On the Effects of Traffic Mixtures in Direct Broadcast Satellite Networks

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ABSTRACT

In this paper we present a multi-criterion control simulation in a realistically complex environment of a satellite network, involving non-symmetric up and downlinks. Direct Broadcast Satellite (DBS) networks carrying heterogeneous traffic is characterized with challenges, such as high traffic burstiness, wireless channel dynamics, and large, but limited capacity. On the other hand, there are system characteristics that can be leveraged to address these challenges such as in centralized topology, different levels in Quality of Service (QoS) and priorities, availability of side information about channel conditions, flexibility in delivery of delay insensitive traffic, etc. We have developed an Adaptive Resource Allocation and Management (ARAM) system that takes the advantage of such characteristics to maximize the utilization of the available capacity on the forward DBS link, while maintaining QoS in the presence of channel effects and congestion in the network. Since variable-bit-rate (VBR) video traffic is given priority over available-bit-rate (ABR) data traffic in the ARAM concept, in this paper we investigate the impact of the fraction of VBR load in overall load.

Keywords- video and data traffic mixture, wireless channel, satellite networks, link utilization, QoS.

I. INTRODUCTION

In Direct Broadcast Satellite (DBS) networks, the channel quality, topology and traffic mixtures change dynamically, suggesting the need for network management algorithms such that Quality of Service (QoS) can be maintained within a desired range. New generation satellite networks supporting real-time multimedia services also faces this dynamic problem. This paper presents a multi-criterion control simulation in a realistically complex environment of a DBS network, involving non-symmetric up and downlinks. Specifically, Motion Pictures Expert Group (MPEG) coded variable bit rate (VBR) traffic, which is part of DBS traffic mixture, causes the dynamics in the traffic, and varying atmospheric conditions causes the dynamics in the satellite channel.

Due to the high peak-to-mean rate ratio (burstiness) of VBR sources generated at the output of the MPEG encoder, the peak-source-rate allocation results in low utilization for a given link capacity. We rely on statistical multiplexing to achieve utilization corresponding to somewhat higher than the sum of the mean rates, approximately. During peak aggregate rate intervals, however, multiplexing results in congestion in the transmit queue, and, consequently, the QoS may be violated by dropping some packets. To alleviate this problem, buffers may be included within the network that act to smooth out these traffic peaks with an additional cost of end-to-end delay for real-time video applications. However, a possibility of buffer overflow remains and packets will be lost and/or dropped.

The problem becomes more challenging when the transmission of this VBR traffic is required on a satellite channel, which suffers from random variability in quality due to atmospheric conditions such as slow fading caused by rain. This means channel problems must be included in the network design to combat or mitigate their effect on QoS. Traditionally, in packet-switched data networks, if full duplex communication is possible, error control may be performed using an automatic repeat request (ARQ) protocol even for limited real-time applications as in [1]. Also, in the wireless environments, if the receiver and transmitter are close enough (where instantaneous feedback is available), ARQ may become a feasible solution for the error recovery [2]. However, in a satellite application with a 250ms of forward link propagation delay, the techniques of packet retransmission are not suitable for the transmission of real-time video traffic. While the error concealment is a possible technique for error recovery of the digital video, the most common technique to resolve the channel problems is to use the forward error correction (FEC) noting that error concealment techniques may be employed on top of FEC mechanisms. However, there are two antagonistic effects that may be produced by employing FEC mechanism. The FEC enables erroneous packets to be recovered, or the FEC causes further congestion and more lost packets [3].

In order to find a feasible solution to the above challenging problem, we have developed Adaptive Resource Allocation and Management (ARAM) system [4]. It attempts to maximize statistical multiplexing gain, under varying channel and traffic conditions, by distributing the rate reduction during congestion evenly among services, thus, providing graceful degradation during overload intervals. By using QoS reports and transmit queue monitoring, the ARAM system utilizes adaptive control of transmission attributes (FEC and source rates).

Due to low latency requirements of the VBR video services, ARAM system may give higher priority during transmission as compared to available-bit-rate (ABR) services. This is possible at the expense of delaying the ABR services by reducing the nominal ABR packet rate. Therefore, we explore the impact of the VBR traffic mixture in overall load and their respective QoS measure.

This paper presents the extended version of the work in [5]. The scope of this paper encompasses a description of the system architecture envisioned for deployment of techniques, technical description of ARAM and the heuristic source/FEC rate algorithm, summary of simulation results, and conclusions derived from this work. It is organized as follows. Section 2 briefly describes the general concept of the ARAM system. Section 3 presents VBR load mixture in overall load and channel error correction mechanism. Section 4 presents the simulative works undertaken to investigate the impact of the mixture of the VBR load in overall load. Section 5 summarizes the conclusions.

II. BACKGROUND

We have developed the Adaptive Resource Allocation and Management (ARAM) system for heterogeneous traffic loaded DBS networks. ARAM system integrates a series of algorithms using all the layers of the wireless network in a manner that to achieve high system throughput while maintaining the QoS both at network (minimum packet loss, delay jitter etc.) and application (Signal-to-Noise Ratio) level.

The ARAM system architecture and the physical location of the ARAM system in the DBS topology are portrayed in Figures 1 and 2, respectively. It operates as follows: Once the services are admitted by the Adaptive Resource Allocation (ARA), according to the available resources and estimated statistical multiplexing gain, the Transmit Queue Monitor (TQM), continuously monitors the transmit queue and delay variations of ongoing services. This provides a basis for reallocation of resources and adaptation of transmission attributes (source and channel rates). Also, the QoS Report Processing (QRP) examines the QoS of ongoing services as explained in the next section. These reports are forwarded to the ARAM system by the:

- DBS Field Terminal (DFT) proxies and end-receivers as a part of return link messages and
- uplink router that is monitoring of transmit facilities.

The Control Mechanism (CM), coordinates control and adaptation actions, handles adjustment of QoS parameters and service attributes in the system, and supports command transfers between the ARAM system components. The adaptation of service attributes is performed in the Source & Channel Rate Decision (SCR) block as a response to inquiries from the CM.

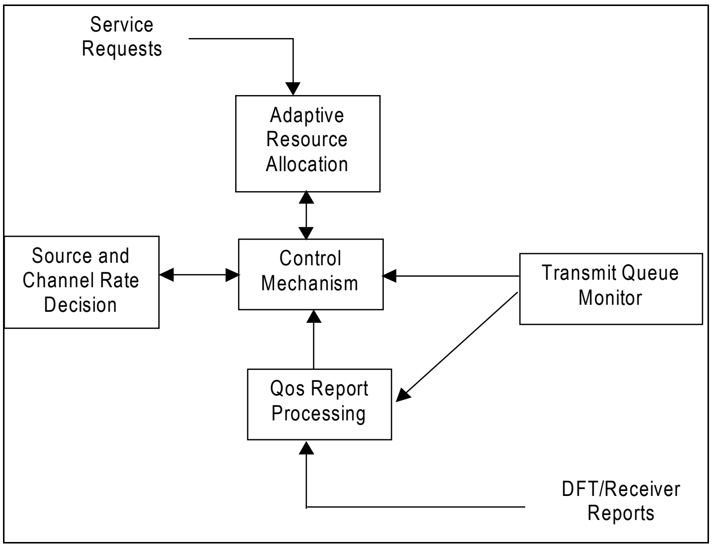

Figure 1. ARAM System Architecture.

The ARAM concept for multiplexing the available bit rate (ABR) and VBR traffic allows for an arbitrary mix of ABR and VBR traffic. Figure 3 depicts the time phased control strategies utilized by ARAM. 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 include Requests Waiting for Service, Queue Sizes, Error Rates, and Jitter Rates.

In response to problems identified by analyzing these symptoms, ARAM uses 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 Figure 3, each control strategy may utilize one or more of the following adjustments: video compression rates, data transmission rates, FEC rates and Resource Allocation.

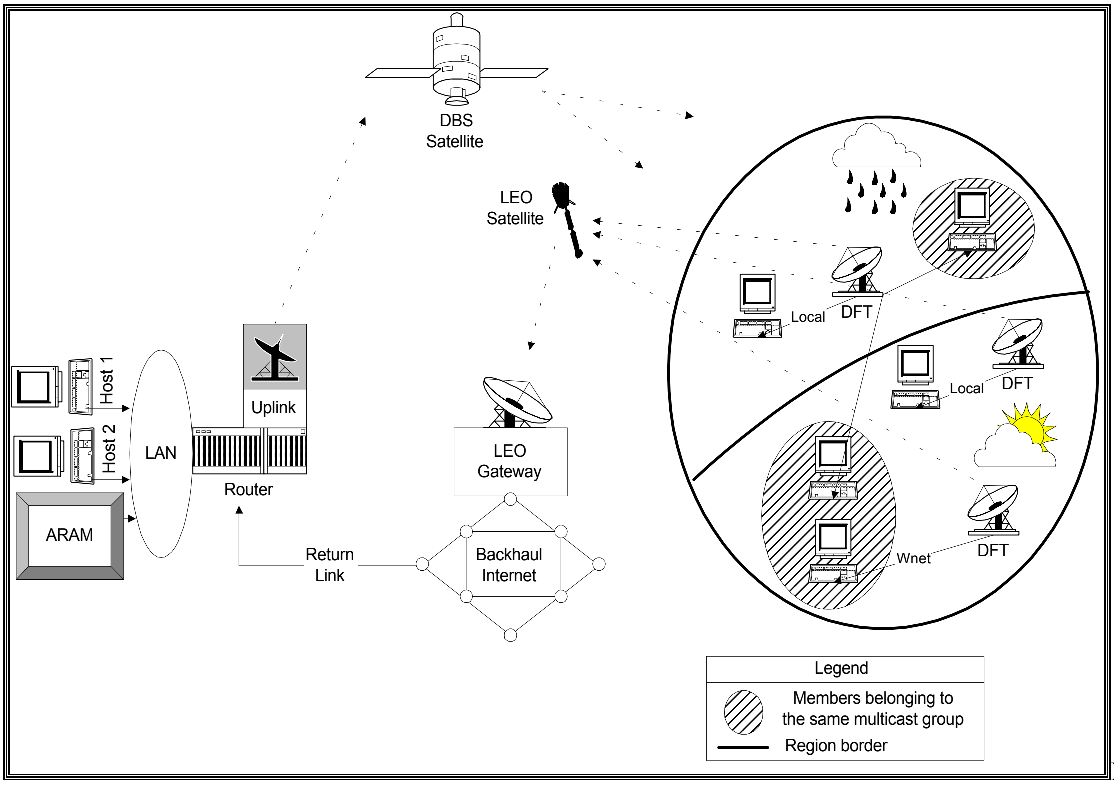

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Figure 2. ARAM Control Algorithm in The DBS Topology.

After resources are assigned to services, short-term control actions are limited to queue management that implements rules for prioritizing packets. In ARAM system, VBR packets are given priority over ABR packets. Also, MPEG I packets are given priority over MPEG P or B packets due to the MPEG coding scheme [6]. For intermediate time periods, ARAM adapts source rate adjustments for both VBR and ABR services and adjusts the FEC coding rate to accommodate small changes in conditions. Long-term control is intended to accommodate bigger changes in network conditions. In this case, ARAM may have to re-allocate resources if network performance degrades below the desired QoS, i.e., drop (terminate) services.

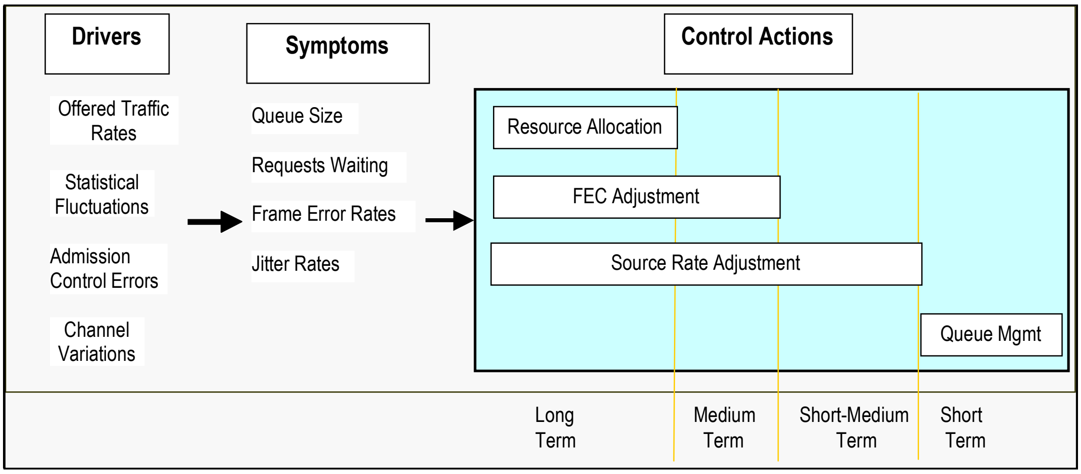

Figure 3. ARAM Algorithmic Concept.

III. SYSTEM MODELS

A. VBR Traffic Mixture

ARAM system can support available bit rate (ABR), constant bit rate (CBR) and variable bit rate (VBR) traffic classes defined by the ATM traffic forum. Although CBR and ABR traffic classes can be characterized deterministically, VBR traffic requires a statistical model. We showed in [7] that the individual VBR streams can be modeled using the mixture of two first order autoregressive processes as follows: Due to periodic refreshment of the MPEG encoding scheme, we model the source on a group of picture (GOP) level (the time interval for one period) using a superposition of two first-order auto-regressive, AR(1), processes; one captures the short, and the other captures the long term dependencies. An AR(1) model of process {X(k)} is used to obtain the number of bits in the k-th GOP,{Y(k)}, generated by an MPEG codec can be expressed as

$$X_i(k) = a_i X(k-1) + b_i w_i(k) \quad \mathrm{i} = 1{,}2$$

$$Y(k) = \sum_{l=1}^{2} f_l X_l(k)$$

where a_i, b_i, and f_l ‘s are coefficients and $w_i\ (k)$ is lognormally distributed noise sequences with mean m_i and variance s_i [8]. In order to test our model to emulate video data with different parameters in a wide range, we used the actual statistics of the *Starwars* movie and the coefficients of the traffic models were chosen to fit the actual GOP level autocovariance function [4]. The following values are found in the model: a_1=.9935, a_2=.6, b_1=.0011, b_2=2.2, f_1=.5 and f_2=.5.

Therefore, in the admission control, 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 specified threshold γ:

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

where, N is number of admitted VBR services, R_i is instantaneous rate of the *i*th VBR service, B_T is fraction of the capacity assigned to VBR services, α is bandwidth expansion factor, R_i^m is average rate of the *i*th VBR service, and $f_x(x)$ is the density function of the aggregate traffic rate. In order to represent the aggregated VBR traffic, we used a Gaussian approximation. We showed in [4,7] that this approximation is somewhat acceptable for superposing a moderate-to-large number of aggregated VBR sources.

In the algorithmic ARAM concept, VBR traffic is given priority due to its delay sensitivity requirements. Therefore, the VBR load in the overall load has impact on the system performance. We simulated several scenarios to present the effect of the ABR and VBR traffic mixture on the system performance using ARAM simulator. We should note that in addition to the VBR traffic characteristics, probability threshold (γ) and the capacity assigned for VBR traffic (B_T) plays a crucial role in the system performance.

B Adaptive FEC Mechanism

In order to combat channel problems, the FEC codes used in Digital Video Broadcasting (DVB) standard are considered [9]. The DVB standard is recently implemented in Europe on the basis of MPEG-2 and employs FEC codes. These are concatenated codes using rate 1/2 convolutional code as the inner code and a (204,188) Reed Solomon code as outer code, whereby the inner code can be punctured to obtain higher rate codes. The performance of probability of bit error rate versus signal to noise ratio can be found followed by [10] and corresponding frame error probabilities for specific frame sizes can be achieved following [11].

The digital video may be completely degraded when the signal falls below some threshold that may be adjusted by varying the rate of forward error correction, by modifying the symbol rate, or by adjusting power levels of the signals. In this work, we examine the system under two-state (good and bad) AWGN channel environment to easily observe the effect of this degradation. Therefore, we model the DBS channel as Semi-Markov process (SMP) with Rayleigh distributed transition-state times that is used to have more flexibility in determining the period of each transition which alleviates the problem of sudden channel transition [8]. The model characteristics are given in Table 1, the states are named (good or bad) based on their performance results.

Channel characteristics	
Channel Process: Semi-Markov Process	$(Eb/No)_{good}$=5.3 dB, $(Eb/No)_{bad}$=3.6 dB , Transition-state Probability: $P_{good/good} = 0.9$ Transition-state Probability: $P_{bad/bad} = 0.85$
DVB FEC rates	1 / 2, 2 / 3, 3 / 4, 5 / 6, 7 / 8

Table 1. Channel Characteristics Used in Simulations.

As we can observe from Table 2, the difference in good and bad channel state (1.7dB) may represent a scenario of heavy rain at which the approximate rainfall attenuation at 12.5 GHz, given that one mile of the path from satellite to the receivers passes through rain [4].

Rainfall (inch/hour)	Attenuation	Description
0.02”	0.01 dB	Drizzle
0.10”	0.10 dB	Light rain
0.50”	0.90 dB	Heavy rain
1.00”	2.30 dB	Very heavy rain

Table 2. Some rainfall attenuation values.

C Adaptive FEC Assignments

The adaptive rate control algorithm (source/FEC assignments) calculates all the possible source and FEC rates and decides a best suited pair of source and FEC rate depending on the congestion in the network, request queue buildup and QoS report metrics. Thus, the required actions may be a change in source rate only or a change in FEC rate only or a change in overall rate, i.e. source rate divided by the FEC coding rate. However, whatever the required change is, the refinement of these rates are performed based on heuristics approach since there is no mathematical analysis that has been reached yet to determine the optimum source/FEC rate pairs. Therefore, our main concern when updating the rates is to find a pair such that it should satisfy the QoS agreements while resolving the problem detected by the

QoS analysis. The detailed algorithm to find the best source/FEC rate pair is given in [4]. Figure 4 illustrates several new pair assignments for an initial pair (Pij) based on the heuristics given below:

- The metrics (ΔB, *ARFER*, etc.), which are calculated by the QRP block, shown in Figure 1, will be used to determine the number of steps when changing FEC rate, are defined as follows:

For a given multicast group in a DBS system which is portrayed in Figure 2, let the indices *m*, *i*, *j* and *k* correspond to end-receiver, DFT, region and multicast group, respectively. Then, we define,

- ΔB; estimate of the excessive overall rate exceeding the link capacity
- FDR_k; fraction of dropped frames for the k-th multicast group;
- $FERC_{ijk}=FER_{ijk}-FDR_k$; frame error rate due to channel only;
- FFR_{jk}; ratio of the number of DFTs in k-th multicast group having unacceptable FER to total number of DFTs in k-th multicast group;
- $ARFER_k$;average ratio of $FERC_{ijk}/FER_k^{max}$ for DFTs experiencing $FERC_{ijk}>FER_k^{max}$.

$$ARFER_k = \frac{1}{N_k FER_k^{\max}} \sum_{\substack{i,j \\ \text{if } FERC_{ijk} > FER_k^{\max}}} \mathrm{FERC}_{\mathrm{ijk}}$$

where N_k is the total number of DFTs experiencing $FERC_{ijk}>FER_k^{max}$.

- If the source rate is decreased by N steps, then we may need to decrease FEC rate as well to compensate the error sensitivity of compressed video stream [12];
- When increasing FEC rate, we consider the past QoS reports to determine if the chosen FEC rate provided acceptable FER in the past.

These parameters are defined in such a way to reflect a quantitative measure of the cause of the problem (such as congestion or poor channel) for a multicast group and, thus, this enable us to identify the problem. In this algorithm, we first detect if the problem is due to the ARAM system (e.g. misestimated statistical multiplexing gain), and/or channel conditions and/or DFT problem. Then, we further investigate the main cause of the problem. In [4], we present the details of the algorithm for multicast services and for unicast services, respectively. FFR_k and $ARFER_k$ reflect the number of DFTs experiencing unacceptable error and the level of the error respectively; thus based on these two measures we can determine the necessary changes in the overall rate, i.e., source rate divided by FEC rate.

For example, let Pij (SR,FEC) denotes current Source Rate/FEC Rate pair and the estimated excessive overall rate is ΔB, thus, depending on the problem identified by the QoS analysis, we calculate best suitable pair using the several discrete thresholds for the FFR and ARFER measures. Unfortunately, this approach only recognizes a solution based on the above quantitative analysis. However, once the test-bed is completed, and we may choose the best pair on the basis of quantitative and qualitative analysis.

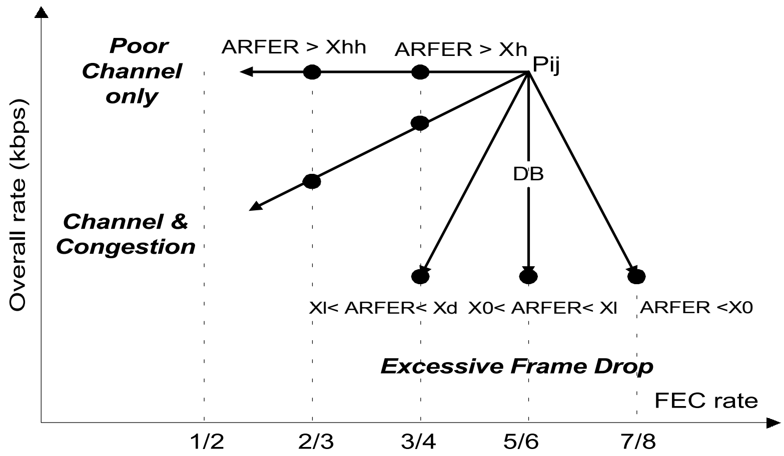

Figure 4. Source and FEC Rate Assignments.

Although the higher FEC rates (1/2, 2/3) are essential to combat channel problems, we may observe that the FEC is over coded in good channels. This may cause frame loss due to congestion in the transmit queue and reduced multiplexing efficiency resulting in fewer services being admitted. These relations can be extended further; however, it is sufficient to examine the performance of our heuristic algorithm.

IV. PERFORMANCE RESULTS

To assess the impact of VBR traffic mixture in total load, we have conducted comprehensive sets of simulation experiments, on the frame basis, using the previously developed ARAM simulator [4]. The simulator performs all the control actions discussed above and includes both traffic (VBR and ABR) and channel-state generators. 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 streams. While VBR traffic is generated on a frame-basis as explained in Section 3, ABR data traffic is packetized with a packet size that is same as average MPEG frame size. Table 3 presents the traffic characteristics used in the ARAM simulator. The DBS channel is modeled as described in Section 3 with maximum allowable FER of 10^{-4}.

Traffic characteristics	ABR	MPEG coded VBR
Generated traffic	250kbps	*Starwars* (GOP: IBBPBBPBBPBB) Mean = 187kbits/GOP , Std = 72kbits/GOP Maximum source reduction rate = 47kbits/GOP
Service time	NA	600 seconds
Request time epoch	NA	600 seconds

Table 3. Traffic Characteristics Used in Simulations.

The following performance metrics were quantified by the simulation. Average DBS link utilization measures the average transmitted information (bit/sec) divided by the link capacity (bit/sec). Fraction of dropped frames measures the ratio of the number of dropped frames to the total number of frames assigned to be delivered. Total number of completed services at the end of the simulation. It is worth noting that when the simulation is ended, the services in progresses are aggregated based on their completion percentage and integer part of this number is included to the number of completed services.

Quality assessment of digital video sequences is very important issue and received large attention from the networking community. Several metrics are used to specify QoS at the network layer. However, at the application layer, the choice of parameters has not been adequately addressed. This is mostly due to a wide diversity in the types of applications and their communication requirement [13,14]. An objective evaluation of these cognitive related issues is out of the scope of this paper, however, based on our experimental observations, we believe that the defined measure should correlate with subjective QoS. Therefore, QoS is measured as the rate loss seen by the end-receiver and is defined as [4]:

$$QoS_T = 1 - \frac{\Delta R_s + \Delta R_d + \Delta R_c}{R_n}$$

where ΔR_s accounts for changes in source rate due to congestion and/or FEC rate changes to keep overall rate constant, ΔR_d accounts for rate reduction due to dropped frames, ΔR_c accounts for sum of rates of all frames that are received with error or lost in the channel and R_n is the nominal rate. Note that ΔR_s, ΔR_d, and ΔR_c are parameters dependent on the traffic rate, channel state, and control actions. This metric will be calculated for each service in every second. If a service is terminated, as to reflect *penalty* to the performance measure, the 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. This can be considered as a measure of the respective subjective video quality with more freedom of scaling.

In these simulations, we consider both ABR and VBR traffic with varying mixtures. The fraction of VBR traffic in the total traffic has been changed from 50% to 100%. The same channel characteristics (Table 1) and the same traffic characteristics (Table 3) are fed into the ARAM simulator. We use Gaussian approximation to represent the aggregate VBR traffic [7].

We present two sets of simulation results: short term control (STC) and Long Term Control (LTC). Since STC consists of primitive control actions, it is used to reflect the traffic characteristics and the load on the ARAM system. Figure 5-8 show the set of simulation results for different mixtures of ABR and VBR traffic (fraction of VBR traffic in total traffic) for DBS link capacity of 22Mbps and probability threshold (γ) of 0.01.

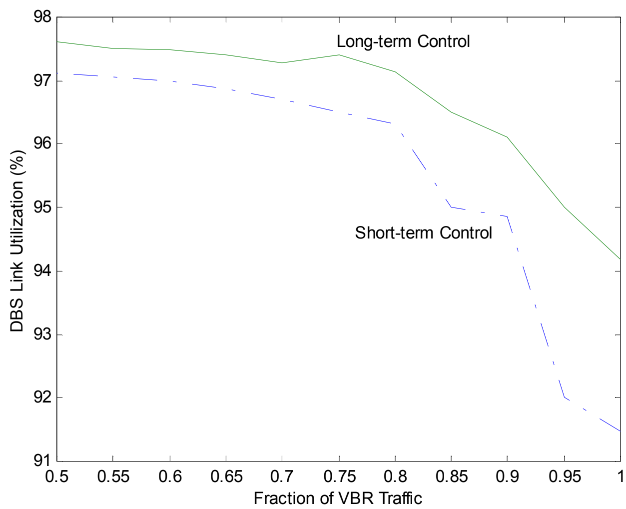

Figure 5. DBS Link Utilization.

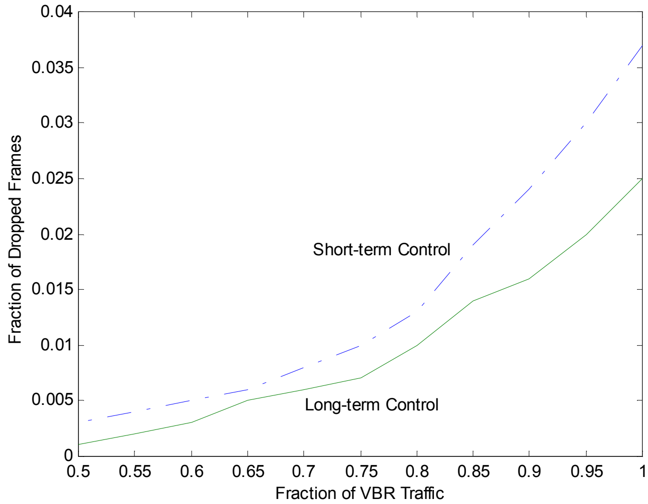

Figure 6. Fraction of Dropped Frames.

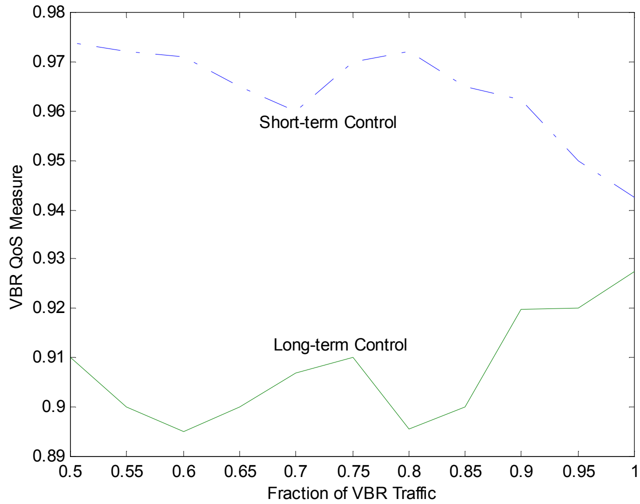

Figure 7. VBR QoS Measure

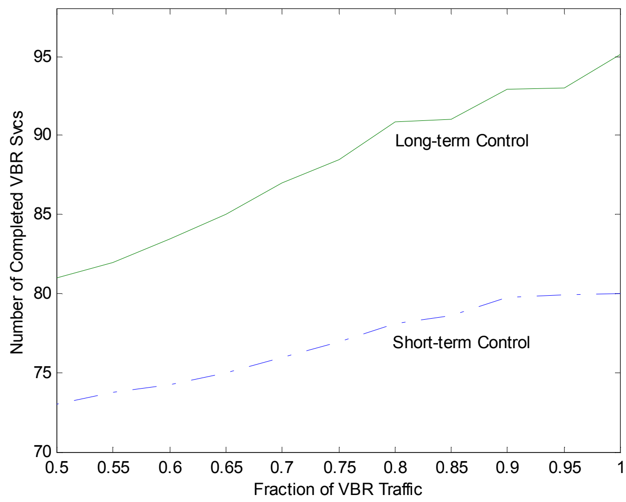

Figure 8. Number of Completed VBR Services.

As we can observe from the results for the LTC simulations, when VBR traffic mixture increases DBS link utilization decrease and fraction of dropped frames increase, yet QoS_T measure can be kept within the acceptable range using ARAM system. For example, if the load is only VBR traffic, the highest frame loss observed both for STC and LTC, yet, the lowest QoS_T measure for the STC is due to the fraction of dropped frames. The QoS_T measure for LTC, somewhat reduces when the ABR traffic mixture increases, this is mostly due to the source rate reduction filtering to accommodate more and more users, while STC do not reduce source rate to admit more user and thus its QoS_T measure increases. DBS link utilization, as expected, increase when ABR traffic mixture increases.

We also simulated different numbers of transition-state probabilities other than the ones given in Table 1, i.e., we have changed the percentage of bad channel duration in the total simulation time. Our observations are as follows: The higher the time spent in bad channel-state, the higher the variation in QoS_T measure for STC, where as the QoS_T measure for LTC gradually decreases yet it lies within acceptable range. Table 4 presents the simulation results when the channel is in the bad channel-state for 600 seconds. The results show that the QoS_T measure for STC is around 0.2 since most of the frames are lost due to bad channel conditions. Table 4 also suggests that the QoS_T measure for LTC is kept high at the expense of lower number of completed service due to higher FEC rate use to compensate for the channel errors. Note also that the number of completed VBR services for STC is 70, but its QoS_T measure suggests that the perceptive QoS will be very annoying.

	STC	LTC
Average DBS link utilization (%)	94.1	95.6
Number of completed VBR services	70	78
Average fraction of dropped frames	0.05	0.02
Average QoS_T measure	0.19	0.83

Table 4. Summary of the bad-channel simulations statistics.

In order to control the QoS variations, we may use lower probability threshold (γ =0.01 in the simulations) at the expense of lower admitted VBR services when the mixture is highly VBR loaded. On the other hand, we may use higher probability threshold (γ) to admit more VBR services at the expense of further delaying ABR services when the mixture is highly ABR loaded .

We also simulated for lower and higher values of DBS link capacities (C=22 Mbps in the simulations). Our conclusions remain same except that the higher the capacity the lower the fraction of dropped frames (higher QoS/utilization), and the lower the capacity the higher the fraction of dropped frames (lower QoS/utilization). Note also that the performance results presented above are characterized by other channel and source parameters. We simulated several other scenarios to exploit their performance, and some results can be found in [4].

5. CONCLUSION

In this paper we presented a multi-criterion control simulation in a realistically complex environment of a satellite network, involving non-symmetric up and downlinks. Direct Broadcast Satellite (DBS) network carrying heterogeneous traffic is characterized with challenges especially, high traffic burstiness and wireless channel dynamics. We first presented Adaptive Resource Allocation and Management (ARAM) algorithms developed to manage DBS networks. We investigated one of the key features of ARAM system; effect of the variable bit rate (VBR) load mixture in total load.

We observed that when VBR load mixture increases, the DBS link utilization decreases and the fraction of dropped frames increases, in general, yet Quality of Service (QoS_T) measure can be kept within the acceptable range by using adaptive admission control strategy. We can further control the QoS variations by using lower values of probability threshold for the aggregate instantaneous rate exceeding link capacity for highly VBR loaded networks. The worst case performance occurs in bad channel and highly VBR loaded conditions. While the short term control (STC) suffers in bad channel conditions regardless of the traffic mixture, the long term control (LTC) combats the channel and traffic mixture problems by using time scale based control strategies and by introducing graceful degradation during these adverse conditions.

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Vitae

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