Fixed Versus Adaptive Admission Control in Direct Broadcast Satellite Networks With Return Channel Systems

Fatih Alagöz, *Member, IEEE*, Branimir R. Vojcic, *Senior Member, IEEE*, David Walters, *Member, IEEE*, Amina AlRustamani, *Member, IEEE*, and Raymond L. Pickholtz, *Fellow, IEEE*

***Abstract*—In this paper, as part of the adaptive resource allocation and management (ARAM) system (Alagöz, 2001), we propose an adaptive admission control strategy, which is aimed at combating link congestion and compromised channel conditions inherent in multimedia satellite networks. We present the performance comparisons of a traditional (fixed) admission control strategy versus the new adaptive admission control strategy for a direct broadcast satellite (DBS) network with return channel system (DBS-RCS). Performance comparisons are done using the ARAM simulator. The traffic mix in the simulator includes both available bit rate (ABR) traffic and variable bit rate (VBR) traffic. The dynamic channel conditions in the simulator reflect time variant error rates due to external effects such as rain. In order to maximize the resource utilization, both for fixed and adaptive approaches, assignment of the VBR services are determined based on the estimated statistical multiplexing and other system attributes, namely, video source, data transmission, and channel coding rates. In this paper, we focus on the admission control algorithms and assess their impact on quality-of-service (QoS) and forward link utilization of DBS-RCS. We show that the proposed adaptive admission control strategy is profoundly superior to the traditional admission control strategy with only a marginal decrease in QoS. Since the ARAM system has several parameters and strategies that play key roles in terms of the performance measures, their sensitivity analysis are also studied to verify the above foundations.**

***Index Terms*—Admission control, multimedia satellite networks, quality-of-service (QoS), source and channel rate, utilization.**

I. Introduction

RECENTLY, many satellite systems, including low earth orbiting (LEO), medium earth orbiting (MEO), and geostationary (GEO) satellites, have been proposed to support worldwide multimedia and interactive services [1]–[5]. The involvement of satellites in Internet protocol (IP) networks is due to new trends in global telecommunications, where the Internet traffic may hold a dominant share in the total network traffic [5]. The multimedia satellite networks may facilitate interconnectivity, minimize the required wiring, and provide broadband Internet services to both fixed and mobile users. The Internet packets based on digital video broadcast (DVB)/MPEG-2 in the forward direction and asynchronous transfer mode (ATM) in the return link are standardized by ETSI for the direct broadcast satellites (DBS) with return channel systems (DBS-RCS) [2]–[5]. In order to improve both the performance and capability of the multimedia satellite networks, researches on multiple fronts are under investigation. These researches include but not limited to integrated satellite architectures, beam scheduling, on board signal regeneration, adaptive modulation and coding, multiple access, flow control, and resource allocation, etc.

In this paper, we focus on the flow control and resource allocation for a DBS network with return channel system (DBS-RCS). Without loss of generality, in the return link channel, we consider quality-of-service (QoS) reports including frame-error rates, frame delay jitter, etc. With reference to this satellite network architecture, an early study on this topic focusing on the effect of adaptive channel coding is reported in [1]. In this paper, further enhancement is introduced into the admission control and resource allocation scheme with a view to reduce both of the intrinsic impairments caused by the misestimated statistical multiplexing gain and bad channel conditions, and the achievement of better performance levels and QoS[1] guarantees.

The DBS-RCS has to overcome two major obstacles to sustain throughput in the forward link while attaining QoS. The first is the variable bit rate (VBR) traffic that cannot be exactly modeled due to inherent traffic characteristics and the second is the variation in the channel quality that continuously changes over time due to fading, propagation anomalies, jamming, etc. Specifically, in regard to the former, if capacity allocations were based on the peak-source-rates, these networks would have very low utilization due to the high burstiness of the traffic such as in the case of moving pictures expert group (MPEG) coded video [7]. Alternatively, if these networks rely on statistical multiplexing and overload links, congestion may occur during peak periods [8]. In regard to the channel constraint, the channel

Manuscript received December 15, 2002; revised July 1, 2003 and September 20, 2003. This work was supported in part by the DARPA Global Mobile Information Systems Program under Contract DABT-95-C-0103 to the U.S. Army of Intelligence Center, Fort Huachuca, AZ.

F. Alagöz was with the United Arab Emirates University, AlAin, United Arab Emirates. He is now with Harran University, Şanlıurfa, Turkey (e-mail: alagoz@harran.edu.tr).

B. R. Vojcic and R. L. Pickholtz are with The George Washington University, Washington, DC 20052 USA (e-mail: vojcic@gwu.edu; pickholt@gwu.edu).

D. Walters was with the Orbital Space Communications, MD. He is now with λ Optical Systems (e-mail: dhwalters@ieee.org).

A. AlRustamani is with the Dubai Internet City, Dubai, United Arab Emirates (e-mail: amina.alrustamani@samacom.com).

Digital Object Identifier 10.1109/JSAC.2003.819972

[1]We assume that the small amount of changes in the Moving Pictures Expert Group (MPEG) source rates and forward error correction (FEC) rates are linear functions of QoS. Thus, given that control strategies maintain the delay and frame error rate within acceptable range, QoS is considered as a linear function of overall rate loss. We use $\mathrm{QoS_T}$ (7) and $\mathrm{QoS_D}$ (8) metrics for QoS of both VBR services and ABR services, respectively.

bit-error rate may range from having almost no impact on performance to dramatically degrading performance depending on the channel conditions. Among the channel error recovery techniques used in wireless environments are automatic repeat request (ARQ) and FEC techniques, or their combinations. However, in a DBS application, the ARQ techniques may not be suitable due to latency constraints of real-time traffic. On the other hand, there is a trade off when employing the FEC mechanism; while the FEC may enable the system to recover erroneous packets in adverse channel conditions, the FEC overhead may cause further congestion and more packet loss in the network [1], [9]. It is, therefore, of particular interest to study the integration of admission and rate control of the system transmission attributes, namely, video source (MPEG compression), data transmission and channel coding (FEC) rates for DBS-RCS.

There are many proposals for admission control and bandwidth reservation to guarantee a reasonable QoS level for wireless/mobile networks [8], [10]–[17]. Traditional admission control schemes based on Poisson process models may provide sufficient precision for 2G cellular networks [17]. Similarly, the admission control schemes for IP-based networks may provide guaranteed QoS for Differentiated Services networks [14]–[16]. However, there are stringent requirements for the multimedia satellite networks. Recently, the admission control and resource management schemes for particular satellite network architectures are presented in [1], [5], [6], [18]–[25]. *Bohm et al.* [18] present the performance of a movable boundary accessing technique, detailing the admission control and resource allocation procedure, in a multiservice satellite environment. *Koraitim et al.* enhance this approach and provide performance results for both conventional and dual movable boundary schemes [19]. *Rose et al.* present the simulation results for an end-to-end connectivity planning and admission control for a multibeam satellite network with on-board cross-connectivity [20]. *Iera et al.* propose an adaptive call management system for real-time (low-interactive) VBR traffic over GEO satellite links [21], [22]. *Zein et al.* present simulation results for the performance of the combined/fixed reservation assignment scheme for aggregated traffic [23]. *Connors et al.* model and simulate the medium access control of the broadband satellite networks [24]. *Açar et al.* present the performance of end-to-end resource management in ATM GEO satellite networks [25].

Unfortunately, most of the assumptions in the above works are either defective for the considered system, or disregard the channel problems, or ignore resource management by considering only the admission control problem. This is because, as often the case in actual situations, it is not only unclear how to achieve the "good" but even what the "good" is. In this paper, we analyze and examine this situation without unduly simplifying it. First, the adaptive resource allocation and management (ARAM) system has been developed to manage the DBS-RCS supporting ABR and VBR traffic mixtures and operating under dynamic channel conditions [26]. This requires the integration and implementation of several well-addressed and standardized subsystems to build a general resource manager. Second, we propose an admission control strategy that adaptively estimates the bandwidth expansion factor that determines the number of admitted services and integrate it into the ARAM system. Finally, we perform extensive set of simulations both for traditional (fixed) and the proposed adaptive admission control approaches.

The rest of the paper is organized as follows. Section II presents the description of DBS-RCS architecture and protocol stack. Section III presents an overview of the ARAM system for use in DBS-RCS. Section IV describes the alternative admission control concepts; fixed and adaptive. Section V presents the simulation results and sensitivity analysis. Section VI presents the conclusions of this work.

II. DBS-RCS Architecture

Many multimedia satellite systems have been proposed to support worldwide multimedia services. In general, the following three system architectures for multimedia satellite networks may be distinguished; one-way communication, two-way communication with telephony return channel, and two-way communication with a transmitter at the user location. For the last architecture, one may have different options: return link via a high-speed DBS, or lower speed MEOs or LEOs [2], [4]–[6]. Fig. 1 depicts the DBS-RCS architecture. In multimedia satellite networks, a return link may follow Link 1, Link 2, and Link 3 in Fig. 1, [1], [2], [4], [5], [19]. In this paper, we consider a low-speed return link (Link 2) that may be provided by a constellation of LEO satellites [1]. Although any backhaul network would work, the LEO constellation enables the user to set up a DBS field terminals (DFT), where there is no terrestrial backhaul and have immediate interconnection to a remote backhaul network. Moreover, we consider a perfect return link, i.e., QoS reports are delivered error-free via the return link channel.

The overall objective of DBS-RCS traffic management is to deliver high volumes of information from source systems hosts to application platforms (APs). The DBS-RCS uses a high capacity forward link provided by a DBS to multicast voice, video, and data packets from source system hosts to DBS field terminals (DFTs) located at the satellite downlink facility. Upon receipt of these packets, the DFT routes them to the user AP which may be a multimedia personal computer or workstation.

The DFT is a combination of DBS antenna, RF system, and set-top box with an IP router. The interface between the DFT and APs may be either a local attachment by a serial link or local area network, or a remote connection by a terrestrial wireless network (Wnet in Fig. 1). Since the area of coverage of the LEO satellite may not include the source systems, the LEO downlinks return packets to an in-theater gateway. Then, the LEO gateway transmits the packets to the source system via a terrestrial backhaul network.

A. Services and Protocol Stack

The DBS-RCS supports two types of services both of which use multicast IP as the underlying protocol, but with their own individual upper layer protocols. These are ABR service for the reliable multicast of data, and VBR service for the multicast of MPEG coded video.

The ABR service is implemented using the reliable DBS multicast protocol (RDMP) that guarantees the reliable delivery of messages to receivers or identifies an error condition [27]. The

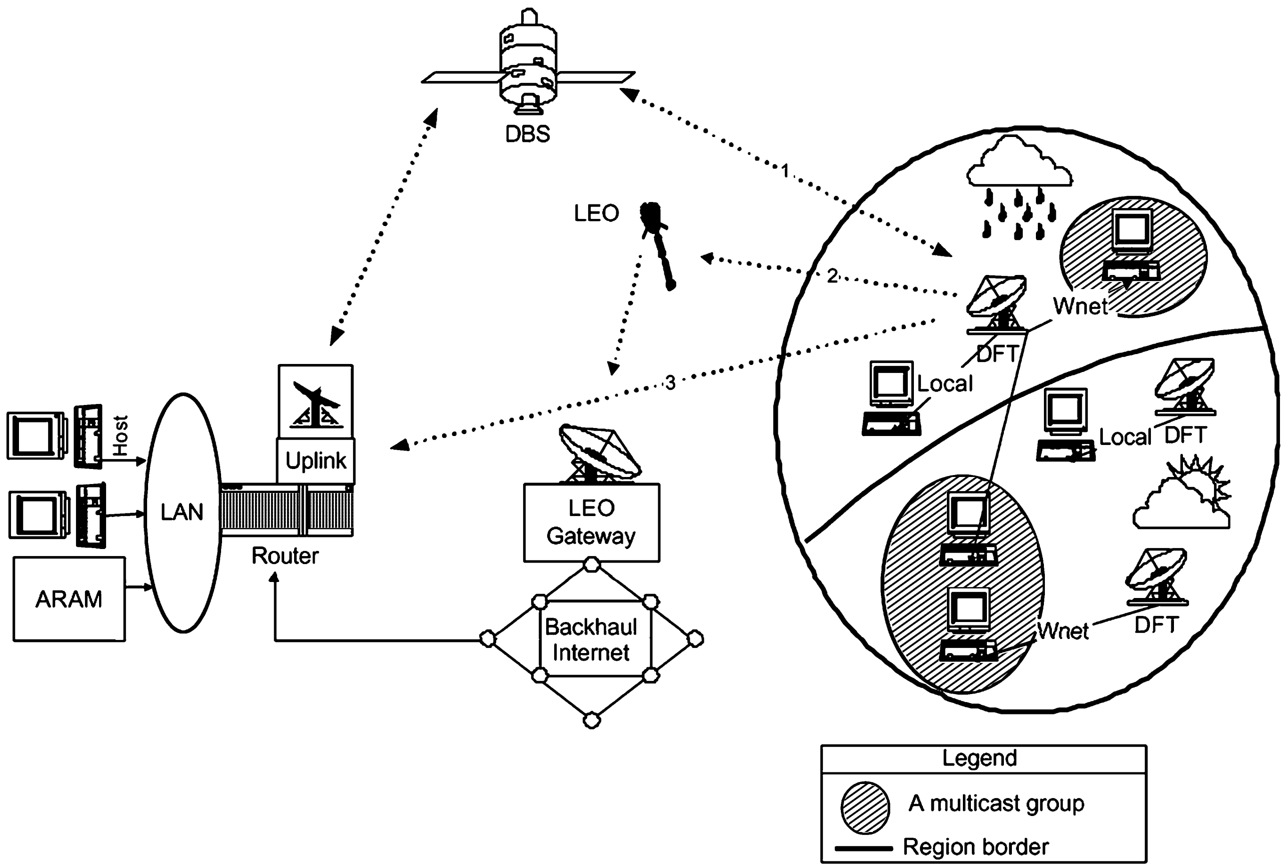

Fig. 1. Architecture of DBS networks with return channel systems.

RDMP is used for delay insensitive multicast services (e-mails, files, images, or objects). It provides error-free delivery of data to each receive protocol entity resident in the APs or notifies the sending application and network management that an error has occurred. The major feature of RDMP relevant to this work is its capability to operate as an ABR service, i.e., with an adaptive transmit data rate at an agreed upon error rate. It will transmit its packets at a rate ranging from $(\mathrm{T_{min}}, \mathrm{T_{max}})$. The ARAM sets and adjusts the values of certain RDMP operational parameters based on the resources allocated to a service, current system status (congestion, excessive frame errors, etc.), and corrective measures taken (packets dropped, FEC rate decreased).

The VBR service provides the delivery of MPEG video via IP multicast from one source to many receivers. Since this service operates in a real-time mode, it does not utilize any acknowledgment techniques but relies on the quality of the underlying network to ensure an acceptable error rate. In this study, MPEG video traffic is characterized by constant transmission rate of two groups of picture (GOP) per second and 12 frames per GOP. Since the number of bytes in a frame is dependent upon the content of the video, the actual bit rate is variable over time. There are three types of frames [7]:

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

Analogous to the ABR service, the major feature of the VBR service in this work is its capability to adjust its offered data rate based on network conditions. While it requires the fixed delivery of two GOP's per second, the length of the frames may be adjusted over a range to control the transmitted data rate. This may be implemented by dynamically adjusting the quantization level in the MPEG algorithm [7].

In what follows, we describe the interaction between the ARAM system and the protocol stack (PS) of the network. The ARAM system and PS are integrated by exchanging notifications and information. For example, the ARAM notifies PS of queue buildups, change of $\mathrm{QoS_T}$, terminating a service, etc. Similarly, PS notifies the ARAM of the status of a service (via QoS reports), new requests, etc. The following is a list of interactions in the system.

From PS (and User) to ARAM:

1) *User_ARAM.ServiceRequest(svcHandle, ServiceParameters).* Issued when requesting admission of a service.
2) *User_ARAM.CloseServiceRequest(svcHandle).* Issued when a service has completed or is to be terminated or suspended.
3) *PS_ARAM.AlterServiceParameter(svcHandle, Parameter Name, NewValue).* Issued when a service parameter is to be changed. For example, the receiver requests higher $\mathrm{QoS_T}$, or the DFT moves under coverage of another transponder.
4) *PS_ARAM.QoSReport(svcHandle, QoSreport).* QoS report of ongoing services.

5) *PS_ARAM.OpenFailed(svcHandle)*. Indicates problems during service initiation. ARAM may respond by increasing *ACK_Timer_Period* by issuing *ARAM.RDMPSetParm(ACK_Timer_Period, NewValue)* if the service is experiencing excessive queueing delays.

From ARAM to PS:

1) *ARAM_PS&User.ServiceAdmitted(svcHandle, StartTime)*. Issued in response to *User_ARAM.Service Request*. Indicates service has been admitted to the system. Start time maybe used by PS to advertise the service on the bulletin board channel.
2) *ARAM_User.ServiceRefused(svcHandle, Reason)* Indicates a service request has been refused for the shown reason.
3) *ARAM_PS.NetworkInitiatedMulticast(svcHandle, New MulticastMemberAddress)*. Indicates a request by ARAM to create (or append) a multicast group as a result of merging a service request with other requests for the same service.
4) *ARAM_PS.ServiceParameterAltered(svcHandle, ParameterName, NewValue)*. Notifies PS of a change to a service parameter.
5) *ARAM_PS&User.ServiceClosed(svcHandle, Reason)*. Notifies PS that the service has been stopped (dropped) for the shown reason. Note that if *User* issues a *User_ARAM.CloseServiceRequest(svcHandle)*, then ARAM will respond with *ARAM_PS&User. Service Closed(svcHandle)* which confirms service closure.
6) *ARAM_PS.RDMPSetParm(RDMPParameterName, New Value)*. Used to set RDMP parameters.

The real-time transport protocol (RTP) is based on application level framing and, hence, operates on top of existing transport protocols, primarily user datagram protocol (UDP). The real-time control protocol (RTCP) is used for monitoring and distributing information on the current level of QoS transmitted and received in a session. Moreover, to support real-time data transfer RTP protocol header has a number of important fields: payload type, sequence number, and time stamp, etc. The payload specifies the media type, e.g., MPEG video, PCM audio, etc. The time stamp may be used by a receiver to resynchronize data and to monitor packet arrival jitter. The sequence number may be used to monitor packet loss and reordering.

In the DBS-RCS, the QoS reports from an application (DFTs or receivers) for each multicast maybe gathered based on RTCP protocols, which may rule the application to send their reports asynchronously to utilize the return link channel.

III. ARAM System Overview

The ARAM system has three main goals in performing traffic management for the multimedia satellite networks: efficient utilization of available capacity, fair access to system resources (within priority constraints), and graceful degradation of $\mathrm{QoS_T}$ during congestion and bad channel conditions [26]. The challenge in achieving these goals is managing the dynamic bandwidth needs of VBR traffic, as well as channel dynamics. The ARAM system addresses these goals in three ways.

1) Leverage the statistical multiplexing effects, (not all VBR peaks occur at the same time).

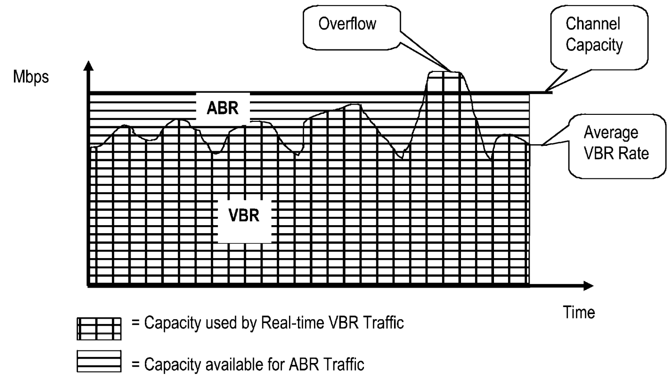

Fig. 2. Concept of multiplexing of VBR and ABR traffic.

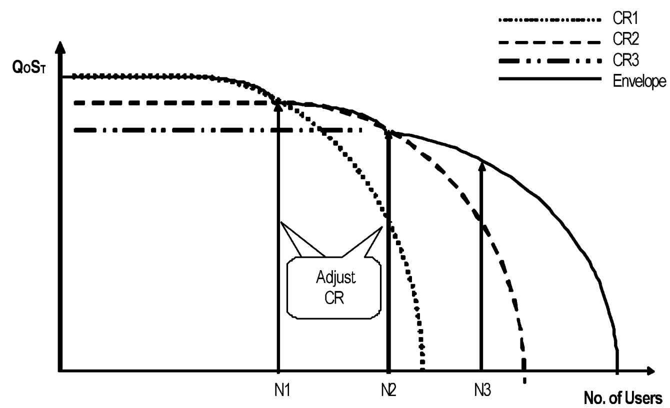

Fig. 3. ARAM $\mathrm{QoS_T}$ performance. CR denotes the MPEG compression rate changes with respect to nominal rates. $\mathrm{QoS_T}$ metric is given in (7).

2) Adjust the rates of ABR traffic with less stringent latency requirements.
3) Scale the MPEG video source rate and channel rate to operate within the bandwidth if all else fails.

Using this approach, the ARAM system maintains a balance between meeting user needs without over designing the network.

Fig. 2 depicts the multiplexing concept utilized in the ARAM system. First, the ARAM system adjusts the ABR traffic rate such that it can be accommodated in the capacity not used by the VBR traffic. The dark area in the figure conceptually indicates this. When the aggregate VBR traffic, which is the sum of bit rates of admitted VBR sources, is less than its allocated capacity, the ABR rates are increased such that capacity is not wasted. Alternatively, when VBR rates increase, ABR rates are decreased. When traffic rates exceed the gains in capacity provided by the statistical multiplexing causing an overflow condition, the rates of both ABR and VBR traffic are scaled to operate within the available capacity. Here, the MPEG video compression rate is increased and the ABR transmission rate is reduced. While this results in some degradation in $\mathrm{QoS_T}$, the ARAM system provides a graceful and fair reduction to its users.

The resulting performance of the ARAM system as the number of users increases is conceptually depicted in Fig. 3. It shows the $\mathrm{QoS_T}$ as a function of the number of users for three compression rates $(\mathrm{CR1} < \mathrm{CR2} < \mathrm{CR3})$. The $\mathrm{QoS_T}$ gradually decreases as the number of users increases until a break point is reached when the $\mathrm{QoS_T}$ sharply degrades. In this

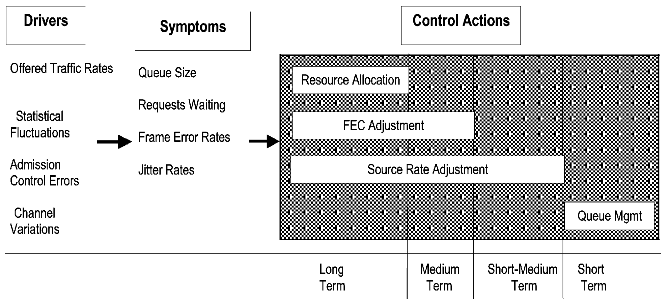

Fig. 4. ARAM algorithmic concept.

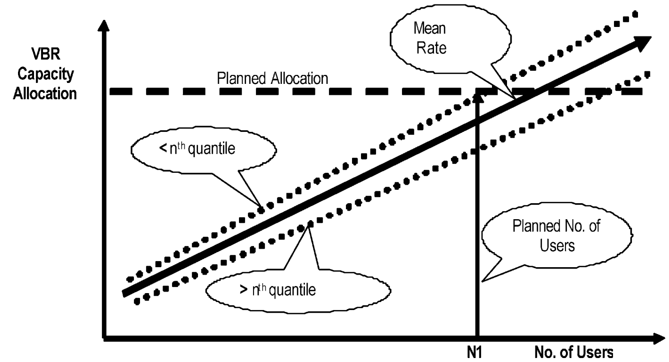

Fig. 5. Admission control concept.

example, the ARAM system would use CR1 until the number of users exceeds N1. Then, it switches to CR2 and uses that compression rate until the number of users exceeds N2. When this occurs, it switches to CR3. It limits the maximum number of users to N3 such that the $\mathrm{QoS_T}$ does not degrade below acceptable levels. As the ARAM system adjusts the compression rate in response to increased number of users, it ensures the system operates on the top envelope of the individual $\mathrm{QoS_T}$ performance curves indicated by the solid line in the figure.

Fig. 4 depicts the time phased control strategies utilized by the ARAM system. These control strategies are intended to respond to events in the network such as increases in the offered traffic, statistical fluctuations in individual services, or changes in the DBS channel conditions. The symptoms used to identify problems in network conference include requests waiting for service, queue sizes, error rates, and jitter rates.

In response to problems identified by analyzing these symptoms, the ARAM system activates the four time phased control strategies with different time scales: short (STC), short-to-medium (SMTC), medium (MTC), and long (LTC) term controls. As shown in the figure, each control strategy may utilize one or more of the following control actions: MPEG video compression rates, data transmission rates, FEC coding rates, and resource allocation.

After resources are assigned to users, the STC actions are limited to queue management of the ARAM system that implements rules for prioritizing packets. Following the MPEG coding scheme [7], MPEG coded VBR packets are given priority over ABR packets, and MPEG I frame packets are given priority over P or B packets. For intermediate time periods, the ARAM system adapts traffic rate adjustments for both VBR and ABR services and adjusts the FEC rate to accommodate small changes in conditions. The LTC is intended to accommodate bigger changes in network conditions. In this case, the ARAM system may have to reallocate resources if network performance degrades below the desired $\mathrm{QoS_T}$, i.e., drop services. It is worth noting that alternative approaches for performing this allocation in response to either user requests or changes in network conditions are the thrust of this paper.

IV. Admission Control

We present two alternative approaches to admission control: fixed and adaptive. The resource allocation algorithm is responsible for assigning service requests to a DBS transponder, thereby admitting or blocking a service request based on $\mathrm{QoS_T}$ parameters of the requests and estimation of the available resources. Since the VBR services operate with a variable data rate, this allocation assumes some statistical multiplexing of VBR services will occur. The number of services allocated is based on the assessment that the capacity allocated to the VBR services will only exceed the assigned capacity a fixed percentage of the time. As depicted in Fig. 5, the ARAM system determines the capacity to be allocated to VBR traffic and applies its admission control to determine the number of users (N1).

However, due to variability of traffic sources, the actual offered traffic will fluctuate around the mean traffic level (solid curve). Based on its statistical traffic characteristics, the offered traffic will exceed the upper quantile (dotted line) no more than for a fixed percentage time based on the $\mathrm{QoS_T}$. Thus, the number of users N1 that can be supported with the desired $\mathrm{QoS_T}$ is determined by the intersection of the upper threshold (dotted line) with the horizontal planned allocation. In addition, reallocation of system resources is required to adapt to changes in traffic and channel conditions.

A. Fixed Admission Control

Fixed admission control uses the same algorithm independent of the past traffic characteristics. The bandwidth expansion factor (BEF) for VBR traffic is determined such that the probability of the aggregate instantaneous rate exceeding the fraction of the capacity assigned to the admitted VBR services will not be greater than a prespecified probability value (γ)

$$\Pr\left\{\sum_{i=1}^{N} R_i \geq B_T\right\} = \int_{B_T}^{\infty} f_x(x)\,dx \leq \gamma$$

$$B_T = \alpha \sum_{i=1}^{N} R_i^m \tag{1}$$

where

N	total number of admitted VBR services;
R_i	instantaneous rate of the ith VBR service;
B_T	fraction of the capacity assigned to VBR services;
α	bandwidth expansion factor;
R_i^m	average rate of the ith VBR service;
$f_x(x)$	probability density function (pdf) of the aggregate rate.

The pdf of the aggregate VBR traffic cannot be found analytically and its complexity depends on the model used to represent the VBR source encoder [29]. Therefore, in the ARAM simulator, the aggregate VBR traffic rate is assumed to have a Gaussian distribution with mean equal to the sum of the individual VBR traffic means and a variance equal to the sum of their variances [26]

$$\Pr\left\{\sum_{i=1}^{\mathrm{N}} R_i \geq B_T\right\} = \int_{B_t}^{\infty} \frac{1}{\sqrt{2\pi\sigma_{\mathrm{agr}}^2}} \times \exp\left(\frac{(x-\mu_{\mathrm{agr}})^2}{2\sigma_{\mathrm{agr}}^2}\right) dx \tag{2}$$

$$\mu_{\mathrm{agr}} = \sum_{i=1}^{\mathrm{N}} R_i^m \quad \text{and } \sigma_{\mathrm{agr}}^2 = \sum_{i=1}^{\mathrm{N}} \sigma_i^2.$$

This estimate is an approximation of the aggregate rate, but it provides reasonably good results for moderate to large number of multiplexed bitstreams [29]. The accuracy of this approximation strongly depends on the value of predefined probability parameter (γ) since this value determines the number of admitted request, as well as the BEF (α).

In the ARAM system with fixed admission control, γ is estimated (γ_{e}) by measuring the fraction of dropped frames or packets over every SMTC period (10 s in the simulation). Also, the average excessive rate is estimated $(\Delta\omega)$ as the difference between the sum of the instantaneous rates minus the capacity allocated to VBR services averaged over the SMTC period

$$\Delta\omega = \overline{\left(\sum_{i=1}^{\mathrm{N}} R_i\right) - B_T}. \tag{3}$$

Then, the average overall rates (source rate divided by FEC coding rate) of individual services are adjusted

$$W_i^{\mathrm{new}} = \begin{cases} W_i^{\mathrm{old}} - \dfrac{\Delta\omega}{\mathrm{N}}, & \text{if } \gamma_{\mathrm{e}} > \gamma \\ W_i^{\mathrm{old}} + \dfrac{|\Delta\omega|}{\mathrm{N}}, & \text{if } \gamma_{\mathrm{e}} < \gamma \end{cases} \tag{4}$$

where W_i^{new} and W_i^{old} are the new and old overall rates of the ith service, respectively. If $\gamma_{\mathrm{e}} > \gamma$, there is more dropping than estimated and the average overall rates of the services are reduced by increasing compression rates and/or FEC rates.

The condition $\gamma_{\mathrm{e}} < \gamma$ indicates that inequality (1) holds and no control actions are necessary; instead the $\mathrm{QoS_T}$ can be improved by increasing overall rate allocation (if previously reduced) and resetting the compression rate. However, the inequality may hold due to under estimating the statistical multiplexing gain. This leads to low utilization and waste of system resources. Therefore, adaptive admission control is proposed to overcome this problem.

B. Adaptive Admission Control

This approach recognizes that the admission control can only approximately estimate the statistical multiplexing and attempts to use the characteristics of past traffic streams to better estimate the gain that can be achieved. Therefore, we argue that optimal bandwidth management scheme should employ adaptive admission. This is achieved by estimating the BEF adaptively for every time scale which is determined depending on the system conditions and also integrating monitored traffic measurements at the transmit queue into this estimation. Unlike the fixed admission control, the adaptive admission control adjusts the BEF such that the actual value of γ is close to the desired value that is restricted by the acceptable $\mathrm{QoS_T}$ limits. As in the fixed admission control, the control actions are activated when the inequality (1) is violated. Nevertheless, when low utilization of system resources is detected, adaptive control admission adjusts the BEF to correct the underestimation of statistical multiplexing gain. Furthermore, it is difficult to predict the statistical parameters of VBR traffic and the source processes might be nonstationary [8]. Therefore, the estimation of statistical multiplexing gain should be updated periodically.

Estimation of new average overall rates and corresponding BEF in adaptive admission control is determined as follows, in (5) and (6) at the bottom of the page, where B_p is the bandwidth required for the pending request. If B_p is less than the wasted fraction of the capacity due to underestimation of statistical multiplexing gain $(|\Delta\omega|)$, then the average overall rate of the services need not to be reduced. Otherwise, if $B_p > |\Delta\omega|$, then reduction in overall rate is necessary, as indicated in the second line in (5). Note that if the required reduction in the overall rate violates $\mathrm{QoS_T}$ limits, then the pending request is not admitted. Admission of a pending request is done every MTC period (every 30 s in the simulation) only if $\mathrm{QoS_T}$ of ongoing services are not violated due to congestion and channel problems.

Based on traffic and available resources, the BEF is initially estimated by resource allocation and it can be decreased if pending requests are admitted, as presented in (6). Whenever there are changes in traffic (service completion and new arrivals) the BEF is reestimated. On the other hand, if statistical multiplexing gain is overestimated and the adaptive control strategies satisfy (1) at the expense of $\mathrm{QoS_T}$ violation, then, in LTC, the BEF is increased resulting in the termination of an ongoing service.

The underlying concept of the adaptive approach is that the choice of γ does not significantly affect the ARAM system performance when LTC is employed [26]. This is because the adaptive control strategies mitigate error in statistical multiplexing gain by fairly and gracefully degrading $\mathrm{QoS_T}$ in small steps.

$$W_i^{\mathrm{new}} = \begin{cases} W_i^{\mathrm{old}} - \frac{\Delta\omega}{\mathrm{N}}, & \text{if } \gamma_{\mathrm{e}} > \gamma \\ W_i^{\mathrm{old}} - \max\left(0, \frac{B_p - |\Delta\omega|}{\mathrm{N}}\right), & \text{if } \gamma_{\mathrm{e}} < \gamma \quad \text{and there is a pending request} \\ W_i^{\mathrm{old}} + \frac{|\Delta\omega|}{\mathrm{N}}, & \text{if} \gamma_{\mathrm{e}} < \gamma \quad \text{and no pending request.} \end{cases} \tag{5}$$

$$\alpha^{\mathrm{new}} = \alpha^{\mathrm{old}} - \frac{B_p}{\mathrm{N}R_i^m} \tag{6}$$

For small values of $\gamma(10^{-4})$, request admission is more conservative and more services are admitted after examining the conditions under which the system operates (congestion and channel conditions). If the $\mathrm{QoS_T}$ of ongoing services is not violated, which is the typical case for low values of γ (due to underestimation of statistical multiplexing gain), then their $\mathrm{QoS_T}$ is marginally degraded to reallocate bandwidth for a pending request. For high values of $\gamma(>0.1)$, on the contrary, the ARAM system admits more requests, and then it tries to mitigate queue build ups due to the overestimation of the statistical multiplexing gain by adaptive rate (source and channel) control. Therefore, the ARAM system may override the admission control algorithm as the queue builds up for high probability values of γ. This further proves the need that the traffic resource management schemes and admission control algorithms should be harmonized to utilize the system resources. A similar harmony on this topic is presented in [21] and [22].

V. Simulation Results

To compare the fixed and adaptive admission control approaches, we have conducted two comprehensive sets of simulation experiments using a discrete event simulator [26]. In Section V-A, we present the first set aimed to analyze the simulation results for only VBR traffic with nonfading channel conditions. In Section V-B, we present the second set aimed for the sensitivity analysis of the simulator under varying traffic and channel conditions.

The ARAM simulator performs all the control actions discussed above and includes both ABR and VBR traffic generators. We use the source model presented in [29]. The model is based on the MPEG-1 coded *Starwars* movie statistics [28], on GOP level using a superposition of two first-order autoregressive processes with lognormally distributed noise sequences (2LAR). The 2LAR source model is also verified for several other MPEG empirical bitstreams including a twenty-four hours long MPEG-2 Cable TV bitstream [29]. Once the generated GOP sizes are determined using the 2LAR model, the corresponding frame sizes are extrapolated from the generated GOP sizes using the first order statistics of the empirical bitstream. Moreover, to simplify the analysis and get a better understanding of the performance results, we have chosen the same statistical characteristics for all VBR traffic. Table I presents the traffic characteristics used in the ARAM simulator. Unless a new arrival or service completion occurs, the adaptive control algorithms are initiated at the time scales of 1, 10, 30, and 90 seconds for STC, SMTC, MTC, and LTC, respectively.

The DBS channel is modeled by semi-Markov process (SMP) with two additive white Gaussian noise (AWGN) states, good and bad, with Rayleigh distributed transition times [26]. In order to combat channel errors, we adopted the FEC codes used in digital video broadcasting (DVB) standard [2] presented in Table II.

To examine the accuracy of the Gaussian approximation, as well as lognormal approximation, Monte Carlo simulation was used to obtain the probability of aggregate rate exceeding the link capacity using the 2LAR model [29]. Fig. 6 depicts the performance results for the number of aggregated bitstreams (admitted services). We observe that the both Gaussian and lognormal approximations are accurate for high probabilities $(>10^{-2})$. For low probabilities $(<10^{-2})$, on the other hand, the Gaussian approximation significantly over estimates statistical multiplexing gain and this error in estimation is larger for smaller number of admitted services. A detailed comparison of these approximations and their impact on admission control algorithms are presented in [29].

The following performance metrics were quantified by the simulation: the average DBS link utilization is calculated as the average transmitted information (bit/s)/link capacity (bit/s), the average number of active services is calculated as the average of number of services in progress per second, the total number of rejected services is calculated as the number of services rejected due to request time epoch, the fraction of dropped frames is calculated as the ratio of the number of dropped frames to the total number of frames assigned to be delivered, and the number of completed services.[2]

TABLE I
Traffic Characteristics of the ARAM Simulator

Traffic	ABR "data"	MPEG coded VBR "*Starwars*"
Generated traffic	Nominal Rate: 250kbps Minimum Rate: 50kbps	GOP pattern: IBBPBBPBBPBB. Frame duration: 1/24 seconds. Mean bitrate =187kbits/GOP, Standard Deviation =72kbits/GOP $\mathrm{QoS_T}$ limit: 25% of the nominal rate
RST	NA	250 seconds (or 6000 frames)
RTE	NA	250 seconds

$\mathrm{QoS_T}$ limit is the maximum allowable rate reduction. RST and RTE are the requested service time, and request time epoch, respectively, for each service.

TABLE II
Channel Characteristics of the ARAM Simulator

Channel States: Good and Bad (Semi-Markov Process)	$(Eb/No)_{good}$=5.3 dB, $(Eb/No)_{bad}$=3.6 dB $P_{good/good}$ =0.9, $P_{bad/bad}$ =0.85 T_{good} = 80 sec., T_{bad} = 35 sec.
DVB standard's FEC rates	1/2, 2/3, 3/ 4, 5/6, 7/8
Maximum Allowable FER	10^{-4}
DBS forward link capacity	22 Mbps

The channel transition probabilities are deliberately assigned as above such that multiple number of transitions occurs during the simulation.

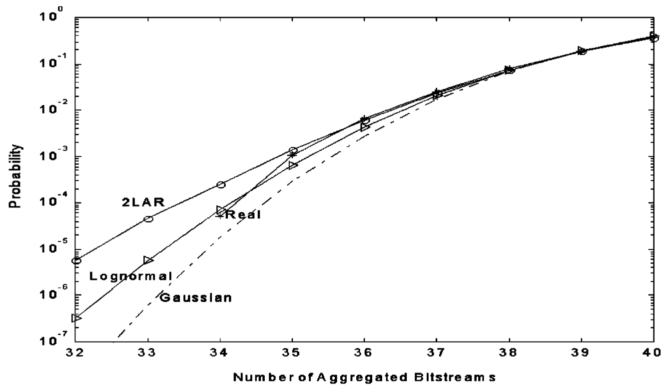

Fig. 6. Probability of aggregate rate exceeding the link capacity.

[2]When the simulation is ended, the services in progresses are aggregated based on their completion percentage and counted for the completed services.

Quality assessment of digital video sequences is a very important issue and has received large attention from the networking community. In order to quantify the QoS, in [1], we introduced the following definitions for QoS of both VBR ($\mathrm{QoS_T}$) and ABR services ($\mathrm{QoS_D}$)

$$\mathrm{QoS_T} = 1 - \frac{\Delta R_s + \Delta R_d + \Delta R_c}{R_n} \quad (7)$$

and

$$\mathrm{QoS_D} = 1 - \frac{\Delta R_s + R_c}{R_n} \quad (8)$$

where

- ΔR_s rate reduction of source rate for VBR and ABR;
- ΔR_d rate reduction of due to filtering for VBR;
- ΔR_c rate reduction due to lost/erroneous packets in the channel;
- R_n nominal rate.

ΔR_s accounts for changes in source rate due to congestion and/or FEC rate changes to keep the overall rate fixed, while ΔR_c accounts for sum of rates of all frames that are received with error or lost in the channel. The performance metrics are calculated for each service in every second. If a VBR service is terminated, the $\mathrm{QoS_T}$ is set to zero for the remaining service time since no frames are delivered for that service. Thus, given that control strategies maintain the delay and frame error rate within acceptable range, this metric captures the overall rate loss. In the case of VBR, for example, this can be considered as a measure of the respective perceptual video quality with more freedom of scaling. Relative to the original coded video sequence, $\mathrm{QoS_T}$ below 50% may be considered very annoying, 60% is annoying, 70% is slightly annoying, 80% is perceptible but annoying, and 90% is imperceptible. In the case of ABR service, $\mathrm{QoS_D}$ may be considered as normalized delay.

A. Set 1: Performances of Adaptive and Fixed Admission Controls

In the first set of simulations, to simplify the analysis and show the benefits of adaptive admission control strategy, the channel was in the good state throughout the simulation and only VBR traffic was considered. Moreover, since STC consists of primitive control actions, it is used as a baseline reflecting the traffic characteristics and the load on the system. We performed simulations for STC, LTC with fixed admission control, and LTC with adaptive admission control. The total simulation time for each set is 3000 s.

The simulation results are shown in Fig. 7 depicting the performance of each approach as a function of the predefined probability (γ) given in (1).

Since STC does not utilize any adaptive rate-control strategies except slowing down ABR traffic and dropping VBR frames, it implicitly uses a fixed admission control (constant BEF). On the other hand, constant BEF reflects changes in transmission attributes as a consequence of employing adaptive rate-control strategies (SMTC, MTC, and LTC) to overcome system overloads. The number of admitted (active) services is the same for both sets and the only difference is the use of adaptive rate control strategies in LTC with fixed admission control. Therefore, a comparison between STC and LTC with fixed admission control reflects the benefits of employing adaptive rate control strategies to mitigate congestion. For $\gamma < 10^{-2}$, adaptive rate control improves $\mathrm{QoS_T}$ less than 0.1% and decreases fraction of dropped frames by 0.3% but reduces utilization by 3%. On the other hand, for $\gamma > 10^{-2}$, adaptive rate control improves $\mathrm{QoS_T}$ by 2% and reduces fraction of dropped frames by more than 4% while decreasing the utilization by 4%. Therefore, utilizing adaptive rate control is beneficial when aggressive strategy of admission control is employed. This suggests that $\gamma \geq 10^{-2}$ is the desired operating region.

The comparison of LTC with fixed admission control and LTC with adaptive admission control shows that the adaptive admission control provides superior performance relative to the fixed admission control subject to a small degradation in $\mathrm{QoS_T}$. For all probabilities (γ), the ARAM system integrated with the adaptive admission control increases system throughput (number of completed services) by 18%, and utilization by 9% at the expense of decreasing $\mathrm{QoS_T}$ by less than 1.4% and fraction of dropped frames by less than 1%. Moreover, the worst throughput performance in terms of completed services of the adaptive approach (for the most restrictive value of γ equal to 10^{-5}) exceeds the best performance for the fixed approach (least restrictive value of γ equal to 0.4). Similarly, the worst utilization for the adaptive approach nearly exceeds the best utilization for the fixed approach. Therefore, by adaptively changing the BEF, the system throughput and utilization can be significantly increased at the expense of subtle degradations in $\mathrm{QoS_T}$.

The adaptive admission control approach rejects far fewer services than the fixed approach because the adaptive admission control utilizes system resources in a more efficient way than the fixed admission control by admitting more services. However, the fixed admission control approach drops marginally fewer frames than the adaptive admission control approach. Because, in the fixed admission control transmission attributes (source and FEC rates) are adjusted only when the inequality (1) is violated and this results in lower utilization than for the adaptive admission control approach. Nevertheless, these results are obtained for a good channel state with no ABR service. The benefits of the adaptive rate control are more prominent in the presence of channel state variations and including ABR services.

B. Set 2. Sensitivity Analysis and Further Issues

The ARAM system has several parameters and strategies that play key roles in terms of the performance measures and thus, their sensitivity analysis are required to verify the previous achievements. These are VBR source parameter for maximum allowable degradation defined by minimum $\mathrm{QoS_T}$, ABR/VBR load ratio in total traffic, sensitivity of time scales, channel fading dynamics, etc. We have performed extensive set of simulations and experimental results for the verification purposes [26]. In this section, we present a set of simulations that is aimed for the sensitivity analysis. In this set, the ARAM

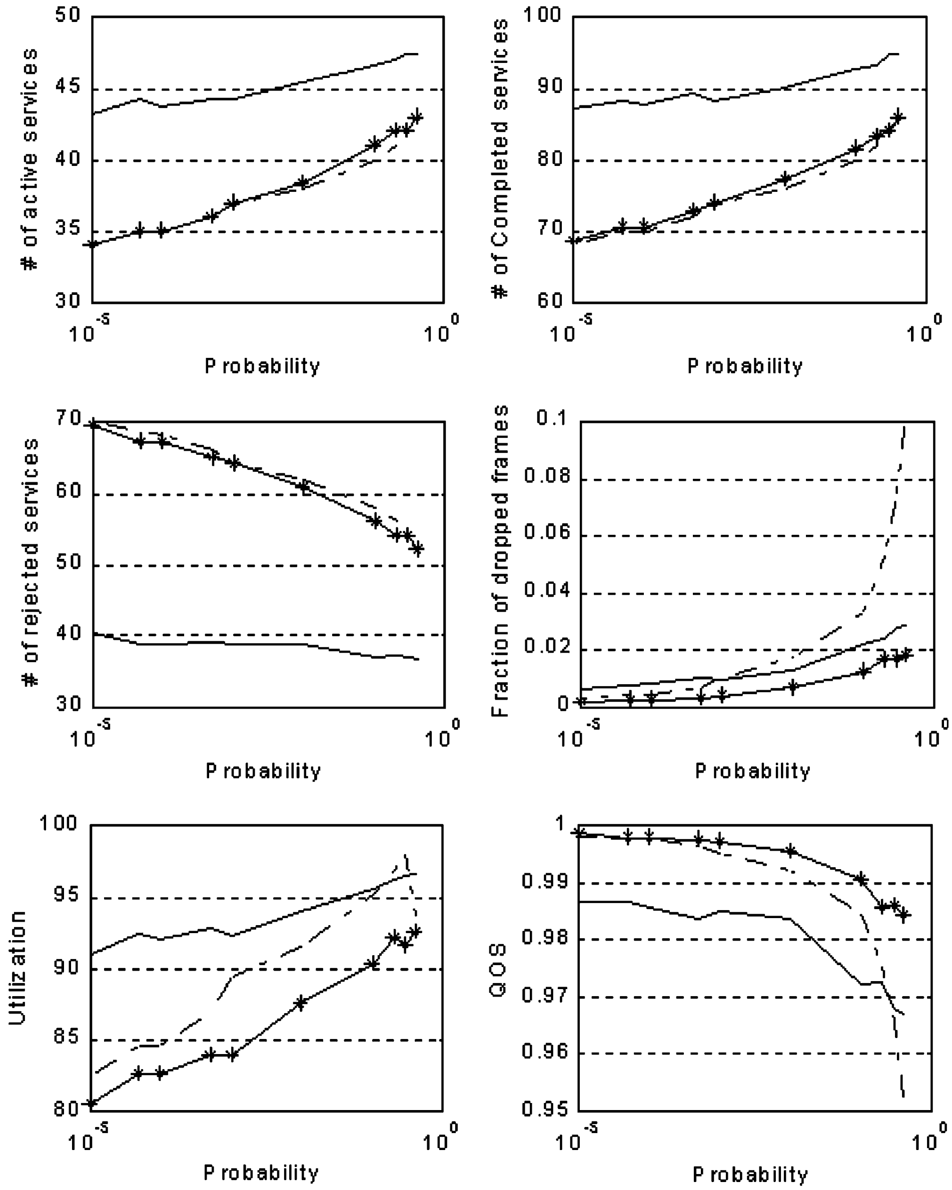

Fig. 7. Simulation results. Legend: STC (dashed), fixed admission control (star), adaptive admission control (line). The term "probability" labeled on the x axes of all figures implies the predefined probability value (γ) given in (1).

simulator is conducted by running simulation five times for the same traffic (VBR and ABR traffic) and channel (good and bad channels) characteristics, given in Tables I and II, respectively, and total duration of 3000 s. In the first simulation, no adaptive rate control is employed, the "no-control" (NC) case, in which the aggregate instantaneous rate of the admitted services is constrained to be less than or equal to the link capacity, by dropping the most recently admitted services, in the case of congestion. This criterion is chosen as a base line for comparison purposes. We tested STC, SMTC, MTC, and LTC to show the benefits of each of the time scaled control strategies discussed above.

Table III presents the statistics of the simulations for the five simulations, respectively. During the simulation, the channel initially starts in the good state and changes to the bad state several times: at time 601 to 950, 1819 to 1960, and 2101 to 2200 s, respectively. The simulation results are obtained for 62 ABR and 563 VBR traffic arrivals.

1) Overview of the Simulation Results: STC and SMTC are the primary control strategies for congestion problems. We observed that while the former helps managing real-time events as they occur, the latter smoothes out the statistical fluctuations in MPEG video traffic. In addition to preventing the congestion, MTC helps managing the admission control and resolves temporary channel problems. While LTC gives the last shape of the control strategies by completely resolving the channel problems and maintaining the $\mathrm{QoS_T}$ with an efficient DBS link utilization. All control strategies achieve much higher utilization than NC, yet with considerably smaller number of terminated VBR services. This is achieved with occasional dropping of VBR frames during congestion intervals. However, this represents relatively small fraction (3–6%) of the total number of frames delivered and, correspondingly, will have negligible impact on perceptual video quality. Also, the average number of admitted (active) and total number of completed services is larger when control strategies are employed. Evidently, the con-

TABLE III
SUMMARY OF SIMULATION STATISTICS FOR 563 VBR AND 62 ABR SERVICE ARRIVALS

	NC	STC	SMTC	MTC	LTC
The number of admitted ABR services					
Mean	0.55	2.99	2.82	1.58	1.39
Standard Deviation	0.92	1.96	1.59	1.30	1.32
The number of					
admitted VBR services	32.86	36.98	37.11	38.9	37.6
Mean	4.39	1.15	0.94	4.35	3.38
Standard Deviation					
Total number of Completed ABR services	62	58	59	60	60
Total number of Completed VBR services	293	352	357	353	351
Total number of rejected VBR requests	0	88	83	43	59
Total number of terminated VBR services	211	23	28	62	53
Average fraction of dropped frames (%)	NA	5.2	3.7	2.9	3.8
DBS link utilization (%)					
Mean	86.0	97.1	96.9	94.8	95.1
Standard Deviation	10.80	3.77	3.72	5.43	5.04
Percentage of time VBR services using FEC rate of:					
1/2	0	0	0	3.1	0.1
2/3	0	0	0	21.2	40.6
3/4	100	100	100	16.7	15.6
5/6	0	0	0	14.2	18.3
7/8	0	0	0	44.8	25.4
Average QoS_D	1	0.265	0.202	0.445	0.481
Average QoS_T	0.474	0.464	0.531	0.888	0.934
QoS_T Statistics:					
(only completed services)	0.811	0.793	0.785	0.907	0.950
Mean	0.340	0.340	0.337	0.163	0.061
Standard Deviation					
Mean (Good Channel)	0.999	0.980	0.970	0.958	0.959
Mean (Bad Channel)	0.206	0.192	0.188	0.743	0.922
QoS_T Statistics: (with terminated services)					
Mean	0.631	0.746	0.734	0.797	0.850
Standard Deviation	0.279	0.321	0.320	0.176	0.080
Mean (Good Channel)	0.777	0.919	0.906	0.858	0.862
Mean (Bad Channel)	0.161	0.188	0.180	0.603	0.811

trol strategies enable utilization of the statistical multiplexing gain to the maximum possible extent, with graceful degradation in $\mathrm{QoS_T}$ during congestion and bad channel conditions. Moreover, Table III shows that by integrating all the adaptive rate control mechanisms we can achieve and maintain high $\mathrm{QoS_T}$ and increase the number of completed services in the presence of traffic, and channel variations.

2) Admission Control: The effectiveness of admission control strategy also depends on acceptable $\mathrm{QoS_T}$ limits (up to 25% of the nominal rate is assumed as the maximum allowable degradation in $\mathrm{QoS_T}$). The adaptive admission control algorithm is intentionally designed to apply an aggressive multiplexing strategy that might often lead to overloads due to misestimated soft margin, which is mitigated by employing STC and SMTC strategies. As shown in Table III, the average link utilization for control strategies is at least 10% higher than that for NC. More importantly, when we compare the average number of admitted (services in progress), total number of rejected, total number of terminated, and total number of completed services for these simulations, we conclude that the proposed adaptive admission control strategy works successfully.

3) Congestion Effects: We analyze the performance results for the NC, STC, and SMTC simulations. These strategies do not respond to the channel problems, thus, we can only resolve the congestion problems. Due to error in estimation of statistical multiplexing and the burstiness of generated VBR traffic, the congestion occurs during heavy loads. The highest number of suspended services is observed in NC, 211 VBR services, whereas it is 23 and 28 for STC and SMTC, respectively. It is also observed that NC has the worst performance for VBR services, in terms of the average number of admitted and total number of completed services and average DBS forward link utilization. For example, when STC is employed, a significant improvement in average utilization (13% more than for NC) and the number of completed VBR services (59 more services than for NC) is apparent in Table III. Although this is achieved with occasional frame dropping, with rate of 0.052 during congestion intervals, SMTC reduces it to 0.037 by handling the statistical fluctuations more effectively. On the other hand, the average delay for ABR services is zero (i.e., $\mathrm{QoS_D}$ is one) for NC, whereas it is 0.265 and 0.202 for STC and SMTC, respectively. This is because the ABR services in NC are not delayed during congestion intervals, while the first stage of the control strategy for STC and, thus, for SMTC was to delay the ABR services.

When we compare the average $\mathrm{QoS_T}$ for VBR metric presented in Table III, we observe that for SMTC and STC they are close to each other 0.74, and they are about 17% superior to that of NC. However, the average $\mathrm{QoS_T}$ values will be misleading unless we observe the fluctuation in the results. STC and SMTC outperform NC in the good channel environments, while they all suffer in bad channels. Moreover, the average $\mathrm{QoS_T}$ values for these simulations are around 0.2 in the bad channels, which is due to the initially assigned FEC coding rate of 3/4. If one initially assigns more powerful FEC rates, the average $\mathrm{QoS_T}$ value may change (increase or decrease), yet the numbers of both admitted and completed services decrease. It is worth reminding that the excessive FEC overhead may cause low $\mathrm{QoS_T}$ as a result of more packet loss due to congestion in the network [1].

4) Channel Effects: We analyze the impact of channel on MTC and LTC strategies. In MTC, we resolve channel and congestion problems jointly by integrating source and channel rate adaptations with the admission control strategy. The $\mathrm{QoS_T}$ for VBR increases gradually in the bad channel period by gradually improving the FEC rate in MTC. When a bad channel state lasts duration of several QoS reports, MTC fails to respond this scenario since MTC may not respond to this variation unless physical layer (signal strength) estimates are not gathered. LTC reduces the degradation of $\mathrm{QoS_T}$ by being able to exploit more information in system optimization.

Finally, from Table III, we can see that by providing more information to the system the FEC rate distribution becomes more peaked, i.e., it exhibits less randomness. Consequently, the average $\mathrm{QoS_T}$ is improved with slight degradation in the total number of completed, rejected and dropped services.

In this paper, we assume that the changes in the source rates and FEC rates are linear functions of $\mathrm{QoS_T}$. For practical purposes, this would be reasonable only if the changes are within a small threshold. We may need to investigate a new QoS metric at the application layer that integrates both source and FEC changes based on experimental evaluations. Both Gaussian and lognormal approximations for QoS threshold was only acceptable for high probabilities $(>10^{-2})$, however, when we want to operate at lower probabilities, we need to develop more accurate aggregate traffic approximations [29]. Finally, the channel dynamics and resolving the channel problems play an important role in determining the number of admitted services. Unfortunately, the ARAM system can respond to the channel problems caused by the slow fading (shadowing) channel. Unless a proactive strategy at the expense of waste of resources is integrated into the ARAM system [26], the problems caused by the fast fading channel can not be combated due to the latency of the return link messages.

VI. Conclusion

In this paper, we explore the admission control strategies for a DBS-RCS. The system supports ABR and real-time VBR traffic mixtures and operates under dynamic channel conditions. The ARAM system is introduced to manage the DBS-RCS. We proposed a new admission control strategy that adaptively estimates the bandwidth expansion factor that determines the number of admitted services and integrated it into the ARAM system.

We have performed extensive set of simulations. The simulation results show that the performance of the adaptive admission control is superior to the traditional (fixed) admission control strategy in terms of the performance measures. Since the ARAM system has several parameters and strategies that play key roles in terms of the performance measures, their sensitivity is studied to verify the above foundations for different traffic and channel conditions.

Acknowledgment

The authors would like to thank the anonymous reviewers for their very useful comments and D. Lee for his major contribution in building the testbed.

References

[1] F. Alagöz, A. AlRustamani, B. Vojcic, R. Pickholtz, and D. Walters, "Adaptive rate control and QoS provisioning in direct broadcast satellite networks," *ACM Wireless Networks*, vol. 7, no. 3, pp. 269–281, 2001.

[2] DVBRCS001, Interaction channel for satellite distribution systems, [Online]. Available: http://www.etsi.org, ETSI draft EN 301 790, Apr. 2000. Rev. 14.

[3] D. J. Bem, T. W. Wieckowski, and R. J. Zielinski, "Broadband satellite systems," *IEEE Commun. Surveys Tutorials*, vol. 3, no. 1, 2000.

[4] J. Neale, R. Green, and J. Landovskis, "Interactive channel for multimedia satellite networks," *IEEE Commun. Mag.*, vol. 39, pp. 192–198, Mar. 2001.

[5] A. Jamalipour, "Satellites in IP networks," *Wiley Encyclopedia Telecommun.*, Jan. 2003.

[6] I. F. Akyildiz and S. H. Jeong, "Satellite ATM networks: A survey," *IEEE Commun. Mag.*, vol. 35, pp. 30–43, July 1997.

[7] B. Haskell, A. Puri, and A. Netravali, *Digital MPEG: An Introduction to MPEG-2*. London, U.K.: Chapman & Hall, 1997.

[8] M. Krunz, "Statistical multiplexing," in *Encyclopedia of Electrical and Electronic Engineering*, J. Webster, Ed. New York: Wiley, Apr. 1999, vol. 20, pp. 479–492.

[9] M. Zorzi, "Performance of FEC and ARQ error control in bursty channel under delay constraints," in *Proc. IEEE VCT98*, Ottawa, ON, Canada, May 1998, pp. 1390–1394.

[10] M. Naghshineh and M. Willebeeklemair, "End-to-end QoS provisioning in multimedia wireless/mobile networks using an adaptive framework," *IEEE Commun. Mag.*, vol. 35, pp. 72–81, Nov. 1997.

[11] J. M. Peha and A. Sutivong, "Admission control algorithms for cellular systems," *ACM Wireless Networks*, vol. 7, no. 2, pp. 117–125, 2001.

[12] E. W. Knightly and N. B. Shroff, "Admission control for statistical QoS: Theory and practice," *IEEE Network Mag.*, vol. 13, pp. 20–129, Apr. 1999.

[13] C. Oliveira, J. B. Kim, and T. Suda, "An adaptive bandwidth reservation scheme for high-speed multimedia wireless networks," *IEEE J. Select. Areas Commun*, vol. 16, pp. 858–874, Aug. 1998.

[14] H. Zhang and E. Knightly, "Integrated and differentiated services for the Internet," *IEEE Network*, vol. 13, pp. 7–8, Sept.–Oct. 1999.

[15] B. Pang, H. Shao, W. Zhu, and W. Gao, "An admission control scheme to provide end-to-end statistical QoS provision," in *Proc. 21st IEEE Int. Performance, Computing, Communications Conf. (IPCCC 2002)*, Phoenix, AZ, Apr. 2002, pp. 399–403.

[16] J. Qiu and E. Knightly, "Measurement-based admission control using aggregate traffic envelopes," *IEEE/ACM Trans. Networking*, vol. 9, pp. 199–210, Apr. 2001.

[17] K. D. Lee and S. Kim, "Optimization for adaptive bandwidth reservation in wireless multimedia networks," *Computer Networks*, vol. 38, no. 5, pp. 631–643, 2002.

[18] S. Bohm, A. K. Elhakeem, K. M. S. Murthy, M. Hachicha, and M. Kadoch, "Analysis of a movable boundary access technique for a multiservice multibeam satellite system," *Int. J. Satellite Commun.*, vol. 12, pp. 299–312, 1994.

[19] H. Koraitim and S. Tohm'e, "Resource allocation and connection admission control in satellite networks," *IEEE J. Select. Areas Commun.*, vol. 17, pp. 360–372, Feb. 1999.

[20] R. Q. Hu, J. Babbitt, H. Abu-Amara, C. Rosenberg, and G. Lazarou, "Call admission control in a multi-media multi-beam satellite cross-connect network," presented at the IEEE ICC 2003, Anchorage, AK, May 2003.

[21] A. Iera, A. Molinaro, and S. Marano, "Call admission control and resource management issues for real-time VBR traffic in ATM-satellite networks," *IEEE J. Select Areas Commun.*, vol. 18, pp. 2393–2403, Nov. 2000.

[22] ——, "Traffic management techniques to face the effects of intrinsic delays in geostationary satellite networks," *IEEE Trans. Wireless Commun.*, vol. 1, pp. 145–155, Jan. 2002.

[23] T. Zein, G. Maral, T. Brefort, and M. Tondriaux, "Performance of the combined/fixed reservation assignment (CFRA) scheme for aggregated traffic," in *Proc. COST 226 Integrated Space/Terrestrial Networks Final Symp.*, Budapest, Hungary, May 1995, pp. 183–199.

[24] D. P. Connors, B. Ryu, and S. Dao, "Modeling and simulation of broadband satellite networks part 1: Medium access control for QoS provisioning," *IEEE Commun. Mag.*, vol. 37, pp. 72–79, Mar. 1999.

[25] G. Açar and C. Rosenberg, "Performance study of end-to-end resource management in ATM geostationary satellite networks," in *Proc. IEEE MILCOM 2001*, vol. 2, McLean, VA, Oct. 2001, pp. 764–769.

[26] F. Alagöz, "Managing Heterogeneous Traffic in One-to-Many Links Wireless Networks," D.Sc dissertation, The George Washington Univ., Washington, D.C., Jan. 2000. 20 052.

[27] D. Walters, "Reliable multicast in the digital battlefield," presented at the SPIE Aerosense, Orlando, FL, Apr. 1998.

[28] M. W. Garret and M. Vetterli, "Joint source/channel coding of statistically multiplexed real time services on packet networks," *IEEE/ACM Trans. Networking*, vol. 1, pp. 71–80, Feb. 1993.

[29] F. Alagöz, "Approximations on the aggregated MPEG traffic and their impact on admission control," *Turkish J. Elec. Eng. Comput. Sci.*, vol. 10, pp. 73–84, 2002.

Fatih Alagöz (M'01) received the B.Sc.degree in electrical engineering from the Middle East Technical University, Gazi Antep, Turkey, in 1992, and the M.Sc. and D.Sc. degrees in electrical engineering from The George Washington University, Washington, DC, in May 1995 and January 2000, respectively.

He is an Assistant Professor in and Chairman of the Department of Electrical and Electronics Engineering, Harran University, Şanlıurfa, Turkey. During 1993, he was a Research Engineer in a missile manufacturing company, Muhimmatsan AS, Turkey. From September 2001 to June 2003, he was with the Department of Electrical Engineering, United Arab Emirates University, AlAin. He has edited two books and published more than 30 scholarly papers in selected journals and conferences. His research interests are in the areas of wireless/mobile/satellite networks, teletraffic modeling, performance evaluation and modeling of terrestrial and satellite mobile communications, neural networks, and multiuser detection.

Dr. Alagoz is a Member of the Satellite and Space Communications Technical Committee. He has numerous awards including the YOK Scholarship achieved based on a nationwide selection examination and the Association for Computing Machinery (ACM) MSWIM'99 Best Paper Award.

Branimir R. Vojcic (M'86–SM'96) received the Diploma in electrical engineering, and the M.S. and D.Sc. degrees in 1980, 1986, and 1989, respectively, from the Faculty of Electrical Engineering, University of Belgrade, Yugoslavia.

In 1986 and 1987, he was a Visiting Scholar at The George Washington University(GWU), Washington, DC. Since 1991, he has been on the Faculty of GWU, where he is a Professor in and Chairman of the Department of Electrical and Computer Engineering. His current research interests are in the areas of communication theory, performance evaluation and modeling of terrestrial and satellite mobile communications, spread spectrum, multiuser detection, adaptive antenna arrays, and wireless data networks. He has also been an industry consultant in these areas.

Dr. Vojcic was an Associate Editor for the IEEE COMMUNICATIONS LETTERS. He was Vice-Chairman of Washington and Northern Virginia IEEE Communication Society Chapters. He is a recipient of the 1995 National Science Foundation CAREER Award.

David Walters (M'79) received the Ph.D. degree in operations research from the University of California, Berkeley.

He has over 20 years of experience working on the design and analysis of packet and circuit switched networks using both satellite and terrestrial infrastructure. Currently, he leads the development of routing and protection algorithms in λ Optical Systems, a new startup optical networking company.

Amina AlRustamani (S'93–M'96) received the B.Sc, M.Sc., and D.Sc. degrees in electrical engineering from The George Washington University, Washington, DC, in 1993, 1996 and 2001, respectively.

She is with Dubai Internet City, Dubai, AlAin. Her areas of interest are spread spectrum, multiuser detection and wireless/mobile networks.

Raymond L. Pickholtz (S'54–A'55–M'60–SM'77–F'82–LF'96) received the Ph.D. degree in electrical engineering from the Polytechnic Institute, Brooklyn, NY, in 1966.

He is a Professor in and former Chairman of the Department of Electrical Engineering and Computer Science at The George Washington University, Washington, DC. He was a Researcher at RCA Laboratories, and ITT Laboratories. He was on the Faculty of the Polytechnic Institute and Brooklyn College, Brooklyn, NY. He was a Visiting Professor at the Universite' du Quebec, Montreal, Canada, and the University of California. He is a Fellow of the American Association for the Advancement of Science (AAAS). He was a Guest Editor for special issues on *Computer Communications*, *Military Communications Spread Spectrum Systems*, and *Social Impacts of Technology*. He is Editor of the *Telecommunication Series* for Computer Science Press. He has published scores of papers and holds six U.S. patents.

Dr. Pickholtz was an Editor of the IEEE TRANSACTIONS ON COMMUNICATIONS. He was elected a Member of the Cosmos Club and a Fellow of the Washington Academy of Sciences in 1986. In 1984, he received the IEEE Centennial Medal. In 1987, he was elected as Vice President, and in 1990 and 1991 as President of the IEEE Communications Society. He received the Donald W. McLellan Award in 1994. He was a visiting Erskine Fellow at the University of Canterbury, Christchurch, New Zealand, in 1997. He was awarded the IEEE Millenium Medal in 2000.