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Voice, Data, and Video Integration for Multi-Access in Broadband Satellite Networks

by B. Ghaffari and E. Geraniotis

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VOICE, DATA, AND VIDEO INTEGRATION FOR MULTI-ACCESS IN BROADBAND SATELLITE NETWORKS

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Abstract

Multi-media integration of broadband services in a broadband satellite network is considered. Voice, data, video teleconferencing, and television with broad range of service (bit) rates are multiplexed through a broadband satellite channel in a multiple-access fashion. Large (but finite) population sizes are considered with arrivals modeled by binomial distributions. A two-state mini-source model is used for voice signals. For video, variable rate interframe coding is utilized to reduce the bandwidth requirements, and Markov phase processes model the modulation of the rates of the video teleconferencing and television signals.

Among these services, video and voice are real-time signals and can not tolerate large random delays. In our attempt to satisfy this, video and voice use the Synchronous Transfer Mode (STM) with a frame structure, while the data users (with their bursty traffic) send (and retransmit, if necessary) their packets randomly within a frame. The video and voice users make their schedules in advance by using a pre-assigned slot (status slot). The first portion of a frame is assigned to the variable rate video users, while the variable rate voice users fill up the last portion of the frame. Data packets fill up the remaining slots between these two movable boundaries in a random-access fashion. In this protocol, the delay introduced by the satellite is taken into consideration. This multiple-access integration protocol is optimized with respect to performance measures, such as the blocking probabilities for voice and video, the average delay for data, and the average throughput for voice, video, and data.

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VOICE, DATA, AND VIDEO INTEGRATION FOR MULTI-ACCESS IN BROADBAND SATELLITE NETWORKS

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1. Introduction

Multi-media integration of broadband services in a broadband satellite network is considered. Voice, data and video telephony with a broad range of service (bit) rates are multiplexed through a broadband satellite channel in a multiple-access fashion. Each type of traffic, according to its characteristics, demands its own requirements, which may conflict or impose some constraints on other types of traffic. The specification of an access protocol, which integrates all these services while keeping the demand requirements of each service within an acceptable limit, is the purpose of this paper.

Among these services, video and voice are real-time signals and can not tolerate large random delays. In an attempt to satisfy this, video and voice use the Synchronous Transfer Mode (STM) with a frame structure [1], while the data users (with their bursty traffic) transmit (and retransmit, if necessary) their packets randomly within a frame. The video and voice users make their schedules in advance by using a pre-assigned slot (status slot). The first portion of a frame is assigned to the variable rate video users, while the fixed rate voice users fill up the second portion of the frame. Data packets fill up the remaining slots in a random-access fashion (see Figure 1).

There is a group of voice, data, and video terminals on the ground communicating with each other through a geostationary satellite. A propagation delay of τ frames (approximately 0.27 sec, see [2]) is considered for the packet transmission. This propagation delay imposes some constraints on system model and performance evaluation for this protocol, which in turn makes the analysis more challenging. The extension of this scheme to a ground-based radio network is obvious ($\tau = 0$).

The models and the arrival processes of three types of traffic (namely, data, voice, and video) are introduced. The arrival processes are Binomial with their corresponding parameters. The parameters characterizing these models and arrival processes (the system variables) are specified.

Voice users follow the simple on-off model. Each inactive voice station becomes active in a frame with some probability. When a voice call is established, it generates geometrically distributed number of fixed length packets and transmits each packet within a frame.

Similarly, idle video terminals become active with some probability and the active ones become inactive with some other probability within a frame. In contrast to voice, the number of generated video packets is not fixed within a frame and it follows a stochastic process, conditioned on the number of active video terminals.

Video users demand large bandwidth according to motion-picture activity requirements. Recent advances in variable bit rate coding algorithms allow us to exploit the interframe activity nature of motion pictures in order to lower the high bit rate requirements. The characterization of an appropriate model for video users activities is an essential task. The precision of such a model relies on two very basic and important principles. First, it needs to match the practical and experimental requirements for motion scenery. Indeed it has to capture the very basic and essential characters of the real video activities. Second, this model must be simple enough in order to lend itself to a feasible and meaningful analysis. We propose a discrete-time random walk to model the activity of a group of independent video telephone scenes. The continuous time version of this, namely the continuous-time death-birth process, was considered in a queuing statistical multiplexing context in [3] and [4]. However, we use the experimental data reported in [3] for bit rate collected with a conditional replenishment interframe coding scheme.

Voice and video users upon their arrivals send information to the status slot to inform others of their arrivals. After τ frames, this information is accessible to all users. If this reservation is confirmed by the structure of global information, which is available to the sender of the reservation as well as to others, then it starts to send its packets. Video users, in addition to their presence, also send the information regarding the degree of their activities.

When the total aggregate video traffic is high, several video packets may be dropped. Packet dropping mechanism can distribute the number of dropping packets among different video users equally. However, this packet dropping should not cause any considerable degradation in the picture quality in the receiver. The average packet dropping probability is derived in a frame at the steady state situation. Voice and video user blocking is introduced and the limiting probabilities are obtained. The voice and video throughput are also evaluated.

Data terminals access the channel randomly. The aggregate data traffic attempts to transmit in a frame, according to the framed Aloha protocol (see for example [5]-[8]). The number of available (or free) slots, in which these data users contend for packet transmission, is a random variable. A user that is idle in a frame generates a packet with some probability and sends it in that frame. If this packet is successful τ frames later, the user will be idle, otherwise it is considered to be backloged. In this way, there exist $\tau + 1$ interleaved processes that are disjointed from each other. Each of them has access to the channel every $\tau + 1$ frames.

The characterization of this system is two Markov processes of dimensions $\tau + 3$ and $\tau + 4$. A well justified approximation reduces these dimensions to 3 and 4, respectively. The first process consists of the following elements: 1) number of active video terminals in a frame, 2) number of packets generated by that number of video terminals, and 3) number of voice calls in progress (or number of voice packets) in a frame. The second process has the number of backloged data packets in addition to the elements of the first process.

The performance of this protocol is examined through the numerical evaluation for performance measures, such as video packet dropping probability, voice and video blocking probability, average video, voice and data throughput, and average data packet delay.

This paper is organized as follows. In Section 2, the models for different traffic types are introduced. In Section 3, we give a thorough description of the integrated protocol. In Section 4, we introduce the arrival processes as well as the Markovian model for video and voice. We also define the performance criteria for video and voice, and evaluate them by obtaining the limiting probabilities of Markov chains. Section 5 is dedicated to the study of data. Delay and throughput for data are introduced and evaluated in this section. In Section 6, we present some numerical results, and finally, in Section 7, we conclude this paper with a summary.

2. System Model

In this section, the models for voice and video are introduced and the parameters that characterize these models (the system variables) are specified. In description of the models, we start with video, which is the more complex and dynamic one.

- **Video Model**

We propose a discrete-time random walk for J independent video telephone scenes. The continuous time version of this, namely continuous-time death-birth process, was utilized in a queuing context in [3] and [4]. However, we use the experimental data reported in [3] for bit rate collected with a conditional replenishment interframe coding scheme.

Figure 2 illustrates a discrete-time and discrete-state space Markov chain. Time is discretized with the frequency of 30 samples/sec, which is equivalent to the picture frame frequency. Each state corresponds to the level in which video sources generates that many packets in a frame. In state i for example, i video packets are generated, which is equivalent to the iL bits/frame. In order to find the parameters of this model, the experimental first- and second-order statistics are matched with those of the random walk. It is easy to show that, in steady state, the distribution of this process is binomial with parameters M and p, where p is $\alpha/(\alpha+\beta)$. The first- and second-order statistics are

$$\overline{L} = MpL$$

$$\sigma^2 = Mp(1-p)L^2$$

$$c(n) = \sigma^2 \lambda^n$$

where $\overline{L}$ is the average number of bits/frame, σ^2 the variance, $c(n)$ the autocovariance, and λ equals $1-\alpha-\beta$. Solving these equations for α, β, and L yields

$$\alpha = \frac{(1-\lambda)\overline{L}^2}{\overline{L}^2 + M\sigma^2}$$

$$\beta = \frac{(1-\lambda)M\sigma^2}{\overline{L}^2 + M\sigma^2}$$

$$L = \frac{\overline{L}^2 + M\sigma^2}{M\overline{L}}.$$

The experimental data from [3] is

$$\overline{L} = 0.52J \quad \text{bits/pixel}$$

$$\sigma^2 = 0.0536J \quad (\text{bits/pixel})^2$$

$$\lambda = \exp(\frac{-3.9}{30}) = 0.8781$$

where J represents the number of independent sources multiplexed. The number of pixels per picture frame was assumed to be 250,000. The parameters α, β, and L depend on J and the number of levels M remains to be chosen. However, the value of M must satisfy the conditions of $M\alpha \leq 1$ and $M\beta \leq 1$ in Figure 2. Due to the packetized nature of the protocol under study, when aggregate video traffic is in state i, it generates i packets, each of length $L_J = \lceil \frac{L}{b} \rceil$ slots, where b is the number of bits that the channel can carry in the duration of a slot. The index J emphasizes that the video packet length depends on the number of video users transmitting.

•Voice Model

Voice users follow the simple on-off model. Each inactive voice becomes active in a frame with probability p_a. When a voice call is established, it generates a geometrically distributed number of fixed length packets with parameter $1 - p_o$, where p_o is the probability that any packet is the last one in a frame. This is in fact the discrete version of the exponential assumption for the continuous on-off model, in which p_a and p_o represent the probabilities that the "off call" and the "on call" end in a frame, respectively.

The more efficient model, in terms of bandwidth occupancy, is the one which considers the talkspurt and silence situations when a voice call is already established. This achieves almost 60% efficiency over the previous model. But, in a multi-media integrated environment, where on average a video bit rate is almost 100 times higher than a voice bit rate, the efficient model for voice using a speech activity detector does not contribute to a considerable bandwidth reduction. This, together with the simplicity of the model which provides a tractable analysis, is the motivation for choosing the simple on-off model.

3. Protocol Description

Video, voice and data are transmitted on a single common channel. Each of them has its own transmission characteristics and requirements. For example, synchronization properties of video and voice packet generation require prompt delivery of packets, while some amount of distortion, which might be due to the packet loss, can be tolerated. On the contrary, data traffic usually can not tolerate any packet loss and the accuracy of the packet delivery is of a great importance. However, finite and variable delays incurring in data packet delivery is acceptable.

In order to ensure that video and voice are delivered in a timely manner, packet transmission format is based on a frame structure. Voice and video users make reservations and data users contend for transmission.

- **Frame and Slot Structure**

The common channel time is divided into frames and slots. The frequency of frames is chosen to be the picture frame frequency, that is, 30 frames/sec. This means that the duration of a frame is $T = 1/30$ sec. Each frame is divided into N slots with each slot capable of carrying one voice packet. The length of a voice packet is $L_a = R_a T$ bits, where R_a is the voice (audio) bit rate in bits/sec. Combining all these together yields

$$N = \lfloor \frac{R_c T}{L_a} \rfloor = \lfloor \frac{R_c}{R_a} \rfloor$$

where R_c is the channel bit rate. The length of a slot is

$$b = \frac{R_c T}{N} \quad \text{bits}$$

The video packet length is $L_J = \lceil L/b \rceil$, which is equivalent to the number of mini-packets (or slots) that comprise the incremental level activity of video sources when J of them are active. For example, when J video users are active and the total aggregate transmission in a frame is at level $i \epsilon \{0, 1, \ldots, M\}$, then the total number of slots occupied is iL_J. If we consider each slot worth of information as a mini-packet, then the number of mini-packets transmitted is iL_J. Although this does not tell us how many packets (or mini-packets) are associated with any individual user, from the performance objectives point of view, it does satisfy our purposes.

Traffic allocation on the channel is performed dynamically. As Figure 1 indicates, each frame is divided into three portions for various types of traffic, in addition to the slot at the beginning of the frame, the status slot used for reservation purposes. These segments are separated by two boundaries which move dynamically on the basis of a frame-by-frame traffic allocation. The voice and video users send their packets in reserved slots and whatever is left is dedicated to the data users contending for transmission. Next we study the reservation and contention policy.

- **Reservation**

We assume a round-trip propagation delay around .26 sec in the satellite networks. We translate this value into frames as

$$\tau \triangleq \lceil \frac{0.26}{T} \rceil = 8.$$

Voice and video users, upon their arrivals, send information in the status slot to inform others of their arrivals. After τ frames, this information is accessible to all users. If this reservation is confirmed by the structure of the global information available to the sender of the reservation as well as to others, then it starts to send its packets.

We assume that a minimum number of slots is dedicated to voice users. This ensures some minimum throughput for voice users in the case of heavy traffic due to the presence of high bit rate video users. This minimum is called A_{min}. However, when A_{min} is not filled by voice users, the remaining empty slots are still available to the video and data users.

If several video and voice calls arrive in a frame simultaneously, they all send their reservations in the status slot of the next frame. If the number of requests is more than what the capacity of the channel can handle, then the following policy is pursued. First, the voice users fill A_{min}. Second, the video users are given priority, because in this way the channel is utilized more efficiently. For example, if only one video and one voice are left to be allocated and if only one of them can be served, by making the voice call the primary choice, the channel is underutilized and resource are wasted. Third, if after all video users that could be allocated ensure their reservations, there are still some available slots, then voice calls are accommodated. Notice that, this does not rule out the possibility that not all video users receive reservations, but some slots are still available for voice users to reserve. This becomes more clear when packet dropping and user blocking is explained in the next section.

The probability that several video and voice users arrive in a frame simultaneously is very low. However, if it occurs and if not all can be allocated, in order to resolve the contention situation between any type of users, let say voice users (or video users), we can assign some priority classes among them. In any event, this situation does not put any burden on the performance analysis of the protocol if we make the appropriate assumptions on the arrival processes.

After a video source establishes its reservation, it starts to inform other users of the degree of activities it possesses at any time. For example, at frame t, it sends the information on the number of packets it has ready to transmit. Then it waits for τ frames and, at frame $t+\tau$, sends its packets. These packets are received by their destinations at frame $t+2\tau$. It takes almost 2τ frames for a packet from generation to reception, τ frames for reservation and τ frames for propagation delay. This also requires a buffer of length τ picture frames at each video station.

The situation for voice users is slightly different. A voice call sends its fixed length packet as soon as it establishes its reservation. The packet generated in frame $t-1$ is sent in frame t and received in frame $t+\tau$. This ensures a fixed delay of almost τ frames, which is due only to propagation delay. There is no need for buffering at the voice stations.

- **Contention**

In contrast to video and voice, data users contend for transmission. Each idle data user generates a packet with probability p_d. At each frame, all users, including data users are informed about the reserved slots. For example, the available slots for contention in frame t is

$$F_t = N - 1 - L_{V_t} Q_t - A_t$$

where Q_t and A_t are the number of video and voice packets transmitted in frame t by video users and voice users, respectively. V_t is the number of established video users. The channel access scheme is framed Aloha with virtual variable length frame F_t.

When a user has a packet, it transmits it in frame t and waits for τ frames to see the result. If this transmission is successful, the user becomes idle in frame $t+\tau+1$, and is ready to generate a packet with probability p_d. If collision occurs, the collided packet is backloged and is retransmitted in frame $t+\tau+1$. Notice that between frames t and $t+\tau+1$ the same process happens for each user. The user which has a packet in frame $t+1$, it is informed of its transmission in frame $t+1+\tau$, and so on. In fact, there are τ interleaved processes which are independent of each other and this requires a buffer of size τ for each user.

4. Performance Study for Video and Voice

In this section, performance evaluation of this integrated multiple-access protocol for voice and video is obtained. Various types of performance measure are introduced and analyzed. Video packet dropping, video user blocking, voice user blocking, video throughput, and voice throughput are defined and corresponding formulations derived. First we elaborate on the video and voice arrivals as well as on the Markovian model of this protocol.

•Voice Arrivals

There are M_a voice terminals which become on and off in a frame with probabilities p_a^a and p_o^a, respectively. If A_t is the number of established voice (or audio) users in progress, then the evolution of this process is

$$A_{t+\tau+1} = A_{t+\tau} + U_t - W_t \tag{4.1}$$

where U_t and W_t are the number of arrivals and departures in frame t and are binomially distributed as follows:

$$Pr\{U_t = i\} = \binom{M_a - A_t - R_t}{i} (p_a^a)^i (1 - p_a^a)^{M_a - A_t - R_t - i} \tag{4.2}$$

$$Pr\{W_t = i\} = \binom{A_t}{i} (p_o^a)^i (1 - p_o^a)^{A_t - i} \tag{4.3}$$

where R_t is the number of voice terminals which are neither established nor idle. In fact, R_t is the number of voice terminals whose reservations are in transmission. This yields

$$R_t = \sum_{i=1}^{\tau} U_{t-i}. \tag{4.4}$$

In order to calculate the one step transition probability of A_t, we make an approximation under the following assumptions:

i) $A_{t+\tau} \approx A_t$

ii) $R_t \approx 0$

These two assumptions are justified. The voice process is slow and it is very likely that $A_{t+\tau}$ and A_t are the same (see [2]); this justifies both assumptions, although we quantify the validity of the second assumption in Appendix A. Yet we opt to drop these assumptions and make another reasonable one which makes the computations easier. If we consider that the number of arrivals or departures in a frame is rarely more than one, then the total aggregate process A_t can be approximated by a random walk. This is justified as follows: Let the time that a voice user being in either state "on" or "off" be exponentially distributed with a mean of 5 min. Then the probability that either a voice call arrives or a voice call terminates in a frame is as small as

$$p_a^a = p_o^a = 1 - \exp(1/(5 \times 60 \times 30)) \approx 10^{-4}.$$

In the worst case, when all users are idle, the probability of only one arrival is

$$M_a p_a^a (1 - p_a^a)^{M_a - 1}$$

whereas the probability of two arrivals in a frame is

$$\frac{M_a(M_a - 1)}{2}(p_a^a)^2(1 - p_a^a)^{M_a - 2}.$$

If

$$\frac{M_a(M_a - 1)}{2}(p_a^a)^2(1 - p_a^a)^{M_a - 2} << M_a p_a^a (1 - p_a^a)^{M_a - 1}$$

or

$$p_a^a << \frac{2}{M_a + 1}$$

then the assumption of the random walk is well made. Also notice that the same arguments is applied for two departures in a frame or one arrival and one departure in a frame. The reduction of the problem to a random walk is depicted in Figure 3, where K, x_j, and y_j are

$$K = M_a$$

$$x_j = (M_a - j)p_a^a(1 - p_a^a)^{M_a - j - 1}, \quad j = 0, 1, \ldots, M_a - 1$$

$$y_j = jp_o^a(1 - p_o^a)^{j-1}, \quad j = 1, 2, \ldots, M_a.$$

The process (A_t) is sufficient to characterize the entire voice activities. This is due to the fact that every established voice call generates a fixed-length packet in every frame during its entire duration. Therefore, A_t is also the total number of packets generated in frame t.

• Video Arrivals

There are M_v video terminals which become active and inactive in a frame with probabilities p_a^v and p_o^v, respectively. This is similar to the on-off model for voice, except that, when a video user is active it generates a random number of packets in a frame that follows the statistical activity of the motion picture. Let V_t be the total active video terminals in frame t. As is the case with voice calls, we assume that V_t is a random walk with parameters

$$K = M_v$$

$$x_j = (M_v - j)p_a^v(1 - p_a^v)^{M_v - j - 1}, \quad j = 0, 1, \ldots, M_v - 1$$

$$y_j = jp_o^v(1 - p_o^v)^{j-1}, \quad j = 1, 2, \ldots, M_v.$$

Notice that the process that characterizes the entire video activity is a double stochastic process (V_t, Q_t), where Q_t is the number of video packets generated in frame t by V_t video terminals. This process was studied earlier .

•Markovian Model

The system presentation of video and voice for this integrated scheme is characterized by a $\tau + 3$ dimensional Markov chain as $(\overline{V}_t, Q_t, A_t)$, where

$$\overline{V}_t = (V_{t-\tau}, V_{t-\tau+1}, \ldots, V_t). \tag{4.5}$$

Due to the variable rate nature of video traffic, a reservation is required for video activity. A video packet generated at the beginning of frame $t - \tau$ is transmitted in frame t. Therefore, the video activity process Q_t depends on $V_{t-\tau}$ but is independent of A_t. Similarly, $\overline{V}_t$ and A_t depend on each other and are independent of Q_t. The computation of the limiting probabilities for this $\tau + 3$ dimensional Markov process is prohibitively difficult. However, since the process V_t is slow, we may assume that it does not change (or that it changes with very low probability) its value in the course of τ frames. More specifically, we write the transition probability as

$$\begin{aligned} &Pr\left\{\overline{V}_{t+1}, Q_{t+1}, A_{t+1}/\overline{V}_t, Q_t, A_t\right\} = \\ &Pr\left\{Q_{t+1}/\overline{V}_{t+1}, A_{t+1}, \overline{V}_t, Q_t, A_t\right\} Pr\left\{\overline{V}_{t+1}, A_{t+1}/\overline{V}_t, A_t\right\}. \end{aligned} \tag{4.6}$$

This reduces to

$$Pr\left\{Q_{t+1}/V_{t-\tau}, \ldots, V_t, V_{t+1}, Q_t\right\} Pr\left\{V_{t+1}, A_{t+1}/V_{t-\tau}, \ldots, V_t, A_t\right\}. \tag{4.7}$$

Since (V_t, A_t) is Markovian and Q_{t+1} depends only on Q_t and $V_{t-\tau}$, (4.7) is reduced to

$$Pr\left\{Q_{t+1}/V_{t-\tau}, Q_t\right\} Pr\left\{V_{t+1}, A_{t+1}/V_t, A_t\right\}. \tag{4.8}$$

Because the process V_t is very slow, any change within τ frames is unlikely. This allows us to substitute $V_{t-\tau}$ by V_t in (4.8). Therefore, (4.6) is finally written as

$$Pr\left\{\overline{V}_{t+1}, Q_{t+1}, A_{t+1}/\overline{V}_t, Q_t, A_t\right\} = Pr\left\{Q_{t+1}/V_t, Q_t\right\} Pr\left\{V_{t+1}, A_{t+1}/V_t, A_t\right\}. \tag{4.9}$$

This is equivalent to the transition probability of the process (V_t, Q_t, A_t), wherein the information about the video packets generated in a frame is available to all users within that frame (immediate reservation). Consequently, Q_t is the number of video packets generated and transmitted in frame t.

•Video Packet Dropping

When the total aggregate video traffic is high and when the intensity of the other traffic and the channel bit rate constraints do not allow full accommodation of video traffic, several video packets may be dropped. This situation arises in the cases in which most of video terminals reach their high activity rate in a frame.

A packet-dropping mechanism can distribute the number of dropping packets among different video users equally. However, such packet dropping should not cause any degradation in the picture quality in the receiver. When a number of video users are active, a minimum number of slots must be guaranteed, such that any overflow of video activity beyond this minimum occurs at steady state with a probability less than some value (say 1%, see Figures 4 and 5). If the number of active video terminals is i, we need to find the corresponding minimum number of video packets q_i, such that the worst case packet dropping probability is

$$P^i_{W-drop} = \lim_{t\to\infty} Pr\{Q_t > q_i/V_u = i, A_u = N-1-q_iL_i, u\epsilon\{0,1,\ldots,t\}\} < 1\%. \quad (4.10)$$

Since Q_t only depends on V_t, we have

$$P^i_{W-drop} = \lim_{t\to\infty} Pr\{Q_t > q_i/V_u = i, u\epsilon\{0,1,\ldots,t\}\} < 1\%. \quad (4.11)$$

Notice that $q_i\epsilon\{0,1,\ldots,M\}$. The process Q_t is a random walk as in Fig. 2; therefore, the limiting distribution is a binomial with parameters M and $p = \alpha/(\alpha+\beta)$. Using this distribution yields

$$P^i_{W-drop} = \sum_{j=q_i+1}^{M} \binom{M}{j} p^j(1-p)^{M-j} < 1\%. \quad (4.12)$$

Next we define m_0 as the maximum number of video users that can be accommodated under the best circumstances. We choose m_0 as the maximum number to satisfy

$$N-1-L_{m_0}q_{m_0} \geq A_{min}. \quad (4.13)$$

In order to find m_0 and the q_i s, we need to satisfy (4.11) and (4.13). After they are determined, we compute the packet-dropping probability when the number of video users is $i\epsilon\{1,2,\ldots,min(m_0,M_v)\}$. This probability is defined as

$$P^i_{drop} = \lim_{t\to\infty} Pr\{V_t = i, L_iQ_t > N-1-A_t\} \quad (4.14)$$

and can be written as

$$P^i_{drop} = \sum_{k=0}^{min\{M_a,N-1-L_iq_i\}} \lim_{t\to\infty} Pr\{V_t = i, L_iQ_t > N-1-k, A_t = k\} \quad (4.15)$$

or

$$P^i_{drop} = \sum_{k=0}^{min\{M_a, N-1-L_i q_i\}} \sum_{j \epsilon \{j: jL_i > N-1-k\}} \pi_{ijk} \tag{4.16}$$

where $\{\pi_{ijk}\}$ is the limiting probabilities of the three dimensional process (V_t, Q_t, A_t). Finally, the video packet-dropping probability is evaluated from

$$P_{drop} = \lim_{t \to \infty} Pr\{L_{V_t} Q_t > N - 1 - A_t\} = \sum_{i=1}^{min(m_0, M_v)} P^i_{drop}. \tag{4.17}$$

Thus we only have to compute these limiting probabilities $\{\pi_{ijk}\}$. This task is performed in the later sections.

• Video and Voice Throughput

The throughput for video or voice is defined as the expectation of the number of video or voice packets in a frame. If we divide by N, the normalized throughput (number of packets in a slot) is obtained.

$$\eta_v = \frac{1}{N} E[Q_t L_{V_t}] = \frac{1}{N} \sum_{i=1}^{min(m_0, M_v)} \sum_{j=0}^{M} \sum_{k=0}^{min\{M_a, N-1-L_i q_i\}} j L_i \cdot \pi_{ijk} \tag{4.18}$$

$$\eta_a = \frac{1}{N} E[A_t] = \frac{1}{N} \sum_{i=0}^{min(m_0, M_v)} \sum_{j=0}^{M} \sum_{k=0}^{min\{M_a, N-1-L_i q_i\}} k \cdot \pi_{ijk}. \tag{4.19}$$

• Video Blocking

A video user finds itself blocked upon its arrival, if $V_t = i$ and

$$N - q_{i+1} L_{i+1} \leq A_t \leq min\{M_a, N - 1 - q_i L_i\}.$$

This probability is

$$P^v_{block} =$$

$$\sum_{i=n_0}^{min(m_0, M_v)} \lim_{t \to \infty} Pr\{V_t = i, N - q_{i+1} L_{i+1} \leq A_t \leq min\{M_a, N - 1 - q_i L_i\}\} \tag{4.20}$$

where n_0 is such that

$$N - q_{n_0+1}L_{n_0+1} \leq min\left\{M_a, N-1\right\} \leq N - 1 - q_{n_0}L_{n_0} \tag{4.21}$$

where $q_{m_0+1}L_{m_0+1}$ and q_0L_0 are assumed to be N and 0, respectively. This corresponds to the situation where a video user arrives and finds the system in the boundary states as indicated by a dashed line in Figure 5. By using the limiting probabilities of the process (V_t, A_t), this blocking probability is expressed as

$$P^v_{block} = \sum_{i=n_0}^{min(m_0,M_v)} \sum_{k=N-q_{i+1}L_{i+1}}^{min\{M_a,N-1-q_iL_i\}} \pi_{ik}. \tag{4.22}$$

• **Voice Blocking**

Similarly, a voice call finds itself blocked upon its arrival, if $V_t = i$ and $A_t = N - 1 - q_iL_i$. This probability is

$$P^a_{block} = \sum_{i=n_0+1}^{min(m_0,M_v)} \lim_{t\to\infty} Pr\left\{V_t = i, A_t = N - 1 - q_iL_i\right\} \tag{4.23}$$

or

$$P^a_{block} = \sum_{i=n_0+1}^{min(m_0,M_v)} \pi_{i,(N-1-q_iL_i)}. \tag{4.24}$$

This corresponds to the system being in one of the black circles in Figure 5.

• **Steady State Distribution of (V_t, Q_t, A_t)**

The three-dimensional process (V_t, Q_t, A_t) is a finite state aperiodic irreducible Markov chain and hence ergodic. The limiting probabilities exist and are nonzero [9].

$$\pi_{ijk} \triangleq \lim_{t\to\infty} Pr\left\{V_t = i, Q_t = j, A_t = k,\right\} \tag{4.25}$$

for $(i,k)\epsilon S_0$ and $j\epsilon\left\{0,1,\ldots,M\right\}$, where

$$S_0 = \left\{(i,k) : 0 \leq i \leq min(m_0, M_v),\ 0 \leq k \leq min(N-1-q_iL_i, M_a)\right\}.$$

This limiting probability is obtained from

$$\pi_{ijk} = \sum_{i',j',k'} P_{i'j'k' \rightarrow jkl} \cdot \pi_{i'j'k'} \tag{4.26}$$

where

$$P_{i'j'k' \rightarrow ijk} = Pr\{V_{t+1} = i, Q_{t+1} = j, A_{t+1} = k / V_t = i', Q_t = j', A_t = k'\} \tag{4.27}$$

or

$$P_{i'j'k' \rightarrow ijk} =$$

$$Pr\{Q_{t+1} = j / V_t = i', Q_t = j'\} \, Pr\{V_{t+1} = i, A_{t+1} = k / V_t = i', A_t = k'\}. \tag{4.28}$$

The variables i', j' and k' in the triple summation in (4.26) take the values

$$max(i-1,0) \leq i' \leq min(i+1, m_0, M_v)$$

$$max(j-1,0) \leq j' \leq min(j+1, M) \quad \text{if } i' \neq 0$$

$$0 \leq j' \leq M \quad \text{if } i' = 0$$

$$max(k-1,0) \leq k' \leq min(k+1, N-1-q_iL_i, M_a).$$

In the matrix representation, (4.26) reduces to

$$\pi = P\pi \tag{4.29}$$

where π is the vector of steady state probabilities and P the transition matrix whose entries are $P_{i'j'k' \rightarrow ijk}$. The size of this matrix for values of $min(m_0, M_v) = 2, M = 10$ and $M_a = 30$ can be up to 1023 by 1023. However, many of these entries are zero. This reduces the size of calculations if we use the iteration method instead of solving (4.29). The first term in (4.28) takes 363 and the second term takes 8649 entries in the computer memory. The limiting probabilities are obtained through iteration as follows:

$$Pr\{V_{t+1} = i, Q_{t+1} = j, A_{t+1} = k\} =$$

$$\sum_{i',j',k'} P_{i'j'k' \rightarrow ijk} \cdot Pr\{V_t = i', Q_t = j', A_t = k'\} \tag{4.30}$$

where $P_{i'j'k' \rightarrow ijk}$ is the one-step transition probability as in (4.28). This iteration is performed until convergence is achieved. For this convergence we require that

$$\frac{|\pi_{ijk}(t+1) - \pi_{ijk}(t)|}{\pi_{ijk}(t)} \leq 5 \times 10^{-4}, \quad \forall i, j, k. \tag{4.31}$$

5. Performance Study for Data

In this section, performance evaluation of this integrated multiple-access protocol for data is obtained. First data arrival process and Markovian model of the system is explained and then performance measures such as data throughput and delay are introduced and obtained.

• Data arrivals

There are M_d data terminals accessing the channel randomly. A user that is idle in a frame generates a packet with probability p_d and sends it based on framed Aloha. If this packet is successful, the user will be idle at τ frames later. Otherwise, it will be backloged. It is possible that a terminal sends a packet in frame t which is in collision, while the terminal is still idle in frames $t+1, t+2, \ldots, t+\tau$. In fact, there exist $\tau+1$ interleaved processes that are disjointed from each other. Each of them has access to the channel every $\tau+1$ frames. For example, $t, t+\tau+1, t+2\tau+2, \ldots$ is one of them and $t+1, t+\tau+2, t+2\tau+3, \ldots$ is another one.

Let us define B_t as the number of backloged terminals in frame t. All new arrivals in frame t join B_t and all together attempt to transmit in frame t. Therefore, the aggregate number of packets in frame t is

$$G_t = B_t + D_t \tag{5.1}$$

where D_t is the new packet arrivals in frame t and given that $B_t = i$, D_t is binomials with parameters $M_d - i$ and p_d.

The way that the aggregate data traffic G_t attempts to transmit in frame t is based on the framed Aloha operation. The number of available (or free) slots, in which these data users contend for packet transmission is

$$F_t = N - 1 - L_{V_t} Q_t - A_t. \tag{5.2}$$

An idle data user generates a packet with probability p_d within a frame. First, if this packet is generated while the video and voice portions of the frame are in progress, it joins the backloged packets and waits until the data portion of the frame is available. Then, it chooses one of F_t slots for transmission with equal probability. Second, if this packet is generated in the data portion, it is transmitted in the first available slot. In any case, any packet generated in a frame is transmitted with probability one in that frame and chooses one of the available slots with equal probability.

•Markovian Model

The system presentation of this integrated video, voice and data is a $\tau + 4$ dimensional Markov process $(\overline{V}_t, Q_t, A_t, B_t)$, which is approximated to the four-dimensional Markov process (V_t, Q_t, A_t, B_t), in the same manor that was discussed for video and voice case earlier.

•Steady State Distribution of (V_t, Q_t, A_t, B_t)

Similarly, the four-dimensional process (V_t, Q_t, A_t, B_t) is a finite state aperiodic irreducible Markov chain and hence ergodic. Therefore, the limiting probabilities exist. Let

$$\pi_{ijkl} \triangleq \lim_{t \to \infty} Pr\{V_t = i, Q_t = j, A_t = k, B_t = l\} \tag{5.3}$$

where

$$i \epsilon \{0, 1, \ldots, min(m_0, M_v)\}$$

$$j \epsilon \{0, 1, \ldots, M\}$$

$$k \epsilon \{0, 1, \ldots, min(N - 1 - q_i L_i, M_a)\}$$

and

$$l \epsilon \{0, 1, \ldots, M_d\}.$$

This limiting probability is obtained from

$$\pi_{ijkl} = \sum_{i',j',k',l'} P_{i'j'k'l' \to ijkl} \cdot \pi_{i'j'k'l'} \tag{5.4}$$

where

$$P_{i'j'k'l' \to ijkl} =$$
$$Pr\{\ V_{t+\tau+1} = i, Q_{t+\tau+1} = j, A_{t+\tau+1} = k, B_{t+\tau+1} = l /$$
$$V_t = i', Q_t = j', A_t = k', B_t = l'\ \}. \tag{5.5}$$

The size of the transition matrix for the values $min(m_0, M_v) = 2$, $M = 10$, $M_a = 30$, and $M_d = 20$ can be up to 21483 by 21484. However, some of the entries are zero. This again reduces the size of calculations. We further elaborate on this by realizing that the number of data-backloged packets in frame $t + \tau + 1$ depends on the number of data-backloged packets, the number of established voice calls, the number of established videos, and the video activity process in frame t. But, video and voice in frame $t + \tau + 1$ do not depend on data. By using these facts the transition probabilities reduce to the two smaller sizes as

$$P_{i'j'k'l' \to ijkl} = Pr\{B_{t+\tau+1} = l / V_t = i', Q_t = j', A_t = k', B_t = l'\}$$
$$Pr\{V_{t+\tau+1} = i, Q_{t+\tau+1} = j, A_{t+\tau+1} = k / V_t = i', Q_t = j', A_t = k'\}. \tag{5.6}$$

We use the iteration method which is denoted as

$$\pi(t + \tau + 1) = P \cdot \pi(t) \tag{5.7}$$

where $\pi(t)$ is the state vector at frame t and t is a multiple integer of $\tau + 1$. We also write this as

$$Pr\{V_{t+\tau+1} = i, Q_{t+\tau+1} = j, A_{t+\tau+1} = k, B_{t+\tau+1} = l\} =$$
$$\sum_{i',j',k',l'} P_{i'j'k'l' \to ijkl} \cdot Pr\{V_t = i', Q_t = j', A_t = k', B_t = l'\} \tag{5.8}$$

where $P_{i'j'k'l' \to ijkl}$ is as (5.6). For the calculation of this transition in (5.6) we need up to $3 \times 11 \times 31 \times 21 \times 21 = 451143$ entries for the first term and $(3 \times 11 \times 31)^2 = 1046529$ for the second term. This reduces the number of storage in the computer from $(3 \times 11 \times 31 \times 21)^2 = 461519289$ to $451143 + 1046529 = 1497672$. The iteration in (5.8) is performed until convergence is achieved. We assume this convergence is reached when

$$|\pi_{ijkl}(t + \tau + 1) - \pi_{ijkl}(t)| \leq 10^{-5}, \quad \forall i, j, k, l. \tag{5.9}$$

This iteration is very time-consuming. Therefore, establishing a method for obtaining an appropriate initial value for this iteration is of great importance. This issue is further discussed in Appendix B.

Next we evaluate the terms in (5.6). The second term is the $\tau+1$-step transition probability of the three-dimensional process (V_t, Q_t, A_t). This is computed from the first step transition probability, which is available as (4.27) or (4.28) (see also Appendix B). The first term corresponds to the probability of backloged data packets in frame t. This is evaluated as follows. The total aggregate data packets in frame t is

$$G_t = B_t + D_t = S_t + B_{t+\tau+1} \tag{5.10}$$

where D_t is the number of arrived packets in frame t and S_t the number of successful packets being transmitted in frame t. D_t has a binomial distribution with parameters $M_d - B_t$ and p_d. According to the framed Aloha protocol, the probability of a successful packet, given the frame length $F_t = f$ and the number of attempting packets $G_t = g$ [10], is

$$Pr\left\{S_t = s/F_t = f, G_t = g\right\} = \frac{(-1)^s g! f!}{f^g s!} \sum_{h=s}^{min(f,g)} \frac{(-1)^h (f-h)^{g-h}}{(h-s)!(f-h)!(g-h)!}. \tag{5.11}$$

The first term in (5.6) is

$$\begin{aligned} &Pr\left\{B_{t+\tau+1} = l/V_t = i', Q_t = j', A_t = k', B_t = l'\right\} = \sum_{max(0,l-l')\leq r \leq M_d - l'} \\ &Pr\left\{B_{t+\tau+1} = l/F_t = max(N-1-L_{i'}j' - k', 0), B_t = l', D_t = r\right\} \\ &\cdot Pr\left\{D_t = r/B_t = l'\right\} \end{aligned} \tag{5.12}$$

where the second term in the summation is

$$Pr\left\{D_t = r/B_t = l'\right\} = \binom{M_d - l'}{r} p_d^r (1-p_d)^{M_d - l' - r} \tag{5.13}$$

and, according to (5.10), the first term is equivalent to

$$Pr\left\{B_{t+\tau+1} = l/F_t = max(N-1-L_{i'}j' - k', 0), G_t = l' + r\right\} \tag{5.14}$$

or equivalent to

$$Pr\left\{S_t = l' + r - l / F_t = max(N-1-L_{i'}j'-k',0), G_t = l'+r\right\} \tag{5.15}$$

which is computed from (5.11). The final result for (5.12) is

$$\begin{aligned}
& Pr\left\{B_{t+\tau+1} = l / V_t = i', Q_t = j', A_t = k', B_t = l'\right\} = \sum_{max(0,l-l')\leq r\leq M_d-l'} \\
& Pr\left\{S_t = r + l' - l / F_t = max(N-1-L_{i'}j'-k',0), G_t = l'+r\right\} \\
& \cdot Pr\left\{D_t = r / B_t = l'\right\}.
\end{aligned} \tag{5.16}$$

•Data Throughput and Delay

The average throughput for data traffic is defined as the average number of data packets that are successfully transmitted during a frame in a steady state situation. Dividing this throughput by N yields the normalized throughput, which is the average successful data packets transmitted in a slot. By using this definition and (5.10) the average data throughput η_d becomes

$$\eta_d \triangleq \frac{1}{N} E\left\{\lim_{t\to\infty} S_t\right\} = \frac{1}{N} E\left\{\lim_{t\to\infty} D_t\right\} \triangleq \frac{1}{N}\overline{D}_t. \tag{5.17}$$

The average delay for a data packet is defined as the number of frames from the time of generation of a packet to the time of successful transmission of the packet. We proceed with the following approximate analysis for delay. When a packet arrives in a frame, it waits until the next data frame (data portion of a frame) and then joins the backloged users. This means that, after the lapse of a residual time, the packet is being treated as a backloged packet. If this packet arrives in the data frame, then it is immediately considered to be backloged. The delay incurred to any data packet then consists of two portions: the residual time and the delay from the contention process, which is obtained from Little's result. If we let d_n be the delay of the n th packet, then

$$d_n = w_n + \frac{\overline{B}_t}{\overline{D}_t} \tag{5.18}$$

where w_n is the residual time of the nth packet and $\overline{B}_t$ the average number of backloged data packets in a steady-state situation. The average data packet delay is obtained by taking the expectation of (5.18)

$$d = \overline{w} + \frac{\overline{B}_t}{\overline{D}_t}. \tag{5.19}$$

First we evaluate $\overline{w}$. If the nth packet arrives in the data frame, then $w_n = 0$; otherwise, $w_n = w_1$, where w_1 is the time delay between the arrival time and the beginning of the data frame. In other words,

$$w_n = \begin{cases} w_1 & \text{w.p.} \quad \text{a} \\ 0 & \text{w.p.} \quad \text{1-a} \end{cases} \tag{5.20}$$

where a is the probability that this packet arrives in the first portion of the frame (status slot+video+voice). This probability is

$$a = min\{\frac{L_{V_t} Q_t + A_t + 1}{N}, 1\}. \tag{5.21}$$

Conditioned on the values of w_1, and of video and voice activities, the average of w_n is

$$\overline{w}_n = w_1 a. \tag{5.22}$$

Conditioned on the values of video and voice activities only, w_1 is a uniform random variable distributed in $[0, a]$; hence its average is $a/2$. Therefore, conditioned on the values of video and voice, (5.22) yields

$$\overline{w}_n = \frac{1}{2} a^2. \tag{5.23}$$

In order to remove the last conditions, we need to take the expectation of (5.23) with respect to the video and voice activities, as $t \to \infty$. The final result is

$$\overline{w} = \frac{1}{2} \sum_{i=0}^{min(m_0, M_v)} \sum_{j=0}^{M} \sum_{k=0}^{min\{M_a, N-1-L_i q_i\}} \left(min\{\frac{jL_i + k + 1}{N}, 1\} \right)^2 \cdot \pi_{ijk}. \tag{5.24}$$

Now it remains to obtain $\overline{B}_t$ and $\overline{D}_t$. The first one is

$$\overline{B}_t = \sum_{l=0}^{M_d} l \cdot \pi_l \tag{5.25}$$

where $\{\pi_l\}$ is the steady-state distribution of the process (B_t) and is obtained as

$$\pi_l = \sum_{i,j,k} \pi_{ijkl}. \tag{5.26}$$

Similarly, $\overline{D}_t$ is

$$\overline{D}_t = \sum_{n=0}^{M_d} n \cdot \pi_n^D \tag{5.27}$$

where $\{\pi_n^D\}$ is the steady state distribution of the process (D_t) and is defined as

$$\pi_n^D = \lim_{t \to \infty} Pr\{D_t = n\}. \tag{5.28}$$

It is easy to show that

$$\pi_n^D = \sum_{l=0}^{M_d - n} Pr\{D_t = n/B_t = l\} \cdot \pi_l \tag{5.29}$$

where $Pr\{D_t = n/B_t = l\}$ is binomial as in (5.13).

6. Numerical Results

The status of this integrated video, voice, and data protocol is very dynamic. Several performance criteria were introduced during the study of this access protocol. These performance criteria depend on the various types of system parameters. The numerical results, for all range of values, is a prohibitive task and we do not intend to provide such a complete set of results. However, we provide some typical numerical results which give a better picture of the way this protocol performs.

There are two sets of results: First, the ones which evaluate the performance of the system based on the video and voice population only, and they merely use the steady-state distribution of the three dimensional Markov process. Performance measures, such as video packet dropping probability, video and voice blocking probability, and video and voice throughput, lie in this category. The second set of results deals with the performance of the protocol based on the access of the data to the channel. Performance measures, such as data delay and throughput, are studied in this category. We first proceed with the first set of numerical results.

In Fig. 8a, the following parameters are set to be

$R_c = 15\ Mbps$

$R_a = 64\ Kbps$

$A_{min} = 5$

$M_v = 5$

$M_a = 30$

$M = 10$

$p_o^v = p_o^a = p_o = 10^{-4}$

Based on these values of parameters, it turns out that at most two ($m_0 = 2$) video users can be accommodated (see Fig. 7). The corresponding threshold levels, over which packet dropping may occur, are $q_1 = 7$ and $q_2 = 9$. The video packet lengths are $L_1 = 19$ slots and $L_2 = 25$ slots, respectively. Number of slots per frame N is 234. Packet dropping occurs whenever the number of active voice users is more than $N - 1 - q_2 L_2 = 8$, and this happens only when the number of active video users is two. We set $p_a^v = p_a^a \triangleq p_a$ in Fig. 8a, and study different performance curves for the range of values between 10^{-5} and 10^{-3}. Typical value for the activity of a source which in the average stays idle for five minutes, is around $p_a = 10^{-4}$. As Fig. 8a indicates, P_{drop} increases first and then drops. According to the Fig. 7, the system reaches to the top faster than to the right, as p_a increases, but then the tendency towards right prevails, because of the higher number of voice users. The same analogy is true for the behavior of the voice-blocking probability, which is the probability of the system being in the black circle in Fig. 7. The video blocking probability is monoton increasing, because the system is pushed to its boundary as users become more active. The voice throughput is also a monoton function of p_a, because the system overally moves to the right. However, the situation for video throughput is slightly different. First, it increases because the system tends to go to the higher number of active videos, but then it decreases because the system is pushed further to the right or lower number of active video users, and finally it stays almost constant.

Figures 8b and 8c depict the situations in which only one type of traffic activity is changing

(p_a^a or p_a^v) and the other one is fixed. The analytical trend of these curves are justified in the same manor as before.

In Figure 8d, the channel bit rate and the number of voice calls are increased to $R_c = 20$ Mbps and $M_a = 50$, respectively. The parameters turns out to be as follows. $m_0 = 3, q_1 = 7, q_2 = 9, q_3 = 9, L_1 = 19, L_2 = 25, L_3 = 31$.

The second part of the results concerns the performance of the protocol for data. In Fig. 9, the normalized throughput for data is depicted versus the data activity p_d for $M_d = 20$. All video and voice parameters are the same as the ones in Fig. 8a. In addition to that, $p_a^a = p_a^v = p_a = 10^{-4}$. This curve is an increasing function and does not achieve a local maximum for $M_d = 20$. Nevertheless, we expect to observe such a maximum point for a higher number of data users. In fact, the average combined video and voice throughput from Fig. 8a at the operating point $p_a = 10^{-4}$ is 0.315, which leaves 0.685 packets per slot for data users to contend. Therefore, it is a reasonable conclusion that, for this number of data users, the channel is underutilized. For higher number of users, however, the sizes of computations would increase to a point where our computer facility could not afford the sufficient memory and speed for this task.

Fig. 10 illustrates the average data packet delay versus p_d for the same values of parameters. For higher values of M_d, we expect a sharp increase around the point, where the average throughput is maximum.

Fig. 11 is the illustration of delay versus throughput. For this light data traffic, this function is increasing. Similarly, for larger number of data users, we expect a sharp increase in delay around the maximum throughput.

There is a remarkable point to make here. For the given channel bit rate, which perhaps could support only two video users in a dedicated line, we have been able to accommodate 30 more voice calls in addition to 20 data users. And yet we are able to increase these numbers under a reliable acceptable performance.

7. Summary

In this paper, we considered multi-media integration of broadband services in a broadband satellite network. Voice, data, and video telephony with a broad range of service (bit) rates were multiplexed through a broadband satellite channel in a multiaccess fashion. The specification of an access protocol, which integrates all these services while keeping the demand requirements of each service within an acceptable limit, was a significant contribution of this paper.

The transmission format was based on frame by frame transmission in which video and voice users (synchronous sources) made reservations and data users (asynchronous sources) accessed the channel randomly by using a framed Aloha protocol. Two movable boundaries separated different traffic types.

The models and the arrival processes of different traffic types (i.e., data, voice, and video with finite population sizes) were introduced. The arrival processes were Binomial with their corresponding parameters. The parameters characterizing these models and arrival processes (the system variables), were specified. Voice and video users both followed the simple on-off model. Each inactive user became active in a frame with some probability, and each active one became idle with some other probability. When a voice call is established, it generated a geometrically distributed number of fixed length packets and transmitted each packet within a frame. In contrast to voice, the number of generated video packets was not fixed within a frame and followed a stochastic process, conditioned on the number of active video terminals. The characterization of such stochastic process to model video user activity was another important contribution to this paper. We proposed a discrete-time random walk to model the activity of a group of independent video telephone scenes. We used the experimental data reported in [3] for bit rate collected with a conditional replenishment interframe coding scheme.

We introduced the notion of video packet dropping when the total aggregate video traffic was high. Based on packet dropping mechanism and its probability, we were able to come up with the switching curve specifying the dynamic of movable boundary between voice and video traffic.

The propagation delay of satellite was taken into consideration. This delay made the Markov chains characterizing the performance model of the integrated protocol, to be of high dimensions. A well justified approximation reduced these dimensions to sizes at which numerical evaluation of limiting probabilities was feasible. We obtained the limiting probabilities of these Markov chains through iteration method which proved to perform better than other methods.

The calculation of limiting probabilities allowed us to examine the quality of this protocol for performance measures, such as video packet dropping probability, voice and video blocking probability, average video, voice and data throughput, and average data packet delay.

Appendix A

In this appendix, we want to show that the assumption of $R_t \approx 0$ is a good one. By taking the expectation of (4.4) the mean value of R_t is obtained as

$$\overline{R}_t = E\{\sum_{i=1}^{\tau} U_{t-i}\}. \qquad (A1)$$

All terms in the summation are binomially distributed and their means are upperbounded by $M_a p_a^a$, therefore,

$$\overline{R}_t \leq \tau M_a p_a^a. \qquad (A2)$$

For typical values of $M_a = 30$ and $p_a^a = 10^{-4}$ and $\tau = 8$, this is

$$\overline{R}_t \leq 0.024 \qquad (A3)$$

The variance of R_t can also be shown to be upperbounded by

$$Var(R_t) \leq \{\sum_{i=1}^{\tau} \sqrt{U}_{t-i}\}^2 \leq \tau^2 M_a p_a^a. \qquad (A4)$$

By using the Tchebyshev inequality we can write

$$Pr\{|R_t - \overline{R}_t| \geq \delta^2\} \leq \frac{Var(R_t)}{\delta^2} \leq \frac{\tau^2 M_a p_a^a}{\delta^2} \qquad (A5)$$

or, equivalently,

$$Pr\{\overline{R}_t - \delta \leq R_t \leq \overline{R}_t + \delta\} \geq 1 - \frac{Var(R_t)}{\delta^2} \geq 1 - \frac{\tau^2 M_a p_a^a}{\delta^2}. \qquad (A6)$$

For the typical values above and $\delta = 1 - 0.024 = 0.976$,

$$\Pr\{R_t = 0\} \geq 0.80 \qquad (A7)$$

or

$$\Pr\{R_t > 0\} < 0.20. \qquad (A8)$$

Similarly,

$$\Pr\{R_t > 1\} < 0.05 \qquad (A8)$$

$$\Pr\{R_t > 2\} < 0.02 \qquad (A9)$$

$$\Pr\{R_t > 3\} < 0.01. \qquad (A10)$$

Appendix B

In this appendix we show how to choose the initial values for the steady-state vector in (5.8) in order to reduce the number of iterations in computer. We write (5.8) as follows:

$$\pi_{ijkl} = \sum_{i',j',k',l'} Pr\{B_{t+\tau+1} = l/V_t = i', Q_t = j', A_t = k', B_t = l'\} \cdot P^{\tau+1}_{i'j'k' \to ijk} \cdot \pi_{i'j'k'l'} \tag{B1}$$

where $P^{\tau+1}_{i'j'k' \to ijk}$ is the $(\tau + 1)$ step transition probability of the process (V_t, Q_t, A_t) which is obtained from the first transition probability matrix whose entries are defined as (4.27). The entries of this matrix (let say A(m,n)) correspond to probability transitions $P_{i'j'k' \to ijk}$ as follows:

$$n = R_i + [min(N - 1 - q_i L_i, M_a) + 1]j + k + 1 \tag{B2}$$

where

$$R_i = \sum_{l=0}^{i-1} (M + 1)[min(N - 1 - q_l L_l, M_a) + 1] \tag{B3}$$

$$R_0 = 0$$

m is also expressed in terms of i', j' and k' in a manner similar to that of (B2).
For any values of i, j, k and l, the variables i', j', k' and l' in the quadruple summation in (B1) take the values

$$max(i - \tau - 1, 0) \leq i' \leq min(i + \tau + 1, m_0, M_v) \tag{B4}$$

$$0 \leq j' \leq M \tag{B5}$$

$$max(k - \tau - 1, 0) \leq k' \leq min(k + \tau + 1, N - 1 - q_i L_i, M_a) \tag{B6}$$

$$0 \leq l' \leq M_d. \tag{B7}$$

In order to get the desired convergence, for each iteration in (B1), the summation should span over all the values of i', j', k' and l'. This takes large cpu time for each iteration. However, if the number of iterations is reduced by an intelligent choice of the initial value, the computation of this iteration process is feasible. In order to do that we reduce the span of variables in the summation in (B1) and hence, reduce the iteration time. The iteration process is performed with the reduced set of variables until the desired convergence is reached. Then, this result is used as the initial value for the iteration process with the complete set of variables in that summation. By using this method, we observed considerable reduction in cpu time. It remains to show how to reduce the set of variables. According to the voice and video models, jumps from any state only to the neighboring states are permitted. But in the $(\tau+1)$th step, jumps of maximum length $(\tau+1)$ is possible for voice as well as video. This fact is clearly indicated in (B4) and (B6). However, the likelihood of lengthy jumps is very low. This allows us to disregard this possibilities in the reduced set case. For example, we can limit the jumps to be maximum two. The following examples illustrate the reduction in (B4) and (B6).

1) If $i' = i$, then

$$max(k-2,0) \leq k' \leq min(k+2, N-1-q_iL_i, M_a)$$

2) If $i' = i-1,\ or\ i+1$, then

$$max(k-1,0) \leq k' \leq min(k+1, N-1-q_iL_i, M_a)$$

3) If $i' = i-2,\ or\ i+2$, then

$$k' = k.$$

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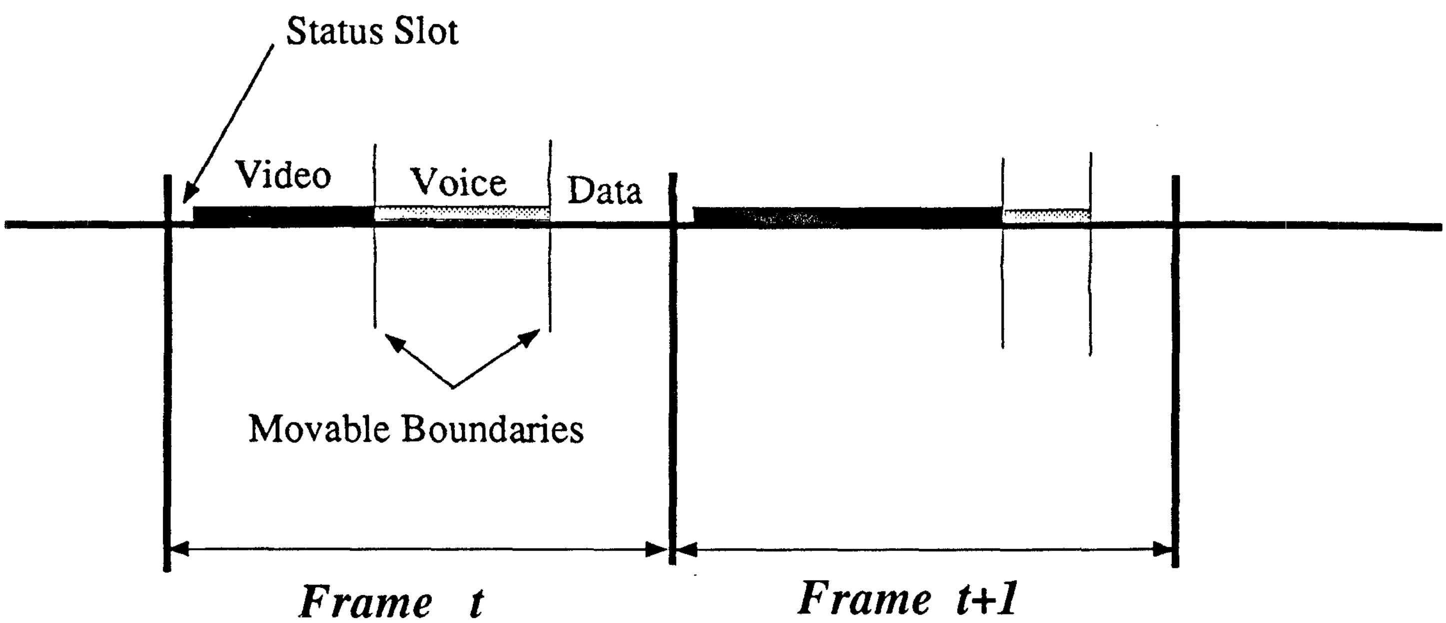

Fig. 1. Traffic allocation on a frame

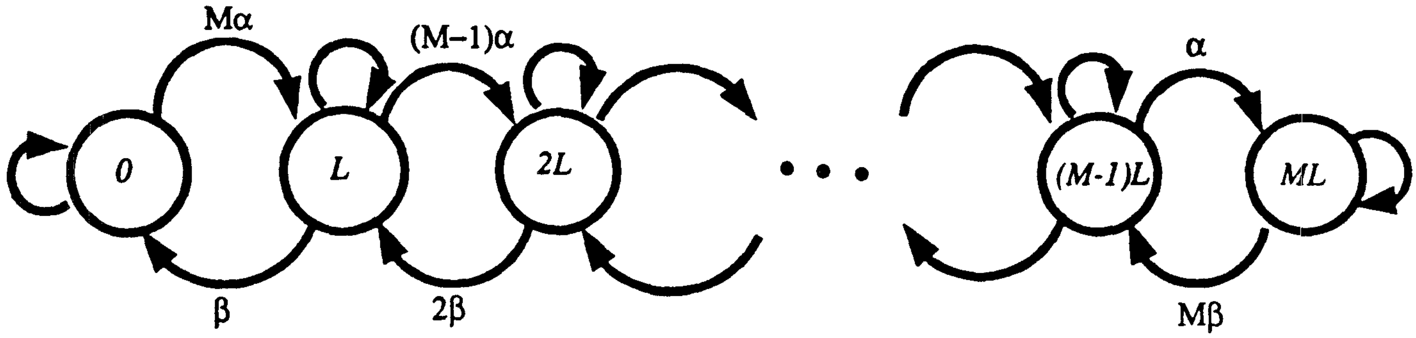

Fig. 2. Random walk for video activity

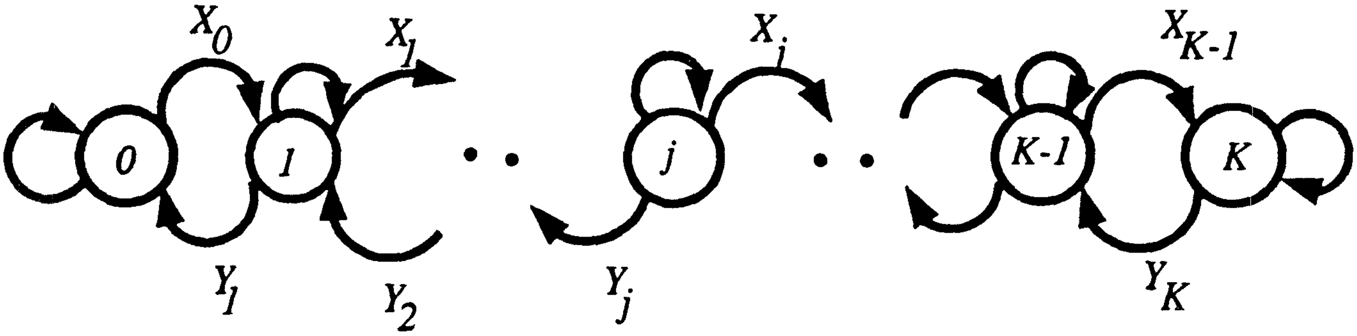

Fig. 3. Random walk for video or voice arrival process

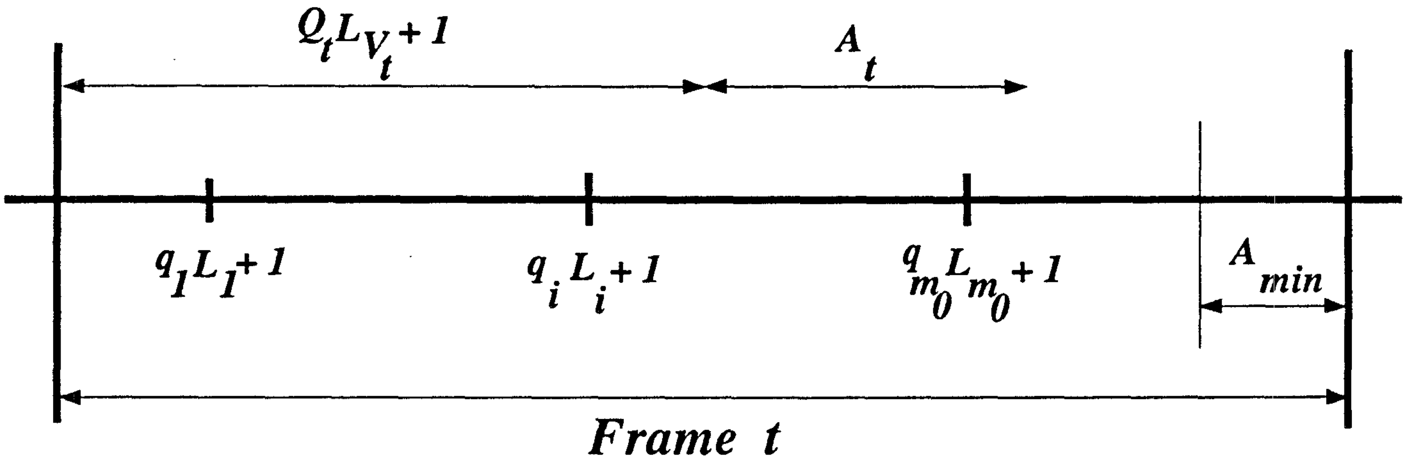

Fig. 4. Video and voice allocation on a frame

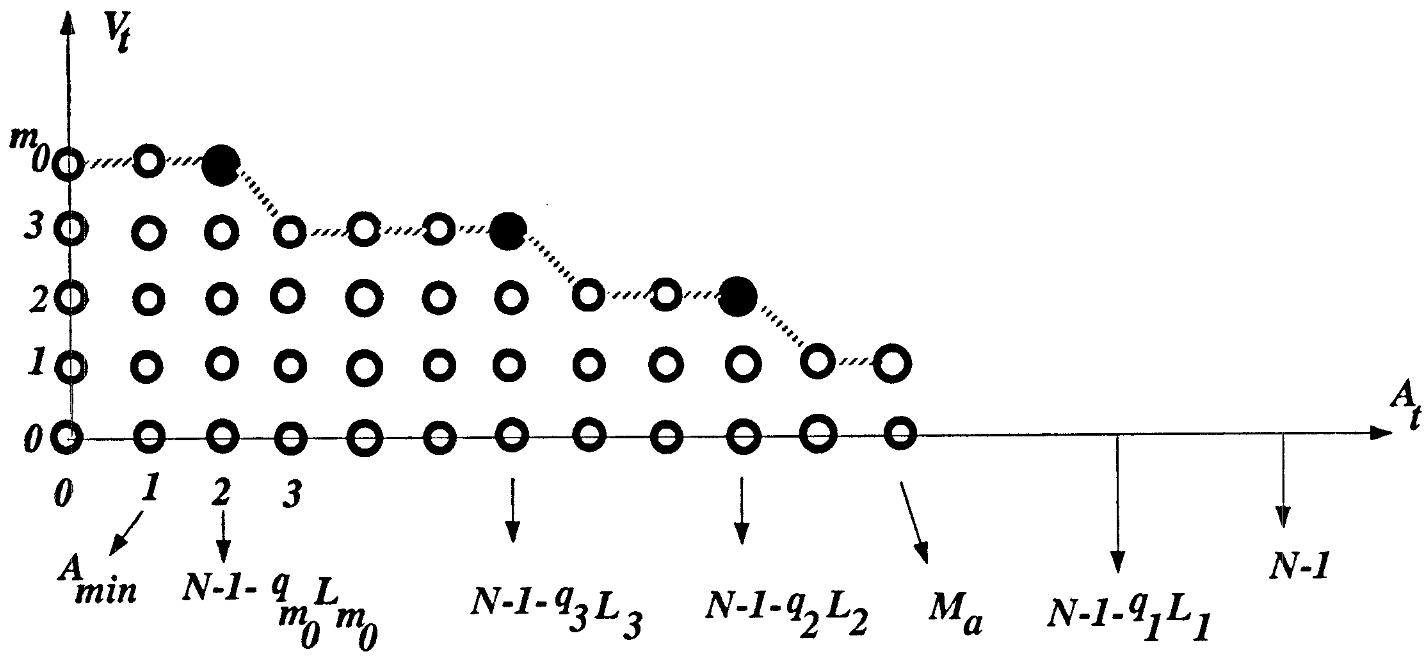

Fig. 5. Discrete-time discrete-state apace Markov chain (V_t, A_t)

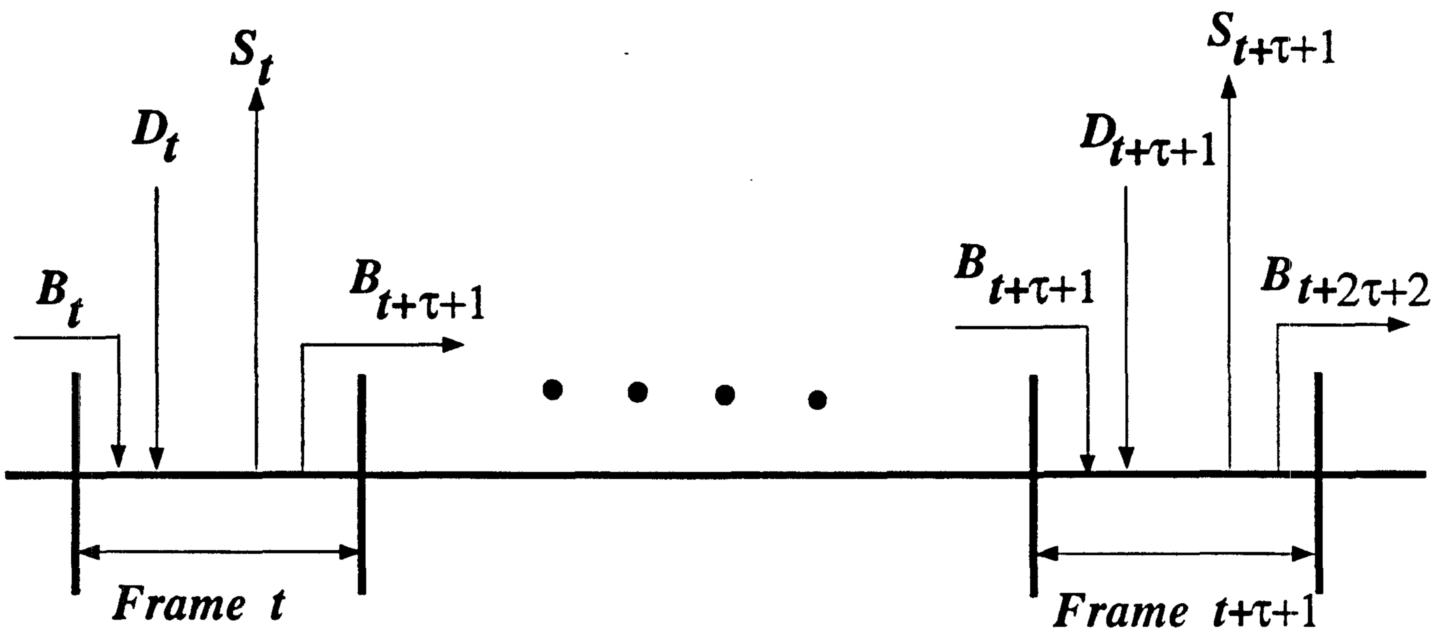

Fig. 6. Data access to the channel

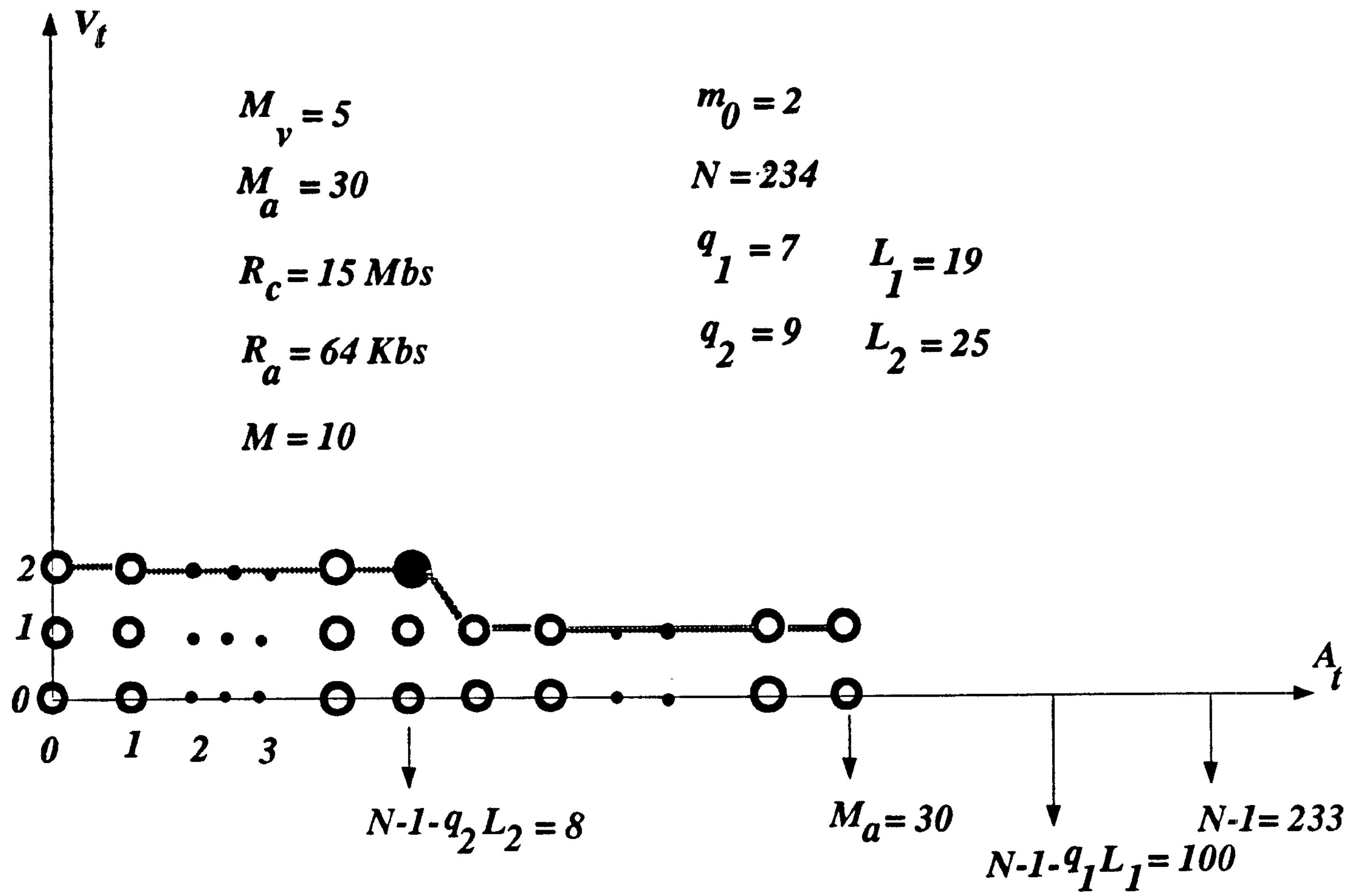

Fig. 7. Discrete-time discrete-state apace Markov chain (V_t, A_t)

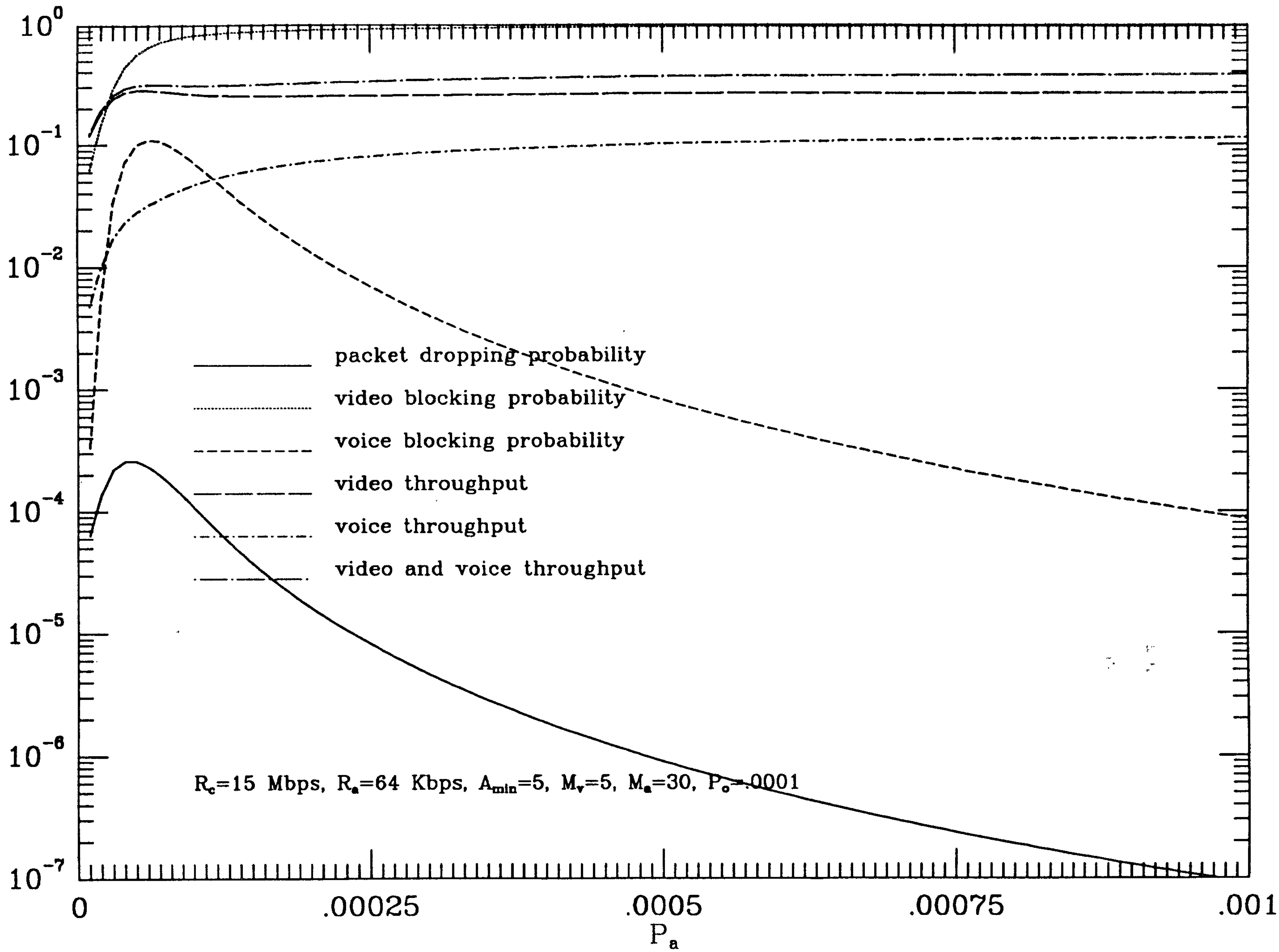

Fig. 8a. Packet dropping, blocking probabilities, and throughput versus user's activity P_a

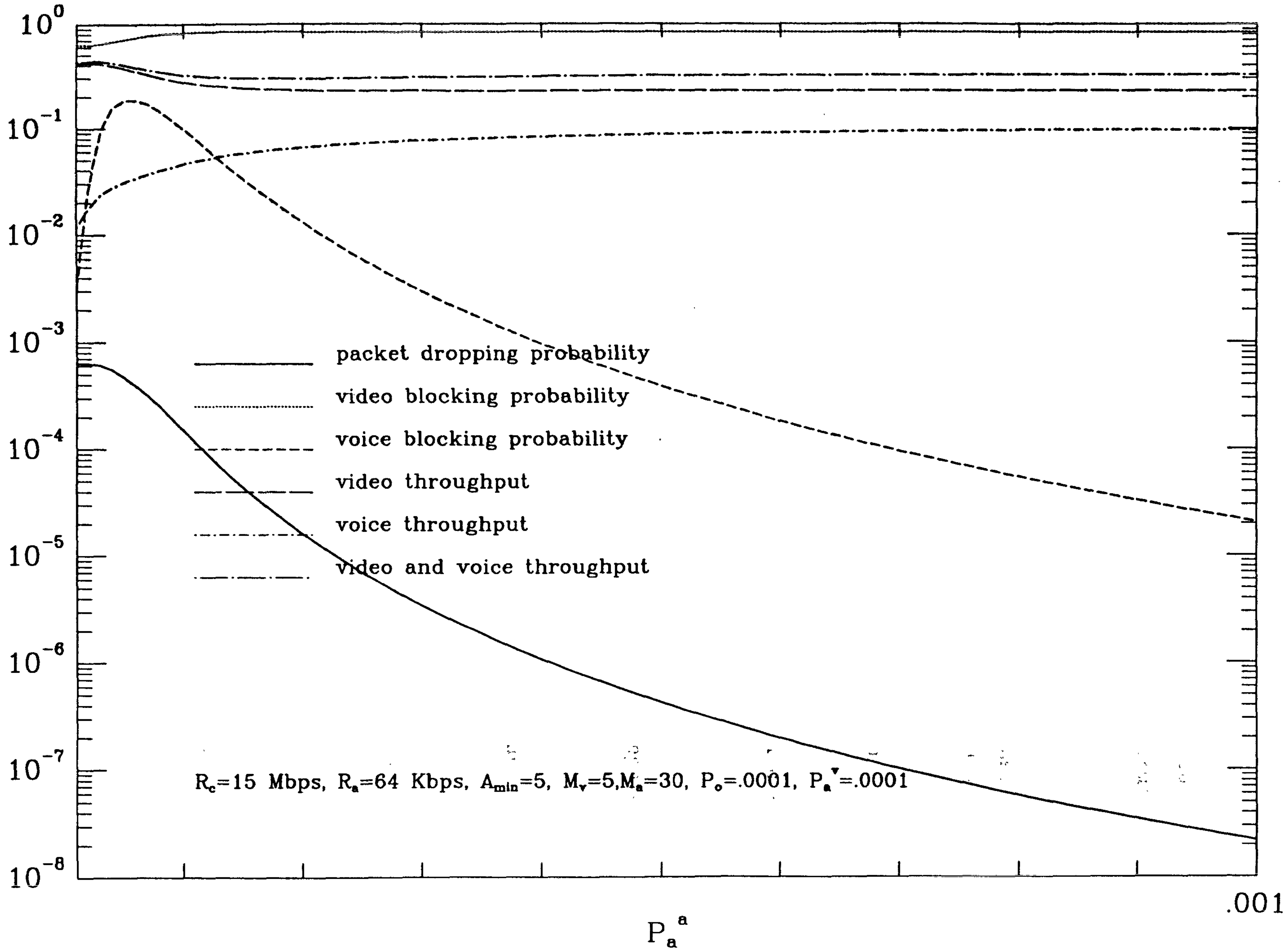

Fig. 8b. Packet dropping, blocking probabilities, and throughput versus voice activity P_a^a

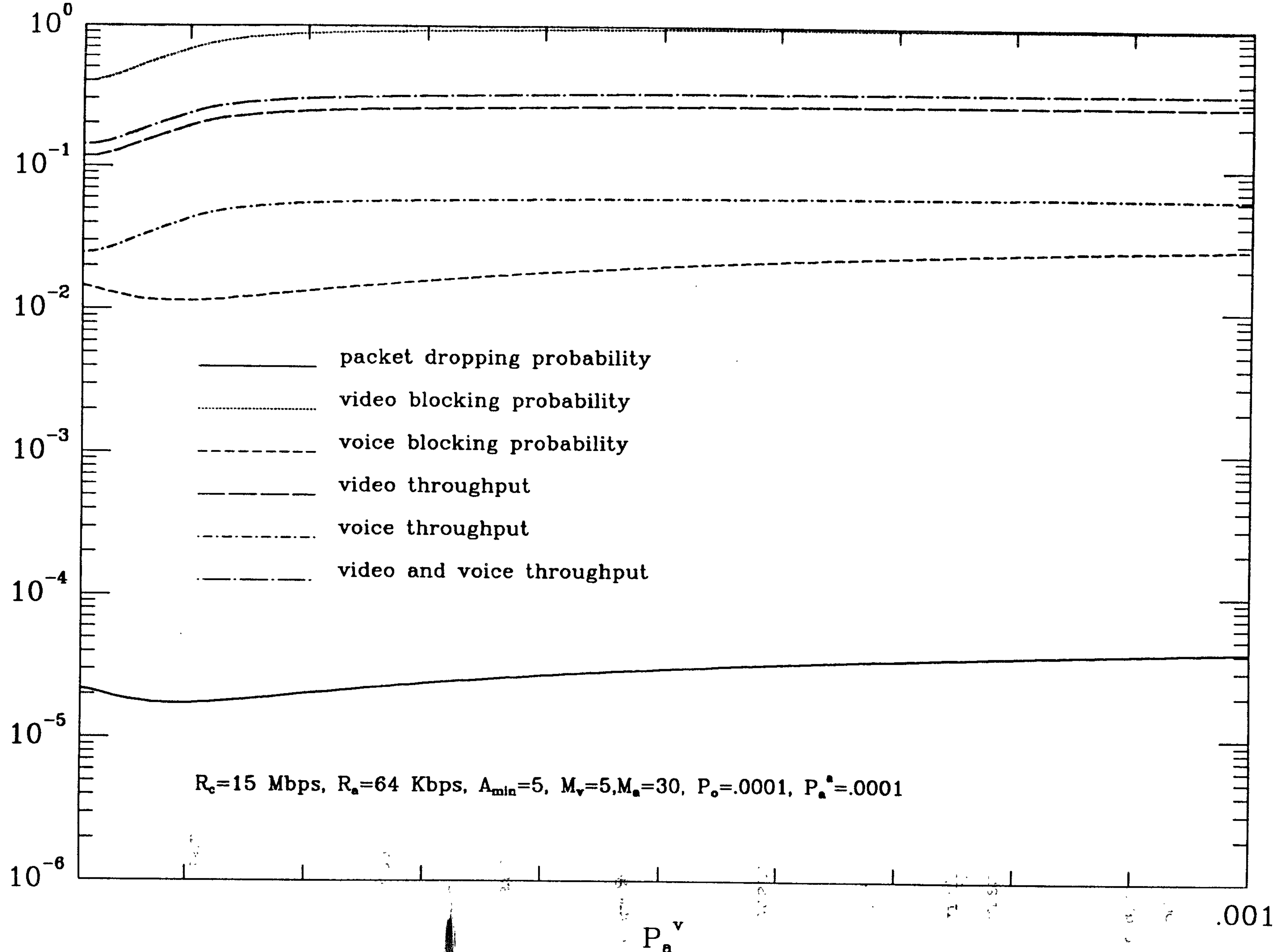

Fig. 8c. Packet dropping, blocking probabilities, and throughput versus video activity P_a^v

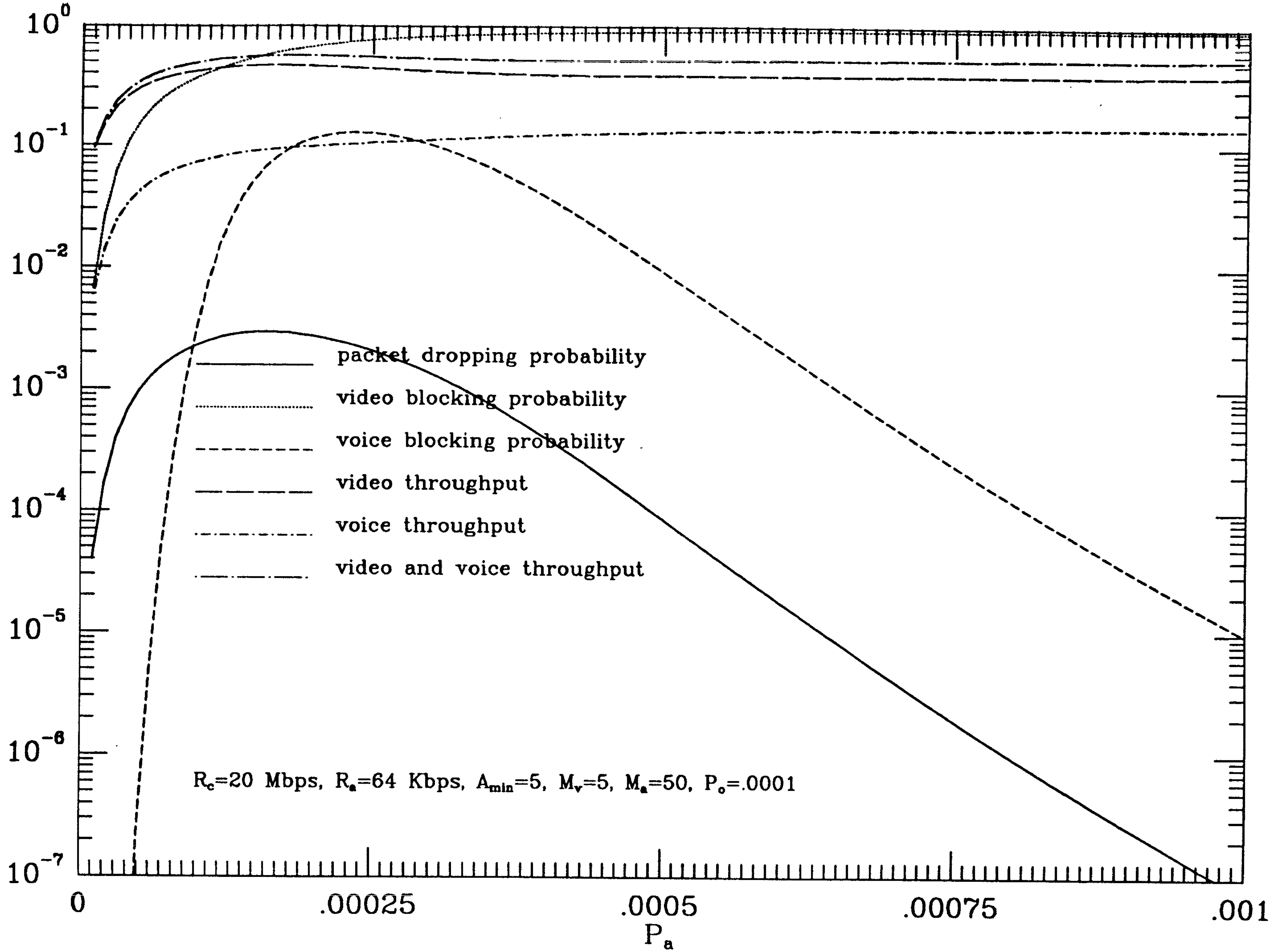

Fig. 8d. Packet dropping, blocking probabilities, and throughput versus user's activity P_a.

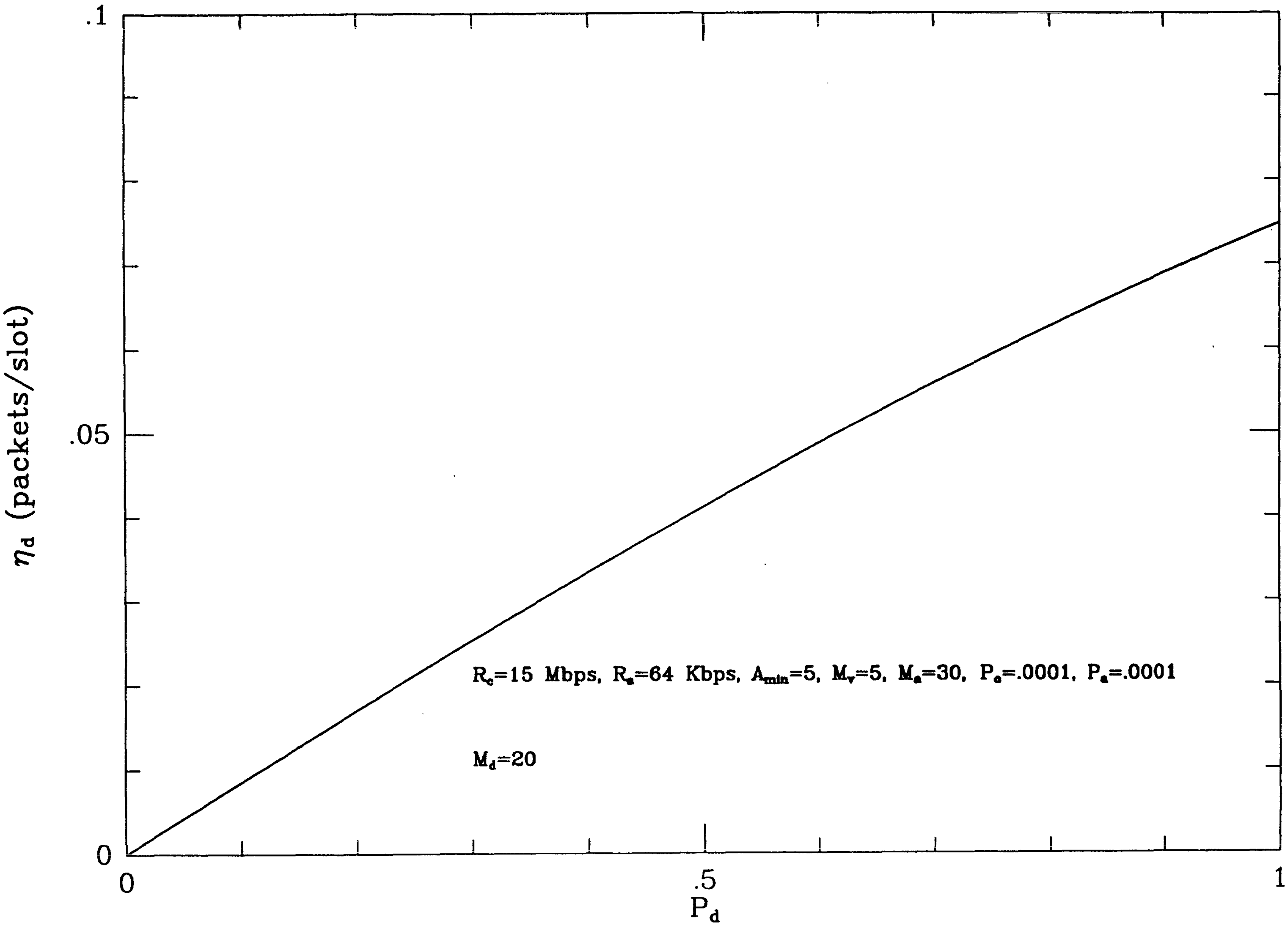

Fig. 9. Throughput versus packet arrival activity

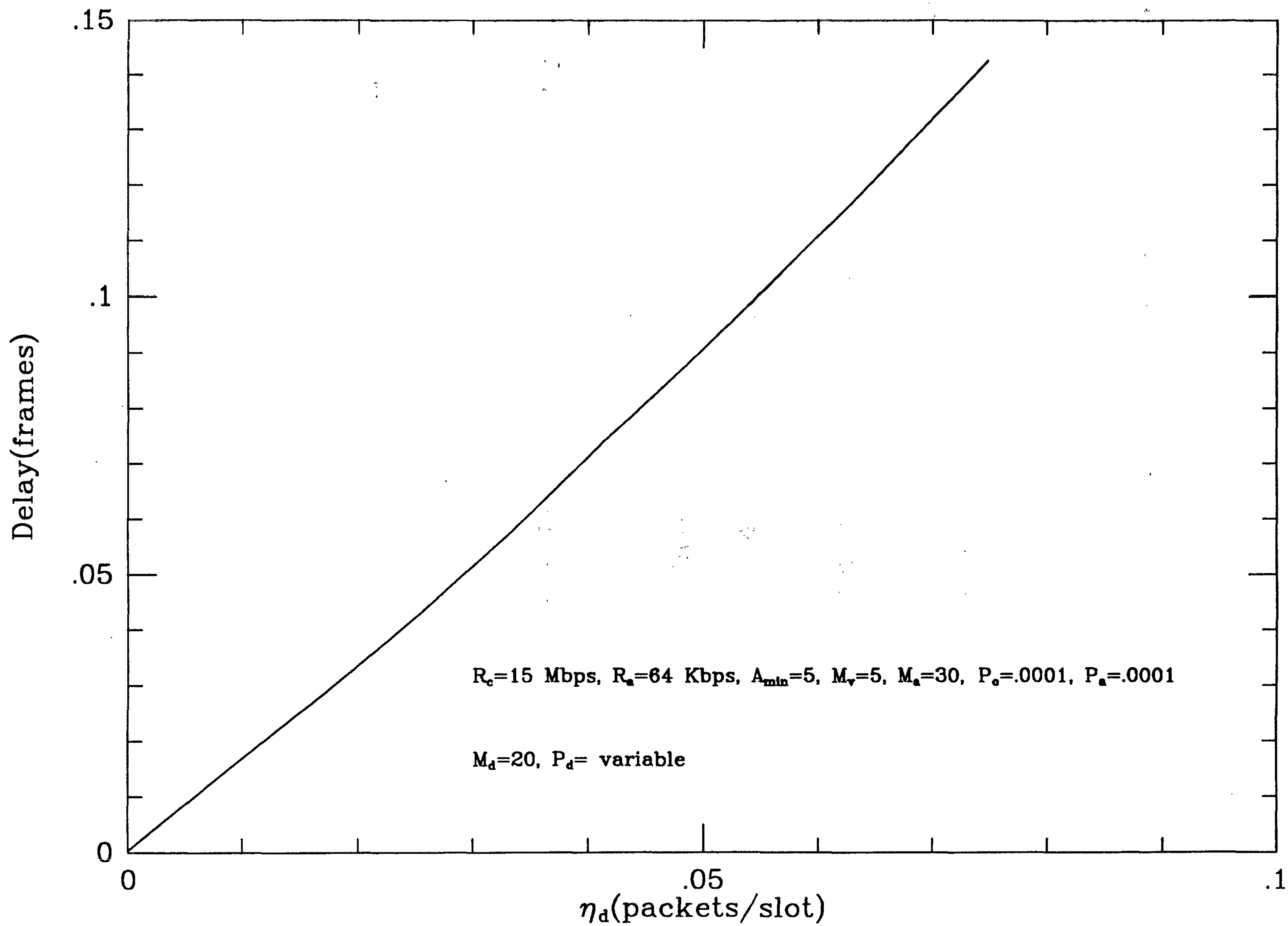

Fig. 11. Delay versus throughput for data