Performance Evaluation of Multi-Access Strategies for an Integrated Voice/Data CDMA Packet Radio Network

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13. ABSTRACT *(Maximum 200 words)*

The problem of voice/data integration in a random-access radio network employing the ALOHA protocol in conjunction with retransmission control is investigated. Channel-access control strategies are introduced that take advantage of the capability of the code-division multiple-access (CDMA) channel to accommodate several voice calls simultaneously, while the data users contend for the remaining (if any) multiple-access capability of that channel. The retransmission probabilities of the backlogged data users are updated based on estimates of data backlog and number of established voice calls, which are obtained from the side information about the state of channel activities. A two-dimensional Markovian model is developed for the voice and data traffic. Based on this model, the voice-call blocking probability, the throughput of both traffic types, and the delay of the data packets are evaluated and the tradeoffs between the parameters of different traffic types are quantified. It is observed that by taking advantage of the multiple-access capability of the CDMA channel in the control of data traffic, we may achieve movable-boundary channel access in the code domain.

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1. INTRODUCTION

The evolution of present communication networks toward integrated services digital networks (ISDN) to accommodate random demands for service from a population of heterogeneous users, has presented new problems to communication engineers. One of the basic problems is the integration of a variety of data types (such as interactive data, digital voice, video) over a common channel. Integration of different traffic types over a common channel requires a method to determine how the users should coordinate their transmissions to avoid the destructive interference which occurs when the traffic load over the channel increases. In other words, a channel-access scheme has to be followed by the terminals of possibly different traffic types in order to make efficient use of the channel.

Different approaches to channel access have been pursued in single traffic-type networks. Circuit switching techniques have been used extensively in telephony, while packet-switched networks have become the main carriers of data traffic (see [1]). Different methods of packet communications in data networks have been described in [2]. The main idea behind the integration of voice and data traffic in a single network has been the use of hybrid circuit/packet switching techniques (see [3]-[4]).

In this report, we consider the problem of voice and data integration in a code-division multiple-access packet radio environment. The scenario for such networks may be of terrestrial based packet radio integrated networks that support mobile terminals. They can also be a subnet of a larger system of interconnected integrated networks that provide radio interface for some users of a backbone packet-switched network. Among the fringe benefits of spread-spectrum techniques are the selective addressing/reception capability and the antijamming capability. They may also support several transmissions simultaneously, and this is exactly the situation investigated in this report.

Effective integration of voice and data onto one network requires the advent of transmission protocols for voice and data and a dynamic data flow control scheme. Flow control is necessary to prevent the degradation of voice quality. Voice traffic is characterized by the need for real-time delivery, while voice packets show greater tolerance to transmission

errors. On the other hand, data traffic requires very low packet error probability and there is no need for real-time delivery of data packets.

Consideration of these issues leads to the conclusion that voice traffic demands contention-free channel access once a voice call has been established. On the other hand, data terminals can contend for the use of a channel and adapt to the availability of communication resources, as the need arises. We consider the slotted ALOHA protocol, in conjunction with retransmission control via channel load sensing, for data nodes (see [7],[11]). At the end of each slot, the (re)transmission probability of data nodes is updated on the basis of estimates of the data backlog and the number of established voice calls currently using the channel. We present several different methods for the control of traffic and compare their performance. The multiple-access capability of code-division multiplexing is taken into consideration in the control of both traffic types. Voice-call blocking is used as a way to control the load of voice traffic. We consider two cases: one in which the blocking is done solely on the basis of number of established voice calls and another in which the blocking depends on the state of the data traffic, as well as on the voice traffic. We will see that the former case provides priority for the voice traffic at the expense of lower data throughput and higher delay for data packets, while the latter serves both traffic types more fairly. Variations of these two schemes are also considered involving different feedback information obtainable from the channel and different maximum allocations of the channel multiple-access capability to the voice and data traffic. A problem in the transport of packetized speech is the stringent delay requirement for maintaining a reasonable quality of conversation. Since each voice call generates a random number of data equivalent packets whenever a voice call is established, the voice node sends its packets in successive slots until the call terminates. In this way, packets of established voice calls never experience delays. Voice calls are blocked on the basis of the feedback information about the number of established voice calls and the number of data nodes in backlog mode.

Due to the graceful degradation of the performance of the CDMA channel, imperfections in the estimates of data backlog and number of established voice calls (i.e., inaccurate or even different estimates of these quantities by the various users) does not severely affect the performance of the channel access protocol. Nevertheless, we consider two distinct

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scenarios regarding the channel feedback information: according to the first, only the total number of transmitting data and voice users is available to all users (e.g., via direct channel sensing); according to the second, both the data backlog and the number of established voice calls become available to the all the users (e.g., by a central controller).

The other important issue addressed in this report is that of movable-boundary access of the CDMA channel for data/voice integration. In data/voice integration schemes that use movable boundary channel access in the time or frequency domains, there is typically an initial maximum allocation of time slots (or frequencies) to voice and a corresponding (considerably smaller) one for data; thus, some kind of boundary separates the resources allocated to the two traffic types. In most scenarios, the data can use some of the resources allocated to voice, if the temporary voice traffic requirements are below the initial allocation; thus the boundary can move to the one direction. According to the most flexible scenario, the boundary can also move to the other direction; that is, if the initial data allocation is not fully used, the voice traffic is allowed to use the surplus temporarily. All the schemes for data/voice integration in CDMA packet radio networks introduced in this report take advantage of the multiple-access capability of the CDMA channel in ways that achieve truly (two-directions) movable-boundary channel access in the code domain.

This report is organized as follows. Section II presents the system model and introduces the global information available to all nodes in the network. In Section III, we develop a Markovian model for voice and data traffic. Section IV introduces three different control policies for both traffic types. Based on the Markovian model, the voice-call blocking probability, voice-call throughput, and the throughput and delay of data nodes are evaluated for a system with a finite number of voice and data nodes. The tradeoffs for the aforementioned performance measures among the different traffic types are discussed in Section V, which contains the numerical results.

2. SYSTEM MODEL

In the network model of this report it is assumed that the nodes are divided into two classes, one of which generates voice exclusively and the other data exclusively. Specifically, there are M_d data users and M_v voice users. The voice users transmit continuously (un-

slotted) using CDMA. The data users employ a controlled slotted ALOHA random-access packet broadcast protocol. No buffering of data takes place at the nodes. Each inactive data and voice node follows the Bernoulli process with parameters P_d and P_v, respectively. Packet transmissions start at common clock instances and packets have constant length of L symbols. Each user (data or voice node) employs a random frequency hopping pattern for the transmission of its packets. The frequency spectrum is divided into q frequency slots and each symbol is transmitted at a frequency chosen from the q frequencies with equal probability, independently of the frequencies chosen for other symbols. We assume that some form of Forward Error Control (FEC) coding is employed, for which up to ν symbol errors can be corrected. We also assume that a packet consists of exactly one codeword. Therefore, a packet is declared successfully transmitted, if at most ν symbol errors occur.

One of the main advantages of CDMA techniques, such as frequency hopping, is that the channel can support several transmissions simultaneously. This is called the multiple-access capability (MAC) of CDMA techniques (see [5]). We define the MAC indices K_d and K_v as the maximum number of users (data and/or voice) that can be accommodated simultaneously, so that the packet error probability remains below a specified threshold for the corresponding traffic type. This means that, given P_E^d and P_E^v as the maximum tolerable data and voice packet error probabilities, respectively, the MAC indices K_d and K_v for the respective traffic types are obtained from the following relationships:

$$P_E(k) \leq P_E^d \quad \forall\, k \leq K_d$$

$$P_E(k) \leq P_E^v \quad \forall\, k \leq K_v \tag{2.1}$$

where $P_E(k)$ is the packet error probability in the presence of k simultaneous packet transmissions, where k includes both voice and data users. In other words, if the total number of simultaneous users is $k \leq K_d$, then all k data or voice packets are received with acceptable error probability; if $K_d < k \leq K_v$, then among the k packets the voice packets are received with acceptable error probability, whereas the data packets have higher than the acceptable error probability; finally, if $K_v < k$, then all voice or data packets are received with unacceptable error probabilities. In practice, $P_E^v > P_E^d$ and therefore

$K_v > K_d$. Also notice that this discussion assumes the same code rate for both voice and data packets. It could be extended to systems with different code rates for the data and voice packets.

The channel access protocol for data nodes is the delayed-first-transmission (DFT) protocol, under which new data packets join the backlog before their first transmissions are attempted, and each packet is independently transmitted in slot t ($(t, t+1]$) with probability f_t (see [6]). It is assumed that global information about the state of channel activities can be obtained to update the retransmission probability f_t at the end of slot t. Once a data node enters the backlog mode, it does not generate new packets until its backlogged packet is successfully transmitted; in other words there is no buffering of data packets.

All voice calls are accepted if their number is less than the threshold. In particular, new voice calls enter the system until the interference level in the channel passes the threshold. New call attempts may be blocked and cleared on the basis of the global information available. There is also no buffering of voice calls. Each voice call generates a random number of packets, geometrically distributed with parameter p. Established voice calls send their packets in successive slots until the call terminates. In other words, except for the time to packetize the speech signals, established calls experience no delays.

As described above, the flow of both traffic types is controlled according to the global information available to all nodes in the network. In conventional ALOHA-type data networks, the flow of data traffic is usually controlled according to some information about the value (estimate) of the data backlog process (see [6]-[8]). Here we base our information on the data backlog and the number of established voice calls in progress. Specifically, the global information available to all nodes in the network at time t is given by $\Psi(N_t^d, N_t^v)$, where N_t^d represents the value of the data backlog at time t and N_t^v the number of established voice calls in progress at time t.

A number of different forms for $\Psi(N_t^d, N_t^v)$ may be considered, depending on the mechanism that is available to produce the feedback information. Complete feedback information corresponds to a case in which

$$\Psi(N_t^d, N_t^v) = (N_t^d, N_t^v),$$

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i.e., both N_t^d and N_t^v are available to all nodes in the network. The availability of such information implicitly assumes the existence of either a central controller that keeps track of the number of voice calls in progress, or some form of additional control channel that permits the dissemination of this information in a distributed manner. In many applications, such information will not be available. In that case, $\Psi(N_t^d, N_t^v)$ is the total number of users that transmit during the slot; it includes all of the established voice calls plus a fraction of the backlogged data users, that is, we have:

$$\Psi(N_t^d, N_t^v) = \beta N_t^d + N_t^v = N_t^{dv}.$$

Thus, the channel feedback information does not permit the nodes to distinguish between voice and data users in this case, and transmission strategies must be based on incomplete information.

In this report, we assume that, under either of these scenarios, perfect feedback information is available, i.e., exact values of either (N_t^d, N_t^v) or N_t^{dv} are known. In many applications, only estimates of these quantities are usually available. These estimates may be inaccurate and even different for the individual users. However, studies of the control of ALOHA systems indicate that control policies are relatively robust, i.e., errors in the estimates of channel backlog can be tolerated without severely impacting on system performance. Moreover, further robustness is provided by the graceful degradation properties of the CDMA channel. Finally, the performance evaluation of the perfect state information case considered here provides us with upper bounds to the best achievable performance of the network.

3. MARKOVIAN MODEL OF VOICE/DATA TRAFFIC

As discussed earlier, the control of voice and data traffic is based on the global information $\Psi(N_t^d, N_t^v)$, which in turn depends on the data backlog and the number of established voice calls in progress. Therefore, it is natural to represent the state of the system by these two processes. As we shall see, this representation enables us to derive and evaluate different performance measures of the system and describe the system both dynamically and in steady state.

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To this end, we introduce the two-dimensional process $\underline{N}_t \overset{\triangle}{=} (N_t^d, N_t^v)$. The state space of this process is given by the set $S_{M_d} \times S_{M_v}$, where $S_{M_i} = \{0, 1, 2, \ldots, M_i\}$ for $i = d$ or v. It can be shown that the two-dimensional process $\underline{N}_t$ is a Markovian one. This derives from the observation that, conditioned on $\underline{N}_t$, both N_{t+1}^d and N_{t+1}^v are independent of values of $\underline{N}_s$, for $s < t$, which leads to the conclusion that the joint statistics of the vector-valued process obey the Markovian property. The evolution of the state variables $\{N_t^d\}$ and $\{N_t^v\}$ is given by

$$N_{t+1}^d = N_t^d - S_t^d + Y_{t+1}^d$$

$$N_{t+1}^v = N_t^v - T_t^v + Y_t^v \tag{3.1}$$

where S_t^d represents the number of successfully transmitted data packets in slot t, Y_{t+1}^d the number of new data packet arrivals in slot t, T_t^v the number of voice calls terminating in slot t, and Y_t^v the number of new voice calls established in slot t. Since the (re)transmission probabilities of data packets and the blocking of voice calls depend only on $\Psi(N_t^d, N_t^v)$ and the length of voice calls have geometric (memoryless) distributions, the Markovian property follows.

Once the Markovian nature of $\underline{N}_t$ is established, one can exploit this property to obtain the various statistics of the process. To evaluate the performance of the channel access schemes considered in this report, we need to compute the stationary probability distribution of $\underline{N}_t$. An equilibrium distribution for $\underline{N}_t$ does exist, as long as the chain is ergodic [9]. First, we compute the transition probability matrix Q^{dv} of the Markov process $\underline{N}_t$. The transition probability expressions of $\underline{N}_t$ are given in Appendix A. Q^{dv} is a matrix of size $(M_d + 1)(M_v + 1) \times (M_d + 1)(M_v + 1)$. The stationary distribution of the process is then evaluated from

$$P\left[\underline{N}_{t+1} = (n, m)\right] = \sum_{i=0}^{M_d} \sum_{j=0}^{M_v} P\left[\underline{N}_{t+1} = (n, m) \mid \underline{N}_t = (i, j)\right] \cdot P\left[\underline{N}_t = (i, j)\right] \tag{3.2}$$

for all $0 \leq n \leq M_d$ and $0 \leq m \leq M_v$. The limiting distribution of the process is given by

$$\Pi^{\underline{N}}(n, m) \overset{\triangle}{=} \lim_{t \to \infty} P\left[\underline{N}_t = (n, m)\right].$$

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When the Markov process is ergodic, the equilibrium distribution is same as the stationary distribution of the process (see [9]). It is observed that as long as the (re)transmission probability f_t of data packets is not identically zero, $\underline{N}_t$ is an irreducible, aperiodic Markov chain and its equilibrium distribution does in fact exist.

Once the probability distribution of the state vector of the system is obtained, it is possible in principle to derive the different statistics of important processes, such as the number of successful data packet transmissions and the composite (voice & data) packet transmissions.

4. VOICE/DATA TRAFFIC CONTROL POLICIES

Three different control schemes are introduced in this section. Scheme A characterizes the situation where the nodes of the network observe the channel activities in each slot and obtain some feedback information based on the number of active voice and data terminals. In Scheme B, we assume that a central controller provides the nodes of the network with the exact number of established voice calls and data nodes in backlog. Scheme C is a variation of Scheme B, where the voice terminals do not use the information about the data backlog to regulate their access to the channel. We also consider variations of Schemes A, B, and C according to which data and voice share the multiple-access capability of the CDMA channel in a movable-boundary fashion with varying degrees of fairness in the initial and time-varying allocation of the MAC between them. All three schemes and their variations perform movable boundary voice/data integration in the code-domain which is similar in principle to but more flexible than standard movable-boundary schemes operating in the time or frequency domain.

4.1. Scheme A. Limited Channel Feedback : $\Psi(N_t^d, N_t^v) = \beta N_t^d + N_t^v$

This scheme may characterize situations where the nodes of the network monitor the radio channel in each slot, and based on the observed level of traffic in that slot produce an estimate of the composite (voice & data) traffic level in that slot. Examples of this idea are given in [7],[11] for a frequency-hopped packet radio data network.

Since the interarrival times and the call-holding times of voice calls are considerably longer than a slot duration, the number of established voice calls N_t^v is a slowly varying

quantity. On the other hand a fraction of data nodes in backlog βN_t^d $(0 < \beta \leq 1)$ transmit in each slot with some probability. Therefore, the feedback obtained by observing the radio channel in slot $t - 1$ (i.e., the time interval $(t - 1, t]$) should provide an estimate of the quantity $\Psi(N_t^d, N_t^v) = N_t^v + \beta N_t^d$. It should be noted that since the (re)transmission probabilities of data packets are updated at the end of each slot, β should be time dependent as well. In fact, the total number of voice and data packets transmitted in time slot $t - 1$ is given by

$$X_{t-1}^T = N_{t-1}^v + Y_{t-1}^v + X_{t-1}^d \tag{4.1}$$

where X_{t-1}^d represents the number of data packet (re)transmissions in slot $t - 1$ with conditional expected value of $f_{t-1} N_{t-1}^d$, where f_{t-1} is the packet retransmission probability for slot $t-1$. We assume that by observing the channel, the nodes of the network can obtain the feedback information N_t^{dv}, which approximates X_{t-1}^T. For the reasons stated above, it is reasonable to assume that $N_t^v \approx N_{t-1}^v + Y_{t-1}^v$ for the voice traffic, and $X_{t-1}^d \approx f_{t-1} N_{t-1}^d \approx f_{t-1} N_t^d \approx \beta N_t^d$, since $N_t^d = N_{t-1}^d - S_{t-1}^d + Y_t^d \approx N_{t-1}^d$ when small size changes in the data backlog occur. We would obviously prefer to use the time-varying retransmission probability f_{t-1} instead of β in the above approximation of X_{t-1}^d but this would result in loss of the Markovian property of the two-dimensional process $\underline{N_t} = (N_t^d, N_t^v)$. Thus, we let $N_t^{dv} \overset{\triangle}{=} \beta N_t^d + N_t^v$. In other words, by listening to the channel, the nodes of the network observe N_t^{dv} and use it in their control policies.

The arrival process at each inactive voice node is Bernoulli with parameter P_v. A voice call in progress will be completed in any slot with probability p, also a Bernoulli process. We use a policy under which the new voice calls are blocked, whenever the value of the global information N_t^{dv} is greater than or equal to the voice MAC index K_v. Consequently the voice-call blocking probability in steady-state is

$$P_B \overset{\triangle}{=} \lim_{t \to \infty} P\left[N_t^{dv} \geq K_v\right] = \sum_{i=0}^{M_d} \sum_{j \geq \max(0, K_v - \beta i)}^{M_v} \Pi^{\underline{N}}(i, j). \tag{4.2}$$

The conditional distribution of number of new calls that go through is given by

$$P\left[Y_t^v = \ell \mid \underline{N_t} = (i, j)\right] = b(M_v - j, \ell, P_v), \quad \beta i + j < K_v \tag{4.3}$$

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where $b(N, n, p)$ is the binomial distribution defined by

$$b(N, n, p) \triangleq \binom{N}{n} p^n (1-p)^{N-n}; \quad 0 \le n \le N. \tag{4.4}$$

Note that

$$P\left[Y_t^v = \ell \mid \underline{N}_t = (i, j)\right] = \begin{cases} 1 & \ell = 0 \\ & \\ 0 & \ell > 0 \end{cases} ; \quad \beta i + j \ge K_v. \tag{4.5}$$

With this method of voice-call blocking, it is possible that the number of established voice calls exceeds the MAC index K_v, because all new calls in slot are accepted as long as the number of previously established calls is less than K_v. However, the threshold will be exceeded relatively infrequently and, due to the graceful degradation of the performance of the CDMA channel, will not severely affect the packet error probabilities of either traffic type.

The new voice-call throughput in steady-state η_v is the expected number of new voice calls that establish communication in each slot. This is given by

$$\eta_v = \lim_{t \to \infty} E\left\{ E\left(Y_t^v \mid \underline{N}_t\right) \right\}. \tag{4.6}$$

The conditional expectation of Y_t^v is given by

$$E\left(Y_t^v \mid \underline{N}_t = (i, j)\right) = \sum_{\ell=0}^{M_v - j} \ell \, P\left[Y_t^v = \ell \mid \underline{N}_t = i(i, j)\right]; \quad \beta i + j < K_v \tag{4.7}$$

and $E\left(Y_t^v \mid \underline{N}_t = (i, j)\right) = 0$, for $\beta i + j \ge K_v$. Averaging with respect to $\underline{N}_t$ gives us

$$\eta_v = \sum_{i=0}^{M_d} \sum_{j < K_v - \beta i} P_v \left(M_v - j\right) \Pi^{\underline{N}}(i, j). \tag{4.8}$$

Simplifying the above formula, we obtain

$$\eta_v = M_v P_v \left(1 - P_B\right) - P_v \sum_{i=0}^{M_d} \sum_{j < K_v - \beta i} j \, \Pi^{\underline{N}}(i, j) \tag{4.9}$$

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where η_v is the long-run average of number of new voice calls that establish communication in a slot. The expected number of voice packets transmitted in a slot is simply the sum of η_v and the expected number of voice calls in progress; i.e.,

$$\bar{X}_v = \lim_{t \to \infty} E\{X_t^v\} = \eta_v + \sum_{i=0}^{M_d} \sum_{j=0}^{M_v} j\, \Pi^{\underline{N}}(i,j). \tag{4.10}$$

Each unbacklogged data node generates a new packet in a slot with probability P_d. Therefore, when the backlog is at state n, the conditional distribution of number of new data packet arrivals is given by

$$P\left[Y_{t+1}^d = \ell \mid N_t^d = n\right] = b(M_d - n, \ell, P_d). \tag{4.11}$$

If the transmission of a new packet is unsuccessful, no new packet is generated by the user until the backlogged packet is transmitted successfully. The backlogged packets are transmitted independently in slot t with probability f_t. Then the conditional distribution of data packets (re)transmissions in slot t is given by

$$P\left[X_t^d = i \mid \underline{N}_t = (n, m)\right] = b(n, i, f_t) \tag{4.12}$$

The following policy is used to update the (re)transmission probabilities:

$$f_t = \begin{cases} 1; & N_t^{dv} \leq K_d \\ \alpha^{N_t^{dv}}; & N_t^{dv} > K_d \end{cases} \tag{4.13}$$

where $0 < \alpha \leq 1$. The motivation for the case where $N_t^{dv} \leq K_d$ is that, as long as N_t^{dv} is less than the MAC index of data traffic, all data packets are (re)transmitted with probability one. When $N_t^{dv} > K_d$, the data nodes employ some form of exponential backoff method, since there is the possibility that there are many voice calls in progress, in which case the channel is not reliable for transmission of data packets. For a given CDMA channel (i.e., given K_d and K_v) and population sizes M_d and M_v, the control parameter α determines the performance of the network in terms of the delay/throughput of data traffic, the blocking of voice calls, and the packet error rate experienced by the users.

In the above form of Scheme A, the new voice calls are blocked if $N_t^{dv} \geq K_v$. This corresponds to an "initial" allocation of the MAC of the CDMA channel that favors the voice traffic. It is possible to consider the following variation of Scheme A, termed Scheme A', which allows for a fairer sharing of the CDMA channel: instead of blocking new voice calls at the MAC index K_v, we block them at the MAC index K_d', which is smaller than the data MAC index K_d. Thus, since $K_d' < K_d$, the data are always allowed to use $K_d - K_d'$ "channels" (actually SS codes) and, in addition, may use some of the CDMA MAC allocated to voice when the voice traffic is low, according to the data retransmission policy of Scheme A. The analysis of Scheme A' is almost identical to that of Scheme A; the only difference is in the evaluation of the voice blocking probability and the average voice throughput, where K_v should be replaced by K_d' in the corresponding expressions provided for Scheme A.

4.2. Scheme B. Full Channel Feedback : $\Psi(N_t^d, N_t^v) = (N_t^d, N_t^v)$

In this scheme, the central controller provides the nodes of the network with exact information about the number of voice calls in progress and the number of data nodes in backlog mode. This may characterize situations where, either due to different format of voice and data packets or to the use of a reservation channel for voice, the central controller is able to obtain separate side information about different traffic types over the channel.

Since the voice terminals are furnished with both N_t^d and N_t^v, the blocking of voice calls can be done based on the information about both traffic types, or may solely depend on the information about the voice traffic. In this scheme, we propose a blocking policy which takes into account the level of data backlog. Again, we do not want the number of established voice calls to exceed the MAC index K_v. Also, in the event that too many data nodes are in backlog, we would like to even block more new calls in order to provide some capacity of the channel for use by the data traffic. Therefore, the blocking threshold should vary with the number of data nodes in backlog. One such threshold is given by

$$\Gamma(N_t^d) \stackrel{\triangle}{=} K_d + (K_v - K_d)\left(1 - \frac{N_t^d}{N_t^d + \gamma}\right) \tag{4.14}$$

where $\gamma \geq 1$. Then, the new voice calls are blocked whenever the number of established

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voice calls equals and/or exceeds $\Gamma(N_t^d)$. Notice that $K_d \leq \Gamma(N_t^d) \leq K_v$. In other words, the boundary of blocking moves as a function of data traffic. γ is a control parameter that can be chosen to meet a specification on the voice-call blocking probability.

The voice-call blocking probability is given by

$$P_B \overset{\triangle}{=} \lim_{t \to \infty} P\left[N_t^v \geq \Gamma(N_t^d)\right] = \sum_{i=0}^{M_d} \sum_{j \geq \Gamma(i)} \Pi^{\underline{N}}(i,j). \qquad (4.15)$$

Following the same steps as the last section, η_v is obtained as

$$\eta_v = M_v P_v (1 - P_B) - P_v \sum_{i=0}^{M_d} \sum_{j < \Gamma(i)} j \Pi^{\underline{N}}(i,j). \qquad (4.16)$$

In conventional ALOHA-type data networks, the optimal control strategies are shown to have the retransmission probability f_t as an inverse function of the data backlog N_t^d (see [6],[8]). We consider the following policy:

$$f_t = \begin{cases} 1; & N_t^v < K_d, \; N_t^d \leq K_d - N_t^v \\ \frac{K_d - N_t^v}{N_t^d}; & N_t^v < K_d, \; N_t^d > K_d - N_t^v \\ \alpha^{N_t^v}; & N_t^v \geq K_d \end{cases} \qquad (4.17)$$

where again $0 < \alpha \leq 1$. With this policy, as long as the number of established voice calls is below the data MAC index K_d and the number of data nodes in backlog is less than $K_d - N_t^v$, each of the backlogged nodes transmits with probability one. $K_d - N_t^v$ is the available capacity of the CDMA channel for the data traffic. When $N_t^d > K_d - N_t^v$, the data nodes in backlog transmit with similar methods as data-only networks; that is, f_t is an inverse function of data backlog N_t^d. When the number of established calls exceeds the MAC index K_d, the data nodes employ the exponential backoff as a function of the number of established calls. As the number of established calls increases, the CDMA channel becomes less reliable for transmission of data packets, and therefore data nodes transmit less frequently.

In Scheme B, the movable threshold for blocking new voice calls $\Gamma(N_t^d)$ given by (4.17) ranges between K_d and K_v. Although this is a more balanced allocation of the

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initial CDMA MAC than that provided by Scheme A, it may still not leave sufficient MAC for the data for some applications. An alternative, termed scheme B', is to define the movable threshold of voice blocking via the equation

$$\Gamma'(N_t^d) \stackrel{\triangle}{=} K_d' + (K_v - K_d')\left(1 - \frac{N_t^d}{N_t^d + \gamma}\right)$$

where γ is as in (4.14) and the MAC index K_d' is smaller than the data MAC index K_d. In this way, $K_d' \leq \Gamma'(N_t^d) \leq K_v$, and additional MAC of the CDMA channel can be allocated to the data. The retransmission policy of backlogged data for Scheme B' remains identical to that of Scheme B. Again, the analysis of Scheme B' is almost identical to that of Scheme B; the only difference is in the evaluation of the voice blocking probability and the average voice throughput, where the voice blocking threshold $\Gamma(N_t^d)$ should be replaced by the new threshold $\Gamma'(N_t^d)$ in the corresponding expressions provided for Scheme B.

4.3. Scheme C. Voice Independent of Data : $\Psi(N_t^d, N_t^v) = (N_t^d, N_t^v)$

As mentioned in the introduction, there are situations where one might want to provide priority to a class of traffic in the network. Specifically, we would like to consider the scenario, in which the voice traffic has priority over its data counterpart regarding the allocation of multiple access capability of the physical link.

Scheme C is same as Scheme B except that the blocking of the new voice calls is based only on the information about the number of established calls in progress N_t^v. Therefore, the evolution of the voice traffic process is independent of data traffic. In this case, the blocking of new voice calls is enforced according to whether N_t^v is greater than or equal to K_v or not.

It can be shown that the process $\{N_t^v\}$ is a Markov chain due to the memoryless property of geometric distribution of the duration of voice calls. Hence, the statistics of the voice traffic can be computed independently of the data traffic. The transition probabilities of $\{N_t^v\}$ are evaluated as follows:

$$P\left[N_{t+1}^v = k \mid N_t^v = m\right] = \sum_{i=0}^{M_v - m} P\left[N_{t+1}^v = k \mid N_t^v = m, Y_t^v = i\right]$$

$$\cdot P\left[Y_t^v = i \mid N_t^v = m\right] \qquad (4.17)$$

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where

$$P\left[N_{t+1}^v = k \mid N_t^v = m, Y_t^v = i\right] = b(m+i,k,1-p). \qquad (4.18)$$

The voice-call blocking probability P_B is then given by

$$P_B \overset{\triangle}{=} \lim_{t \to \infty} P\left[N_t^v \geq K_v\right] = \sum_{m \geq K_v} \Pi^v(m) \qquad (4.19)$$

where $\{\Pi^v(m)\}$ is the long-term state occupancy distribution of the process N_t^v. Similarly, η_v is given by

$$\eta_v = M_v P_v (1 - P_B) - P_v \sum_{j=0}^{K_v - 1} j \Pi^v(j). \qquad (4.20)$$

As an alternative to this scheme, one can set the voice threshold at a value less than than K_d, say at K_d' (as for Scheme A'), so that some of the channel will always be available to the data. The new scheme, termed Scheme C', has an analysis identical to that of Scheme C, where we only need to replace the voice blocking threshold K_v by K_d' in the expressions for the voice blocking probability and the average voice throughput.

4.4. Throughput/Delay of Data Traffic

In order to evaluate the data throughput, we need the marginal distribution of the number of successful data packet transmissions in a slot. The probability distribution of S_t^d is derived in Appendix B. The data throughput is then given by

$$\eta_d = \lim_{t \to \infty} \sum_{k=0}^{M_d} k \, P\left[S_t^d = k\right] \qquad (4.21)$$

and the steady-state delay $\bar{D}$ is given by Little's formula

$$\bar{D} = \frac{\sum_{n=0}^{M_d} \sum_{m=0}^{M_v} n \Pi^N(n,m)}{\eta_d}. \qquad (4.22)$$

We may also evaluate the probability distribution of the total traffic over the channel in slot t represented by X_t^T and derived in Appendix C. This probability distribution

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enables us to compute the steady-state expected number of packet transmissions and the average packet error probability in a slot given by

$$\bar{P}_E = \lim_{t \to \infty} \sum_{m=0}^{M_d + M_v} P_E(m) \, P\left[X_t^T = m\right] \qquad (4.23)$$

where $P_E(m)$ is the conditional probability of packet error in the presence of m simultaneous packet transmissions. The expected number of packet transmissions in a slot is given by

$$\bar{X}_T = \lim_{t \to \infty} E\left\{X_t^T\right\} = \lim_{t \to \infty} \sum_{m=0}^{M_d + M_v} m \, P\left[X_t^T = m\right]. \qquad (4.24)$$

5. NUMERICAL RESULTS

For the integrated random access system described above, an extended (64,32) Reed-Solomon code of rate 1/2, codeword length $n = 64$, and number of information symbols per codeword $k = 32$ was used for error correction purpose. This code can correct up to $\nu = 16$ symbol errors. Each code symbol carries 6 bits of information and is transmitted by employing 64-ary FSK modulation with noncoherent demodulation ($M = n = 64 = 2^6$). We have assumed that both voice and data packets are using the same code, although in practice, the voice packets usually do not require an error-correction code as powerful as that needed by the data packets. The specification on the packet error probabilities of both traffic types were set at $P_E^d = 5 \times 10^{-3}$ for data packets and $P_E^v = 5 \times 10^{-2}$ for voice packets. It was also assumed that the average bit error probability due only to additive white Gaussian noise (AWGN) is $P_b = 10^{-6}$. The frequency spectrum was divided into $q = 80$ frequency slots for frequency hopping purpose. The resulting MAC indices for the given P_E^d and P_E^v are $K_d = 5$ and $K_v = 8$. We chose $\beta = 0.5$ as the fraction of data nodes to the total number of voice and data nodes. We also let $\gamma = K_v$ in (4.14) for Scheme B. It represents the nominal number of voice calls that the channel may support. Smaller values of γ provides more priority for data traffic, when the number of data nodes in backlog becomes large. For convenience, we let $\lambda_d \triangleq M_d P_d$ and $\lambda_v \triangleq M_v P_v$ represent the expected arrival rate of data packets and voice calls, respectively.

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Figure 1 presents the voice-call blocking probability for the three schemes introduced above with retransmission probability parameter α as a parameter. It is observed that Scheme A is more sensitive to the choice of α, while Scheme C is not affected by the value of α, as expected. For Scheme B, where the blocking boundary moves as a function of the number of data nodes in backlog mode, the smaller value of α increases the blocking probability. This is due to the fact that, as α decreases, the number of data nodes in backlog increases, which in turn moves the blocking threshold from K_v toward K_d. Figure 2 illustrates the voice-call blocking probability with call-holding time as parameter. As expected, larger holding time (i.e., smaller p) produces larger blocking probability, since established calls stay in the channel for a longer period of time. Figure 3 presents the voice-call blocking probability with the data traffic load λ_d as parameter. Again Scheme C is unaffected by the change in the level of data traffic, while the blocking probabilities of Schemes A and B increase as the data traffic load into the channel increases. Scheme A is more sensitive to this increase in the level of offered data traffic than Scheme B. Figure 4 illustrates the percentage of new voice calls that are accepted by the system, with the data traffic input rate as parameter. For $P_v < 0.01$, all three schemes perform similarly. Scheme A shows more sensitivity to the change in the level of data traffic and this is due to the uncertainty in the feedback information as which type of traffic has more contribution in the feedback information. Figure 5 illustrates the degradation in the blocking performance of the three schemes as we double the number of voice and data terminals while keeping the same parameters of the CDMA channel as before.

From the above results we conclude that, if the primary objective of the network designer is to satisfy the needs of the voice traffic, then the control of the voice traffic should be independent of data traffic as much as possible. Therefore, Scheme C is the best among the three schemes considered as far as voice-call blocking is concerned. Scheme B does almost as well as Scheme C, while adjusting to the level of data traffic equally well.

Figure 6 shows the throughput of data traffic for different schemes with α as parameter. It is observed that the data throughput of Scheme A is much more sensitive to the choice of α than Schemes B and C. For $\alpha = 0.8$ all three schemes perform similarly, with Scheme B doing better than Schemes C and A. Decreasing α to 0.5 results in substantial degradation

in the data throughput performance of Scheme A. It achieves its maximum throughput at $P_d = 0.05$. On the other hand, Schemes B and C are not as sensitive to the choice of α as Scheme A is. They maintain an almost constant throughput rate for a wide range of P_d. Another way to observe the sensitivity of these schemes to the choice of α is, by observing the throughput/delay profile of Figure 7. Figure 8 presents the average packet error probability experienced for the same system. As we can see, the increase in traffic load of data nodes has different effects on the packet error performance of different schemes. For $\alpha = 0.8$, Scheme A shows greater control over the error rate performance and as the data traffic load becomes large, it actually forces the number of packet transmissions to decrease in order to keep the reliability of the channel intact. Scheme A also does well in terms of delay/throughput of data traffic for $\alpha = 0.8$. This superior performance comes at the expense of larger voice-call blocking probabilities. For $\alpha = 0.5$, the packet error performance of Scheme A improves, but that is due to the fact that the number of packet (re)transmissions and the number of established voice calls decreases more rapidly and the channel is idle most of the time. This can be seen from Figures 6-7 where the throughput decreases and the delay goes up exponentially. For $\alpha = 0.5$, Schemes B and C show more control over the level of traffic over the channel as the average packet error probability remains almost constant over the wide range of offered data traffic.

Figures 9-10 illustrate the expected number of total (voice & data) and voice packet transmissions in a slot (i.e., $\bar{X}_T$ and $\bar{X}_v$) for the three schemes considered. For a given λ_v, the increase in probability of data packet arrival results in decrease of voice traffic in Schemes A and B. For $\alpha = 0.8$, the decrease in the level of voice traffic is accompanied by an increase in the level of data traffic in Scheme A. In other words, the channel capacity is allocated to the traffic with more demands as the need arises. This is similar to dynamic movable-boundary methods of voice/data integration in time domain, while here we may interpret the channel capacity allocation in the code domain. Schemes B and C show more control over the total traffic level for smaller value of α. Figure 11 presents the same results for $\lambda_v = 2.0$.

Figure 12 shows the throughput/delay performance with the call-holding time as parameter. As expected, larger call-holding time causes a degradation of data performance.

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Figure 13 illustrates the throughput/delay performance of data traffic with offered voice traffic as parameter. Comparison of Figures 12-13 indicates that the degradation in the performance of data traffic due to the doubling of λ_v is almost the same as the degradation due to the doubling of average call-holding time $1/p$. Figure 14 illustrates the average packet error probability achievable by Scheme A for different parameters of voice process. The effect of control over the error rate of this scheme for a wide range of P_v is apparent. Figure 15 shows the throughput/delay performance of data traffic as we double the number of data and voice terminals, using the same CDMA channel. Figure 16 shows the throughput performance of data traffic as a function of offered voice traffic. For small values of P_v, the new calls establish communications which results in rapid decrease in the data throughput. For larger values of P_v, most new calls are blocked and for the given data traffic load λ_d, the data throughput does not change drastically.

Figures 17-18 illustrate the sensitivity of the voice-call blocking probability and throughput/delay of data to the initial choice of blocking threshold. As can be seen the throughput/delay of data traffic for Schemes A and C greatly improves at the expense of larger blocking probability as we lower the value of blocking threshold from K_v to K_d'. However, Scheme B is not as sensitive to the choice of K_d', since the blocking boundary still moves between K_v and K_d'. When the data traffic load becomes large, the advantage of having $K_d' < K_d$ is that there will be $K_d - K_d'$ of the MAC capacity available for data nodes.

6. CONCLUSIONS

The problem of voice/data integration over a radio channel was considered in this report. Code-division multiplexing techniques were employed as an alternative to traditional FDMA and TDMA schemes to offer resource sharing to both traffic types. We developed a Markovian model of the voice and data traffic. Based on this model, we evaluated the steady-state performance of the network in terms of the blocking probability of voice calls, the throughput/delay profile of data packets, and the expected packet error probability encountered over the channel.

The control policies introduced here are based on the amount of feedback information that is available to the nodes of the network. The different parameters of control policies

can be adjusted to meet the required performance indexes of voice and data traffic. For Schemes A and B, the channel resources are shared in a dynamic way. As the load for one type of traffic increases, the multiple-access capability of the CDMA channel gracefully switches to meet the demands of that traffic. Scheme C provides priority for voice traffic, as voice traffic control acts independently of the level of data traffic. Comparison of different control policies shows that the multiple-access capability of CDMA techniques can be utilized in an efficient way to allocate the communication resources of the network among the different traffic types. Depending on the amount of feedback information available to the nodes of the network, we may develop priority policies for a specific type of traffic. Scheme C considered here clearly gives priority to the voice traffic at the expense of lower throughput and larger delay of data packets. Moreover, variations to Schemes A, B, and C were considered for which the blocking of the voice calls takes place at a lower MAC index of the CDMA channel, so that there is always some MAC available for the data; the new schemes, A', B', and C', provide increased data throughput at the expense of a larger voice blocking probability.

For given performance indices of voice and data traffic, a suitable choice of control parameters can achieve the desired performance objectives. It was also observed that movable-boundary channel access in the code domain is achievable by all schemes introduced in this report.

APPENDIX A

- Transition probability matrix Q^{dv}

The transition probability expressions of the two-dimensional process $\underline{N}_t$ are obtained from

$$P[\underline{N}_{t+1} = (n,m) \mid \underline{N}_t = (i,j)] = \sum_{\ell=0}^{M_v-j} P[\underline{N}_{t+1} = (n,m), Y_t^v = \ell \mid \underline{N}_t = (i,j)]$$

$$(A1)$$

$$= \sum_{\ell=0}^{M_v-j} P[\underline{N}_{t+1} = (n,m) \mid Y_t^v = \ell, \underline{N}_t = (i,j)] \cdot P[Y_t^v = \ell \mid \underline{N}_t = (i,j)].$$

Conditioned on $\underline{N}_t$ and Y_t^v, N_{t+1}^d and N_{t+1}^v are independent. Hence the first term in the summation decomposes into the following two terms:

$$P[\underline{N}_{t+1} = (n,m) | \underline{N}_t = (i,j), Y_t^v = \ell] = P[N_{t+1}^d = n | \underline{N}_t = (i,j), Y_t^v = \ell]$$

$$(A2)$$

$$\cdot P[N_{t+1}^v = m | \underline{N}_t = (i,j), Y_t^v = \ell].$$

The first term on the right-hand side can be obtained from

$$P[N_{t+1}^d = n | \underline{N}_t = (i,j), Y_t^v = \ell] = \sum_{k=i-\min(n,i)}^{i} P[S_t^d = k | \underline{N}_t = (i,j), Y_t^v = \ell]$$

$$(A3)$$

$$\cdot P[Y_{t+1}^d = n - i + k | N_t^d = i]$$

where we have used the fact that the new data packet arrivals are independent of current successful transmissions and the voice traffic processes. The second term on the right hand side of (A2) is simply

$$P[N_{t+1}^v = m | \underline{N}_t = (i,j), Y_t^v = \ell] = b(j + \ell, m, 1 - p). \qquad (A4)$$

Since the voice calls are geometrically distributed in length, the probability that a call is completed in any particular slot is a Bernoulli process. Hence the probability that m out of $j + \ell$ calls are still in progress is given by the above binomial distribution.

APPENDIX B

- Probability distribution of successful data packet transmissions

Successful data packet transmissions in slot t depend on the state of the system at time t (i.e., $\underline{N}_t$) and the number of new voice calls that establish communication in slot t. On the basis of this observation, we may first evaluate the conditional distribution of S_t^d and then remove the conditioning by averaging with respect to the pertinent random variables. Consequently, we may express the probability of k successful data packet transmissions as

$$P\left[S_t^d = k\right] = \sum_{i=0}^{M_d} \sum_{j=0}^{M_v} \sum_{\ell=0}^{M_v-j} P\left[S_t^d = k | N_t^d = i, Y_t^v = \ell, N_t^v = j\right]$$
$$\cdot P\left[N_t^d = i, Y_t^v = \ell, N_t^v = j\right]. \tag{B1}$$

The second term in the summations is given by

$$P\left[N_t^d = i, Y_t^v = \ell, N_t^v = j\right] = P\left[Y_t^v = \ell | \underline{N}_t = (i,j)\right] \cdot P\left[\underline{N}_t = (i,j)\right]$$

whose individual components have been derived before. The first term in the summations is the conditional distribution of number of successful data packet transmissions which is derived as follows:

$$P\left[S_t^d = k | N_t^d = i, Y_t^v = \ell, N_t^v = j\right]$$

$$= \sum_{m=\max(0,k)}^{i} P\left[S_t^d = k, X_t^d = m | N_t^d = i, Y_t^v = \ell, N_t^v = j\right] \tag{B2}$$

$$P\left[S_t^d = k, X_t^d = m | N_t^d = i, Y_t^v = \ell, N_t^v = j\right]$$

$$= P\left[S_t^d = k | N_t^d = i, Y_t^v = \ell, N_t^v = j, X_t^d = m\right]$$

$$\cdot P\left[X_t^d = m | N_t^d = i, N_t^v = j\right].$$

We have used the fact that X_t^d is independent of Y_t^v.

Let $P(\cdot \mid k)$ represent the distribution of number of successes, given that k packets are transmitted in the slot. Then

$$P\left[S_t^d = k \mid X_t^d = m, N_t^v = j, Y_t^v = \ell\right] = P(k \mid m + \ell + j). \tag{B3}$$

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If we make the assumption of independent packet error events, then

$$P(m \mid k) = b(k, m, 1 - P_E(k)) \qquad (B4)$$

where $P_E(k)$ is the packet error probability in presence of k simultaneous packet transmissions. $P_E(k)$ can be evaluated from the methods presented in [10] for frequency-hopped spread-spectrum systems. The assumption of independent packet error events is not accurate in the entire range of values of the parameters of interest due to the correlation among the terms in the multiple access interference. An accurate approximation to $P(\cdot \mid k)$ which actually becomes tight as the number of symbols per codeword (n) increases has been derived in [11]; this approximation is computationally demanding. Here we use the independence assumption (B4) in order to avoid burdening further the already excessive computational effort required for our overall evaluation of the two-dimensional Markov chains. Hence

$$P\left[S_t^d = k | N_t^d = i, N_t^v = j, Y_t^v = \ell, X_t^d = m\right] = P\left[S_t^d = k | X_t^d = m, N_t^v = j, Y_t^v = \ell\right]$$
$$= b(m, k, 1 - P_E(m + \ell + j)).$$
$$(B5)$$

We use the fact that given X_t^d, S_t^d is independent of N_t^d. Note that

$$P\left[S_t^d = k | N_t^d = i, Y_t^v = \ell, N_t^v = j\right] = P\left[S_t^d = k | \underline{N}_t = (i, j), Y_t^v = \ell\right].$$

By putting all these together and by changing the variable on the second summation we get the following for $P\left[S_t^d = k\right]$:

$$P\left[S_t^d = k\right] = \sum_{i=o}^{M_d} \sum_{j=0}^{M_v} \sum_{\ell=0}^{M_v - j} P\left[S_t^d = k | \underline{N}_t = (i, j), Y_t^v = \ell\right] \qquad (B6)$$
$$\cdot P\left[Y_t^v = \ell | \underline{N}_t = (i, j)\right] \cdot P\left[\underline{N}_t = (i, j)\right]$$

APPENDIX C

- Probability distribution of composite (voice & data) packet transmissions

Let $X_t^T \stackrel{\triangle}{=} X_t^d + X_t^v$ denote the total number of packets (voice & data) transmitted in slot t. Then

$$P\left[X_t^T = k\right] = \sum_{n=0}^{k} P\left[X_t^d = n,\ X_t^v = k - n\right]$$

$$= \sum_{n=0}^{k} P\left[X_t^d = n | X_t^v = k - n\right] \cdot P\left[X_t^v = k - n\right]. \tag{C1}$$

The voice traffic over the channel is given by $X_t^v = Y_t^v + N_t^v$. Therefore

$$P\left[X_t^v = m\right] = \sum_{\ell=0}^{n} \sum_{i=0}^{M_d} P\left[Y_t^v = \ell | \underline{N}_t = (i, m - \ell)\right] \cdot P\left[\underline{N}_t = (i, m - \ell)\right]. \tag{C2}$$

The distribution of data packet transmissions conditioned on the voice traffic is given by

$$P\left[X_t^d = n | X_t^v = k - n\right] = \sum_{i=0}^{M_d} \sum_{j=0}^{M_v} P\left[X_t^d = n | N_t^d = i, N_t^v = j, X_t^v = k - n\right] \tag{C3}$$

$$\cdot P\left[N_t^d = i, N_t^v = j | X_t^v = k - n\right]$$

The first term in (C3) is independent of X_t^v, when N_t^d and N_t^v are given and is given by (4.12). The second term in (C3) is derived as follows:

$$P\left[N_t^d = i, N_t^v = j \mid X_t^v = k - n\right] =$$

$$= \frac{P\left[X_t^v = k - n | N_t^d = i, N_t^v = j\right] \cdot P\left[N_t^d = i, N_t^v = j\right]}{P\left[X_t^v = k - n\right]}. \tag{C4}$$

$$= \frac{P\left[Y_t^v = k - n - j | \underline{N}_t = (i, j)\right] \cdot P\left[\underline{N}_t = (i, j)\right]}{P\left[X_t^v = k - n\right]}$$

Substituting for the terms in (C1), we get

$$P\left[X_t^T = k\right] = \sum_{n=0}^{k} \sum_{i=n}^{M_d} \sum_{j=0}^{\min(k-n, M_v)} b(i, n, f_t) \cdot P\left[Y_t^v = k - n - j | \underline{N}_t = (i, j)\right] \tag{C5}$$

$$\cdot P\left[\underline{N}_t = (i, j)\right]$$

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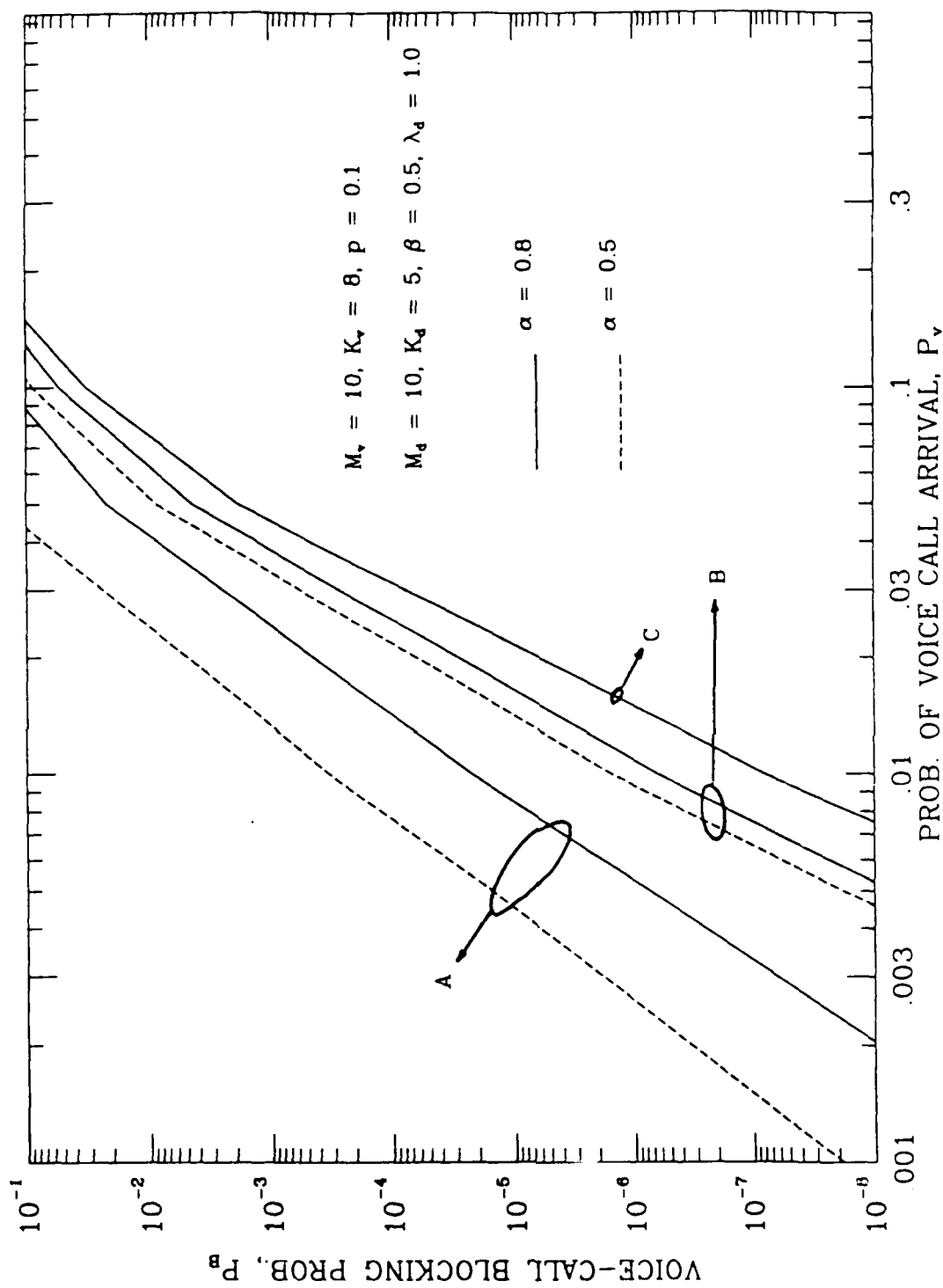
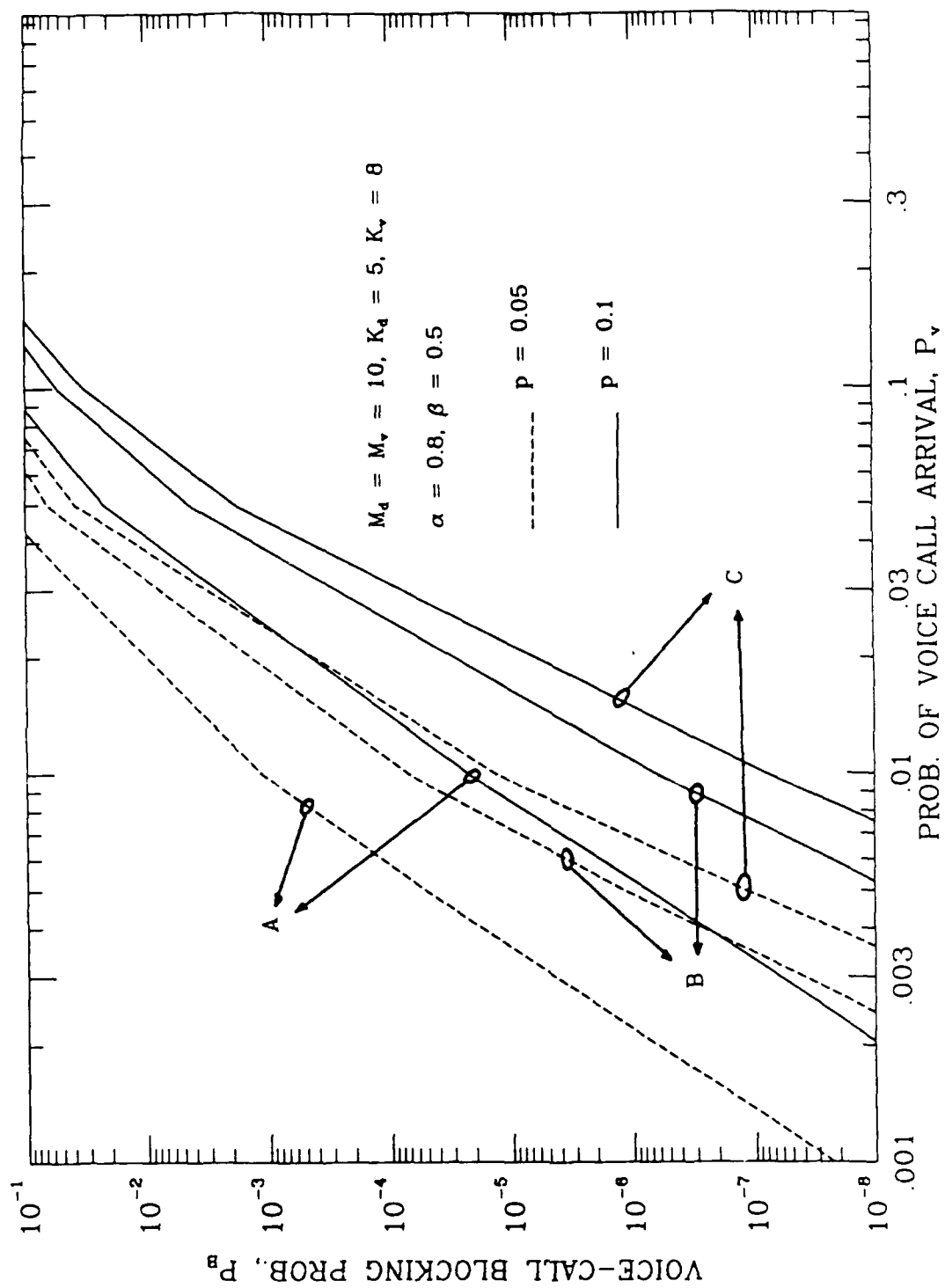Fig. 1 — Voice-call blocking probability for different control schemes ($\lambda_d = 1.0$)

$M_v = 10$, $K_v = 8$, $p = 0.1$

$M_d = 10$, $K_d = 5$, $\beta = 0.5$, $\lambda_d = 1.0$

$\alpha = 0.8$

$\alpha = 0.5$

PROB. OF VOICE CALL ARRIVAL, P_v

VOICE-CALL BLOCKING PROB., P_B

Fig. 2 — Voice-call blocking probability for different control schemes ($\lambda_d = 1.0$)

$M_d = M_v = 10$, $K_d = 5$, $K_v = 8$

$\alpha = 0.8$, $\beta = 0.5$

- - - - - $p = 0.05$

———— $p = 0.1$

PROB. OF VOICE CALL ARRIVAL, P_v

VOICE-CALL BLOCKING PROB., P_B

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Fig. 3 — Voice-call blocking probability for different control schemes (λ_d = 1.0, 2.0)

M_v = 10, K_v = 8, p = 0.1

M_d = 10, K_d = 5, α = 0.8, β = 0.5

λ_d = 1.0

λ_d = 2.0

PROB. OF VOICE CALL ARRIVAL, P_v

VOICE-CALL BLOCKING PROB., P_B

29
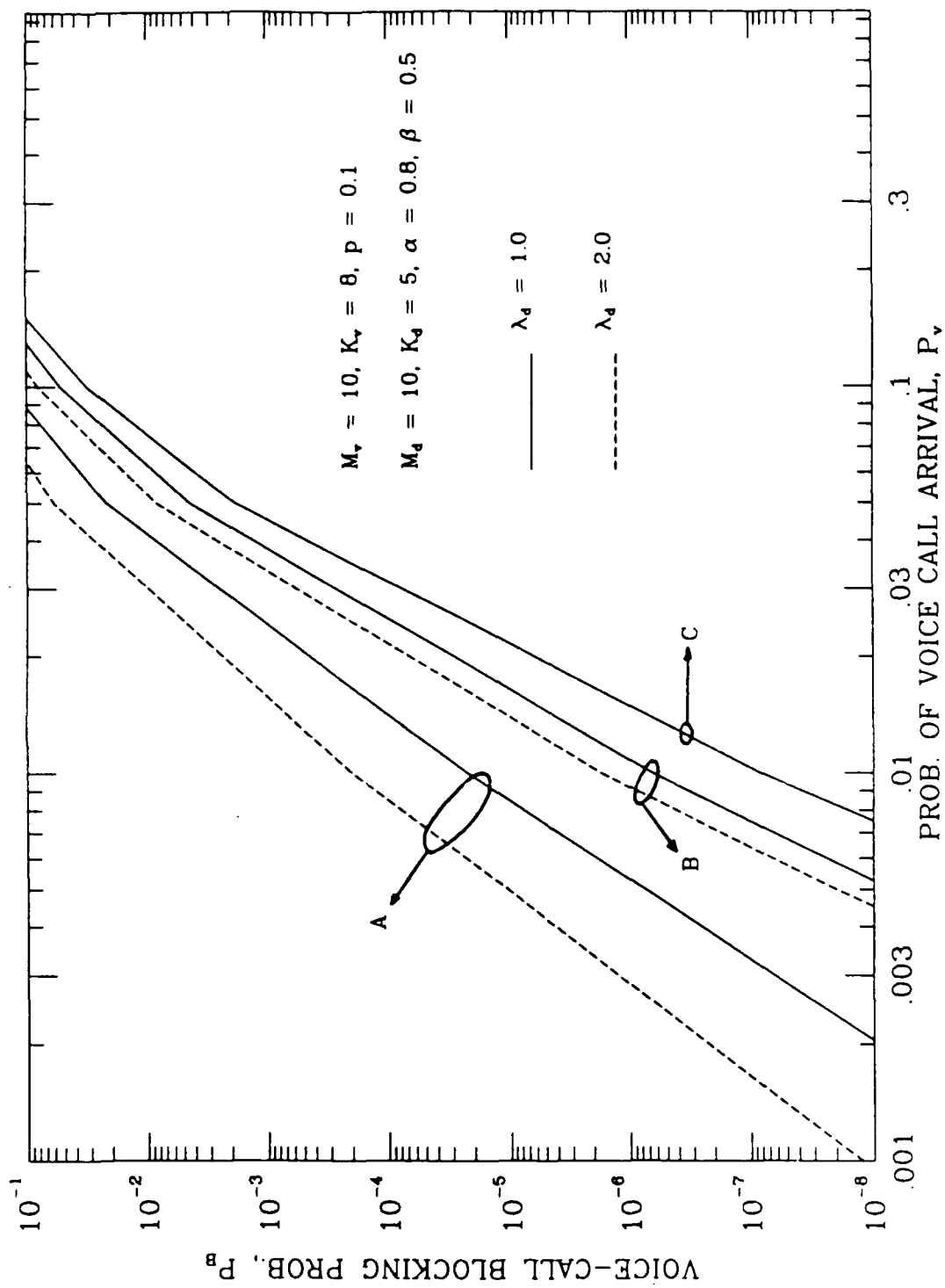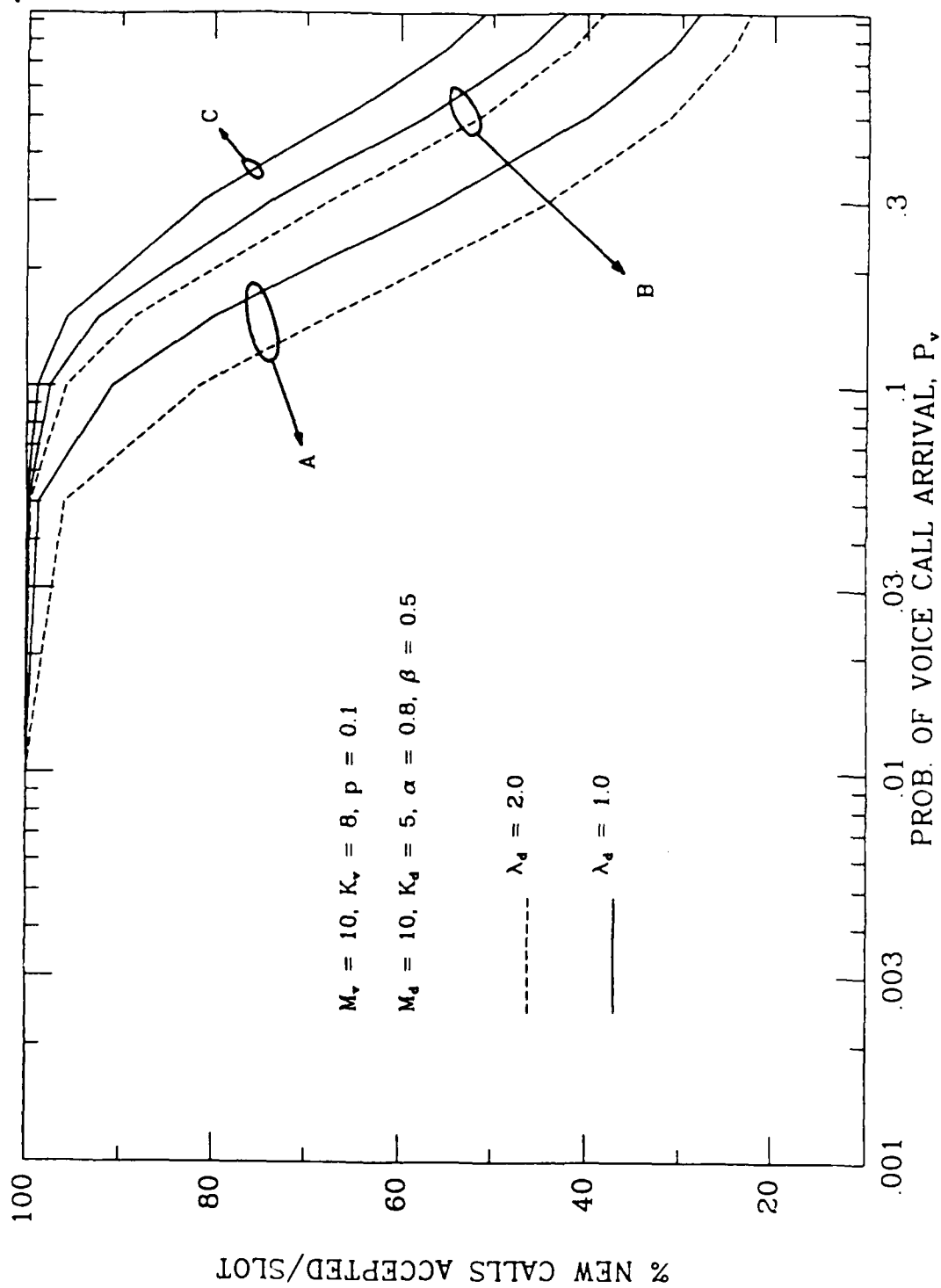
Fig. 4 — Voice throughput for different control schemes (λ_d = 1.0, 2.0)

M_v = 10, K_v = 8, p = 0.1

M_d = 10, K_d = 5, α = 0.8, β = 0.5

- - - - - λ_d = 2.0

———— λ_d = 1.0

% NEW CALLS ACCEPTED/SLOT

PROB. OF VOICE CALL ARRIVAL, P_v
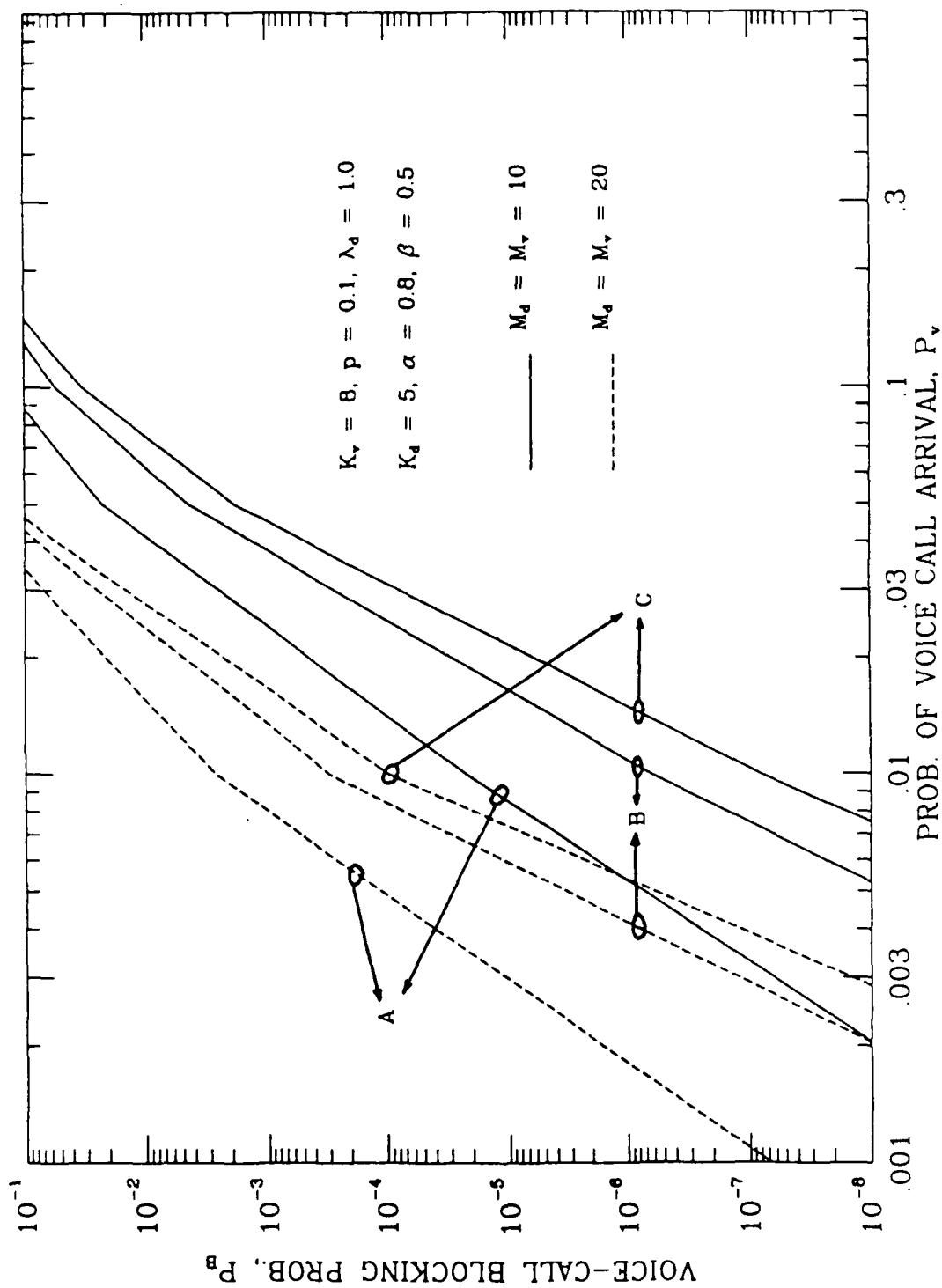
Fig. 5 — Voice-call blocking probability for different population size ($\lambda_d = 1.0$)

The plot axes and legend read:

VOICE-CALL BLOCKING PROB., P_B (vertical axis, from 10^{-1} to 10^{-8})

PROB. OF VOICE CALL ARRIVAL, P_v (horizontal axis, .001, .003, .01, .03, .1, .3)

$K_v = 8$, $p = 0.1$, $\lambda_d = 1.0$

$K_d = 5$, $\alpha = 0.8$, $\beta = 0.5$

$M_d = M_v = 10$

$M_d = M_v = 20$
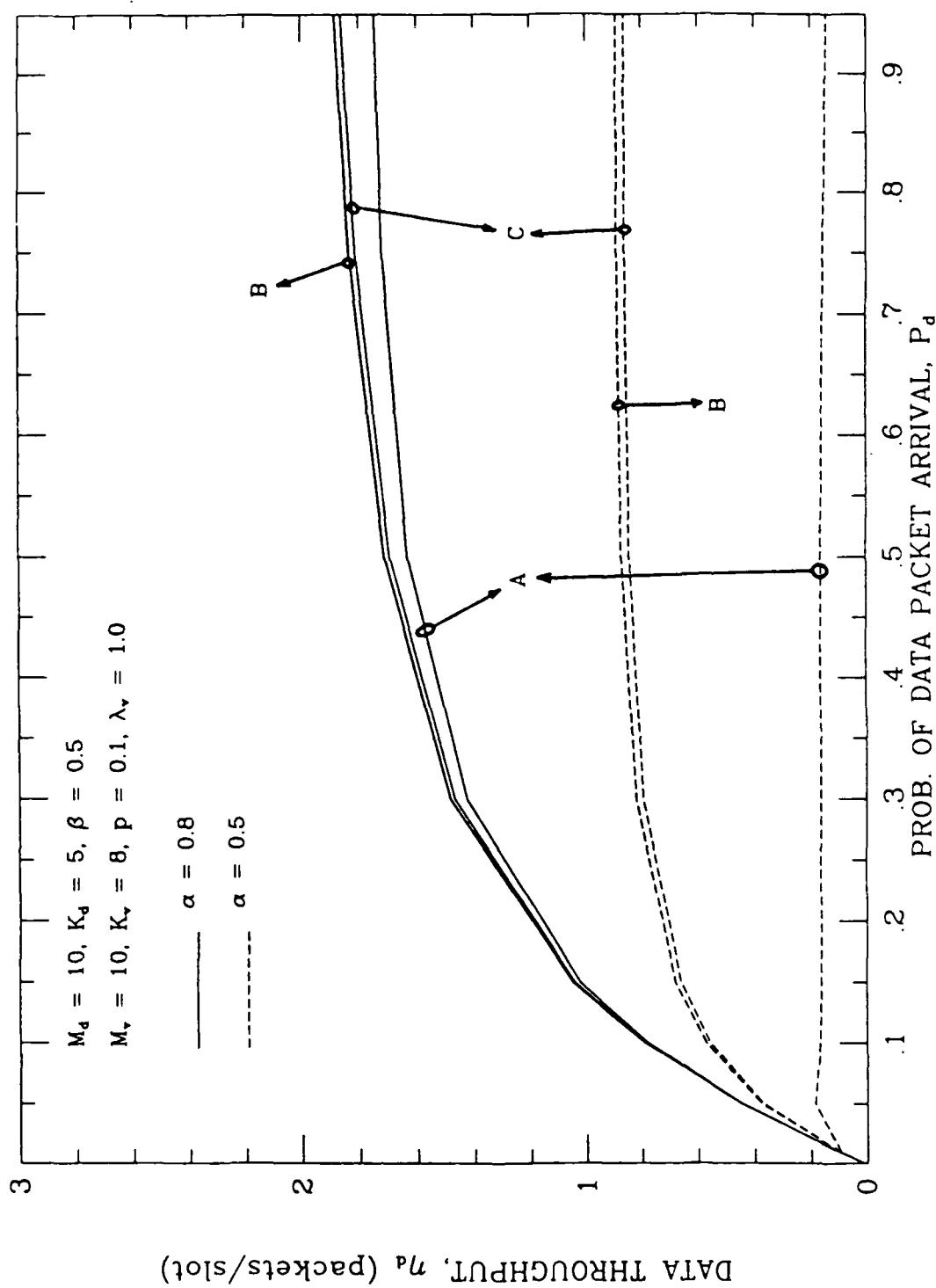
Fig. 6 — Data throughput for different control schemes ($\lambda_v = 1.0$)

$M_d = 10$, $K_d = 5$, $\beta = 0.5$

$M_v = 10$, $K_v = 8$, $p = 0.1$, $\lambda_v = 1.0$

$\alpha = 0.8$

$\alpha = 0.5$

PROB. OF DATA PACKET ARRIVAL, P_d

DATA THROUGHPUT, η_d (packets/slot)

32

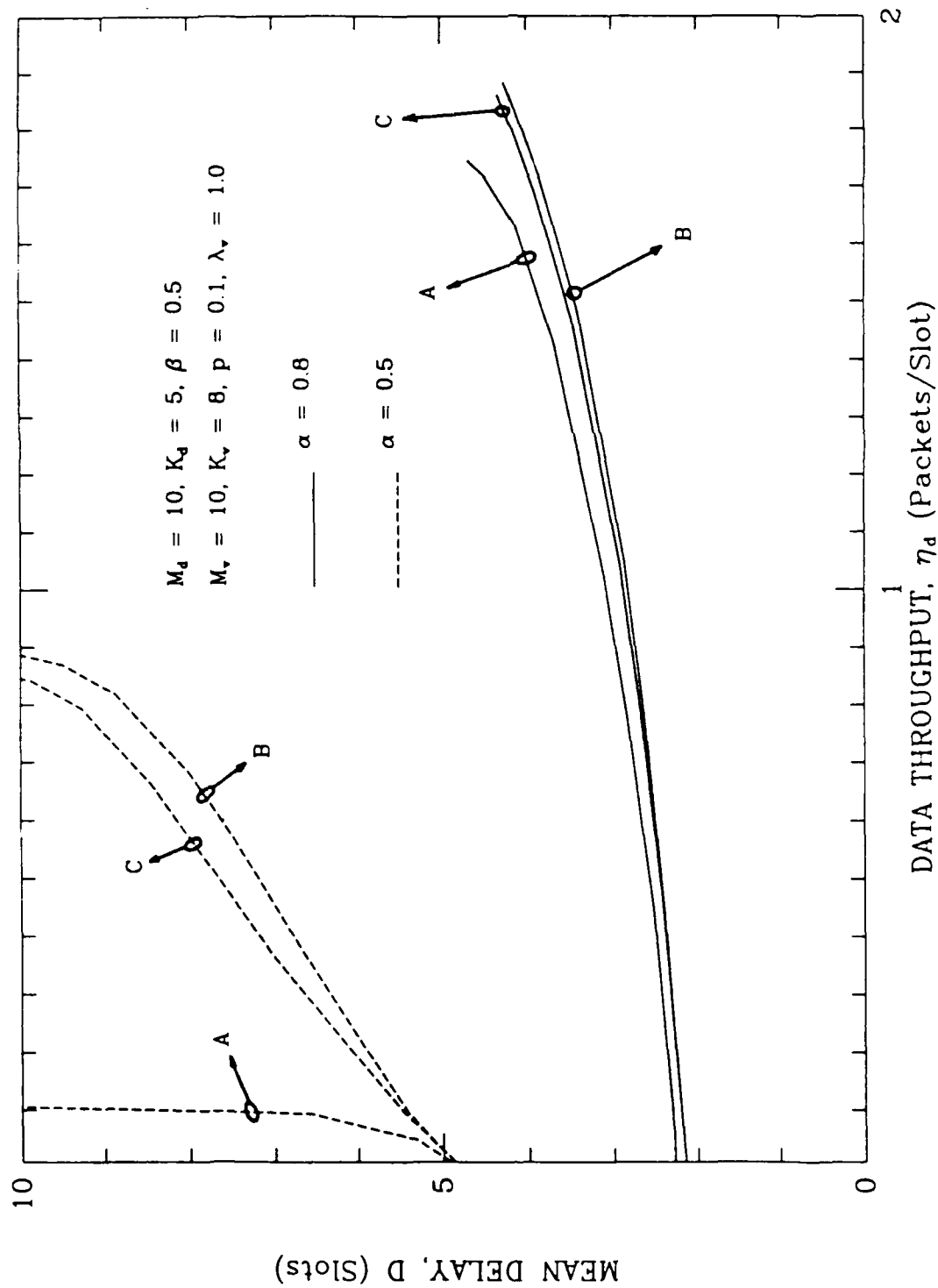DATA THROUGHPUT, η_d (Packets/Slot)

Fig. 7 — Mean data delay for different control schemes ($\lambda_v = 1.0$)

33

Fig. 8 — Packet error probability for different control schemes ($\lambda_v = 1.0$)

$M_d = 10$, $K_d = 5$, $\beta = 0.5$

$M_v = 10$, $K_v = 8$, $p = 0.1$, $\lambda_v = 1.0$

—— $\alpha = 0.8$

----- $\alpha = 0.5$

PROB. OF DATA PACKET ARRIVAL, P_d

AVERAGE PACKET ERROR PROBABILITY, P_E
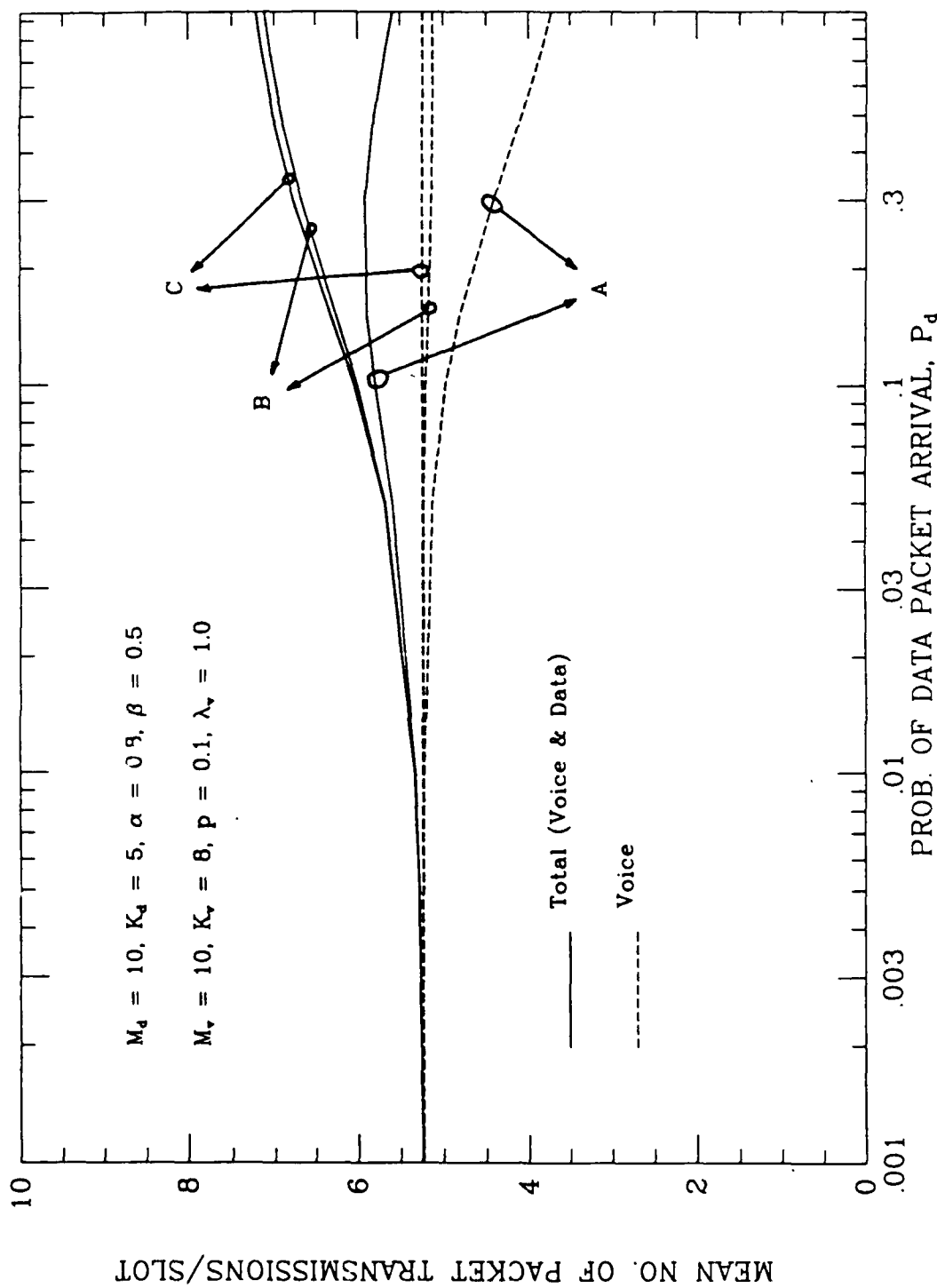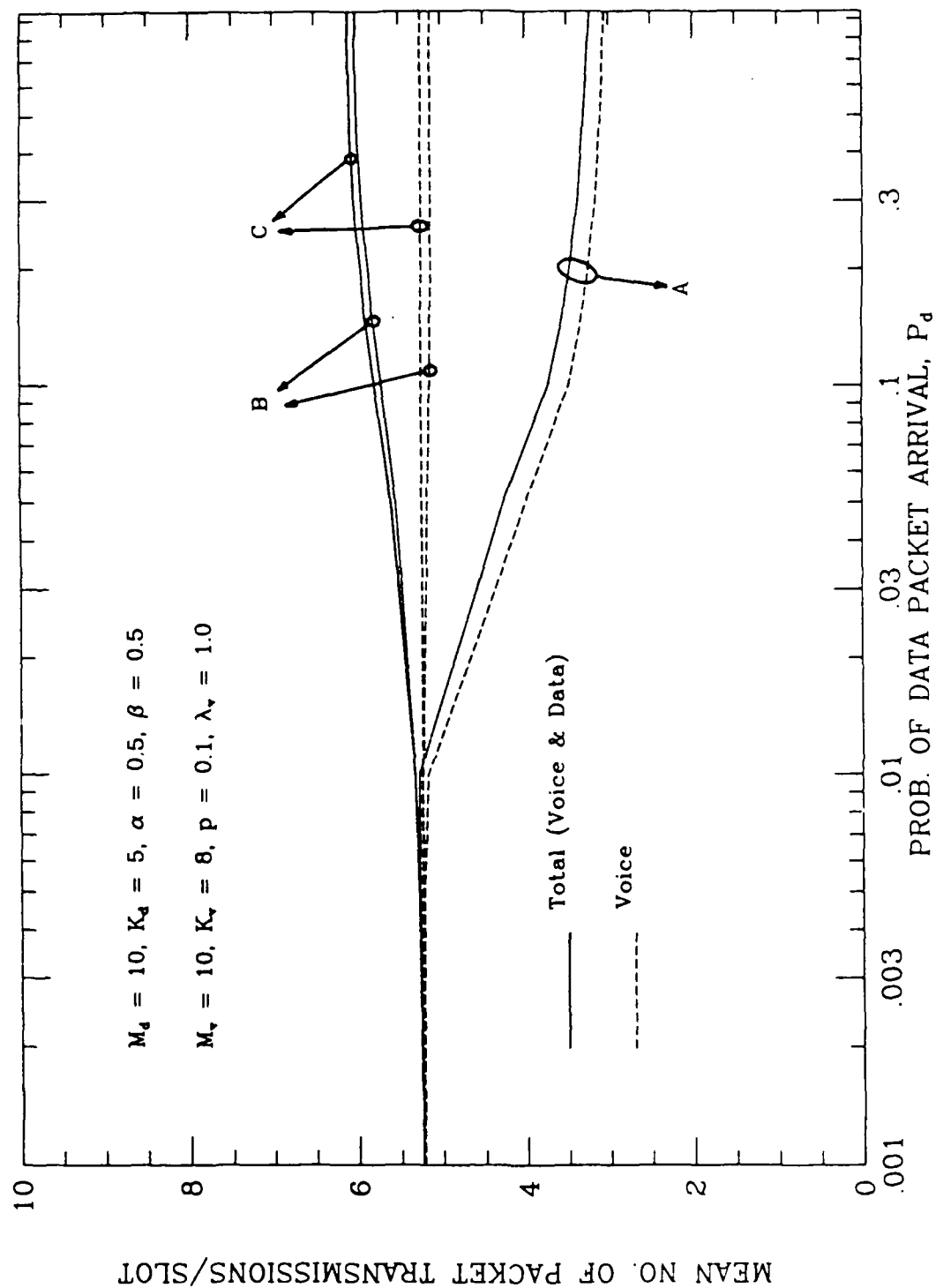
34
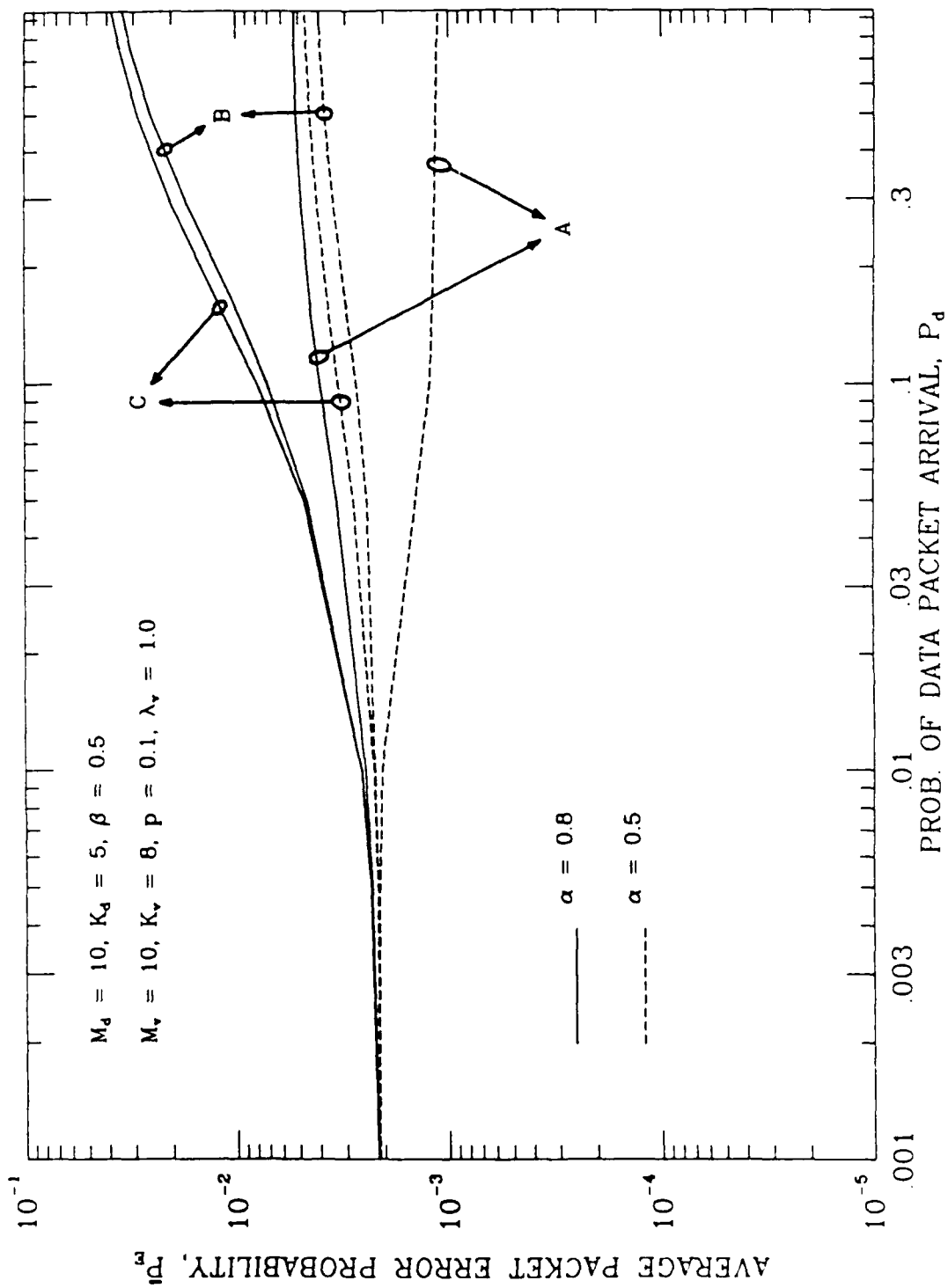
Fig. 9 — Expected number of packet transmissions for different control schemes ($\lambda_v = 1.0$, $\alpha = 0.8$)

Fig. 10 — Expected number of packet transmissions for different control schemes ($\lambda_v = 1.0$, $\alpha = 0.5$)
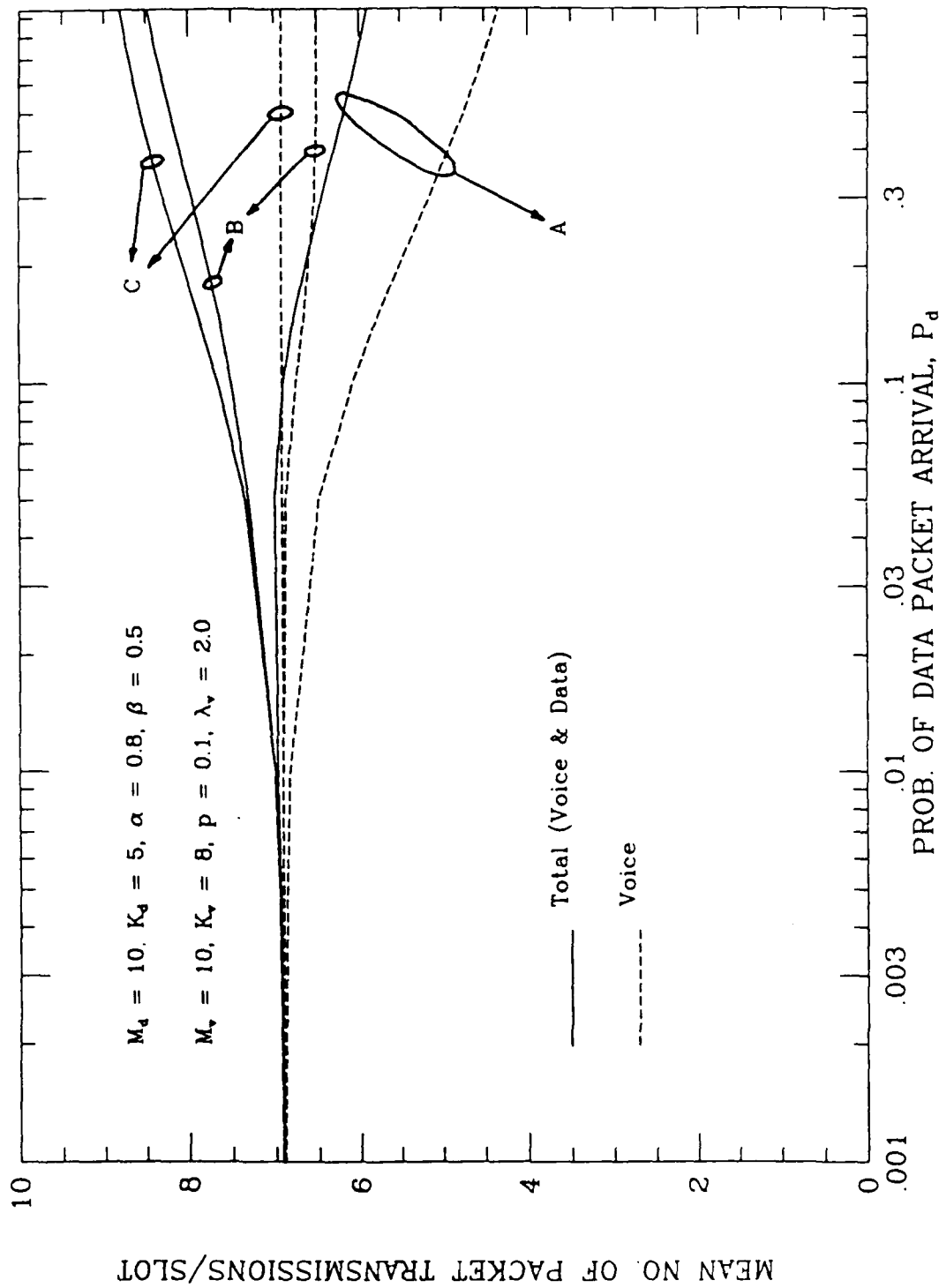
$M_d = 10$, $K_d = 5$, $\alpha = 0.5$, $\beta = 0.5$

$M_v = 10$, $K_v = 8$, $p = 0.1$, $\lambda_v = 1.0$

Total (Voice & Data)

Voice

PROB. OF DATA PACKET ARRIVAL, P_d

MEAN NO. OF PACKET TRANSMISSIONS/SLOT

Fig. 11 — Expected number of packet transmissions for different control schemes (λ_v = 2.0)

M_d = 10, K_d = 5, α = 0.8, β = 0.5

M_v = 10, K_v = 8, p = 0.1, λ_v = 2.0

Total (Voice & Data)

Voice

PROB. OF DATA PACKET ARRIVAL, P_d

MEAN NO. OF PACKET TRANSMISSIONS/SLOT

DATA THROUGHPUT, η_d (Packets/Slot)

Fig. 12 — Mean data delay for different control schemes

$M_d = 10$, $K_d = 5$, $\alpha = 0.8$, $\beta = 0.5$

$M_v = 10$, $K_v = 8$, $\lambda_v = 1.0$

p = 0.1

p = 0.05
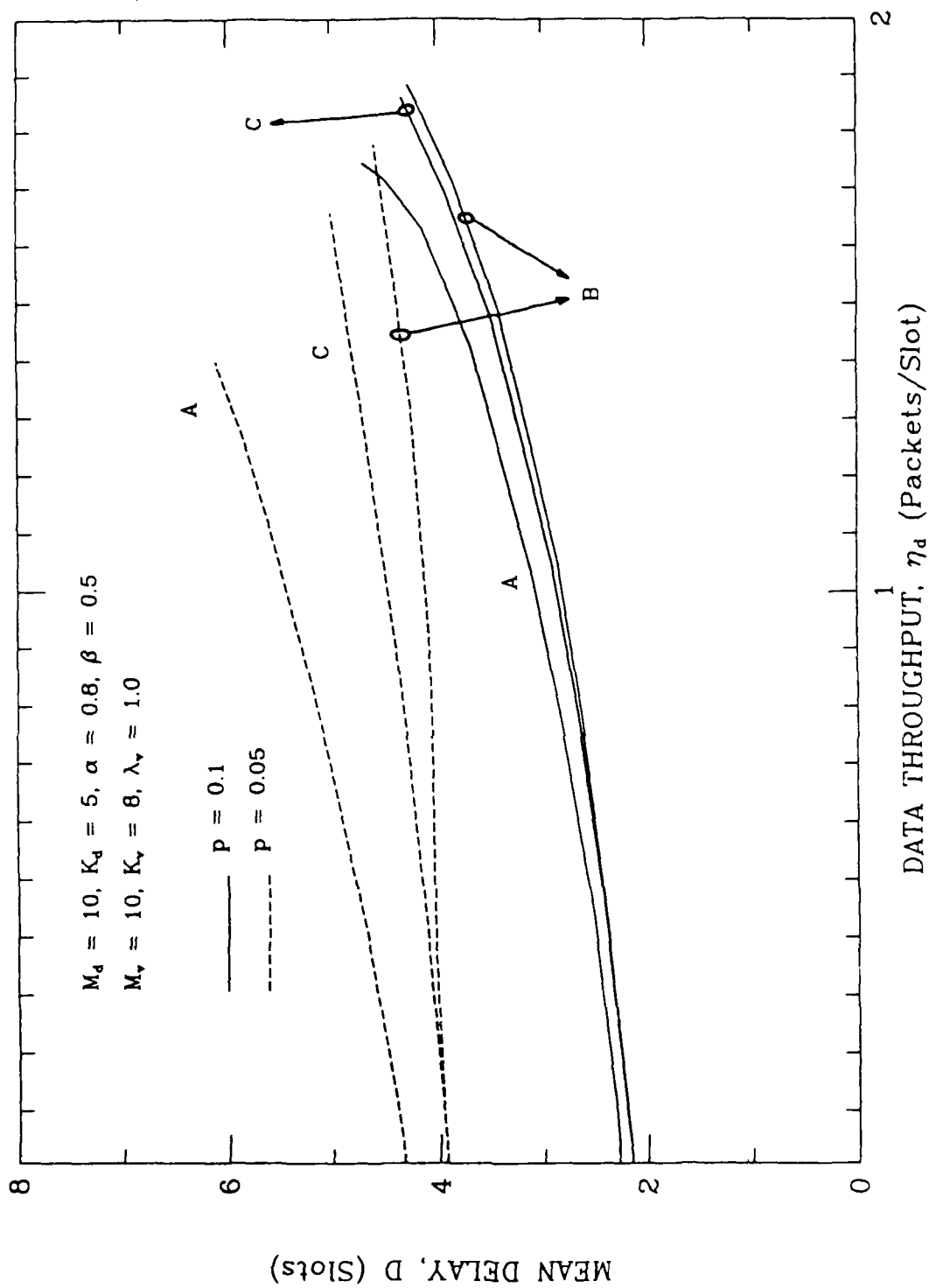
MEAN DELAY, D (Slots)

Fig. 13 — Mean data delay for different control schemes

$M_d = 10$, $K_d = 5$, $\alpha = 0.8$, $\beta = 0.5$

$M_v = 10$, $K_v = 8$, $p = 0.1$

—— $\lambda_v = 1.0$

----- $\lambda_v = 2.0$

DATA THROUGHPUT, η_d (Packets/Slot)
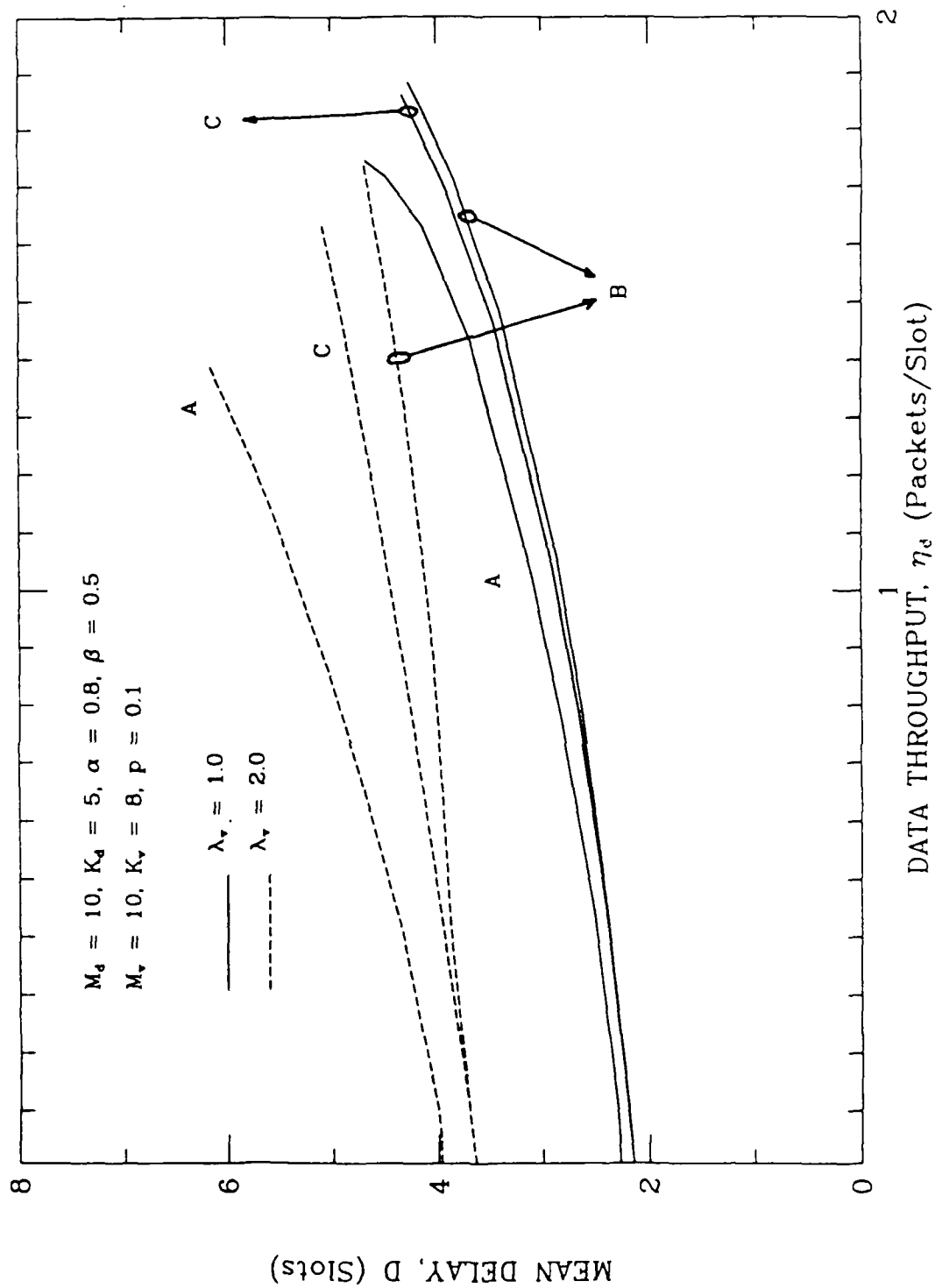
MEAN DELAY, D (Slots)
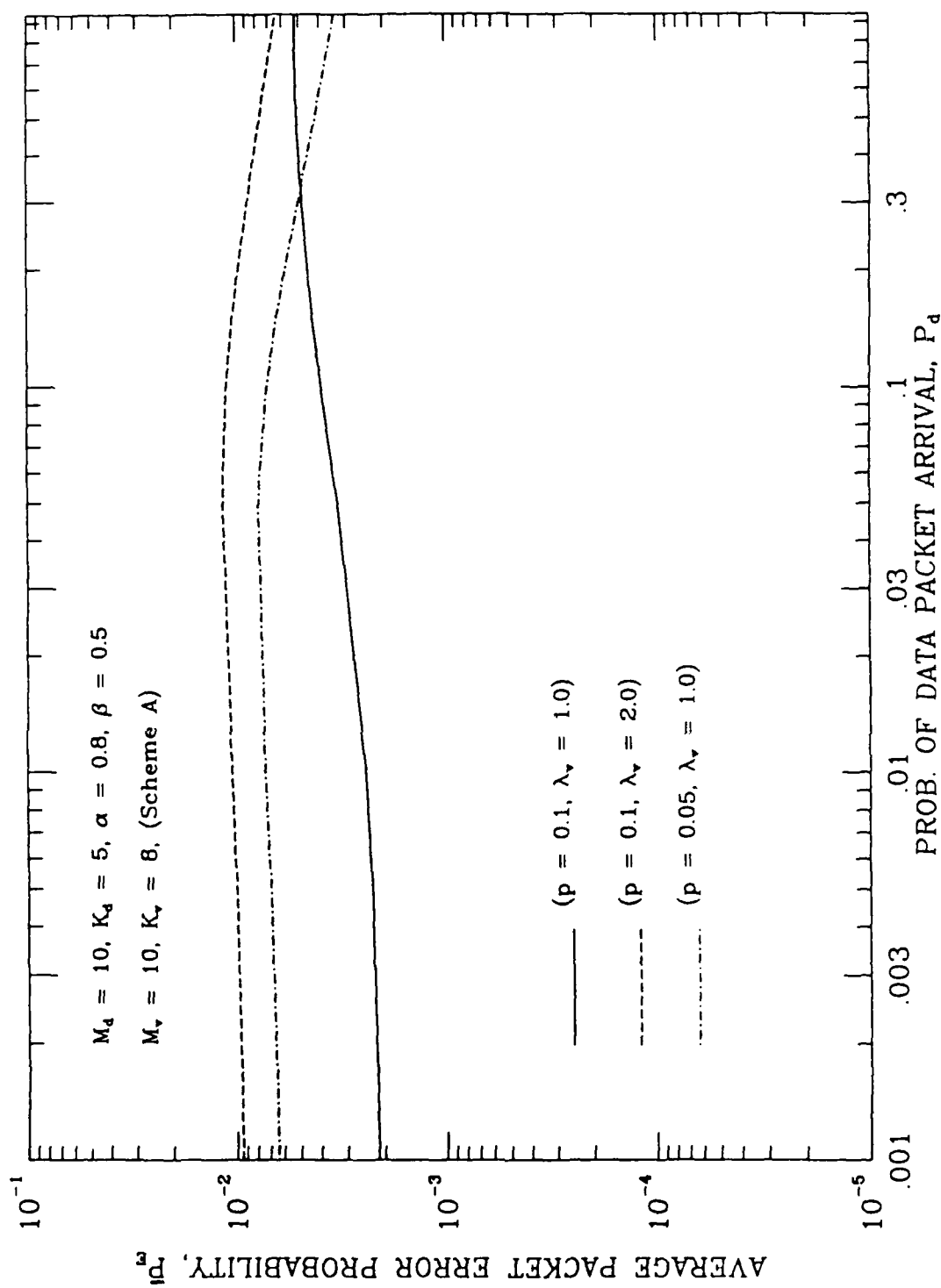
Fig. 14 — Packet error probability for Scheme A

$M_d = 10$, $K_d = 5$, $\alpha = 0.8$, $\beta = 0.5$

$M_v = 10$, $K_v = 8$, (Scheme A)

$(p = 0.1, \lambda_v = 1.0)$

$(p = 0.1, \lambda_v = 2.0)$

$(p = 0.05, \lambda_v = 1.0)$

PROB. OF DATA PACKET ARRIVAL, P_d

AVERAGE PACKET ERROR PROBABILITY, $\bar{P}_\varepsilon$
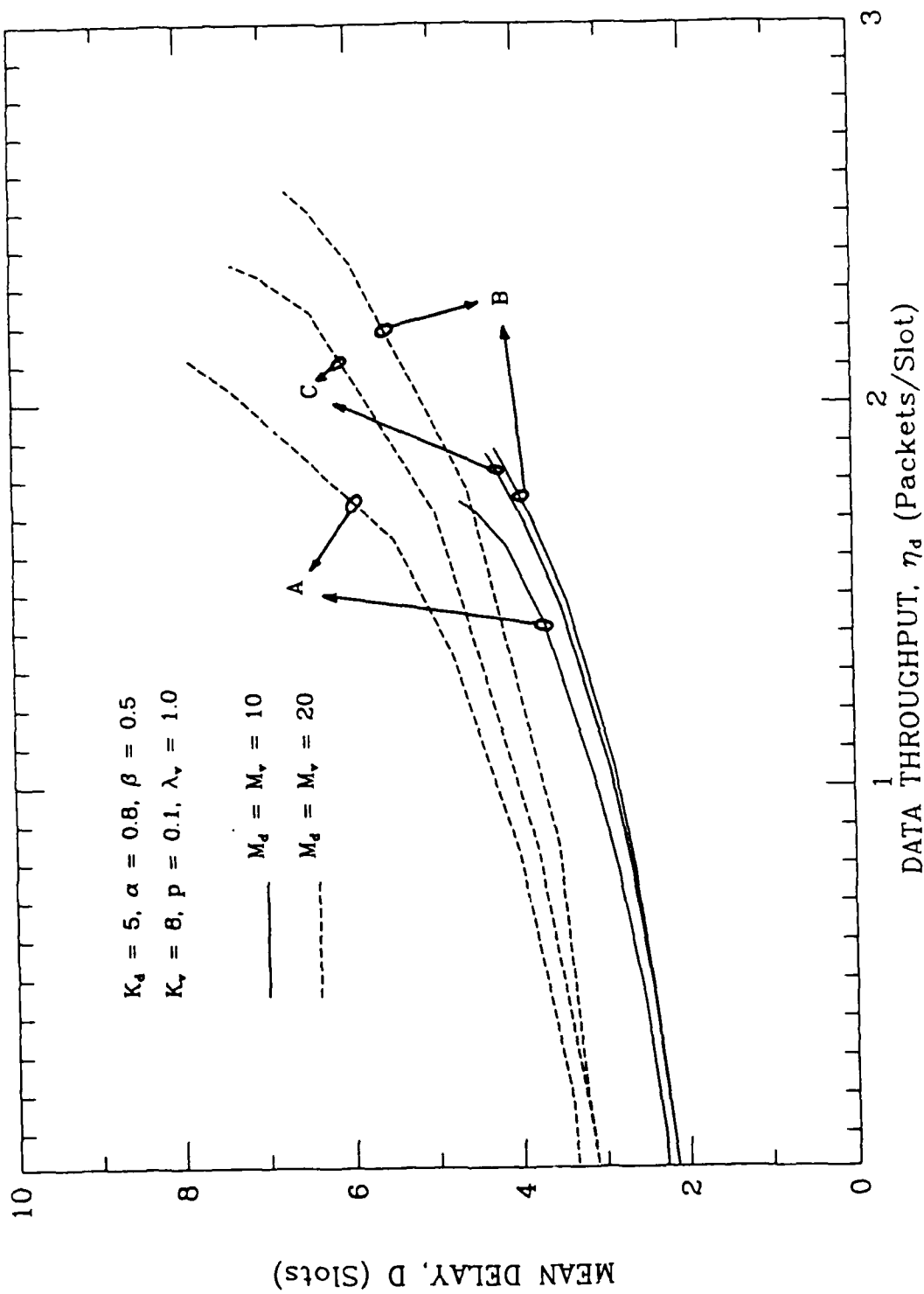
Fig. 15 — Mean data delay for different population size ($\lambda_v = 1.0$)

$K_d = 5$, $\alpha = 0.8$, $\beta = 0.5$
$K_v = 8$, $p = 0.1$, $\lambda_v = 1.0$

——— $M_d = M_v = 10$

- - - - $M_d = M_v = 20$
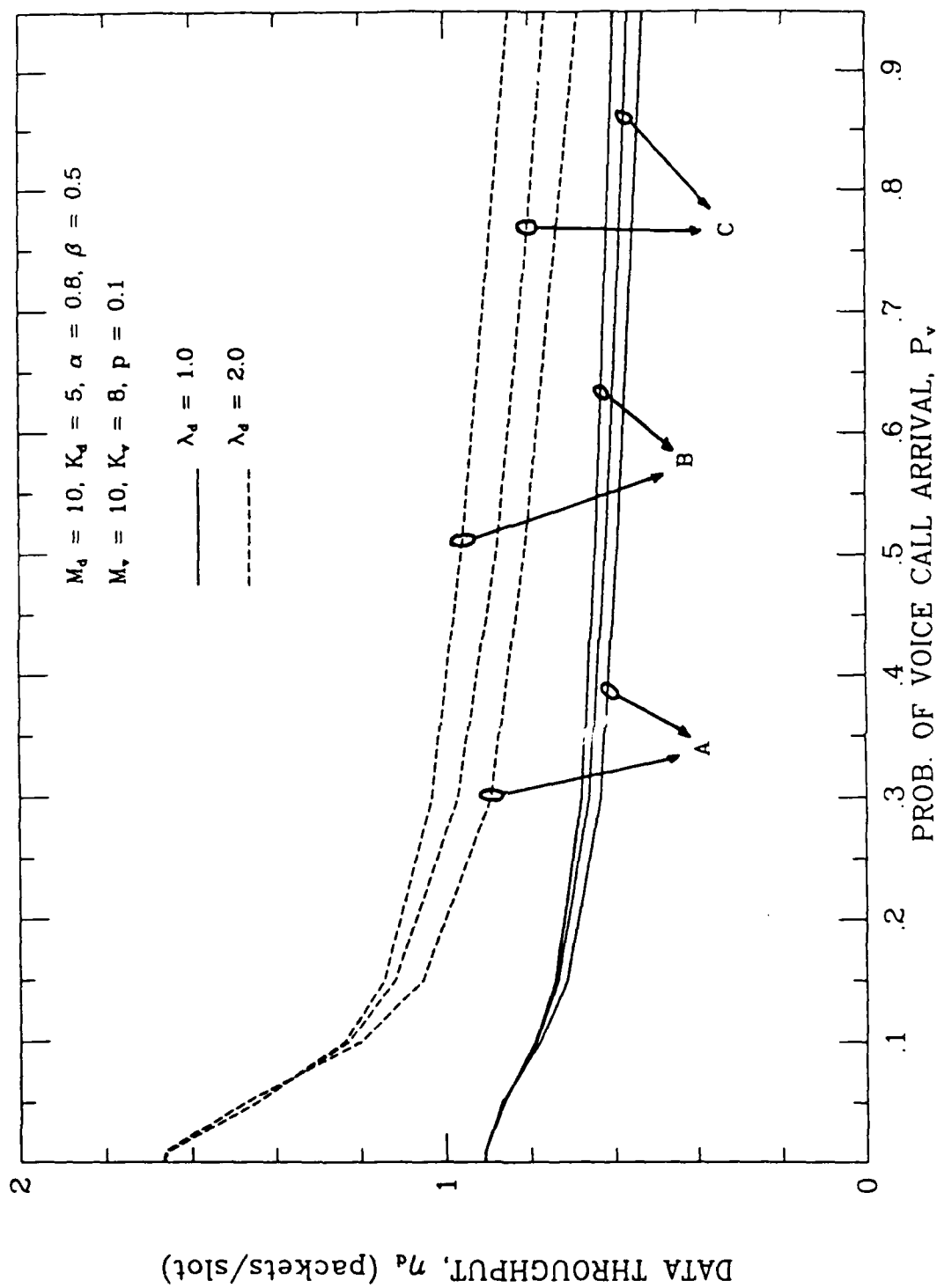
MEAN DELAY, D (Slots)

DATA THROUGHPUT, η_d (Packets/Slot)

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Fig. 16 — Data throughput vs. probability of voice-call arrival
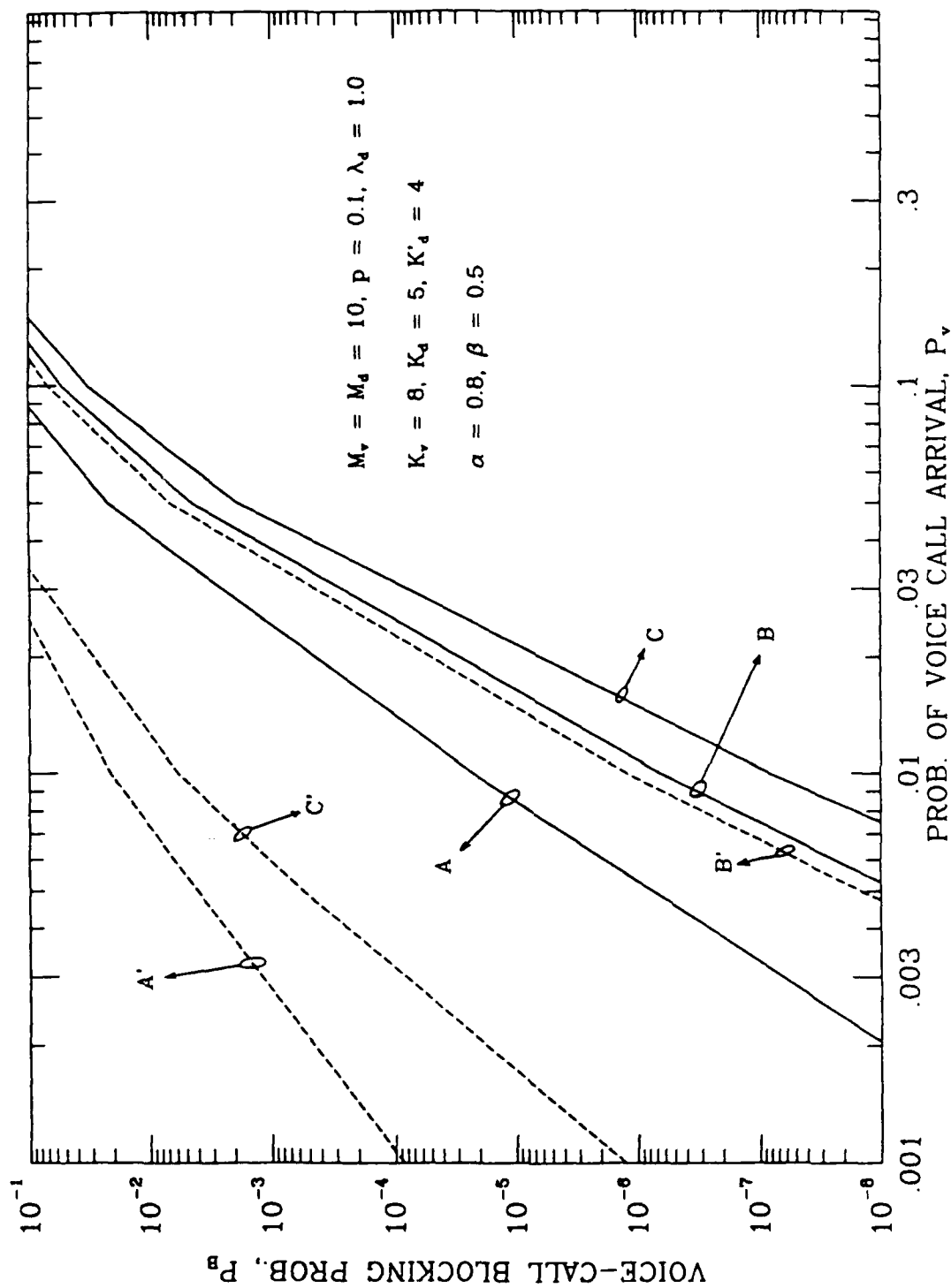
Fig. 17 — Sensitivity of blocking probability to K'_d

$M_v = M_d = 10$, $p = 0.1$, $\lambda_d = 1.0$

$K_v = 8$, $K_d = 5$, $K'_d = 4$

$\alpha = 0.8$, $\beta = 0.5$

PROB. OF VOICE CALL ARRIVAL, P_v

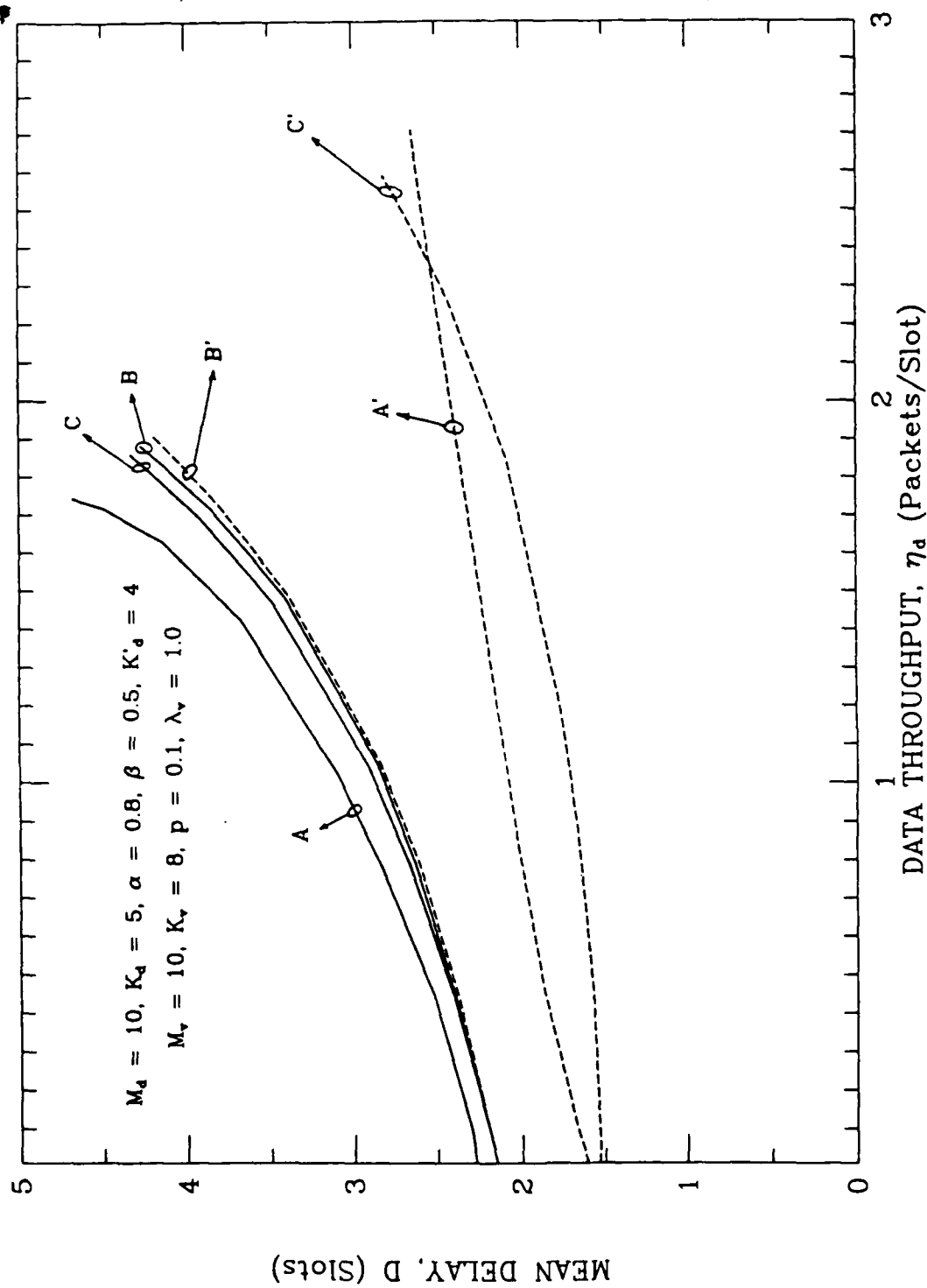VOICE-CALL BLOCKING PROB, P_B

Fig. 18 — Sensitivity of throughput/delay of data to K'_d

$M_d = 10$, $K_d = 5$, $\alpha = 0.8$, $\beta = 0.5$, $K'_d = 4$
$M_v = 10$, $K_v = 8$, $p = 0.1$, $\lambda_v = 1.0$

DATA THROUGHPUT, η_d (Packets/Slot)

MEAN DELAY, D (Slots)