A New MAC Protocol Based on Multimedia Traffic Prediction in Satellite Systems

P. Koutsakis* and A. Lazaris

Dept. of Electronic and Computer Engineering
Information & Computer Networks Laboratory
Technical University of Crete, Chania, Greece
{polk, alazaris}@telecom.tuc.gr

Abstract. The provision of acceptable Quality-of-Service (QoS) for integrated multimedia traffic over a geosynchronous earth orbit (GEO) satellite network demands the existence of a well-designed Medium Access Control (MAC) protocol. This paper proposes a new dynamic satellite bandwidth allocation technique which is based on accurate videoconference traffic prediction. Our work is combined with another work on data traffic modeling and prediction and is shown to provide very good throughput and delay results.

1 Introduction

In recent years, broadband satellite networks have attracted significant attention as a part of the global communications infrastructure. Satellite networks have specific characteristics which make them an attractive solution in order to complement terrestrial networks in providing worldwide access to the present and future generation multimedia services. More specifically, satellite networks include global coverage, providing access for remote users to terrestrial wide area networks from any location within the satellite coverage area, and broadcast/multicasting.

However, at the same time other characteristics of satellite networks, such as long propagation delays (270 ms round-trip delay for a GEO satellite network, i.e., at least 540 ms from the time the signaling information is sent from the station to the satellite until it reaches the receiver terminal), the limited bandwidth to be shared among many users, and limitation in power (which implies stringent use of buffer memory, transponder capacity and processing power) create the need for a well-designed Medium Access Control (MAC) protocol, especially in today's networks which are expected to handle bursty multimedia traffic. Among the above mentioned problems, the propagation delay is the most significant one, as it makes bursty users' (such as video users') current traffic profile rather useless for bandwidth allocation, since the profile will probably have changed significantly by the time the bandwidth allocation is made by the Network Control Center (NCC), which we consider to be integrated in the satellite on-board device.

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Videoconference traffic is expected to be a substantial portion of the traffic carried by emerging networks, hence presenting a challenge for satellite system designers. For Variable Bit Rate (VBR) coded video, statistical models are needed to design networks which are able to guarantee the strict Quality of Service (QoS) requirements of video traffic. Video packet delay requirements are strict, because delays are annoying to a viewer; whenever the delay experienced by a video packet exceeds the corresponding maximum delay, the packet is dropped, and the video packet dropping requirements are equally strict. Therefore, a good statistical model can be very useful in evaluating network performance under various videoconferencing loads.

The most well-known and used video standards for this application today are H.263 and MPEG-4. In this work, we first build a model which accurately captures the behavior of multiplexed H.263 videoconference movies, and we proceed to use this model for predicting the behavior of videoconference traffic in our new MAC protocol proposal for a Digital Video Broadcasting Return Channel Satellite (DVB-RCS) system. Our proposed MAC protocol is shown, via a simulation study, to achieve video packet delays and video packet dropping probability only slightly worse than the ideal scenario in which the actual video user requirements for each video frame were a priori known to the NCC.

Finally, our modeling scheme is combined with the modeling approach used in [10] for predicting self-similar data traffic and the combination is shown to provide once again very satisfying results in terms of minimizing bandwidth waste.

2 H.263 Video Traffic Modeling

H.263 is a video standard that can be used for compressing the moving picture component of audio-visual services at low bit rates. It adopts the idea of PB frames, i.e., two pictures being coded as a unit. Thus a PB-frame consists of one P-picture which is predicted from the previous decoded P-picture and one B-picture which is predicted from both the previous decoded P-picture and the P-picture currently being decoded. The name B-picture was chosen because parts of B-pictures may be bidirectionally predicted from the past and future pictures. With this coding option, the picture rate can be increased considerably without increasing the bit rate much [1].

Our work focuses on the accurate fitting of the marginal (stationary) distribution of video frame sizes of single videoconference traces. More specifically, our work follows the steps of the work presented in [2], where Heyman et al. analyzed three videoconference sequences coded with a modified version of the H.261 video coding standard and two other coding schemes, similar to the H.261. The authors in [2] found that the marginal distributions for all the sequences could be described by a gamma (or equivalently negative binomial) distribution.

In our work, we have studied three different long sequences of H.263 VBR encoded videos, from the publicly available library of frame size traces of long MPEG-4 and H.263 encoded videos provided by the Telecommunication Networks Group at the Technical University of Berlin [3]. We have investigated the possibility of modeling the traces with a number of well-known distributions and our results have shown that the best fit among these distributions for modeling a single movie is achieved for

all traces examined with the use of the Pearson type V distribution (also known as the inverted gamma distribution).
The three traces are, respectively:

1. A video stream extracted and analyzed from a camera showing the events happening within an office ("Office Cam").
2. A video stream extracted and analyzed from a talk-show ("ARD Talk").
3. A video stream extracted and analyzed from another talk-show ("N3 Talk").

All three of these traces are movies with low or moderate motion.

In the work presented in this paper, we will show that, based on the good fit of the Pearson V distribution for modeling a single movie, the behavior of *both single and multiplexed H.263 videoconference movies* from VBR coders can be accurately captured and used for efficient proactive resource management in satellite systems *(the case of modeling multiplexed videoconference streams is especially significant for our study, since numerous sources are multiplexed in the uplink channel*). To do this, we will first briefly explain our results on modeling a single movie.

For each one of the three videos under study we have used the VBR coding version, in which new video frames arrive every 80 msecs. The length of the videos varies from 45 to 60 minutes and the data for each trace consists of a sequence of the number of cells per video frame. Table 1 presents the trace statistics for each trace (packet size=48 information bytes; we use packets of ATM cell size throughout this work, but our mechanism can be used equally well with packets of other sizes, as the nature of our modeling results would not be altered at all), as well as the parameters (α, β) of the Pearson type V distribution fit for each movie. The Probability Density Function (PDF) of a Pearson type V distribution with parameters (α, β) is:

$$f(x)= [x^{-(\alpha+1)}\, e^{-\beta/x}\,]/\, [\beta^{-\alpha}\, \Gamma(\alpha)], \text{ for all } x>0 \tag{1}$$

and zero otherwise.

The mean and variance are given by the equations:

$$\text{Mean}=\beta/(\alpha-1) \tag{2}$$

$$\text{Variance}=\beta^2/[(\alpha-1)^2(\alpha-2)] \tag{3}$$

Table 1. Trace Statistics

Movie	Mean (packets)	Peak (packets)	Standard Deviation (packets)	Pearson type V parameters (α,β)
Office	18.8	109	6.8	(9.64,162.43)
ARD Talk	49.5	277	27	(5.36,215.82)
N3 Talk	53.1	291	30.3	(5.07,216.11)

Autoregressive models have been used in the past to model the output bit rate of VBR encoders, e.g. [4, 5]. A Discrete Autoregressive model of order *p*, denoted as DAR(*p*) [6], generates a stationary sequence of discrete random variables with an arbitrary probability distribution and with an autocorrelation structure similar to that of an Autoregressive model. DAR(1) is a special case of a DAR(p) process and it is defined as follows: let $\{V_n\}$ and $\{Y_n\}$ be two sequences of independent random variables. The random variable V_n can take two values, 0 and 1, with probabilities 1-ρ and ρ, respectively. The random variable Y_n has a discrete state space S and $P\{Y_n = i) = \pi(i)$. The sequence of random variables $\{X_n\}$ which is formed according to the linear model:

$$X_n = V_n X_{n-1} + (1- V_n) Y_n \tag{4}$$

is a DAR(1) process.

A DAR(1) process is a Markov chain with discrete state space S and a transition matrix:

$$\mathbf{P} = \rho\mathbf{I} + (1-\rho)\,\mathbf{Q} \tag{5}$$

where ρ is the autocorrelation coefficient, **I** is the identity matrix and **Q** is a matrix with $Q_{ij} = \pi(j)$ for i, j $\in S$.

Autocorrelations are usually plotted for a range W of lags. The autocorrelation can be calculated by the formula:

$$\rho(W)= E[(X_i - \mu)(X_{i+w} - \mu)]/\sigma^2 \tag{6}$$

where μ is the mean and σ^2 the variance of the frame size for a specific video trace.

As in [2], where a DAR(1) model with negative binomial distribution was used to model the number of cells per frame of VBR teleconferencing video, we want to build a model based only on parameters which are either known at call set-up time or can be measured without introducing much complexity in the network. DAR(1) provides an easy and practical method to compute the transition matrix and gives us a model based only on four physically meaningful parameters, i.e., the mean, peak, variance and the lag-1 autocorrelation coefficient ρ of the offered traffic (these correlations are typically very high for videoconference sources). According to [7], the DAR(1) model can be used with any marginal distribution.

More specifically, in our model the rows of the **Q** matrix consist of the Pearson type V probabilities $(f_0, f_1, \ldots f_k, F_K)$, where $F_K = \Sigma_{k>K} f_k$, and K is the peak rate. Each k, for $k<K$, corresponds to possible source rates less than the peak rate of K.

The lag-1 autocorrelation coefficient ρ is estimated by Equation (2) to be equal to 0.943 for the office camera trace, 0.867 for the ARD Talk trace and 0.872 for the N3 Talk trace.

We proceeded with testing our model statistically in order to study whether it produces a good fit for the trace superposition. For this reason we have used Q-Q plots. The Q-Q plot is a powerful goodness-of-fit test [2, 8], which graphically compares two data sets in order to determine whether the data sets come from populations with a common distribution (if they do, the points of the plot should fall approximately along a 45-degree reference line). More specifically, a Q-Q plot is a plot of the quantiles of the data versus the quantiles of the fitted distribution (a z-quantile of X is any value x such that $Pr(X \leq x) = z$).

In Figure 1, we have plotted the 0.025-, 0.05-, 0.075-,… quantiles of the actual office camera trace versus the respective quantiles of the DAR(1) model for the superposition of the 15 traces. As shown in the Figure, the points of the Q-Q plot fall almost completely along the 45-degree reference line, with the exception of the last

2.5% quantile (right-hand tail), for which the DAR(1) model greatly overestimates the probability of frames with a very large number of cells. The very good fit shows that the superposition of the actual traces can be modeled very well by a respective superposition of data produced by the DAR(1) model.

The same conclusions were deducted by our results for the superposition of various number of sources transmitting the other two traces. The small differences observed in our results were that, in the case of the ARD Talk trace, the overestimation made by the DAR(1) model was respectively smaller than that shown in Figure 1, and that, in the case of the N3 Talk trace, the overestimation starts “earlier”, i.e., it covers the last 5% quantile.

Nevertheless, all our results show that our model captures with great accuracy the behavior of H.263 videoconference traffic.

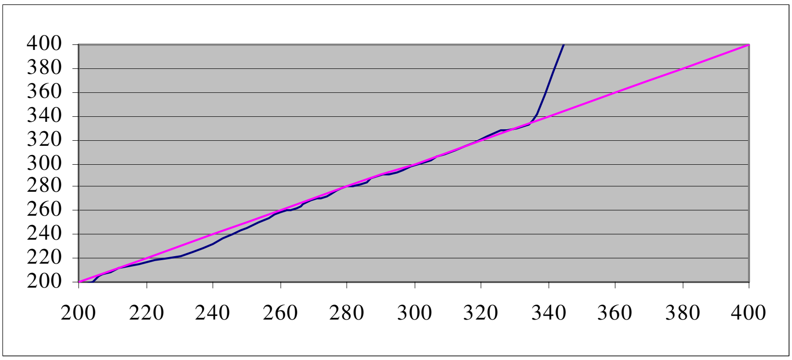

Fig. 1. Q-Q plot of DAR(1) model versus the actual office camera trace for 15 superposed sources

3 Our MAC Protocol Proposal

The Digital Video Broadcasting Return Channel Satellite (DVB-RCS) standard [12] develops a communication system for the return channel (uplink channel), i.e., the link from the user terminal to the network gateway. Due to the expected services features and large delay-bandwidth product of satellite networks, DVB-RCS represents a proper test bed for the proposed resource allocation schemes. The most relevant elements of the DVB-RCS Network are: a) RCSTs (Return Channel Satellite Terminals), i.e., a generic access terminal, b) the NCC (Networks Control Center), a device in charge of managing the access and bandwidth allocation for RCSTs, c) Gateways and Feeders, which are the elements that receive and transmit information outside the network [10].

As in [10], our proposed satellite medium access scheme is based on a Multi-Frequency Time Division Multiple Access (MF-TDMA) approach, according to which a carrier is divided in timeslots (grouped in frames and superframes). MF-TDMA schemes are capable of providing efficient and flexible bandwidth utilization [10, 14]. The system parameters are taken from [10] and are presented in Table 2.

Table 2. System Parameters

Frame Duration	26.5 ms
Carriers	4
Slots/frame/carrier	128
Bytes/slot	53
System global rate	8 Mbps

The NCC allocates to each active RCST a set of timeslots, each characterized by a frequency, bandwidth, start time and duration time. The DVB-RCS standard provides five allocation request types [12, 13] which can be joined in order to satisfy the QoS requirements:

- Continuous Rate Assignment (CRA) is a fixed capacity negotiated between the RCST and the NCC. It is maintained across frames until a new negotiation.
- Rate Based Dynamic Capacity (RBDC) is a capacity allocated to the RCST based on its rate request (bytes/frame), and is subject to a maximum rate limit negotiated between the RCST and the NCC. The last request from a RCST overwrites all previous RBDC requests from the same RCST.
- Volume Based Dynamic Capacity (VBDC) is an assignment strategy in which a terminal signals its request in terms of total number of slots required to empty its queue. The request remains effective as long as not all of the requested time slots have been granted. This strategy is especially suited for bursty traffic [14].
- Absolute Volume Based Dynamic Capacity (AVBDC) request is a request similar to VBDC, with the difference that the last request from a RCST overwrites all previous AVBDC requests from the same RCST.
- Free Capacity Assignment is not a true request, allocating the otherwise unused capacity. It is automatic and does not involve signaling from the RCST to the NCC.

The modeling approach proposed in [10] for self-similar data traffic (such as World Wide Web traffic) lets an RBDC request correspond to the data traffic prediction, plus a corrective factor ζ_N. This approach will also be incorporated in the second part of our results, in Section 4, where data traffic is incorporated into our simulations for the satellite system.

The first difference between the work presented in [10] and our work is that, additionally to the MF-TDMA frame structure, we adopt the idea that, after all requests have been satisfied, the bandwidth left is distributed freely following a certain algorithm (this approach is named in the literature as a Combined Free and Demand Assignment Scheme, CFDAMA scheme); the algorithm implemented in our scheme is a simple round-robin assignment algorithm to all RCSTs which are currently active. This idea of a hybrid protocol was first proposed in [15] and is especially useful in allocating slots to video users, as the difficulty in providing them with adequate bandwidth due to their frequent changes in bandwidth needs could be somewhat alleviated by their acquiring the unused channel bandwidth freely in a round-robin manner. We use the Combined Free and Demand Assignment Scheme with Piggybacking (CFDAMA-PB) version of the protocol, which was shown in [16] to be the most efficient way of making reservations. According to the PB strategy, user stations send their capacity requests embedded in the header of their packets. The free capacity distribution performed by the protocol brings the end-to-end delay performance at low loads close to that obtained with random access protocols, while the demand-based bandwidth allocation at the beginning of each frame guarantees the protocol's stability, robustness and efficient utilization of transmission bandwidth at high loads. Free capacity distribution in an MF-TDMA frame structure was also used in [14], with which we will conceptually compare our work in Section 4.

The second and most important difference of our work with [10] is that an approach similar to the one used in [10] for data traffic (letting the RBDC request correspond to the data traffic prediction) is not enough for real time Variable Bit Rate (rt-VBR) videoconference traffic, which has strict QoS requirements in terms of average video packet dropping (set to a maximum of 0.1% in our work) and end-to-end video packet delays (set to a maximum of 0.6 seconds in our work, which is especially strict considering that, for each possible failure of our prediction due to underassignment, the respective packets which would have to wait for a new assignment will have a minimum end-to-end video packet delay of 0.54 seconds). For this reason, we propose the following different approach in our MAC scheme.

As explained in Section 2, the Pearson V fit provides a good (but not perfect) fit for a single H.263 videoconference trace. Also, a logical assumption for next generation networks is that videoconference users will be allowed to adopt one of just a few specific "modes" (each corresponding to a set of traffic parameters). This is especially plausible for videoconference traffic, as the number of variations between source bandwidth requirements is naturally restricted by the type of application (a much larger pool of "modes" would have to be used in the case of video traffic). Therefore, in this work we consider that a videoconference user can adopt one of the three "modes" presented in Table 1 (a slightly larger pool of modes would have to be used in an actual satellite system scenario). Based on the good model for single videoconference traces and the highly accurate model of multiplexed traffic, *we propose that the "burden" of traffic prediction for the RCSTs should fall on the NCC instead of the RCSTs.* More specifically, the NCC should run a real-time simulation, both for single and for multiplexed videoconference sources. Hence, based on the "mode" declared by the RCSTs at call establishment, the NCC does not need to wait for a request from the RCSTs every channel frame (which would arrive with a delay of more than 5 channel frames, due to the propagation delay); instead, it can start allocating resources to the videoconference terminals, by simulating the single source models with the sources' mean rate as a simulation start point, and by computing the free slots in each channel frame (using the DAR(1) models for multiplexed videoconference traffic, and subtracting the estimated used slots from the total number of slots in the system) in order to allocate the estimated number of free slots in a round-robin manner to all active RCSTs. With this slot allocation scheme, the RCST will not need to send frequent requests to the NCC but it will only need to send a "corrective" AVBDC request every superframe (defined in our work as equal to 6 channel frames, to account for the propagation delay). The reason for sending this request will be for the RCST to help the NCC correct any mistakes (due to either slots overassignment or underassignment) of the models produced at the NCC via online simulation. After receiving the AVBDC request, the NCC will resume its simulation with the current RCST state (in terms of bandwidth requirements) as a start point. As it will be shown from our results, this approach, which minimizes the need for signaling among the video RCSTs and the NCC, provides clearly improved results in terms of videoconference users' QoS requirements satisfaction, due to the quality of the prediction made by our video modeling scheme.

4 Results and Discussion

4.1 Conceptual Comparison with Relevant Work

In [14], the authors propose a satellite MAC protocol based on a CFDAMA MF-TDMA access scheme (i.e., they use the same approach with our work). The protocol, however, proposes the use of a CRA-type assignment for rt-VBR traffic, such as the videoconferencing traffic used in our work, with the difference that the assignment is fixed for the duration of the connection (no new negotiation is needed), and equal to the rt-VBR user's peak transmission rate; the reason for this choice in [14] is that rt-VBR traffic has strict delay and packet dropping constraints, and no accurate traffic prediction mechanism is provided in [14], hence leading to the "defensive" choice of peak cell rate assignment, which leads to significant bandwidth waste, as the assignment is most of the time larger than the rt-VBR user's actual needs (the other problem with this type of assignment is that if the assignment was made for less than the peak transmission rate, it would at times lead to severe packet dropping; the use, in [14], of the CFDAMA policy could probably help to partially alleviate the packet dropping problem, through the use of the free slot assignment, but the authors chose the "safer" solution of peak transmission rate assignment). The inferior performance of such an assignment will also be shown in our simulation results. Finally, it should be noted that video traffic in [14] is generated with the use of a Markov Modulated Poisson Process (MMPP) model, whereas we use actual video traces in our work.

4.2 First Implementation Case: Only Video Traffic Present in the Channel

At the start of our simulation study, we let each videoconference user choose one of the three traffic parameter sets ("modes") which are presented in Table 1, with equal probability.

Three MAC schemes will be compared in this Section. The first is our proposed scheme, the second is a scheme conceptually similar to [14], allocating to each video terminal its peak rate (declared at call establishment) and the third is an "ideal" scheme, as we want to compare our protocol with a similar one in which the NCC would "know", without any information exchange (therefore, no contention is necessary among video RCSTs), exactly what the video RCSTs' bandwidth demands for the next video frame will be.

Figure 2 presents our simulation results for the average video packet dropping metric versus the system utilization. Utilization indicates the traffic load normalized to the uplink capacity, e.g., a traffic load equal to 40% represents 40% of the 8 Mbps uplink capacity, i.e., 3.2 Mbps system throughput. As it is shown in the Figure, the difference in video packet dropping between our scheme and the "ideal" case is so small that it can be considered almost negligible for all normalized video traffic loads. Our scheme can handle up to 76% system load while at the same time satisfying the strict QoS requirement of maximum video packet dropping equal to 0.1%; the respective maximum system load which the “ideal” scheme can handle is 79%. The reason that none of the two schemes can achieve a higher throughput is the high burstiness of video traffic; in certain channel frames, video bursts from more than one RCST happen to take place simultaneously in the uplink channel. Although our traffic modeling scheme can often predict such bursts, the total amount of requested bandwidth in certain channel frames may surpass the system’s available capacity; this will lead to inevitable video packet dropping, as some of the packets may will not be sent within the roughly three channel frames which pass before the arrival of the next video frame (when a new video frame arrives, all packets of the previous video frame which have not yet been sent are discarded).

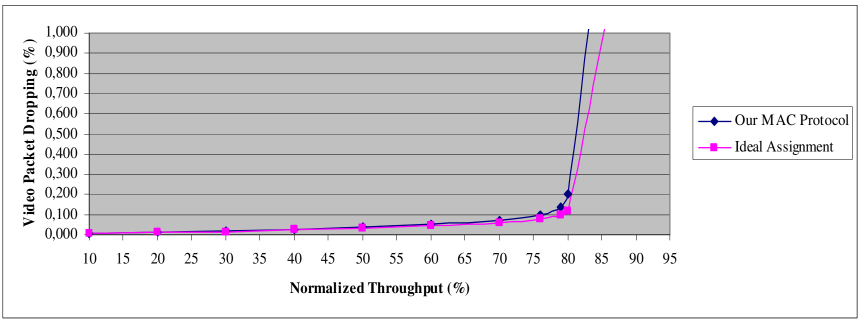

Fig. 2. Average Video packet dropping vs. System Utilization

Figure 3 presents our simulation results for the average end-to-end video packet delay versus the system utilization. The results are similar in nature with those of Figure 2, denoting that our scheme’s results are again very close to the ones achieved by the “ideal” scheme.

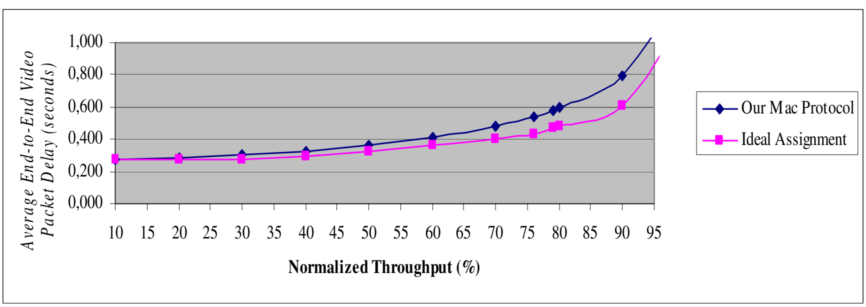

Fig. 3. Average End-to-end video packet delay vs. System Utilization

The scheme which allocates the peak rate to each video terminal is not shown in Figures 2-3, as it achieves zero packet dropping and the minimum possible end-to-end video packet delay (respectively, constant and equal to 0.274 seconds, i.e., equal to the propagation delay plus a very small additional amount of time in order for the video RCST to gain access to the channel). However, these two exceptionally good results come at a high cost. As explained earlier and shown in Table 1, all the video-conference traces used in our study are bursty. More specifically, all three traces used in our study have a peak-to-mean ratio larger than 5.4. This means that the constant allocation of the peak rate to all active video sources causes a double disadvantage to the MAC scheme:

1. it causes significant loss of valuable bandwidth resources, as many slots are left unused,
2. the number of free slots in each channel frame, which could be used for other types of traffic (such as data traffic) is heavily decreased, hence restricting them from accessing the system and failing to efficiently satisfy their QoS requirements (this will be shown through our simulation results in Section 4.3).

For these reasons, the maximum channel throughput achieved by allocating the peak rate to each video RCST is very low, equal to 32%, i.e., less than half the achieved throughput from our scheme.

4.3 Second Implementation Case: Combining Video and Data Traffic Models

Self-Similar traffic modeling has been shown to fit the data traffic of both a typical Ethernet network [17] and of World Wide Web (WWW) applications [18]. For this reason, self-similarity has been widely used in the literature either simply for modeling data traffic (e.g., [14]), or for data traffic prediction [9, 10]. As already explained earlier, in this work we adopt the approach of [10] for data traffic prediction, and we let an RBDC request correspond to the data traffic prediction, plus a corrective factor ζ_N (as in [10], we use the corrective factor ζ_1 in our simulations). Data traffic has lower priority than video traffic, as it is much more delay-tolerant. For this reason, we set an upper end-to-end delay bound of 1 second for data packets.

Figure 4 presents our simulation results for the average end-to-end data packet delay versus the system utilization. The results presented in the Figure are the average of three different "divisions" of the system load: in the first case, 30% of the total load was offered from video traffic (e.g., for a normalized system load of 60%, 18% was offered from video traffic), and 70% from data traffic; in the second case, both types of traffic offered 50% of the total system load; in the third case, 70% of the total load was offered from video traffic and 30% from data traffic. Once again, the results presented in Figure 4 are generally similar in nature with those of Figures 2 and 3, denoting that our scheme's results are very close to the ones achieved by the "ideal" scheme. More specifically, as the system load increases, the "ideal assignment" scheme achieves a lower delay of about 0.1-0.2 seconds for small and medium system loads and more than 0.3 seconds for high loads in comparison to our scheme, due to the lack of contention (and therefore, lack of collisions) in the "ideal assignment" scenario. Regarding the achieved maximum system throughput (maximum throughput for which all the QoS requirements of video and data RCSTs are satisfied), the results are again qualitatively similar with those in Section 4.2: the maximum throughput achieved by our scheme is 83%, by the "ideal assignment scheme" 85% and by the scheme assigning the peak rate to video RCSTs equal to 41%. It is clear that all three schemes achieve a much higher throughput with the addition of data traffic into the system. The reason for this is that a significant portion of the slots left unused in the case when only video traffic exists in the system, are ideally filled with the much less demanding data traffic, which does not have an equally urgent need to be transmitted as video traffic (no data packets are dropped if they are not transmitted within a specified amount of time), therefore it can "compromise" with the use of whichever slots are left unused. Still, the channel throughput results cannot reach beyond 85% even for the ideal assignment scheme; this is once more due to the burstiness of video traffic.

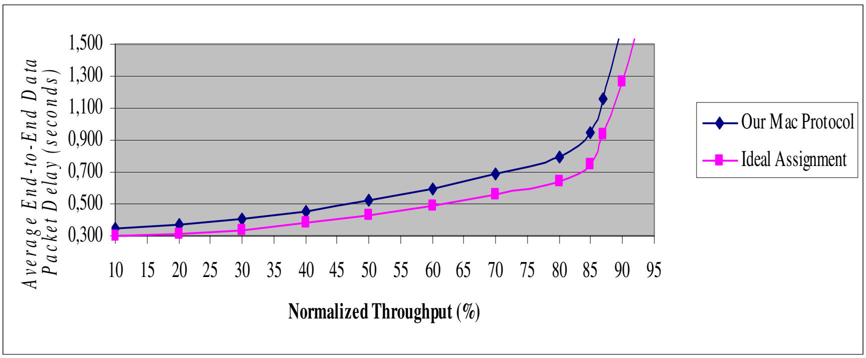

Fig. 4. Average End-to-end data packet delay vs. System Utilization

References

1. ITU-T Recommendation, H.263, 3/1996.
2. D. P. Heyman, A. Tabatabai and T. V. Lakshman, "Statistical Analysis and Simulation Study of Video Teleconference Traffic in ATM Networks", *IEEE Transactions on Circuits and Systems for Video Technology,* Vol. 2, No. 1, pp. 49-59, 1992.
3. http://www-tkn.ee.tu-berlin.de/research/trace/trace.html
4. B. Maglaris, D. Anastassiou, P. Sen, G. Karlsson, and J. D. Robbins, "Performance Models of Statistical Multiplexing in Packet Video Communications", *IEEE Transactions on Communications*, Vol. 36, No.7, pp. 834-844, 1988.
5. C. Shim, I. Ryoo, J. Lee and S. Lee, "Modeling and Call Admission Control Algorithm of Variable Bit Rate Video in ATM networks", *IEEE Journal on Selected Areas in Communications*, vol.12, No.2, pp. 332-344, 1994.
6. P. A. Jacobs and P. A. W. Lewis, "Time Series Generated by Mixtures", *Journal of Time Series Analysis*, Vol. 4, No. 1, pp. 19-36, 1983.
7. T. V. Lakshman, A. Ortega and A. R. Reibman, "VBR Video: Trade-offs and potentials", *Proceedings of the IEEE,* Vol. 86, No. 5, pp. 952-973, 1998.
8. A. M. Law and W. D. Kelton, "Simulation Modeling & Analysis", 2nd Ed., McGraw Hill Inc., 1991.

9. Z. Jiang and V. C. M. Leung, "A Predictive Demand Assignment Multiple Access Protocol for Internet Access over Broadband Satellite Networks", *International Journal of Satellite Communications and Networking*, Vol. 21, No. 4-5, pp. 451-467, 2003.
10. F. Chiti, R. Fantacci and F. Marangoni, "Advanced Dynamic Resource Allocation Schemes for Satellite Systems", *in Proceedings of the IEEE International Conference on Communications (ICC) 2005,* Vol. 3, pp. 1469-1472, Seoul, Korea.
11. H. Peyravi, "Medium Access Control Protocols Performance in Satellite Communications", *IEEE Communications Magazine,* Vol. 37, No. 3, pp. 62-71, 1999.
12. Std., ETSI EN 301 790 V1.3.1 (2003-03).
13. Std., ETSI EN 101 790 V1.2.1 (2003-01).
14. A. Iuoras, P. Takats, C. Black, R. DiGirolamo, E. A. Wibowo, J. Lambadaris and M. Devetsikiotis, "Quality of Service-Oriented Protocols for Resource Management in Packet-Switched Satellites", *International Journal of Satellite Communications,* Vol. 17, No. 2-3, pp. 129-141, 1999.
15. S. V. Krishnamurthy and T. Le-Ngoc, "Performance of CF-DAMA Protocol with Pre-Assigned Request Slots in Integrated Voice/Data Satellite Communications", *in Proceedings of the IEEE International Conference on Communications (ICC) 1995,* Vol. 3, pp. 1572-1576, Seattle, USA.
16. T. Le-Ngoc and S. V. Krishnamurthy, "Performance of Combined Free/Demand Assignment Multiple Access Schemes in Satellite Communications", *International Journal of Satellite Communications,* Vol. 14, No.1, pp. 11-21, 1996.
17. W. E. Leland, M. S. Taqqu, W. Willinger and D. Wilson, "On the Self-Similar Nature of Ethernet Traffic (extended version)", *IEEE Transactions on Networking*, Vol. 2, No. 1, pp. 1-15, 1994.
18. M. E. Crovella and A. Bestavros, "Self-Similarity in World Wide Web Traffic: Evidence and Possible Causes", *IEEE Transactions on Networking*, Vol. 5, No. 6, pp. 835-846, 1997.