Dynamic Resource Allocation for an IP-based Communications Network Supporting Space Exploration

Hui Zeng[*], Mihael Hadjitheodosiou.[†] and John Baras[‡]
Institute of Systems Research, University of Maryland, College Park, MD, 20742, USA

We address issues related to efficiency, fairness, end-to-end delay minimization and Quality-of-Service (QoS) framework in order to enable a flexible access and dynamic mission operation capability in the next generation NASA space-to-ground IP-based communication infrastructure. In our scenario, the downlink channel of the NASA Tracking and Data Relay Satellite System is shared by a number of spacecraft, which we model as streams with different priority levels going through a common queue and a router. Both the current and future potential architectures for this relay system are addressed. We formulate an optimization problem for long-term static bandwidth allocation and present its solution to serve as initial bandwidth allocation algorithm and periodical regulation approach. Along with it, a hybrid Time Division Multiple Access (TDMA)-based protocol is proposed and then an assignment problem for the short-term dynamic optimal timeslot scheduling is studied. By using simulation, the performance of a suitable Medium Access Control (MAC) protocol with timeslot scheduling is analyzed and compared with that of the existing static fixed-assignment scheme.

I. Introduction

WE address issues related to efficiency, fairness, end-to-end delay minimization and Quality-of-Service (QoS) framework in order to enable a flexible access and dynamic mission operation capability in the next generation NASA space-to-ground IP-based communication infrastructure. An end-to-end communication architecture for future space missions, using the Internet Protocol (IP) as the “glue” that connects everything together is clearly feasible. IP provides a basic standardized mechanism for end-to-end communication between applications across a network. This will lead to an environment where most spacecraft could have an IP router on board and instruments on the spacecraft can become addressable nodes, connected with an on-board LAN. To achieve this, the underlying medium access control (MAC) protocol with QoS framework plays a very important role and therefore needs to be suitably designed.

In this paper we focus on the allocation of bandwidth in a space relay network that supports several scientific spacecraft with a number of different streams on-board sharing a broadband satellite channel to send traffic to the ground. Our system model includes a number of mobile spacecraft (MS) in Lower Earth Orbit (LEO), a Geo-synchronous (GEO) relay satellite, and the ground network consisting of several ground stations (GS). The downlink channel of the relay satellite is shared by these spacecraft, which we model as streams with different priority levels going through a common queue and a router. The data will be delivered to the ground station through this relay, and then arrive at the end user, which could be either at a NASA facility or at the edge of a public/private network.

To provide dynamic access with fairness and efficiency, a suitable hybrid-mode Time Division Multiple Access (TDMA) protocol along with frame-wise packet scheduling for bursty data flows was proposed for this network in Ref.1. It is shown that a carefully designed time-varying bandwidth allocation based on the instant or statistical traffic from all users/flows performs better in terms of throughput and end-to-end delay. However, only short-term (time varying) bandwidth allocation may cause instability and will have difficulties in providing QoS guarantees and managing the long-term (average) behavior of all the users/flows. Therefore, we propose a two-level bandwidth allocation in our implemented TDMA scheme.

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[*] Research Assistant, CSHCN, University of Maryland.

[†] Assistant Research Scientist, CSHCN, University of Maryland, AIAA Member.

[‡] Professor, CSHCN, University of Maryland, AIAA Member.

For a more well-coupled framework with per user/flow average bandwidth management, we derive our long-term bandwidth/rate allocation problem from the model discussed by Kelly in Ref.2, and incorporate ideas from other work.[3,4,5] In addition, for instantaneous bandwidth management, we incorporate ideas from some recent work[6,7,8] to formulate the short-term timeslot assignment problem and find the solution for optimal timeslot scheduling.

The paper will be organized as follows: Section II introduces the network architecture. Section III briefly describes the hybrid-mode access protocol. Section IV proposes the architecture of our two-level bandwidth allocation. Then Sections V and VI present the formulations and solutions of long-term and short-term bandwidth allocation, respectively. Section VII shows the simulation results and gives some discussions. Finally Section VIII summarizes our conclusions.

II. Network Architecture

A. Scenario Design

Most ESE missions either use the NASA Tracking and Data Relay Satellite System (TDRSS)[9] for relaying data to the ground or can communicate directly with certain NASA (or other) ground terminals. TDRSS consists of 7 satellites in geostationary orbit around the globe that relay data from satellites in Low Earth Orbit (LEO) and Medium Earth Orbit (MEO) to ground facilities at the White Sands Complex in New Mexico, and Guam. The satellites have the capability to forward and return data in the S and Ku bands at speeds of up to 300 Mbps in the Ku band. These systems were developed in the 1970's and have been heavily used over the past two decades. A new generation of TDRS satellites (called TDRS-H, TDRS-I, and TDRS-J) has recently started to augment the older system and provides additional capacity for users. This new generation TDRS satellite has the additional capability to relay data in Ka-band at up to 300 Mbps without modifications to the ground stations, and up to 800 Mbps with ground station modifications. A new tunable, wideband, high frequency service offered by the 15-foot antennas provides for the capability of these high data rates.

In the architecture shown in Fig. 1, large numbers of spacecraft share the downlink channel of TDRS or other relay satellite system to the ground station, which can provide single access for high data rate channel (up to hundreds Mbps) per TDRS satellite. TDRSS has a Single Access (SA) and Multiple Access (MA) Capability using Spatial Diversity. The total end-to-end architecture is known as the Space Network (SN). Further details about the TDRSS operation and SN can be found in the Space Network Users Guide[10], and it is beyond the scope of this paper.

It is important to note that although we start by considering a GEO relay satellite similar in architecture to TDRSS we are not focusing on the details of the current TDRSS nor are we trying to modify or improve on that design. We are looking into the concept of optimizing future relay systems which could be the next generation of NASA owned relays or other systems that share NASA but also commercial traffic.

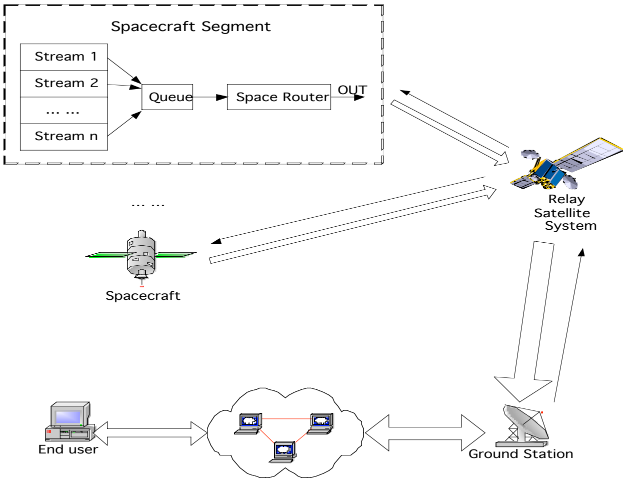

Figure 1. Architecture of multi-access for the downlink channel of relay satellite

For simplification, we consider only one GEO relay satellite to avoid the issues of handover and routing. Those issues along with reliability will be addressed in later work.

The ground station receives the scientific data from all the spacecraft via the TDRS link but also acts as Network Control Center (NCC) performing the bandwidth allocation under certain QoS guarantees by collecting the reservation/dynamic access and statistical information from the data transmission link. We mainly consider LEO spacecraft in orbits common to Earth Observation Science (EOS) missions, and only consider the zone where the spacecraft are in the coverage of TDRSS. Since the orbits of mission spacecraft are known, we know the exact time they "join" (enter the coverage zone) and "depart" (leave the coverage zone) the zone.

The uplink is from the ground station to spacecraft through the TDRS satellite, operated in TDM broadcasting mode. It is used by the ground station to notify all users the bandwidth allocation for downlink. The downlink is the data link, from the users (spacecraft) to ground station through the TDRS satellite. This link has much larger bandwidth, and we need to focus on how the spacecraft can share this dynamically. In the current mode of operation this is done using a static-TDMA process using a priori reservations. There is some Multiple Access capability in the new TDRS but this cannot really support dynamic, on-demand operation.

B. Traffic Sources

We still adopt the multi-state multi-mode traffic source model proposed in Ref.1. The traffic model works as follows: In a specific state, at the beginning of each small period, the active data rate is randomly chosen from two modes by their active probabilities. Then the generator keeps generating source packets in this active data rate during the following period. All these parameters can be set or changed.

We model all the instruments on-board spacecraft in the above way. Several traffic sources are using a common queue with priority queuing. The priority levels are assigned according to their data rate, and the instrument with the highest data rate has the highest priority. Of course the priority levels also can be assigned by other rules, for example, the importance of data. The source traffic generated from one spacecraft is relatively bursty and unpredictable. One modification for the multi-mode multi-state traffic models is that the periods could be exponentially distributed instead of being fixed.

We also need to consider other types of traffic model. For example, since some on-board instruments only generate traffic at constant rate and for part of the time, we could use a Constant Bit Rate (CBR) traffic generator model for some instruments. We are using our own multi-state-multi-mode traffic generators to emulate this instrument traffic that include a Poisson-distributed traffic generator and CBR traffic generator.

III. Hybrid-Mode MAC Protocol

As shown in Fig. 2, we are using a simple hybrid-mode MAC protocol in our scenario. All active spacecraft are using the common downlink channel to send packets to the ground station (NCC). There are request sub-channels in this downlink channel and feedback in control link for bandwidth reservation.

The whole channel is divided into a number of identical sub-channels. A fixed number of these sub-channels are allocated for the static slots and the rest are using for reservation-based slots. This hybrid-mode can guarantee the minimal bandwidth for each user (spacecraft) while assigning reservation channels dynamically for optimum performance.

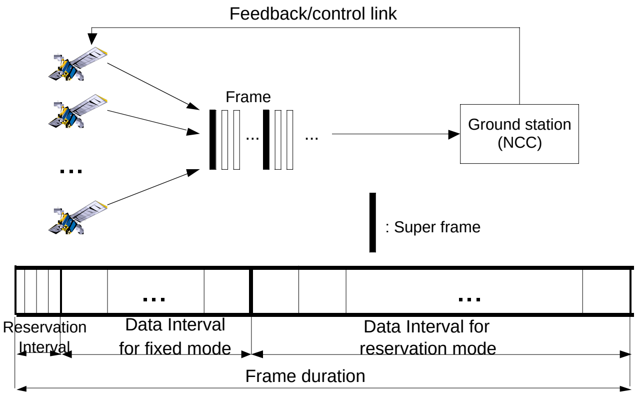

Figure 2. Hybrid-Mode MAC Protocol

A. Triple Request

Every traffic source will be provided with a guaranteed QoS in terms of its triple request: LR (Lower Resource), TR (Targeted Resource) and UR (Upper Resource). Intuitively, the LRs and URs are the minimum and maximum bandwidth assignments to fulfill the data delivery for every connection according to the different requirements. And the TRs are the expected bandwidths to “better” satisfy the QoS requirements of the connections in some sense based on the estimation of the traffic behaviors. Here the “better” means: below the TR, the traffic source is very eager to get more bandwidth assignment if the price is payable; while some way beyond the TR, additional bandwidth assignment is not that in need any more considering the price. In other words, the TR is a measure to describe the start point of the turning zone.

B. Access Control Algorithm

Reservation mini-slots are used for access requests from new users. In the demands, according to the different requirements, every traffic source will provide its triple request: LR, RR and UR, and its priority level and weight when trying to get access to the broadband satellite network. Certainly, for some types of traffic sources, three parameters might be redundant and therefore could be combined. This framework is similar to the studies presented by Hung[3] and BoD protocol[4], but has a different parameter model.

The access control algorithm will be performed as following: the new user will be admitted only if the sum of the LRs of all active users is less than or equal to the total bandwidth of the broadband channel. After admitted to the network, every stream will be assigned new triple: LR, PR (Projected Resource) and UR. The stream will be allocated its PR as a sum of LR and a best-effort share from the rest available bandwidth. The best-effort share will be assigned according to fairness and efficiency by solving an optimization problem. We will present the detailed problem formulation and results in the following Sections.

C. Bandwidth Request Track

Estimation of "future" traffic demands from users. One-step forward linear estimation bases on n-step previous and current information.

Since, the RTD is approximately 480ms or more it cannot be ignored. We chose RTD as the maximum round trip delay during the whole operating time. Typically the frame duration T_f is set to equal RTD. In our case, however, to support the data rate as high as 200Mbps, it is impossible to let T_f =RTD. So, we can set $M \cdot T_f$ = RTD, where M is a given integer. Then, when a user sends a request or makes a transmission attempt in a specific frame, say the frame k, it will know the feedback from uplink before the same slot in the frame (k+M).

The ground station (NCC) broadcasts the feedback packets to spacecraft immediately after processing the incoming frame. And for every M frames, it sends a "super packet" including system time along with the feedback information. When a "super packet" arrives, the spacecraft will perform time synchronization, and we call the following frame as "super frame". And notice that frames and slots are using guarding time to avoid overlapping.

Suppose a frame has N data slots. N_1 slots are used for static slots, while N_2 data slots are used for reservation-based slots. And $N_1 + N_2 = N$. Since the data rates and capacities of all spacecraft are predictable, these N_1 static data slots are allocated to them based on the expected traffic loads and the minimal bandwidth requirements. One reservation request is piggybacked in the first assigned data slot for each user per frame. A reservation request contains the source (spacecraft) ID (MAC address) and the current size of on-board queue. The NCC keeps storing two statistics: the size of on-board queue in the previous adjacent request, and that in the lately super frame. The ground station collects these requests from all the spacecraft and then determines the allocation of the rest N_2 data slots basing on these statistics. NCC assigns the weights to every spacecraft by some rules (for example, nominal data rate, or importance), and then calculating the products of weight and queue size, determine the portions of reserved slots assigned to each spacecraft. The obvious benefit is that the ground station considers the behavior of traffic not only in a very short range (T_f) but also in a relatively long range (RTD). This helps the ground station to make a more fair and optimal decision and decrease the opportunity to waste the bandwidth, therefore approach the bandwidth-efficiency. Our hybrid-mode protocol performs bandwidth optimization on a frame-by-frame basis although the collected information is M frames "out-of-date".

Another option is to send reservation requests in the "super frame" only and operate as a static TDMA protocol based on the previous granted bandwidth allocation. This option can decrease the computing complexity and lower the overhead, but now the optimization must be done on a multi-frame basis.

IV. Two-Level Bandwidth Allocation

To provide per-user/per-flow QoS guarantees for different users with fairness consideration, an efficient bandwidth allocation process along with a good admission control algorithm is necessary. Because of the significant propagation delay, the satellite communications network needs a more adaptive bandwidth allocation and a more systematic admission control. To efficiently use the available channel, it is shown that well-design time-varying bandwidth allocation based on the instant or statistical traffic of all users/flows has better performance in terms of throughput and end-to-end delay[1]. However, only short-term (instantaneous) bandwidth allocation may cause instability and will have difficulties in providing QoS guarantees and managing the long-term (average) behavior of all the users/flows. Besides, the instantaneous and average behavior managements need to be well-coupled with each other. Therefore, we propose a two-level bandwidth allocation.

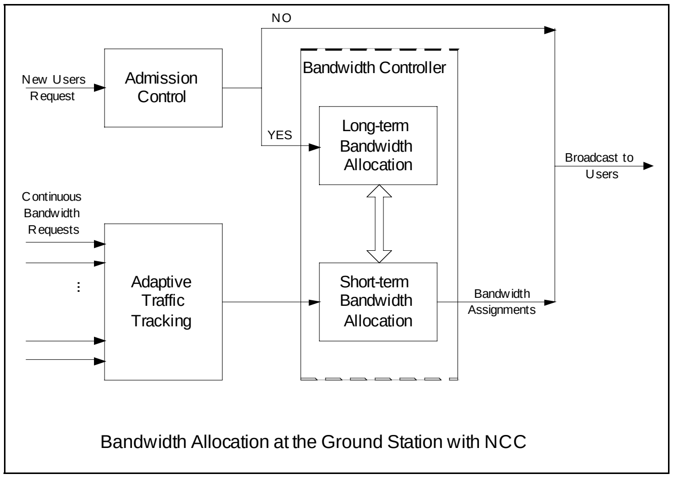

Figure 3. Two-level Bandwidth Allocation at the Ground Station

As shown in the Fig. 3, the two-level

bandwidth allocation is performed by the scheduler at the ground station with Network Control Center (NCC). To access the channel, a new user or new flow first sends a request to the scheduler. After performing the Access Control algorithm mentioned in Section III-B, the scheduler will broadcast its decision to the users. If the user/flow is accepted, a static initial bandwidth allocation is made by the long-term bandwidth allocator. Then the initial allocations will be delivered to the short-term bandwidth allocator as control parameters for the next-level scheduling. Under some other conditions, the long-term bandwidth allocation might be performed and updated to the next level. In the short-term scheduler, according to the continuous bandwidth requests from users/flows, the time-varying bandwidth allocation (or slots assignments) will be obtained and broadcasted. This is another reason why we use the triple request model, which gives us more control for the bandwidth management.

V. Long-Term Bandwidth Allocation

As we mentioned earlier, to provide certain per-steam (and per-user) QoS guarantee, for access request from new stream (or new user), the central scheduler will perform the admission control algorithm to ensure the sum of the contracted bandwidths (rates) of all the users/flows is less than or equal to the targeted bandwidth of the broadband channel. The stream will be allocated its PR as a sum of LR and a best-effort share from the available bandwidth, according to fairness and efficiency by solving an optimization problem.

This long-term optimal bandwidth allocation will be conducted not only when a new user or new connection is requesting the admission to the broadband satellite communications network ("joining'), but also when an active user turns to be inactive or changes to another relay satellite ("leaving"). Also, it could be performed in a fixed or event-driven schedule. Here, long-term is referring to a relatively long time range compared with the dynamic bandwidth allocation, which is performed per frame or in multi-frame basis.

Our long-term optimization problem is derived from the Kelly's model[2], so we will first briefly introduce the original framework in the following subsection, and then propose our formulation and solution thereafter.

A. Kelly's Model

Consider a network with a set L of resources or links and a set I of users (or flows). Let B_l denote the finite capacity of link l Є L. Each user has a route r, which is a non-empty subset of L. Define a 0-1 matrix A, where $A_{l,r} = 1$ if l Є r, and $A_{l,r} = 0$ otherwise. Suppose that if a rate (bandwidth) x_i is allocated to the user then $U_i(x_i)$ represents its utility. Here, the utility $U_i(x_i)$ is an increasing, strictly concave and continuously differentiable function of x_i over the range $x_i \geq 0$ (i.e., a elastic traffic). Also, utilities are additive so that the aggregate utility of rate allocation $x = (x_i, i$Є$I)$ is $\sum_{i\in I} U_i(x_i)$. Let $B = (B_l, l$ Є $L)$ and $U = (U_i(\cdot), i$Є$I)$ and the rate-control optimization problem is formulated as follows:

SYSTEM (U, A, B):

$$\begin{aligned} &\max \sum_{i\in I} U_i(x_i) \\ &subj.\ to\ Ax \leq B, x \geq 0. \end{aligned} \tag{1}$$

From the convexity of the feasible region for x and the strict concavity of the logarithm function, it follows that the solution of (1) is unique and proportionally fair. In Ref.5, it is shown that the optimum solution associated with the logarithm utility function is also a Nash Bargaining Solution. We are interested in the proportional fairness or its variations because of its simpleness and popularity, although there are also other fairness criterions.

B. Utility Functions Discussion

In Kelly's model, there are no definitions for lower resource (LR) guarantee upper resource (UR) bound or targeted resource (TR). To formulate the optimum problem in our case, we need to modify the utility function and the constraints to incorporate all these parameters. For simpleness, we denote c = LR, b = UR, a = TR.

To investigate the alternative utility functions, consider a small feasible perturbation $x = (x_i, i$Є$I) \longrightarrow x + \delta x = (x_i + \delta x_i, i$Є$I)$ in (1). The optimal point x has the property that for any perturbation,

$$\sum_{i\in I} U_i'(x_i) \cdot \ddot{a}x_i \ \leq \ 0 .$$

To incorporate the minimum bandwidth c and also maintain the proportional fairness, we modify the utility function to log $(x-c)$, for which the optimal point x has the property $\sum_{i\in I} \ddot{a}x_i \Big/ (x_i - c_i) \leq 0$ for any perturbation.

Consider the TR (or notation a), we want the optimal solution associated with the modified utility function has the following property: below its TR, the traffic source is very likely to get more bandwidth assignment if the price is payable; while some way beyond the TR, more bandwidth assignment is not that in need any more considering the price. In other words, the TR is a measure to describe the start point of the turning zone for the tradeoff between the resource and the price. Considering the variations of logarithm functions, we list some candidates in Table 1, where k, $k > 0$, is the desired attenuation parameter for the designated source. Note that the utility functions can be written as logarithmic functions. For example, the first one is alternatively $\log[(x-c)\cdot\exp(-(x-a)/k)]$.

Recall that the utility function $U(x)$ is an increasing, strictly concave and continuously differentiable function of x over the range $x \in (c, b]$ for the elastic traffic. So, to keep the strict concavity, the second derivative of $U(x)$ need to be negative, which is clearly correct except one point ($x = a$) for the 3rd candidate utility function. To keep the utility function to be increasing, the first derivative of $U(x)$ is nonnegative over the range $x \in (c, b]$, which leads to the regions specified in Table I respectively.

$U(x)$	$U'(x)$	$U''(x)$	Region
$\log(x-c)$ $-(x-a)/k$	$1/(x-c)$ $-1/k$	$-1/(x-c)^2$	$k \geq b-c$
$\log(x-c)$ $-(x-a)^2/k$	$1/(x-c)$ $-2(x-a)/k$	$-1/(x-c)^2$ $-2/k$	$k \geq$ $2(b-a)(b-c)$
$\log(x-c)$ $-\lvert x-a\rvert/k$	$1/(x-c)$ $-1/k$	$-1/(x-c)^2$ except $x=a$.	$k \geq b-c$

Table 1. Comparison of Utility Functions

For detailed comparison with the above utility functions, we draw all of them and the truncated logarithm function in Fig. 4, with $c = 1$, $b = 8$, $a = 6$. As shown in Fig. 4, before the point $x = a$, the 2nd line is the most steep one among all the utility functions; while after the point $x = a$, it is the most flat one except the 3rd one. However, the 3rd line is the least steep one before the point $x = a$. Therefore we take the 2nd one, which is associated with the utility function

$$\log[(x-c)\cdot\exp(-(x-a)/k)] = \log(x-c)-(x-a)/k .$$

Also note that after the point $x = a$, the 2nd line and the 4th line coincide with each other.

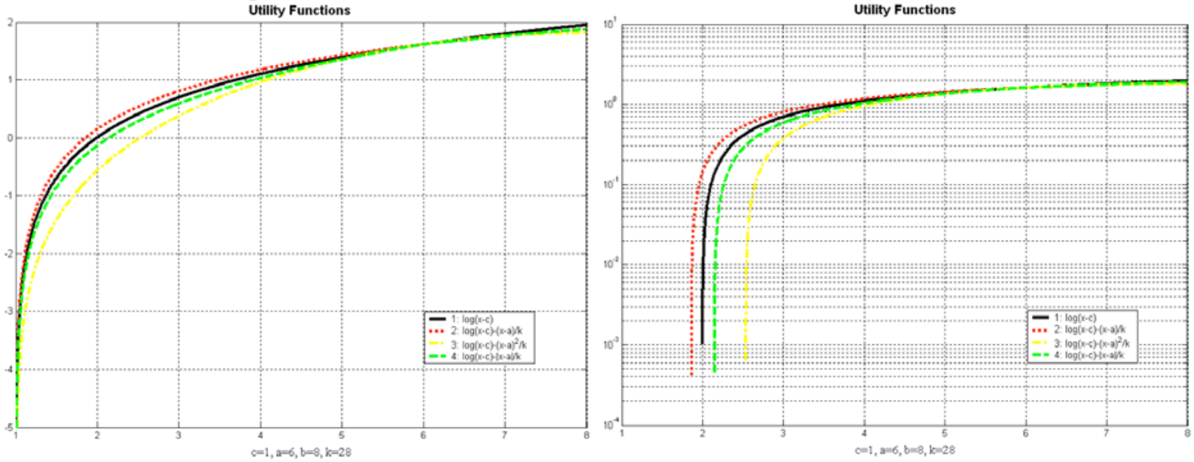

Figure 4. Comparison of the Utility Functions

Left: linear scales, Right: semi-logarithmic scales.

1st : log(x-c), 2nd: log(x-c)-(x-a)/k, 3rd: log(x-c)-(x-a)^2/k, 4th: log(x-c)-|x-a|/k.

C. Problem Formulation

In this section, we will use the chosen utility function in the previous section to formulate the long-term static centralized bandwidth allocation problem. Consider N users or flows and L links or nodes in a network compete with each other for use of the broadband channel. In the following discussion, we will generally use users to refer to users or flows. Each user is associated with a minimum bandwidth LR to be guaranteed by the network, maximum bandwidth UR and targeted bandwidth TR. According to the framework presented in the previous Sections, the feasible rate vector space X is decided by the finite capacity B and the triple parameters of the users, and could be defined as:

$$X = \left\{x: x \in R^N, x \geq LR, x \leq UR \text{ and } Ax \leq B\right\}.$$

Where LR = $[LR_1, LR_2, \ldots, LR_N]^T$ is the vector of lower resource requests of N users, UR = $[UR_1, UR_2, \ldots, UR_N]$ is the vector of upper resource requests of N users, and A is the $L\times N$ 0-1 matrix and B is the vector defined before. Recall that for simplicity we will use c, a, b to denote LR, TR, UR in the equations respectively.

We make an assumption that the available bandwidth for each node is greater than the sum of the LRs in the same node. If for one specific node this assumption does not hold, the long-term bandwidth allocation problem is trivial, i.e., x = LR. We are only interested in the subset of nodes for which the assumption holds in the network. So our assumption is reasonable. Now with our assumption, the feasible rate vector space X is:

$$X = \left\{x: x \in R^N, x > LR, x \leq UR \text{ and } Ax \leq B\right\},$$

and has at least one nonempty interior point.

Now we can formulate our centralized bandwidth allocation problem as follows:

$$\begin{aligned} &\max \sum_{i=1}^{N} [m_i \cdot \log(x_i - c_i) - \frac{x_i - a_i}{k_i}] \\ &\textit{subj. to: } x_i \geq c_i, x_i \leq b_i \\ &\qquad Ax \leq B \\ &\qquad i = 1, 2, \ldots, N \end{aligned} \tag{2}$$

With assumption: $Ax_c < B$, where $x_c = [c_1, c_2, \ldots, c_N]^T$. Because that the concave and injective properties are invariable under the mapping of the logarithm function[5], the objective function in (2) is equivalent to the following one:

$$\max \prod_{i=1}^{N} (x_i - c_i)^{m_i} \cdot \exp(-\frac{x_i - a_i}{k_i}).$$

In the problem, m_i is the weight for the source i, x_i is the allocated bandwidth for the source i, c_i is the minimum bandwidth for the source i, a_i is the targeted bandwidth for the source i, b_i is the maximum bandwidth for the source i, k_i is the desired attenuation parameter for the source i, and A, B represent the other constraints for the capacity.

Before solving the problem, we will first investigate the objective function, or namely our chosen utility function. According to the discussion previously, the optimal point x of the above problem has the property that for any perturbation, when $m_i = 1$,

$$\sum_{i\in I} \ddot{a}x_i \Big/ (x_i - c_i) \leq \sum_{i\in I} \ddot{a}x_i \Big/ k_i,$$

which is similar as the proportional fairness. Then nearby the optimum point, the aggregation of the relative changes of all the sources will be upper-bounded, although not zero. We call it "pseudo-proportional fairness". When (k_i, $i \in I$) are large enough, the upper bound will be small, even close to zero. We will see this property is well-coupled with the short-term time-varying bandwidth allocation in section VI.

D. Problem Solution

Under our assumptions, the feasible rate vector space X has nonempty interior, and the chosen utility function is an increasing, strictly concave and continuously differentiable function of x over the designated range. Then clearly, in (2), the objective function is increasing, strictly concave and continuously differentiable, and the constraints are linear. Therefore, the first-order Kuhn-Tucker conditions are the sufficient and necessary conditions for optimality.[11]

Now we consider the Lagrangian form:

$$L(x, \lambda, \beta, \mu) = \sum_{i=1}^{N} [m_i \cdot \log(x_i - c_i) - (x_i - a_i) \big/ k_i] - \sum_{i=1}^{N} \ddot{e}_i (c_i - x_i) - \sum_{i=1}^{N} \hat{a}_i (x_i - b_i) - \sum_{l=1}^{L} \grave{\imath}_l [(Ax)_l - B_l],$$

$$x_i \geq 0, \ddot{e}_i \geq 0, \hat{a}_i \geq 0, \grave{\imath}_i \geq 0, i = 1, \ldots, N,$$

where λ_i, μ_i, β_i, i=1, …,N, are slack variables associated with LRs, URs and capacity constraints.

And by considering the sufficient and necessary conditions under our assumptions, we obtain the unique solution as follows:

$$\forall i = 1, \ldots, N, \quad \forall l = 1, \ldots, L,$$

$$x_i = c_i + \min\left[(b_i - c_i), \ \frac{m_i}{\frac{1}{k_i} + \sum_{l=1}^{L} \mu_l A_{l,i}} \right] \tag{3}$$

$$Ax \le B, \ (Ax - B)_l \cdot \mu_l = 0, \ \mu_l \ge 0.$$

We have several useful remarks for the obtained optimal solution:

1. The Lagrange multiplier μ_l is the implied cost of unit flow through link l, or the shadow price of additional unit capacity for link l.
2. For one specific user, the assigned bandwidth is explicitly dependent of the link costs and its own parameters, while implicitly dependent of the users in other nodes.
3. m_i is the weight for the user i. The user with higher m_i has better opportunity to get more bandwidth than the user with lower one in the same node.
4. k_i is the desired attenuation parameter for the source i. Assume that k is proportional to *(a-c)* and *(b-c)*, while inversely proportional to *(b-a)*, then we have that $k \propto \frac{a-c}{b-a}(b-c)$. Again, the user with higher k_i has better opportunity to get more bandwidth than the user with lower one.

From the discussions above, we see that our framework has one more parameter (TR) which models the turning point of user's request. And by increasing the utility function before TR while decreasing it after TR, we make the bandwidth allocation more reasonably among all the users/flows while maintaining similar property as proportional fairness. At the same time, the importance of TR is modeled by k_i. With higher k_i, the effect of TR on our framework is smaller.

Now consider the asymptotic property of k_i, and the relation between our model and the model associated with proportional fairness.

Recall that $k_i \ge b_i - c_i, \forall i = 1, \ldots, N$. As all k_i go to ∞, it follows that our objective function

$$\max \sum_{i=1}^{N} [m_i \cdot \log(x_i - c_i) - (x_i - a_i) / k_i]$$

$$\rightarrow \max \sum_{i=1}^{N} [m_i \cdot \log(x_i - c_i)],$$

which is exactly the one with proportional fairness discussed in Ref.2 and Ref.5 when $m_i = 1$. As a result, our optimal solution here is exactly the one in Ref.5 as all k_i go to ∞. Also, with k_i increasing, the attenuation for the source i is decreased, and then the possibility for the source i to get more bandwidth after certain point is increased. This just shows the relation between our model associated with pseudo-proportional fairness with the model associated with proportional fairness.

VI. Dynamic Bandwidth Allocation

In this section we formulate the general time-varying dynamic bandwidth allocation problem for slotted TDMA protocol in space communications network based on the parameters determined by the long-term bandwidth allocation, and then find its solution, which would be used in our proposed hybrid TDMA protocol.

E. Model Description

The multiple-access scheme in the downlink channel is based on TDMA protocol. The frame with duration T_f consists of control slots and data slots. Let M denote the complete set of all MS, and M_a denote the set of active MS (i.e. the spacecraft with generating traffic). MS $k \in M_a$ sends a bandwidth request (BR) packet to the scheduler in the central ground station. There are two different levels of scheduling for dynamic bandwidth allocation: burst-level scheduling and packet-level scheduling. For burst-level scheduling, the central ground station performs the scheduling only once during each frame and allocates timeslots to a stream within a frame in a contiguous fashion. While for packet-level scheduling, the scheduling is performed during each timeslot and one timeslot is assigned at a time. Here we consider burst-level scheduling only. According to all the BR packets, the scheduler generates a bandwidth allocation table (BAT) and sends it back to all the MS in the set M_a. Then each active MS knows its assigned timeslots after reading the BAT. A BAT contains several information fields such as User_ID, First_slot, and Last_slot. User_ID field defines the identifier of the MS. First_Slot and Last_Slot give the number of the first and last timeslot assigned to this specific MS respectively.

F. Problem Definition

The resources to be assigned in our TDMA downlink channel are the total available data slots. Let N denote the number of the total available data slots. Here we focus on the optimal scheduling problem for assigning the N timeslots for all the active MS. The MS now present the different streams on-board itself.

We consider the penalty weights ν_{kl}, k€M_a, l€C for the service class l of the MS k to reflect the QoS and different requirements in our optimal scheduling problem. The penalty weights are determined by the QoS, average waiting time and the amount of waiting packets in queues.

For different slots assignment, the total penalty can be calculated with the definition of these penalty weights and the utility function. Our objective for the optimal scheduling is to find the solution for minimizing the total penalty. It is very convenient to change the penalty weights and utility function to achieve different optimization problems.

G. Input Parameters and Utility Function

Every time before making the BAT, the scheduler collects the updated information including the number of MS (M) and active MS (M_a), the bandwidth demands (D) of active MS and those for calculating the penalty weights. To present the different types of generating traffic, we let C denote the set of service classes. Thus, D is a two dimension matrix $\{D_{kl}\}$, k€M_a, l€C. The demands (D) could be directly given by the MS or estimated by the collected information from the MS. The latter is more practical while more complicate since an estimation step is must. The PR, i.e., (x_i, $i = 1, \ldots, N$), are used as parameters for estimation. We will discuss this later.

We use a matrix $s = \{s_{kl}\}$ to denote the amount of assigned data slots for service class l€C of the MS k € M_a. Therefore, the throughput for MS k is:

$$\sum_{l \in C} s_{kl}$$

We use the proportional utility function with the proportion of 1.

H. Problem Formulation

$$\text{Minimize} \quad \sum_{k \in M_a} \sum_{l \in C} \acute{\imath}_{kl} \, (D_{kl} - s_{kl})^{+} \tag{4}$$

subject to:

$$\begin{aligned}
& s_{kl} \le \min(U_{kl}, D_{kl}), && k \in M_a,\ l \in C \\
& s_{kl} \ge L_{kl}, && k \in M_a,\ l \in C \\
& \sum_{k \in M_a} \sum_{l \in C} s_{kl} \le N, && \\
& \forall\, s_{kl} \in \{0, 1, 2, \ldots, N\} &&
\end{aligned}$$

If a MS requests more timeslots than the available data slots which can be assigned to it, only a portion of its request slots will be actually admitted and the residual packets must wait for the next scheduling. Let U_{kl} and L_{kl} denote the upper bound and lower bound of capacity for the service class l€C of MS k€M_a, respectively. The LR and UR from the user via long-term bandwidth could be used directly here. Some mappings from LR and UR are also allowed. The PR, i.e., (x_i, $i = 1, \ldots, N$), are used as parameters for bounded assignment. The upper bound of waiting time (delay) for the service class l€C of MS k€M_a is set and used in the decision of penalty weights to guarantee the maximum delay if necessary.

I. Problem Solution

The solution for this linear problem can be found by these steps:

1. *Sorting*: Sort the penalty matrix $\{\nu_{kl}\}$ and re-list them in a vector V in the descending order.
2. *Lower Bound assignment*: Determine the number of data slots for the active MS to satisfy the lower bound requirements.
3. *Additional Amount assignment*: After 2nd step, assign the available slots to the active MS according to their order in the vector V until the demand or upper bound is fulfilled.
4. *Final assignment*: Allocate timeslots to each stream within a frame in a contiguous fashion.
5. *Create the BAT*.

Our problem formulation is based on two assumptions: 1). The demands D_{kl}, upper bounds U_{kl} and lower bounds L_{kl} are known or could be determined by the scheduler. 2). The penalty weights ν_{kl} are very important and distinct. Another concern is that our problem should consider the multi-frame condition in the space communications network with long propagation delay.

We make some improvements for allowing for these concerns. Usually the U_{kl} and L_{kl} can be assigned according to the service requirements of the streams and the practical condition of the whole channel, and can be viewed as two adjustable parameters. Let t_0 and t denote the time the request was created in the MS and processed in the scheduler respectively. Between t_0 and t, the total assigned timeslots for the service class l of the MS k is called "credit" and denoted by $Ct_{kl}(t_0, t)$. Similarly, the total incoming packets between t_0 and t plus the number of packets in queue at time t_0 for the service class l of the MS k is called "debit" and denoted by $Dt_{kl}(t_0, t)$. Then, the "balance", which is $[Dt_{kl}(t_0, t) - Ct_{kl}(t_0, t)]^+$, is a very practical determination of the demand D_{kl}. The cumulative bandwidth assignment for one user is upper-bounded by its (TR × frames + U) and lower-bounded by its (TR × frames − L). Notice that it considers the multi-frame condition for the long propagation delay. The penalty weights ν_{kl} are assigned discrete values based on the relations between the "balance" and some prescribed thresholds. When ν_{kl} of some streams are same, the calculated demands D_{kl} are used to determine their order in the first step.

VII. Configuration and Simulation Results

A. Network Configuration

We use OPNET to model the MAC protocol and network scenario. Once again, we do not consider the issue of handover; therefore will use only one relay satellite. And for simplification, we only consider the spacecraft in the coverage of a relay satellite in the current location of TDRS_EAST [longitude of 319 degree, latitude of 0 degree, and altitude of 35,787 km]. And we consider four LEOs in its coverage zone: TERRA, LEO2, LEO3 and LEO4, which have the altitude range of 701-716 km. The White Sands Ground Terminal (WSGT) is located at longitude of -106 degrees west, latitude of 34.5 degree.

Because the number of LEOs in the coverage zone is predictable, we can determine the length of frame and time slot in advance. The calculated RTD is more than 0.66 seconds. We set $M \cdot T_f = 0.68$ seconds according to the analysis in the previous Section. We set the number of data slots per frame as 64 ($N = 64$). The data rate of 200Mbps is assumed to be supported by this common link. However, to simplify the simulation, we do the following transformation: lower the channel capacity to 2Mbps and accordingly take 1/100 of data rate of all spacecraft. For example, the peak data rate of TERRA is taken as 1.08Mbps. Therefore, by combining these parameters and the length of source packet (512 bytes), we set $M=5$, and get $T_f = 0.1372$ sec. Consider the use of guarding time, we can assume the downlink channel as an error-free TDMA common link.

We are particularly interested in the system throughput, defined as the total amount of traffic arrived at the ground station in a given unit of time. This measure, in a sense, provides an indicator of the level of bandwidth-efficiency. Another performance of interest is end-to-end (ETE) delay. By ETE delay of a packet, we mean the time interval between its generation on-board in the spacecraft and its arrival at the ground station. For our hybrid-mode TDMA, in the initial phase, the ETE delay includes the reservation delay, which is more than twice the propagation delay. However, once the system is stable, the packet may not endure this type of long delay by using the allocated data slots.

We assume that the network traffic is diverse, i.e., the traffic loads are unevenly distributed among the spacecraft. Also, as mentioned earlier, the source traffic generating rate in a specific spacecraft varies considerably. Those properties match the unpredictable and dynamic traffic pattern in this environment. There would be times when spacecraft could be completely inactive for a period of time, and an adaptive protocol would be capable to accommodate that. In practice, the inactive spacecraft can notify the ground station of this special status by sending a "negative" reservation request, i.e., set a negative number in the field of queue size. Then the ground station will exclude the assignment of reservation slots to this inactive spacecraft, and free all the reservation slots assigned to it before, except the statically assigned data slots. This may improve the bandwidth-efficiency by assigning the waste slots to the active spacecraft.

B. Simulation Results

Our simulation is run for several minutes to reach steady-state. We try to adjust the simulation time to take within the limit of having the spacecraft inside the common coverage zone under one TDRS relay satellite is limited. Note that the spacecraft are orbiting rapidly (typically their orbit periods are around 95 minutes). Also note that these LEOs have an altitude range of 701-716 km. The propagation delay would vary from 0.24s to 0.30s. The variation of the propagation could be 0.06s, almost half the length of one frame. More detailed discussion on the time-varying propagation delay is in Ref.1.

We first present the performance of our hybrid protocol under unevenly distributed traffic load. Then the ETE-delay and successful throughput performances of a conventional (static) TDMA solution will be compared with this protocol. As shown in Fig.5, the ETE delay is ranging from 0.26 seconds to less than 0.5 seconds under different traffic loads with different numbers of active spacecraft. Considering the large propagation delay and the large variation of it because of the spacecraft mobility, this is very good. A major portion of the ETE delay in this case is introduced by the error of time synchronization. Suppose when the spacecraft has packets to transmit, it will calculate the propagation delay, say t_1, and determine whether it owns the current data slot (or control slot). If the answer is yes then it sends the packet, otherwise it determines the time for its data slot, say t. However, at time t, the propagation delay is t_2, not t_1. The direct result is that the spacecraft misses its data slot. And when the packets are delayed inappropriately in the spacecraft, the ETE delay and the throughput would be affected obviously. From Fig.5, we can obviously see that, the less active spacecraft we have, the better performance the protocol will have.

Now we fix the ratio of expectations of traffic loads of four users (spacecraft) as 3:2:2:1, and study the performance of our hybrid protocol in terms of ETE delay, successful throughput and the fairness under this special scenario. As shown in Fig.6, the hybrid protocol outperforms the fixed TDMA in terms of ETE delay and successful throughput. This is because the hybrid protocol can utilize the data slots once belonging to the inactive spacecraft or spacecraft at low data rate in a short range, while in the fixed TDMA, these data slots are just wasted. Another reason is that in the hybrid protocol, the data slots are dynamically assigned based on the behavior of their traffic, and therefore achieve the better bandwidth utilization. The more bursty and unpredictable the traffic sources are, the more the hybrid protocol will outperform a fixed TDMA solution.

To study the (long-term) fairness among all the users, the successful average throughputs of the total channel and every individual user are shown in Fig.7. As we can see, although obviously the proportional fairness is not achieved, the pseudo-proportional fairness is obtained in some sense. In other words, the average share of the channel for every user is close to its proportional portion according to the expectations of traffic loads of four users, i.e., 3/8, 2/8, 2/8 and 1/8 respectively. Since we use the order of the users to

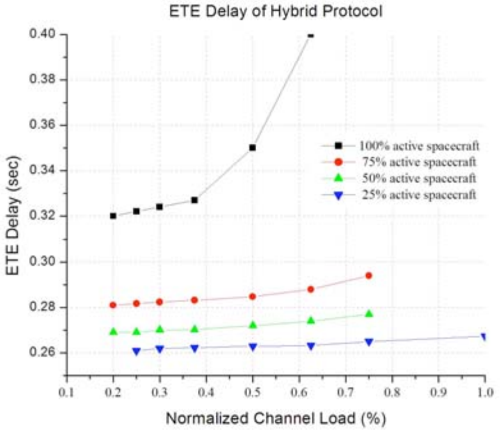

Figure 5. ETE Delay of Hybrid Protocol

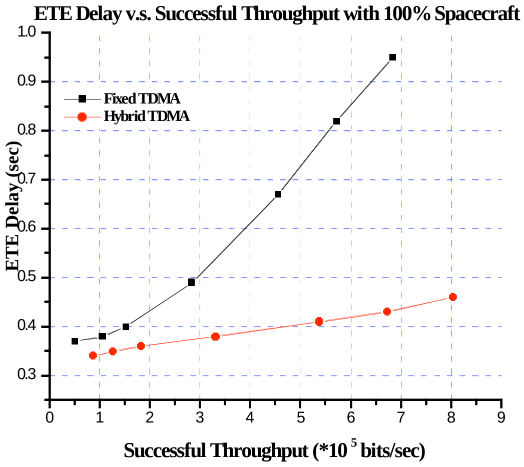

Figure 6. ETE Delay v.s. Throughput

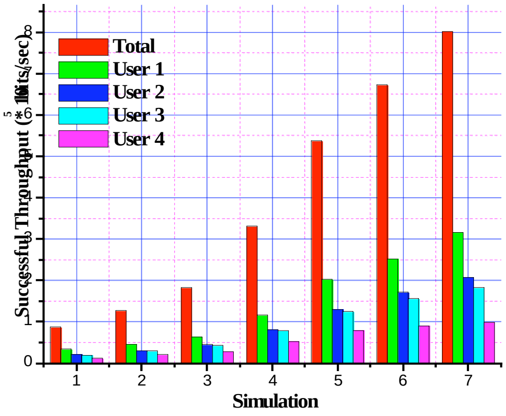

Figure 7. Fairness among Users

break the tie sometimes in the dynamic bandwidth allocation, the average throughput of the user 2 is always a little higher than that of the user 3 despite that they have same traffic loads.

VIII. Summary

To provide optimal or near-optimal efficient utilization and fair allocation of bandwidth of the downlink channel while guaranteeing specific QoS requirements for different service classes, we propose two-level (long-term static and short-term dynamic) bandwidth allocation for a slotted TDMA high data rate satellite communication link. The long-term bandwidth allocation is implemented to provide per-stream/per-user QoS guarantee and shape the average behaviors. In our time-varying short-term bandwidth allocation with threshold regulation, a dynamic allocation is performed by solving an optimal timeslot scheduling problem according to the requests and other parameters. By using simulation, the performance of a suitable Medium Access Control (MAC) protocol with timeslot scheduling is analyzed and compared with that of the existing static fixed-assignment scheme in terms of ETE delay and successful throughput. We also study the fairness among all the users under a special scenario and find that the pseudo-proportional fairness is achieved for our hybrid protocol.

There are still some future works. One is that the aggregation of all the streams on-board spacecraft will not be priority queuing. Instead, the bandwidth allocation, long-term or short-term, will be explicitly determined stream-wise completely in the controller on ground. Weighted Round Robin (WRR) could be a simple improvement from priority queuing. Another is that the effects of changing the period of performing the long-term bandwidth allocation need to be investigated. Long period probably causes out-of-date information and then long-term not-good behavior, while short period might lead to unstable behavior. So an appropriate period is needed. Also, an event-driven variable-period long-term bandwidth allocation could be an alternative.

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