} - 2p_{nl} + p_{nl})) / 13 \), which is equal to 0.0030 and 0.0044 in highway and rural environments, respectively. For the case of \( n_a = 8 \) and RSE(12, 8), we have the following...
three-superframe period: 8(5), 4 + 4(8l), and 8(l). Thus, we have \( SLR(8, (12, 8)) = 2(p_{al} + 3p_{al})/3 \), which is equal to 0.0029 and 0.0043 in highway and rural environments, respectively.

In summary, for the two cases shown, we deduce that, although the RSE(13, 8) code has a redundancy level that is higher than that of RSE(12, 8), the latter code produces a slightly lower SLR; furthermore, RSE(12, 8) is preferred between the two, because it also has lower redundancy.

B. Suitability Criterion

As already mentioned, DSA and DAMA bandwidth allocation algorithms are evaluated with respect to the bandwidth cost and the overall time to complete the connection. In DAMA, the bandwidth cost can be evaluated as the total number of slots allocated by the NCC for that connection, regardless of the number of slots actually used. In DSA, the bandwidth cost is given by the number of successfully used slots divided by the target throughput. The completion time sums up, other than the data transfer time, the duration of the connection opening and closing. We simulate the behavior of single TCP connections relative to single mobile users. Our fundamental assumption is that the reference connection is not able to alter the system-loading condition. This means, for instance, that the DAMA assignment cycle of \( c \) superframes remains unchanged upon the arrival of a new connection.

To select the best access method and related parameters, among the large set of cases investigated, we adopted the following strategy: First, we discarded all cases that exhibited both worse bandwidth cost and completion time. The remaining cases are all suitable for the user, and a choice between them is not feasible without knowing his/her character. Thus, we introduce parameter \( w(0 \leq w \leq 1) \), which expresses the user’s willingness to pay; the higher the \( w \), the more the user is able to pay (in bandwidth, i.e., in money) to speed up the data transfer. To evaluate the highest suitability, we define fitness index \( \zeta_r(0 \leq \zeta_r \leq 1) \), which is able to select the best condition for each case. We have

\[
\zeta_r = \zeta / \zeta_{\text{max}}, \quad \zeta = \frac{1}{w \gamma_{\text{max}} + (1 - w) \gamma_{\text{min}}}
\]  

where \( \gamma_{\text{max}} \) and \( \gamma_{\text{min}} \) are the maximum throughput and the minimum bandwidth cost, which are obtainable in ideal conditions, respectively. Thus, the best condition is given by \( \zeta_r = 1 \). Therefore, if the user wants to maximize his/her savings (\( w = 0 \)), the maximum \( \zeta_r \) is reached for \( \gamma = \gamma_{\text{min}} \), whereas if the user does not bother to spend (\( w = 1 \)), the best selection is given by \( \tau = \tau_{\text{max}} \).

IV. SIMULATION RESULT

The analysis has been carried out by means of simulations based on ns-2 [26]. The simulated setup is represented in Fig. 2: A vehicular user exchanges data with a remote Internet host through a gateway (hub station). Since we are interested in the performance achieved by the two bandwidth-sharing techniques over the return link only, we assume that sufficient bandwidth is available to the hub station and, hence, in the forward link direction. We made simulations by considering the mobile user acting as both server and client for both short- and long-lived connections. For each case, such as the server or client role, the number of segments to transfer or long-lived connections, the environment type, and the system load (i.e., the length of the assignment cycle), we selected the most suitable transfer parameters. These parameters consist of the channel access scheme (DAMA or DSA) and the coding redundancy to apply to data packets. In the case of DAMA, the number of requested slots with the FR option and the value of \( \alpha \) with the VR option are selected as well. These parameters are employed to build lookup tables that allow an RCST that knows its willingness factor \( w \) to directly choose the best transmission modality for each case. As far as the environment type is concerned, we suppose using a Global Positioning System (GPS) to distinguish among four possibilities (highway, rural, suburban, and urban).

Simulations were carried out with TCP selective ACK options, using the default parameters, except for the maximum segment size, which has been set to 1416 B, corresponding to eight MPEG-2 packets. The minimum roundtrip time of the TCP connection is about 600 ms, which accounts for the Geostationary Earth Orbit satellite latency and 100-ms terrestrial network delay. The simulation results reported in all the succeeding figures are obtained by repeating each simulation run for a number of times that is sufficient to obtain a confidence

Fig. 2. Mobile satellite scenario.
interval of ±2% at 95% level. The fitness indexes reported in all figures refer to three different values of $w$ (0.1, 0.5, and 0.9).

### A. Mobile User as Server

In simulative experiments performed with a mobile host acting as server, we observed that the main cause of performance limitation of the TCP throughput is the loss of ACKs over the forward link that cause additional retransmission timeout (RTO) expirations other than the segment losses over the return link. The reason is that the mobile RCST is able to avoid most of packet losses by transmitting only when the reception of a TBTP is correct. A simple mechanism for mitigating this problem at the MAC layer consists of sending more copies of the same ACK, spaced by a certain time interval, which we refer to as the duplicated and delayed ACK (DDA) technique.

Fig. 3 shows an example of our methodology comparing optimal cases (envelope curve) with nonoptimal cases. Subsequent graphs display only the most convenient MAC configurations. The various cases are differently labeled, depending on the user request (either FR or VR). In particular, the $F(n_a, h)$ label denotes an FR equal to $n_a$ slots per superframe and an RSE block size of $h$, i.e., RSE($h$, 8), whereas the $V(\alpha, h)$ label denotes a VR computed as in (1) with an RSE($h$, 8) packet-level coding. We note that the DSA choice is not reported in these graphs in that it shows to be inefficient for long-lived connections.

Figs. 4 and 5 show the throughput versus the relative cost for bulk data transfers (10 000 segments), together with the fitness index, which is expressed by (13). The connection cost of the MPEG-2 packets, which appears in the definition of the fitness index, has been referred to as the cost that would be necessary in ideal conditions (= 80 000 MPEG-2 slots).

As we expected, lower costs are achieved when VR is employed. Indeed, VR reflects the instantaneous need of bandwidth and is able to better track rate variations of TCP. On the other hand, the high latency between requests and assignments slows down the growth of the TCP congestion window and, consequently, reduces the TCP throughput. As far as FR is concerned, this analysis confirms the results carried out in [12] and [27], regarding the optimization of TCP throughput with respect to the applied level of redundancy for a fixed bottleneck rate (i.e., $n_a/c$ slot/superframe). When the redundancy is increased, a lower SLR and, hence, a higher TCP efficiency are achieved, but at the same time, the available bandwidth is reduced. The compromise between the two opposite effects leads to the optimal condition. These graphs show that the required level of redundancy is lower for higher loads, i.e., for $c > 1$, and is not even necessary in most cases. Indeed, when the available bandwidth is small, the loss rate enforced by the channel is as such to allow TCP to saturate the link capacity. Further increase in the packet-level protection would correct more channel errors, but it would not increase the TCP throughput.

Fig. 4(a) also shows the throughput enhancement achieved by using DDA with three ACK copies and a 300-ms interval between them. The spacing interval is clearly a function of the number of copies sent and should be chosen as a compromise between the channel coherence time and the maximum time to avoid TCP RTO expiration. Since similar improvements have been obtained in all the cases investigated, all other results are presented as enabling DDA.
The results for short-lived connections are shown in Fig. 6(a) and (b) for $c = 1$, in highway and rural environments, respectively, whereas Fig. 7(a) and (b) shows highway cases for $c = 2$ and $c = 4$, respectively. For the sake of comparison, Fig. 6(a) also shows DSA cases for two and four segments. (Points with more than four segments would lie outside the figure.) DSA($t, h$) denotes a DSA access scheme with a throughput of $t\%$ and RSE($h$, $8$) coding. These results provide evidence that DSA is not suitable for short-lived connections in both highway and rural environments. Moreover, different from the long-lived-connection case, for short data transfers, the lowest bandwidth cost is reached by an FR option $[F(1, 9)]$, which actually allocates one slot every $c$ superframes.

**B. Mobile User as Client**

According to the results presented so far, the DSA technique is not convenient when the mobile host acts as server. On the contrary, when data flow from the remote host to the mobile RCST, using a contention-based technique permits a considerable reduction in the bandwidth cost. In this case, the mobile RCST has to send only TCP ACKs, which are cumulative; thus, a small packet loss rate due to collisions can be tolerated as long as new ACK messages are able to update information lost in previous ACK messages. Since the mobile client produces light traffic, there are no substantial differences between VR and FR options; the minimum bandwidth guarantee allocation is sufficient to satisfy the bandwidth requirements of the mobile user. The advantages of DSA with respect to DAMA are shown in Fig. 8(a) and (b), which depicts the cost of MPEG-2 slots versus the completion time for two-, four-, eight-, and 16-segment transfers in highway and rural environments, respectively. From these graphs, the superior performance in adopting mini-slots with respect to MPEG-2 slots is evident: The former exhibits a sensible reduction in cost to the detriment of a marginal increase in the completion time. In fact, the DAMA scheme achieves higher costs than DSA in almost all the cases considered. Based on the fitness index, DAMA is the best option only for $\{w = 0.9, c = 4, 8, \text{and } 16\}$ segments in the highway environment and for $\{w = 0.9, c = 4, \text{and } 16\}$ segments in the rural environment. Finally, Table II reports the suitability of DSA with mini-slots versus DAMA for long-lived connections. In many circumstances, the DSA scheme is still preferred over DAMA, particularly when DAMA competes with 20% DSA target throughput and in the rural environment. However, for
long-lived connections, the cost of bandwidth in DAMA is comparable with DSA in that the ACKs’ traffic tends to saturate the minimum bandwidth assignment, and multiple ACKs can be concatenated in the same MPEG-2 packet. The combination of these two occurrences makes the assignment on demand the most suitable in some cases.

V. CONCLUSION AND FUTURE WORK

We have analyzed the behavior of two different access methods, based on DSA and DAMA, respectively, in providing the basis for data transfer services to and from mobile DVB-RCS users. The two methods have been compared, in terms of bandwidth cost and completion time, for cases in which the mobile user acts as both server and client. Simulation results have shown that using DSA is not convenient for mobile users when they work as servers. Thus, the need to adopt DSA MPEG-2 slots to carry data traffic does not arise, but mini-slots can be used, which allows us to save bandwidth. In fact, in the client case, by adopting mini-slots, we found an almost total convenience in using DSA over DAMA for short transfers and, in most cases, for long transfers.

| TABLE II |
| SUITABILITY OF THE ACCESS METHOD FOR LONG-LIVED CONNECTIONS |

<table>
<thead>
<tr>
<th>w</th>
<th>c</th>
<th>highway 10%</th>
<th>highway 20%</th>
<th>rural 10%</th>
<th>rural 20%</th>
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</thead>
<tbody>
<tr>
<td>0.1</td>
<td>1.0</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
</tr>
<tr>
<td></td>
<td>2.0</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
</tr>
<tr>
<td></td>
<td>4.0</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
</tr>
<tr>
<td>0.5</td>
<td>1.0</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
</tr>
<tr>
<td></td>
<td>2.0</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
</tr>
<tr>
<td></td>
<td>4.0</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
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<tr>
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<td>2.0</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
</tr>
<tr>
<td></td>
<td>4.0</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
<td>DAMA</td>
</tr>
</tbody>
</table>

Our results can be stored in lookup tables and employed in the following way: The user is supposed to know the type of environment he/she is crossing by making use of a GPS device. When the user has a file to send or receive, the file length is known; thus, considering the system-loading conditions (parameter $c$ is advertised by the NCC), the environment, and his/her own willingness factor, the user is able to select the optimal condition to operate the transfer.

To improve the TCP throughput in the server case, we proposed RSE coding on data packets and DDAs from the
REFERENCES


Nedo Celandroni received the Dr. Ing. degree in electronic engineering from the University of Pisa, Pisa, Italy, in 1973.

Since 1976, he has been a Researcher with the CNR Institute (which is now the Information Science and Technology Institute), Italian National Research Council (CNR), Pisa. He worked on the realization of the flight dynamic system of the SIRIO Satellite Project. Since 1979, he has been involved in the field of digital satellite communications. He participated in several projects in this field: the Satellite Transmission Experiment Linking Laboratories (STELLA II), the FIFO Ordered Demand Assignment (FODA), FODA/Information Bit Energy Adaptive (IBEA), Progetto Finalizzato Telecomunicazioni, and experiments on the satellites Olympus and ItalSat. He is currently involved in the SatNEx European Network of Excellence and some national projects. He is a coauthor of more than 100 papers published in international journals and conference proceedings. He is the holder of two patents for the design of the FODA and FODA/IBEA systems (in 1989 and 1996, respectively). His research interests include rain fade countermeasure systems; data quality estimation; VSAT systems; GEO, MEO, and LEO satellites for mobile telephony and multimedia systems; wireless networks; TCP congestion management; and TCP over wireless channels.

Dr. Celandroni is a reviewer for many international journals and conferences.

Raffaele Secchi received the Laurea degree in telecommunication engineering (summa cum laude) and the Ph.D. degree from the University of Pisa, Pisa, Italy, in 2002 and 2006, respectively. His Ph.D. dissertation was on traffic modeling and control in high-speed networks.

In 2006, he joined the Information Science and Technology Institute, Italian National Research Council (CNR), Pisa, as a Postdoctoral Fellow. He is currently a Research Fellow with the University of Aberdeen, Aberdeen, U.K.
Suitability of DAMA and Contention-Based Satellite Access Schemes for TCP Traffic in Mobile DVB-RCS

Nedo Celandroni and Raffaello Secchi

Abstract—The problem of optimizing access and bandwidth sharing among Transmission Control Protocol (TCP) connections in the mobile digital video broadcasting return channel via satellite (DVB-RCS) is tackled in this paper. After sketching the general system architecture, we explicitly deal with the dynamic assignment of bandwidth to TCP connections on the return link, which is accomplished by a network control center (NCC) placed onboard the satellite. Mobile users access the satellite in multifrequency time-division multiple access (MF-TDMA), whereas they receive data from the NCC in time-division multiplexing (TDM). Two different techniques, based on deterministic and random access, are compared in terms of bandwidth usage and average completion time per connection, when the mobile user acts as both server and client. In the server case, to increase the TCP throughput, both packet-level forward error correction (FEC) on data sent by mobile users and a duplicated and delayed acknowledgment technique for TCP acknowledgment traffic from the NCC to the mobile users are applied. An analysis of the packet losses and a simulation campaign of file transfers by employing a realistic channel model has been carried out. The results of the analysis show the convenience of adopting a technique, in addition to the optimal data redundancy in different cases, such as the server or client role of users, their willingness to pay, the file size, and the environment type.

Index Terms—Demand assignment multiple access (DAMA), digital video broadcasting return channel via satellite (DVB-RCS), diversity slotted ALOHA (DSA), satellite mobile environment, Transmission Control Protocol (TCP) throughput.

I. INTRODUCTION

THE convergence of video, audio, and Internet access with mobile services in a satellite network, which is currently referred to as quadruple play, is considered to be one of the key factors for the future success of digital video broadcasting (DVB) technology. Future releases of the current DVB return channel via satellite (DVB-RCS) standard [1] are indeed expected to support both Internet mobile access and interactive services across a satellite network. The development of mobile services in DVB-RCS is ongoing, and a substantial research effort is required to enable the medium access control (MAC) layer to mitigate negative effects induced by mobility. The goal of this paper is to propose improvements to the standard by presenting a comprehensive analysis of the suitability of various bandwidth allocation schemes to elastic traffic1 services in a vehicular environment. More specifically, we study the behavior of Transmission Control Protocol (TCP) flows originating from or directed to individual mobile sources (e.g., cars with radio equipment). Different from large mobile traffic terminals (e.g., trains, ships, and aircraft), which may convey traffic from several individual sources, these terminals tend to produce small and bursty flows and require careful management.

The design of a satellite network architecture that is suitable for TCP connections can begin by looking at various proposals presented in the literature (see, for instance, [2]) to mitigate impairments that are mainly due to the large latency. Solutions are generally categorized as end to end and link layer. End-to-end approaches focus on modifications of the transport layer to make TCP aware of the presence of the wireless link; such schemes may require the active participation of the network [3], only TCP sender modifications [4], or specific feedback in the TCP/Internet Protocol (IP) header [5], [6]. On the other hand, link-layer schemes employ forward error correction, automatic repeat request (ARQ), or a combination of both to make the link reliable. These solutions are typically embedded in a more general framework, which is sometimes referred to as the performance-enhancing proxy [7], [8], which may take advantage of dedicated entities, such as spoofing or splitting agents, traffic classification and conditioning, optimized transport protocols (e.g., [9]), and reservation schemes.

We propose a new satellite network architecture for vehicular users oriented to the link-layer paradigm. Starting from the discussions in [10] and [11], which are related to the TCP performance over fixed satellite links adopting DVB-RCS demand assignment multiple access (DAMA) schemes, this paper aims at presenting performance evaluations in terms of bandwidth cost and transfer time for mobile links. The original contribution of this paper consists of evaluating the suitability of the access scheme for TCP over a channel with impairments caused by mobility. Other studies consider the

1TCP flows are also called elastic in that the congestion-avoidance algorithm of TCP causes an elastic variation of the throughput, in reaction to network state variations.
TCP performance optimization over a discontinuous channel (such as [12]) but without taking into account the dynamics of the allocation scheme.

The rest of this paper is organized as follows: In Section II, we provide a description of the system architecture and the details of the analyzed algorithms. System design issues are discussed in Section III, and simulation results are presented in Section IV. Finally, in Section V, we conclude this paper, summarizing main contributions and outlining future work.

II. SYSTEM ARCHITECTURE

The multimedia mobile bandwidth allocation (MUMOBAL) architecture is the outcome of a joint research action undertaken in the framework of the SatNEx II European Network of Excellence [13]. The objective of MUMOBAL is to respond to design issues related to the user mobility in a satellite network. In this section, we summarize the main elements of this architecture.

The RCS terminal (RCST) acts as an access point for users in a multiservice transmission environment. The system is centrally controlled by a network control center (NCC), which may be placed in an Earth station or onboard the satellite (onboard processing). The results presented in this paper refer to a system with the NCC onboard the satellite. However, the architecture is also consistent, although with poorer performance, when the NCC is placed on ground.

MUMOBAL considers various classes of users: fixed, collective mobile (e.g., trains, ships, and aircraft, which collect traffic from several sources), and vehicular users. The last class poses the most challenging constraints in that, while each fixed and collective mobile user offers a moderately variable traffic (being the multiplexing of several sessions), vehicular users are characterized by connections that are highly variable in duration and bandwidth requirements. In fact, we refer to particular vehicles, such as rescue and civil protection vehicles, in addition to the cars of business and government people; the number of such vehicles may be large over a wide coverage area, but each of them uses the network with a very small duty cycle. In addition, the radio channel quality of mobile users is highly variable and subjected to intermittent link failures due to multipath and shadowing; this implies the adoption of mechanisms for compensating channel blockage periods.

MUMOBAL operates a class-based dynamic bandwidth allocation that guarantees a degree of independence among traffic classes by isolating each class of traffic in a different superframe portion. At each superframe, the boundaries between classes may be repositioned, taking into account the traffic load, channel conditions (e.g., users’ environmental positions), and quality-of-service requirements.

The allocation for each RCST is done in the time-frequency space, i.e., in multifrequency time-division multiple access (TDMA), following either a collision-free or a contention-based access scheme. The two alternatives dynamically share the available bandwidth as detailed here; moreover, the collision-free bandwidth portion is subdivided into two parts, which are devoted to carrying streaming (multimedia) and elastic traffic. More specifically, for each of the three bandwidth partitions, we adopt the following: 1) an independent bandwidth allocation scheme for multimedia traffic [14]; 2) a DAMA scheme (Section II-A) with dynamic or constant request of bandwidth [15], [16]; and 3) a contention-based access (Section II-B). After the NCC has allocated the amount of bandwidth required for partitions 1 and 3, the rest of the bandwidth is assigned in best-effort mode to the second partition.

The task of selecting the most suitable partition between the second and third partitions to serve a single TCP connection is demanded to the RCST. The choice is performed by considering the benefits for TCP in terms of bandwidth cost and transfer delay, as detailed in Section III-B. One of the factors that determine this choice is the length of the file to transfer; since this parameter has to be known at the MAC layer, cross-layer actions are performed [17].

Our approach is based on an infinite source asymptotic, i.e., we suppose that a single user is not able to significantly affect the load of the whole system. This hypothesis is justified by the fact that the population of vehicular users is generally vast.

A. DAMA Scheme

Since our objective is to define an allocation scheme that is suitable for TCP connections over a mobile satellite link, we carried out a set of simulations considering the request categories proposed in DVB-RCS [1, Sec. 6.8], i.e., continuous rate assignment (CRA), rate-based dynamic capacity (RBDC), and volume-based dynamic capacity (VBDC). The results of these experiments indicated that none of these categories dominates over the others in all cases of practical interest. A different candidate solution was indeed selected, depending on the environment type, connection length, and loading conditions. Moreover, the indications provided by different performance metrics (i.e., bandwidth consumption or connection completion time) led us to different choices. Thus, we design a flexible DAMA scheme that generalizes the DVB-RCS request categories and exhibits higher performance in many cases. Our approach consists of determining over a wide range of cases the optimal DAMA parameters, storing them in lookup tables, and using them at the run time.

In our system, the RCST has two options for requesting bandwidth: 1) fixed request (FR), which consists of asking for a fixed amount of bandwidth, and 2) variable request (VR). The VR option that we consider has some similarities with the First-in–first-out Ordered Demand Assignment (FODA) system described in [16]. When VR is enabled, the RCST computes the request, taking into account instantaneous queue size $V$ and incoming traffic rate $R$, which is the amount of data entering the RCST queue in each superframe. Requests for bandwidth are sent in each burst using the satellite access control (SAC) field, and new requests coming from RCSTs override the previous FR request.

2FR may be categorized as a CRA scheme. However, whereas CRA may block the assignment when there is not enough capacity available, the FR-based system is allowed, reducing the allocation during congested periods by making it lower than the request.

3DVB-RCS v1.4.1 allows us to insert a SAC request both in an ATM burst after the preamble [1, Sec. 6.2.1.1] or in an MPEG-2 burst by using the MPEG-2 adaption field [1, Sec. 6.6.1.5].
requests. Introducing a coefficient \(0 < \alpha \leq 1\) that weights the rate and volume components, request \(r\) is computed as
\[
r = \alpha R + (1 - \alpha)V
\]
where \(R\) is expressed in hundredths of slot (i.e., 184 B of payload) per superframe, and \(V\) is expressed in hundredths of slot. The RBDC and VBDC request categories can be obtained by setting \(\alpha = 1\) and \(\alpha = 0\), respectively. The results reported in Section IV show the benefits of a proper combination of volume and rate components, which, in our view, might justify the introduction of a new request category.

The NCC performs the bandwidth allocation by visiting in a round-robin fashion all active RCSTs and assigning to each of them a number of slots \(a = \beta r\) proportional to the request. An allocation cycle is the time required to complete a round of allocation. During an allocation cycle, our DAMA scheme guarantees at least one slot to each active RCST. An allocation cycle may be smaller than a superframe or larger in case a single superframe does not have enough slots to satisfy all requests. When the allocation cycle is smaller than the superframe, extra capacity is allocated to VR users, proportionally to their requests (free capacity assignment) to make the allocation cycle equal to the superframe. An RCST switches from idle to active when the NCC receives its bandwidth request for the first time. Conversely, it switches from active to idle upon reception of an explicit closure message at the NCC or when no data are received from the RCST for a certain time. The duration of such a time must be chosen in such a way as not to interfere with TCP timeouts.

Experimental studies [15] showed that the preceding allocation method requires a modest computational power and can be performed within an allocation cycle. When an RCST is going to access the DAMA system, it sends a request in contention mode by using the contention method provided by the standard [1, Sec. 6.6.1.4]. Section IV reports the TCP performance in terms of allocation cycle \(c\) expressed in superframe units.

### B. Contention-Based Scheme

In DVB-RCS, contention-based methods are used only to perform login operations and transmit signaling information. In this paper, we investigate the suitability of a contention-based scheme for user data delivery as well. We consider a variant of the contention-based access method slotted ALOHA (SA), which is called diversity SA (DSA) [18]. In this technique, a mobile RCST transmits multiple copies of the same packet to a pool of slots dedicated to contention-based access. The receiver retains only one of the copies correctly received. The method has been shown to provide better throughput up to a certain load level (in Erlangs) with respect to the basic SA.

In our system, the NCC allocates the number of DSA slots necessary to maintain the throughput to a target threshold. Such an allocation corresponds to the average number of DSA slots successfully transmitted, which can easily be measured by the NCC, divided by the target throughput itself. In [18], throughput and collision probability formulas are provided in the case of Poisson traffic. Here, we follow a different approach by making the most generic assumption that a given number of RCSTs have a packet to send.

Let us consider random variable \(X(s)\), which represents the number of slots occupied when \(s\) packets are independently transmitted from a pool of \(n\) slots. The distribution of \(X(s)\) is given by
\[
Pr\{X(s) = m\} = \frac{m!}{n^m} S(s, m) = \frac{1}{n^m} \sum_{i=0}^{m} (-1)^i \binom{m}{i} (m - i)^s
\]
\[
m = 1, 2, \ldots, n
\]
where \(S(s, m)\) are the Stirling numbers of the second kind [19], which, by definition, represent the number of ways of partitioning a set of \(s\) elements into \(m\) nonempty sets, regardless of their order. The numerator in (2) is indeed the number of ways we can arrange \(s\) elements in a vector of \(n\) positions, so that exactly \(m\) positions are occupied, whereas the denominator expresses the total number of combinations. By using the generating function representation of Stirling numbers, we can derive the first- and second-order statistics of (2) as
\[
E\{X(s)/n\} = 1 - (1 - 1/n)^s
\]
\[
\text{var}\{X(s)/n\} = (1 - 2/n)^s - (1 - 1/n)^{2s} - 1/\left[(1 - 2/n)^s - (1 - 1/n)^s\right].
\]

The second expression of (3) states that, when the number of slots \(n\) tends to infinity and \(s\) is made proportional to \(n\), the variance of the distribution of \(X(s)/n\) tends to zero. This means that, according to Chebyshev’s inequality [20], \(X(s)/n\) tends to a deterministic value that is equal to its mean as \(n\) tends to infinity.

In DSA, \(u\) RCSTs make \(k\) independent draws of slots in the set \(\{1, \ldots, n\}\) and experience a collision event when all the copies they sent collide with the packets sent by other RCSTs. To simplify the analysis, we assume that the same slot can be drawn more than once. Thus, the distribution of superframe occupancy seen by each RCST is given by (2), with \(s = (u - 1)k\), and the corresponding collision probability is
\[
P_{\text{coll}}^{(k)} = \frac{1}{n^{k(u-1)}} \sum_{m=1}^{\min(n,k(u-1))} \frac{m!}{m^m} \sum_{i=0}^{m} (-1)^i \binom{m}{i} (m - i)^{k(u-1)} \left(\frac{m}{n}\right)^{k(u-1)}.
\]
However, we are interested in the system behavior when a large number of RCSTs access the system. Thus, we consider the first expression of (3), from which we are able to calculate a good asymptotic approximation of the collision probability
\[
P_{\text{coll}}^{(k)} \approx \left(E\left\{\frac{X^{(k(u-1))}}{n}\right\}\right)^k = \left(1 - (1 - 1/n)^{kG_u-1}\right)^k
\]
\[
(5)
\]
where \(G_e = u/n\) is the equivalent system load, i.e., the traffic load that is not blocked by the channel and actually contributes in collisions. From this expression, the throughput can be immediately evaluated as

\[
\lambda_n^{(k)} \cong G_e \left(1 - \left(1 - \frac{1}{n}\right)^{-k(nG_e-1)}\right)^k.
\]  

(6)

Finally, taking the limit of (5) and (6) for \(n \to \infty\), we obtain expressions for the collision probability for the case of an infinite number of RCSTs

\[
P_{coll \_lim}^{(k)} = (1 - e^{-kG_e})^k
\]

(7)

and the relative throughput

\[
\lambda_{lim}^{(k)} = G_e \left(1 - e^{-kG_e}\right)^k.
\]

(8)

Equations (7) and (8), for \(k = 1\), are the classical results of SA with a system load of \(G_e\) Erlangs. In Fig. 1, the throughput and collision probability, which are derived from (7) and (8), are reported as functions of the equivalent offered load for values of \(k\) ranging from 1 to 4. The curve intersection points are for load values of 0.481, 0.281, and 0.199, for curves with \(k = 1, 2, 3, 4\), respectively.

III. SYSTEM DESIGN ISSUES

We choose a short TDMA superframe of 20 ms to track frequent variations of the mobile channel. Each slot is able to transport a single MAC packet, i.e., a 53-B asynchronous transfer mode (ATM) cell or a 188-B MPEG-2 packet. Since the bandwidth cost of DSA is directly dependent on the slot size, a smaller slot should be used when DSA is not used to carry large TCP segments. Even if small IP packets could be multiplexed in the same MPEG-2 slot, its size is excessive in transmitting individual acknowledgments (ACKs). Short-lived TCP connections generate few relatively spaced ACKs that tend to be encapsulated in separate slots. The inefficiency of using MPEG-2 can be obviated by employing an ATM stream or the mini-slot feature of the DVB standard [1, Sec. 6.2] with a mini-slot dimensioned to accommodate a full-sized TCP/IP header.

Reference [21] indicates that the general-purpose multiprotocol encapsulation (MPE) header adopted by the DVB standard carries redundant information when used in conjunction with an MPEG-2 transport stream and can be replaced by a 4-B header. This smaller header is, in fact, sufficient to convey per-packet information; the other MAC functionalities of MPE can be provided by extension headers or MAC tables. Assuming a MAC slot of \(TS\_size\) bytes and one-half convolutional coding rate \(EncRate\), the mini-slot size at the physical (PHY) layer is

\[
MiniSlotSize = \frac{EncRate \cdot TS\_size}{2} + PhyOH
\]

(9)

where \(PhyOH\) is the PHY layer overhead associated to each burst (typically some tens of bytes). Equation (9) tells us that the mini-slot size needed for the TCP/IP header in the worst case is about one half the size that would be required to carry an MPEG-2 (184-B payload) packet. However, simulation results reported in Section IV show that the actual bandwidth saving, by employing mini-slots with respect to MPEG-2 slots, is lower than the expected one half. This is because, in many cases, the system that uses MPEG-2 has the ability to pack multiple ACKs in the same slot.

In our reference system, RCSTs receive a terminal burst time plan (TBTP) in TDM mode at each superframe from the NCC. Aside from the TBTP, each RCST receives all other data coming from the hub and the fixed and mobile stations. Upon the correct reception of the TBTP, the RCST is able to evaluate the channel state over the forward link and infer, from this event, the probable state of the return link. Recent studies [22] demonstrated that, indeed, the link quality is strongly influenced by the terminal being able to see the satellite, that is when the terminal is along the line of sight (LOS) of the satellite; this is a symmetrical property of the channel. According to this property, when we do receive the TBTP in TDM mode, we can consider it to be along the LOS. The channel model assumes a negative exponential distribution of the time spent by the RCST in LOS state (and, thus, in good state). Due to the memoryless property of the negative exponential distribution, the sojourn time in a state has the same statistics as the residual sojourn time; thus, we assume that the channel remains in good state for a negative-exponential-distributed time, provided that the TBTP has been received. We base the segment loss estimation in Section III-A on this assumption.
Thus, assuming that the RCST is able to detect the LOS condition, it is able to predict the periods in which a TCP segment has high chances to be successfully received by the NCC. This avoids many segment losses due to return link failures and ultimately limits the performance degradation of TCP due to the channel intermittence. Clearly, every estimation is error prone. To compensate such estimation errors, a packet-level coding can be used. The coding type adopted in this paper is the Reed–Solomon erasure (RSE) [23], [24]. An RSE $(h, i)$ code allows decoding $i$ information packets when $i$ or more packets are correctly received out of the $h$ sent packets, including redundant packets.

### A. Segment Loss Estimation on the Return Channel

The mobile channel is very hostile to TCP: Both forward and return channel impairments can greatly degrade TCP performance. However, the loss rate can sensibly be reduced by making the RCST aware of the return link channel conditions and enabling packet transmissions in nonblocked periods only. Supposing that one channel estimation is done at every superframe, for instance, at the TBTP reception time, a packet loss occurs when the RCST attempts to transmit after a transition between the nonblocked and blocked states before the next channel state estimation. Choosing a frequent sampling of the channel state (e.g., with a superframe duration of 20 ms), the channel state change during the transmission of a segment turns out to be rather unlikely.

1) **Channel Model:** The channel model assumed in this paper is derived from the three-state model described in [22], which applies to the Ku band. According to this model, each state is characterized by a different distribution of the received signal strength and average loss with respect to the LOS condition. We adopted a simplified version of the model by merging the two non-LOS states in a single blocked state and considering the LOS state as loss free. Such an approach is often used in the literature to describe the land mobile satellite channel (see, for instance, [25]). Table I reports the average duration of the nonblocked and blocked states, i.e., $t_g$ and $t_b$, respectively, together with the resulting steady-state probability of channel blocking for the two scenarios used in simulations, i.e., highway and rural.

<table>
<thead>
<tr>
<th>Environment</th>
<th>$t_g$</th>
<th>$t_b$</th>
<th>Blocking prob.</th>
</tr>
</thead>
<tbody>
<tr>
<td>highway</td>
<td>3.02</td>
<td>0.36</td>
<td>0.107</td>
</tr>
<tr>
<td>rural</td>
<td>2.03</td>
<td>0.55</td>
<td>0.215</td>
</tr>
</tbody>
</table>

2) **Loss Analysis:** Assuming that the RCST estimates a nonblocked channel condition at the TBTP reception $t_0$, the probability that the next channel switching from nonblocked to blocked occurs before $t$ is

$$p_a(t) = 1 - e^{-\frac{t-t_0}{t_g}} \approx \frac{t-t_0}{t_g}, \quad (t-t_0) \ll t_g$$

where $t_g$ is the average nonblocked state duration. All the data sent after instant $t$ are lost. Thus, the probability of losing all the packets sent in the assigned slots is

$$p_{al} = \left( t_p + n_b \frac{t_f}{n_{sf}} \right) / t_g$$

where $t_p$ is the time interval between the channel estimation (at TBTP reception) and the beginning of the transmission superframe, $t_f$ is the superframe time, $n_b (1 \leq n_b \leq n_{sf} - n_a + 1)$ is the first assigned slot in the superframe, $n_{sf}$ is the total number of slots in a superframe, and $n_a$ is the number of assigned slots in the superframe. When all the assigned slots are used to send packets, the probability of losing a number of packets equal to $i (i \leq n_a - 1, i \neq 0)$ is

$$p_{al} = \frac{t_f}{t_g n_{sf}}$$

which is independent of $i$, whereas the probability of not losing any packets is

$$p_{al}(n_b) = 1 - (n_b + n_a - 1)p_{al} - t_p / t_g.$$

Due to the nature of the assignment algorithm, we can assume that $n_b$ is uniformly distributed inside its variation interval with a probability mass function

$$p_{nb} = 1 / (n_{sf} - n_a + 1).$$

Thus, by averaging over $n_b$, we get

\[
\begin{align*}
\bar{p}_{al} &= \sum_{k=1}^{n_{sf} - n_a + 1} p_{al}(k) = \frac{(2 + n_{sf} - n_a)t_f}{2n_{sf}t_g} + \frac{t_p}{t_g} \\
\bar{p}_{nl} &= \sum_{k=1}^{n_{sf} - n_a + 1} p_{nl}(k) = 1 - \frac{t_p}{t_g} - \frac{t_f}{2t_g} - \frac{n_a t_f}{2n_{sf}t_g},
\end{align*}
\]

The segment loss process can be assumed to be Bernoulli distributed for an $n_a$ assignment and the RSE$(h, i)$ code adopted. The estimation is made for each case. Considering that $t_p$ is dimensioned for the maximum difference in the distance from the satellite and various stations (assumed to be 3 ms in the succeeding calculations) and taking a superframe length of 20 ms, we have $p_{al} < 1.5t_f / t_g$. This allows us to neglect the events relative to packet losses in two consecutive superframes, because $(\max p_{al})^2 \ll \max p_{al}$.

Equations (11) and (12) can be used to select suitable redundancy levels for a certain allocation in FR mode, i.e., when the allocation is always the same. As an example, let us consider the case of $n_a = 8$ and RSE$(13, 8)$ and express all the possible combinations in which a TCP segment of eight MPEG-2 packets can be fragmented into consecutive superframes. Indicating in parentheses the number of lost packets that can cause a segment loss (nl means that, even if all the packets are lost, no segment loss occurs), we have the following description of the 13-superframe period: $8(6), 5 + 3(nl), 8(6), 2 + 6(6), 7 + 1(7), 8(6), 4 + 4(nl), 8(6), 1 + 7(6), 6 + 2(8), 8(6), 3 + 5(nl)$, and $8(6)$.

Thus, the segment loss rate (SLR) is $\text{SLR}(8, 13, 8) = (8\bar{p}_{al}) + (\bar{p}_{al} - 2p_{al} + \bar{p}_{al}) / 13$, which is equal to 0.0030 and 0.0044 in highway and rural environments, respectively. For the case of $n_a = 8$ and RSE$(12, 8)$, we have the following
three-superframe period: 8(5), 4 + 4(11), and 8(5). Thus, we have \(\text{SLR}(8, (12, 8)) = 2(p_{al} + 3p_{al}) / 3\), which is equal to 0.0029 and 0.0043 in highway and rural environments, respectively.

In summary, for the two cases shown, we deduce that, although the RSE(13, 8) code has a redundancy level that is higher than that of RSE(12, 8), the latter code produces a slightly lower SLR; furthermore, RSE(12, 8) is preferred between the two, because it also has lower redundancy.

### B. Suitability Criterion

As already mentioned, DSA and DAMA bandwidth allocation algorithms are evaluated with respect to the bandwidth cost and the overall time to complete the connection. In DAMA, the bandwidth cost can be evaluated as the total number of slots allocated by the NCC for that connection, regardless of the number of slots actually used. In DSA, the bandwidth cost is given by the number of successfully used slots divided by the target throughput. The completion time sums up, other than the data transfer time, the duration of the connection opening and closing. We simulate the behavior of single TCP connections relative to single mobile users. Our fundamental assumption is that the reference connection is not able to alter the system-loading condition. This means, for instance, that the DAMA assignment cycle of \(c\) superframes remains unchanged upon the arrival of a new connection.

To select the best access method and related parameters, among the large set of cases investigated, we adopted the following strategy: First, we discarded all cases that exhibited both worse bandwidth cost and completion time. The remaining cases are all suitable for the user, and a choice between them is not feasible without knowing his/her character. Thus, we introduce parameter \(w(0 \leq w \leq 1)\), which expresses the user’s willingness to pay; the higher the \(w\), the more the user is able to pay (in bandwidth, i.e., in money) to speed up the data transfer. To evaluate the highest suitability, we define fitness index \(\zeta_r(0 \leq \zeta_r \leq 1)\), which is able to select the best condition for each case. We have

\[
\zeta_r = \zeta / \zeta_{\text{max}}, \quad \zeta = \frac{1}{w \tau_{\text{max}} \gamma_{\text{max}} + (1 - w) \gamma_{\text{min}}} \tag{13}
\]

where \(\tau_{\text{max}}\) and \(\gamma_{\text{min}}\) are the maximum throughput and the minimum bandwidth cost, which are obtainable in ideal conditions, respectively. Thus, the best condition is given by \(\zeta_r = 1\). Therefore, if the user wants to maximize his/her savings \((w = 0)\), the maximum \(\zeta_r\) is reached for \(\gamma = \gamma_{\text{min}}\), whereas if the user does not bother to spend \((w = 1)\), the best selection is given by \(\tau = \tau_{\text{max}}\).

### IV. SIMULATION RESULT

The analysis has been carried out by means of simulations based on ns-2 [26]. The simulated setup is represented in Fig. 2: A vehicular user exchanges data with a remote Internet host through a gateway (hub station). Since we are interested in the performance achieved by the two bandwidth-sharing techniques over the return link only, we assume that sufficient bandwidth is available to the hub station and, hence, in the forward link direction. We made simulations by considering the mobile user acting as both server and client for both short- and long-lived connections. For each case, such as the server or client role, the number of segments to transfer or long-lived connections, the environment type, and the system load (i.e., the length of the assignment cycle), we selected the most suitable transfer parameters. These parameters consist of the channel access scheme (DAMA or DSA) and the coding redundancy to apply to data packets. In the case of DAMA, the number of requested slots with the FR option and the value of \(\alpha\) with the VR option are selected as well. These parameters are employed to build lookup tables that allow an RCST that knows its willingness factor \(w\) to directly choose the best transmission modality for each case. As far as the environment type is concerned, we suppose using a Global Positioning System (GPS) to distinguish among four possibilities (highway, rural, suburban, and urban).

Simulations were carried out with TCP selective ACK options, using the default parameters, except for the maximum segment size, which has been set to 1416 B, corresponding to eight MPEG-2 packets. The minimum roundtrip time of the TCP connection is about 600 ms, which accounts for the Geostationary Earth Orbit satellite latency and 100-ms terrestrial network delay. The simulation results reported in all the succeeding figures are obtained by repeating each simulation run for a number of times that is sufficient to obtain a confidence
interval of $\pm 2\%$ at 95% level. The fitness indexes reported in all figures refer to three different values of $w$ (0.1, 0.5, and 0.9).

A. Mobile User as Server

In simulative experiments performed with a mobile host acting as server, we observed that the main cause of performance limitation of the TCP throughput is the loss of ACKs over the forward link that cause additional retransmission timeout (RTO) expirations other than the segment losses over the return link. The reason is that the mobile RCST is able to avoid most of packet losses by transmitting only when the reception of a TBTP is correct. A simple mechanism for mitigating this problem at the MAC layer consists of sending more copies of the same ACK, spaced by a certain time interval, which we refer to as the duplicated and delayed ACK (DDA) technique.

Fig. 3 shows an example of our methodology comparing optimal cases (envelope curve) with nonoptimal cases. Subsequent graphs display only the most convenient MAC configurations. The various cases are differently labeled, depending on the user request (either FR or VR). In particular, the $F(n_a, h)$ label denotes an FR equal to $n_a$ slots per superframe and an RSE block size of $h$, i.e., $RSE(h, 8)$, whereas the $V(\alpha, h)$ label denotes a VR computed as in (1) with an $RSE(h, 8)$ packet-level coding. We note that the DSA choice is not reported in these graphs in that it shows to be inefficient for long-lived connections.

Figs. 4 and 5 show the throughput versus the relative cost for bulk data transfers (10 000 segments), together with the fitness index, which is expressed by (13). The connection cost of the MPEG-2 packets, which appears in the definition of the fitness index, has been referred to as the cost that would be necessary in ideal conditions ($= 80 000$ MPEG-2 slots).

As we expected, lower costs are achieved when VR is employed. Indeed, VR reflects the instantaneous need of bandwidth and is able to better track rate variations of TCP. On the other hand, the high latency between requests and assignments slows down the growth of the TCP congestion window and, consequently, reduces the TCP throughput. As far as FR is concerned, this analysis confirms the results carried out in [12] and [27], regarding the optimization of TCP throughput with respect to the applied level of redundancy for a fixed bottleneck rate (i.e., $n_a/c$ slot/superframe). When the redundancy is increased, a lower SLR and, hence, a higher TCP efficiency are achieved, but at the same time, the available bandwidth is reduced. The compromise between the two opposite effects leads to the optimal condition. These graphs show that the required level of redundancy is lower for higher loads, i.e., for $c > 1$, and is not even necessary in most cases. Indeed, when the available bandwidth is small, the loss rate enforced by the channel is such to allow TCP to saturate the link capacity. Further increase in the packet-level protection would correct more channel errors, but it would not increase the TCP throughput.

Fig. 4(a) also shows the throughput enhancement achieved by using DDA with three ACK copies and a 300-ms interval between them. The spacing interval is clearly a function of the number of copies sent and should be chosen as a compromise between the channel coherence time and the maximum time to avoid TCP RTO expiration. Since similar improvements have been obtained in all the cases investigated, all other results are presented as enabling DDA.
The results for short-lived connections are shown in Fig. 6(a) and (b) for $c = 1$, in highway and rural environments, respectively, whereas Fig. 7(a) and (b) shows highway cases for $c = 2$ and $c = 4$, respectively. For the sake of comparison, Fig. 6(a) also shows DSA cases for two and four segments. (Points with more than four segments would lie outside the figure.) DSA($t, h$) denotes a DSA access scheme with a throughput of $t$% and RSE($h$, 8) coding. These results provide evidence that DSA is not suitable for short-lived connections in both highway and rural environments. Moreover, different from the long-lived-connection case, for short data transfers, the lowest bandwidth cost is reached by an FR option $[F(1, 9)]$, which actually allocates one slot every $c$ superframes.

B. Mobile User as Client

According to the results presented so far, the DSA technique is not convenient when the mobile host acts as server. On the contrary, when data flow from the remote host to the mobile RCST, using a contention-based technique permits a considerable reduction in the bandwidth cost. In this case, the mobile RCST has to send only TCP ACKs, which are cumulative; thus, a small packet loss rate due to collisions can be tolerated as long as new ACK messages are able to update information lost in previous ACK messages. Since the mobile client produces light traffic, there are no substantial differences between VR and FR options; the minimum bandwidth guarantee allocation is sufficient to satisfy the bandwidth requirements of the mobile user. The advantages of DSA with respect to DAMA are shown in Fig. 8(a) and (b), which depicts the cost of MPEG-2 slots versus the completion time for two-, four-, eight-, and 16-segment transfers in highway and rural environments, respectively. From these graphs, the superior performance in adopting mini-slots with respect to MPEG-2 slots is evident: The former exhibits a sensible reduction in cost to the detriment of a marginal increase in the completion time. In fact, the DAMA scheme achieves higher costs than DSA in almost all the cases considered. Based on the fitness index, DAMA is the best option only for $\{w = 0.9, c = 4, 8, \text{ and } 16 \text{ segments}\}$ in the highway environment and for $\{w = 0.9, c = 4, \text{ and } 16 \text{ segments}\}$ in the rural environment. Finally, Table II reports the suitability of DSA with mini-slots versus DAMA for long-lived connections. In many circumstances, the DSA scheme is still preferred over DAMA, particularly when DAMA competes with 20% DSA target throughput and in the rural environment. However, for
long-lived connections, the cost of bandwidth in DAMA is comparable with DSA in that the ACKs’ traffic tends to saturate the minimum bandwidth assignment, and multiple ACKs can be concatenated in the same MPEG-2 packet. The combination of these two occurrences makes the assignment on demand the most suitable in some cases.

V. CONCLUSION AND FUTURE WORK

We have analyzed the behavior of two different access methods, based on DSA and DAMA, respectively, in providing the basis for data transfer services to and from mobile DVB-RCS users. The two methods have been compared, in terms of bandwidth cost and completion time, for cases in which the mobile user acts as both server and client. Simulation results have shown that using DSA is not convenient for mobile users when they work as servers. Thus, the need to adopt DSA MPEG-2 slots to carry data traffic does not arise, but mini-slots can be used, which allows us to save bandwidth. In fact, in the client case, by adopting mini-slots, we found an almost total convenience in using DSA over DAMA for short transfers and, in most cases, for long transfers.

Our results can be stored in lookup tables and employed in the following way: The user is supposed to know the type of environment he/she is crossing by making use of a GPS device. When the user has a file to send or receive, the file length is known; thus, considering the system-loading conditions (parameter $c$ is advertised by the NCC), the environment, and his/her own willingness factor, the user is able to select the optimal condition to operate the transfer.

To improve the TCP throughput in the server case, we proposed RSE coding on data packets and DDAs from the
NCC to the mobile user. This study will be completed with the adoption of techniques to improve the throughput in the client case as well. We think that the adoption of RSE coding on data in the client case can give only a marginal increment in the throughput, and we propose to also experiment on ARQ and duplicated and delayed data from the NCC to the mobile user.

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Nedo Celandroni received the Dr. Ing. degree in electronic engineering from the University of Pisa, Pisa, Italy, in 1973. Since 1976, he has been a Researcher with the CNR Institute (which is now the Information Science and Technology Institute), Italian National Research Council (CNR), Pisa. He worked on the realization of the flight dynamic system of the SIRIO Satellite Project. Since 1979, he has been involved in the field of digital satellite communications. He participated in several projects in this field: the

Raffaele Secchi received the Laurea degree in telecommunication engineering (summa cum laude) and the Ph.D. degree from the University of Pisa, Pisa, Italy, in 2002 and 2006, respectively. His Ph.D. dissertation was on traffic modeling and control in high-speed networks. In 2006, he joined the Information Science and Technology Institute, Italian National Research Council (CNR), Pisa, as a Postdoctoral Fellow. He is currently a Research Fellow with the University of Aberdeen, Aberdeen, U.K.