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Performance Evaluation Of Multi-Access Strategies For An Integrated Voice/Data CDMA Packet Radio Network

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PERFORMANCE EVALUATION OF MULTI-ACCESS STRATEGIES FOR AN INTEGRATED VOICE/DATA CDMA PACKET RADIO NETWORK

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ABSTRACT

The problem of voice/data integration in a random-access radio network employing the ALOHA protocol in conjunction with recursive retransmission control is investigated. Code-division multiplexing (CDMA) is used as a suitable modulation in a radio environment to decrease the effect of multiple-acces interference. Multi-access control strategies are introduced which take advantage of the multiple-access capability of the CDMA channel to accomodate several voice calls simultaneously, while the data users contend for the remaining (if any) multiple-access capability of the CDMA 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, with the data backlog and number of established voice calls representing the state of the system. 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.

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I. INTRODUCTION

The evolution of present communication networks toward an integrated services digital networks (ISDN), to accomodate 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 variety of data types (e.g., interactive data, digital voice, video, etc.) over a common channel. Integration of different traffic types over a common channel requires a method to determine how the users should schedule their transmissions to avoid 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 possibley different traffic types in order to efficiently make 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 [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 [3]-[4].

In this paper, we consider the problem of voice and data integration in a packet radio environment. ALOHA protocol [2], in conjunction with retransmission control via channel load sensing is used by data nodes. At the end of each slot the (re)transmission probability of data nodes are updated based on the estimates of data backlog and number of established voice calls over in the channel. We present three different methods for the control of data traffic and compare the performance of them. Voice-call blocking is used as a way to control the load of voice traffic. A problem in the transport of packetized speech is the stringent delay requirement in order to maintain 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 based upon the level of interference sensed present over the channel.

We develop a markovian model for voice and data tarffic. Based on this model, the throughput and delay of data nodes, for a system with finite number of voice and data nodes are evaluated. Voice-call blocking probability and voice-call throughput are presented and the tradeoff among the different traffic types are discussed.

II. SYSTEM MODEL

Consider a slotted ALOHA random-access packet broadcast system with M_d data users and M_v voice users. Packet transmissions start at common clock instances and packets have constant length of L bytes. Each user (data or voice node) employs a random frequency hopping pattern for transmission of its packets. The frequency spectrum is divided into q frequency slots and each byte is transmitted at a frequency chosen from the q frequencies with equal probability, independently of the frequencies chosen for other bytes. We assume that a packet consists of exactly one codeword from a Reed-Solomon code for which up to ν byte errors can be corrected. A packet is therefore declared successfully transmitted if at most ν byte errors occur.

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 [5]. It is assumed that side information about the state of channel activities can be obtained to update the retransmission probability f_t, at the end of slot t.

Voice calls enter the system until the interference level in the channel passes a threshold, where the voice calls will be blocked and cleared. Each voice call generates a random number of packets, geometrically distributed with parameter p. Voice calls which are not blocked send their packets in successive slots until the call terminates.

The following definitions will be used:

Y_t^d = Number of new data packet arrivals in slot $t-1$,

Y_t^v = Number of new voice calls attempting transmission in slot t,

N_t^d = Number of backlogged data packets at time t,

X_t^d = Number of data packets (re)transmissions in slot t,

S_t^d = Number of successfully transmitted data packets in slot t,

N_t^v = Number of established (active) voice calls, having packets to transmit in slot t.

III. MARKOVIAN MODEL OF VOICE/DATA TRAFFIC

There are several possibilities for estimating the channel traffic at the end of each slot, and how these approaches affect the system's performance. In this paper, we present the case where the nodes are able to obtain the estimate of $N_t^{dv} \triangleq N_t^d + N_t^v$ from the observation of channel activities during the slot $(t-1, t]$. In what follows, we assume that all the nodes in the network can estimate the value of $N_t^{dv} = N_t^d + N_t^v$ exactly.

In order to obtain the statistics of N_t^{dv}, we introduce the two-dimensional process $\underline{N}_t \triangleq (N_t^d, N_t^v)$. One question to ask is whether the two-dimensional process $\underline{N}_t$ is Markov. That is whether

$$
\begin{aligned}
&P\left[\underline{N}_{t+1} = (n,m) \mid \underline{N}_0 = (k_0,\ell_0), \underline{N}_1 = (k_1,\ell_1), \ldots, \underline{N}_t = (i,j)\right] \\
&\quad = P\left[\underline{N}_{t+1} = (n,m) \mid \underline{N}_t = (i,j)\right]
\end{aligned}
$$

the above relationship holds. It can be shown 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 $s < t$. This observation leads to the conclusion that the joint statistics of the vector-valued process obeys the Markovian property.

Once the Markovian nature of $\underline{N}_t$ is established, one would desire to exploit this property to get the statistics of the process. As a first step, we would like to be able to compute the transition probability matrix of the Markov process $\underline{N}_t$. This can be done as follows:

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

The individual components on the right hand side will be given in the following sections on voice and data traffic analysis. The stationary distribution of the process is then evaluated from

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

The limiting distribution of the process is given by

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

when the Markov process is ergodic, the limiting distribution is same as the stationary distribution of the process. It can be seen that

$$P\left[N_t^{dv} = k\right] = \sum_{i=0}^{k} P\left[\underline{N}_t = (i, k-i)\right]$$

From the above, we also obtain the steady-state distribution of N_t^{dv}

IV. VOICE TRAFFIC ANALYSIS

The new voice calls are blocked whenever the value of N_t^{dv} exceeds the voice threshold K_v. Therefore, the conditional distribution of number of calls that go through is given by

$$P\left[Y_t^v = \ell | N_t^{dv} = j\right] = \sum_{i=o}^{j} P\left[Y_t^v = \ell | N_t^v = i,\ N_t^{dv} = j\right] \cdot P\left[N_t^v = i | N_t^{dv} = j\right]$$

The first term in summation can be written as

$$\begin{aligned} P\left[Y_t^v = \ell | N_t^v = i, N_t^{dv} = j\right] &= P\left[Y_t^v = \ell | N_t^d = j - i, N_t^v = i\right] \\ &= P\left[Y_t^v = \ell | \underline{N}_t = (j - i, i)\right] \end{aligned}$$

which is given by

$$P\left[Y_t^v = \ell \mid \underline{N}_t = (j - i, i)\right] = \ b(M_v - i, \ell, P_v), \quad j \leq K_v;$$

The second term in the summation is given by

$$P\left[N_t^v = i | N_t^{dv} = j\right] = \frac{P\left[\underline{N}_t = (j - i, i)\right]}{P\left[N_t^{dv} = j\right]}$$

where

$$P\left[N_t^{dv} = j\right] = \sum_{k=0}^{j} P\left[\underline{N}_t = (k, j - k)\right]$$

Note that

$$P\left[Y_t^v = \ell \mid N_t^{dv} = j\right] = \begin{cases} 1 & \ell = 0 \\ & \qquad ; j > K_v. \\ 0 & \ell > 0 \end{cases}$$

The voice-call blocking probability is given by

$$P_B \triangleq \lim_{t \to \infty} P\left[N_t^{dv} > K_v\right] = \sum_{j > K_v} \sum_{k=0}^{j} \Pi^{\underline{N}}(k, j - k)$$

IV-A. Calculation of Voice Traffic Throughput

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

$$\eta_v = \lim_{t \to \infty} E\left\{E\left(Y_t^v | N_t^{dv}\right)\right\}.$$

The above formulas provide us with the conditional distribution of number of voice calls that go through in each slot. The conditional expectation of this random variable is then given by

$$E\left(Y_t^v | N_t^{dv} = j\right) = \sum_{\ell=0}^{M_v - j} \ell \, P\left[Y_t^v = \ell | N_t^{dv} = j\right]; \quad j \le K_v$$

and $E\left(Y_t^v | N_t^{dv} = j\right) = 0$ for $j > K_v$. Substituting in the above, we get

$$E\left(Y_t^v | N_t^{dv} = j\right) = \sum_{i=o}^{j} P_v\left(M_v - i\right) \cdot P\left[N_t^v = i | N_t^{dv} = j\right]; \quad j \le K_v$$

Averaging with respect to N_t^{dv} gives us

$$\eta_v = \sum_{j=0}^{K_v} \sum_{i=0}^{j} P_v\left(M_v - i\right) \Pi^{\underline{N}}(j-i, i)$$

Simplifying the above formula, we obtain

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

The percentage of calls that go through is then given by

$$\frac{\eta_v}{M_v P_v} \cdot 100\%$$

V. DATA TRAFFIC ANALYSIS

Since the data nodes follow the DFT protocol for their first transmissions, the evolution of the data backlog process $\{N_t^d\}$ is given by

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

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

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

If the transmission of new packet is unsuccessful, no new packet will be generated by the user until the backlogged packet is transmitted successfully. The backlogged packets are transmitted independently in slot t with probability f_t. Therefore,

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

The (re)transmission probabilities are updated according to

$$f_t = \begin{cases} \min\left(1, \frac{K_d}{\max(1, N_t^{dv})}\right); & N_t^{dv} \le K_v \\ \Phi(N_t^{dv}, K_v, K_d); & N_t^{dv} > K_v \end{cases}$$

where $\Phi(\cdot,\cdot,\cdot)$ is a suitable control function of channel information N_t^{dv}, and control parameters K_v and K_d. The choice of Φ determines the course of action taken by backlog data nodes when the state of channel activities implies the blocking of new voice calls. Hence, the choice of Φ has direct effect on both the throughput/delay performance of data nodes and blocking performance of voice traffic.

In order to evaluate the data throughput, we need the marginal distribution of number of successful data packet transmissions. This can be obtained via

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

The second term in the summations is given by

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

where the 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:

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

$$\begin{aligned} P\left[S_t^d = k, X_t^d = m | N_t^d = i, Y_t^v = \ell, N_t^{dv} = j\ \right] & \\ &= P\left[S_t^d = k | N_t^d = i, Y_t^v = \ell, N_t^{dv} = j, X_t^d = m\right] \\ &\cdot P\left[X_t^d = m | N_t^d = i, N_t^{dv} = j\right] \end{aligned}$$

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

$$P\left[X_t^d = m | N_t^d = i, N_t^{dv} = j\right] = b(i, m, f_t)$$

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

$$P\left[S_t^d = k \mid X_t^d = m, N_t^v = j - i, Y_t^v = \ell\right] = P(k \mid m + \ell + j - i)$$

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

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

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 [6] for frequency-hoopped spread-spectrum systems. The assumption of independent packet error events is not accurate due to the correlation among the terms in the multiple access interference. Accurate approximations to $P(\cdot \mid k)$ have been derived in [7]. Hence,

$$\begin{aligned} P\left[S_t^d = k | N_t^d = i, N_t^{dv} = j, Y_t^v = \ell, X_t^d = m\right] &= P\left[S_t^d = k | X_t^d = m, N_t^v = j - i, Y_t^v = \ell\right] \\ &= b(m, k, 1 - P_E(m + \ell + j - i)) \end{aligned}$$

Note that

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

Putting all these together and by a change of variable on the second summation, we get the following for $P\left[S_t^d = k\right]$:

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

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

The data throughput is then given by

$$\eta_d = \lim_{t \to \infty} \sum_{k=0}^{M_d} k \; P\left[S_t^d = k\right]$$

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

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

V-A. The control function $\Phi(N_t^{dv}, K_v, K_d)$

The control function Φ determines the (re)transmission probabilities of data nodes when the feedback from channel implies the blocking of new voice calls. One way to proceed is to have very small retransmission probabilities when $N_t^{dv} > K_v$. We may some form of the exponential backoff method. A suitable control function in this case is

$$\Phi(N_t^{dv}, K_v, K_d) = \left(\frac{1}{2}\right)^{N_t^{dv} - K_d} \quad ; \quad N_t^{dv} > K_v$$

In this way, the data nodes follow the exponential backoff while the control gives some room to data nodes via K_d.

On the other hand, we may let the data nodes follow the same rule as when $N_t^{dv} \leq K_v$. That is, we let

$$\Phi(N_t^{dv}, K_v, K_d) = \frac{K_d}{N_t^{dv}} \; ; \qquad N_t^{dv} > K_v$$

Using this rule results in dramatic decrease in the data backlog queue compared to the first rule. This also helps to improve the throughput of voice traffic. This happens since the blocking mechanism of new voice calls depends directly on the size of the data backlog queue as well as the number of established voice calls already in the system. Reducing the size of the data backlog queue results in smaller voice-call blocking probability which in turn improves the voice throughput.

For the purpose of comparison, we have also used a modification of the above rule by ignoring the multiple access capability parameter K_d. This one is given by

$$\Phi(N_t^{dv}, K_v, K_d) = \frac{1}{N_t^{dv}} \; ; \qquad N_t^{dv} > K_v$$

V-B. Transition probability matrix of $\underline{N}_t$

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

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

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

where

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

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

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

where we have used fact that the new data packet arrivals are independent of current successful transmissions and the voice traffic processes. That is

$$P\left[Y_{t+1}^d = n - i + k | N_t^d = i\right] = b\left(M_d - i, n - i + k, P_d\right)$$

The second term on the right hand side is simply

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

VI. DISTRIBUTION OF TOTAL TRAFFIC (VOICE&DATA) AND RELATED PERFORMANCE MEASURES

In this section, we will first derive the probability distribution of the total traffic in a slot, in terms of number of packet transmissions in a slot, and then introduce some performance measures related to this process.

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

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

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

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

$$P\left[X_t^v = m\right] = \sum_{\ell=0}^{m} P\left[Y_t^v = \ell | N_t^v = m-\ell\right] \cdot P\left[N_t^v = m-\ell\right]$$

The first term in the above summation is given by

$$P\left[Y_t^v = \ell | N_t^v = m-\ell\right] = \sum_{j=m-\ell}^{M_d+m-\ell} P\left[Y_t^v = \ell | N_t^v = m-\ell, N_t^{dv} = j\right]$$

$$\cdot P\left[N_t^{dv} = j | N_t^v = m-\ell\right]$$

$$P\left[N_t^{dv} = j | N_t^v = m-\ell\right] = \frac{P\left[\underline{N}_t = (j-m+\ell, m-\ell)\right]}{P\left[N_t^v = m-\ell\right]} \quad ;$$

and

$$P\left[Y_t^v = \ell | N_t^v = m-\ell, N_t^{dv} = j\right] = P\left[Y_t^v = \ell | \underline{N}_t = (j-m+\ell, m-\ell)\right]$$

which is given in the previous section.

The resulting distribution of voice traffic is then given by

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

The conditional distribution of data traffic, given the voice traffic is given by

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

The first term in the summation is independent of X_t^v when N_t^d and N_t^{dv} are given. Therefore.

$$P\left[X_t^d = n | N_t^d = i, N_t^{dv} = j, X_t^v = k-n\right] = P\left[X_t^d = n | N_t^d = i, N_t^{dv} = j\right] = b(i, n, f_t)$$

The second term in the summation is derived as follows:

$$\begin{aligned} P\left[N_t^d = i, N_t^{dv} = j \;| X_t^v = k-n\right] &= \\ &= \frac{P\left[X_t^v = k-n | N_t^d = i, N_t^{dv} = j\right] \cdot P\left[N_t^d = i, N_t^{dv} = j\right]}{P\left[X_t^v = k-n\right]} \\ &= \frac{P\left[Y_t^v = k-n-j+i | \underline{N}_t = (i, j-i)\right] \cdot P\left[\underline{N}_t = (i, j-i)\right]}{P\left[X_t^v = k-n\right]} \end{aligned}$$

Substituting for the terms in the summation, we get

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

The expected number of packet transmissions in a slot is then given by

$$\bar{X}_T = \lim_{t \to \infty} E\left\{X_t^T\right\} = \lim_{t \to \infty} \sum_{m=0}^{M_d+M_v} m \; P\left[X_t^T = m\right]$$

The expected number of voice packet transmissions is also given by

$$\bar{X}_v = \eta_v + \sum_{i=0}^{M_d} \sum_{j=0}^{M_v} j \; \Pi^{\underline{N}}(i,j)$$

We can then compare the change in $\bar{X}_T$ and $\bar{X}_v$ as a function of voice and data arrival rates.

Another performance measure of interest is the average probability of error in a slot given by

$$\bar{P}_E = \sum_{m=0}^{M_d+M_v} P_E(m) \; P\left[X_t^T = m\right]$$

where $P_E(m)$ is the conditional probability of packet error in the presence of m simultanous packet transmissions.

These are evaluated for the three schemes described in V-A.

VII. NUMERICAL RESULTS

For the integrated random access system described above, we considered a system of $M_v = 10$ voice and $M_d = 10$ data nodes. An extended Reed-Solomon code of rate 1/2 ($n = 64, k = 32$) was used for error correction purpose. This code can correct up to $\nu = 16$ byte errors. The frequency spectrum was divided to $q = 50$ frequency slots for frequency-hopping multiplexing. The maximum tolerable voice packet error probability was set at 10^{-2}. This gave us the voice threshold of $K_v = 5$. This means that for the given values of physical link parameters (i.e., code rate, packet length, frequency slots, etc.), the network can support up to 5 active voice calls with moderately good packet error probabilities. The maximum tolerable data packet error probability was set at 10^{-3}. This resulted in $K_d = 4$.

Figure 1. presents the voice-call blocking probability (Loss probability) for the three different schemes given above. Figure 2. illustrates the voice traffic throughput for the same schemes. Figures 3.-4. present the data traffic throughput and delay performance of the given schemes, respectively. It is observed that the exponential backoff scheme has the worst performance for both traffic types. This is due to the fact that the retransmission probabilities depend on the estimate of data backlog through N_t^{dv}, and the exponential backoff of data nodes causes the data backlog to increase and a positive feedback effect occurs. On the other hand, the first scheme seems to fully utilized the resources of the CDMA channel to accomodate both traffic types.

Figure 5. presents the expected number of packet transmissions in a slot for the total and voice traffic, respectively. It is clear that as the offered data traffic increases the voice traffic gives more priority to the data nodes. Figure 6. illustrates the average packet error probability as a function of offered data traffic for the three schemes considered here. It is observed that for the average data traffic of less than one packet per slot the best scheme in terms of throughput and delay also does best in terms of average packet error probability. For $M_dP_d > 1$, the exponential backoff scheme does better than the other two schemes, but that is expected; since the traffic load over the channel decreases dramatically which causes a sharp decrease in average packet error probability for this scheme.

VIII. CONCLUSIONS

The problem of integration of voice and data traffic over a radio channel was considered in this paper. Code division multiplexing techniques were employed, as an alternative to traditional FDMA and TDMA schemes; to offer resouce sharing by both traffic types. The three control strategies considered here do well in terms of average packet error probability experienced by the network users (Fig. 6). The throughput/delay performance of data nodes are more sensitive to the type of control being exercized by the nodes than the voice performance measures (blocking probability, throughput) are. It is observed that multiple access capability of CDMA techniques can be exploited to enhance the efficient use of the channel, and increase the throughput of both traffic types while the degradation of service (in terms of packet error probability) is graceful (see Fig. 1-6).

One drawback of the above model is the fact the retransmission probabilities can not be implemented in a distributed and decentralized fashion. Our work in progress is to remedy this problem. Recursive methods of control introduced in [5] are being investigated for implementation with this protocol.

Our analysis was carried out for a finite-population model (actually, binomial distributions for the populations of voice and data users). The extension to the case of an infinite population of data users (the voice population remains finite) involves deriving the stability region of this protocol for data traffic as a function of variations in the population and duty factor of voice traffic.

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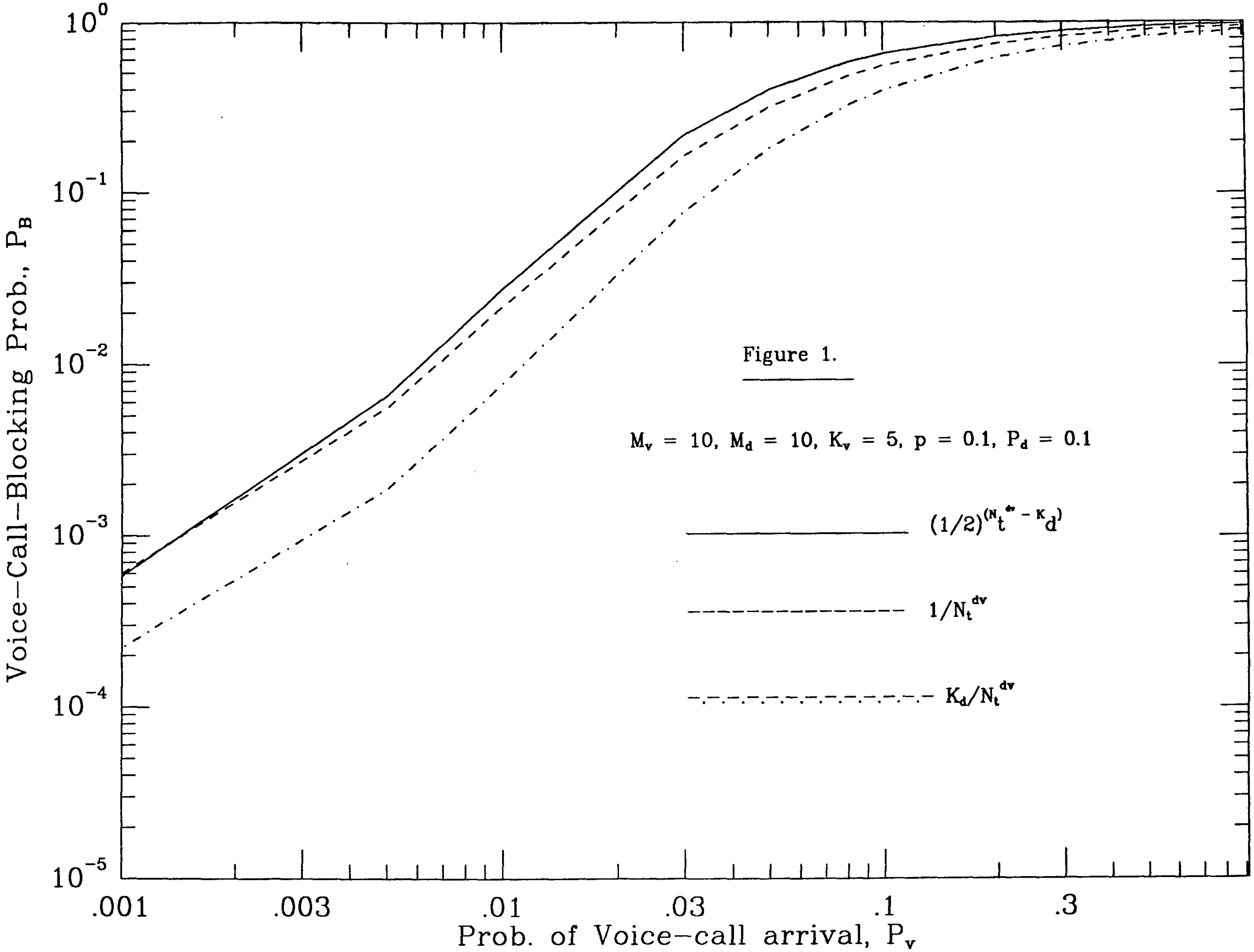

Figure 1.
$M_v = 10,\ M_d = 10,\ K_v = 5,\ p = 0.1,\ P_d = 0.1$
$(1/2)^{(N_t^{dv} - K_d)}$
$1/N_t^{dv}$
K_d/N_t^{dv}
Voice-Call-Blocking Prob., P_B
Prob. of Voice-call arrival, P_v
10^0
10^{-1}
10^{-2}
10^{-3}
10^{-4}
10^{-5}
.001
.003
.01
.03
.1
.3

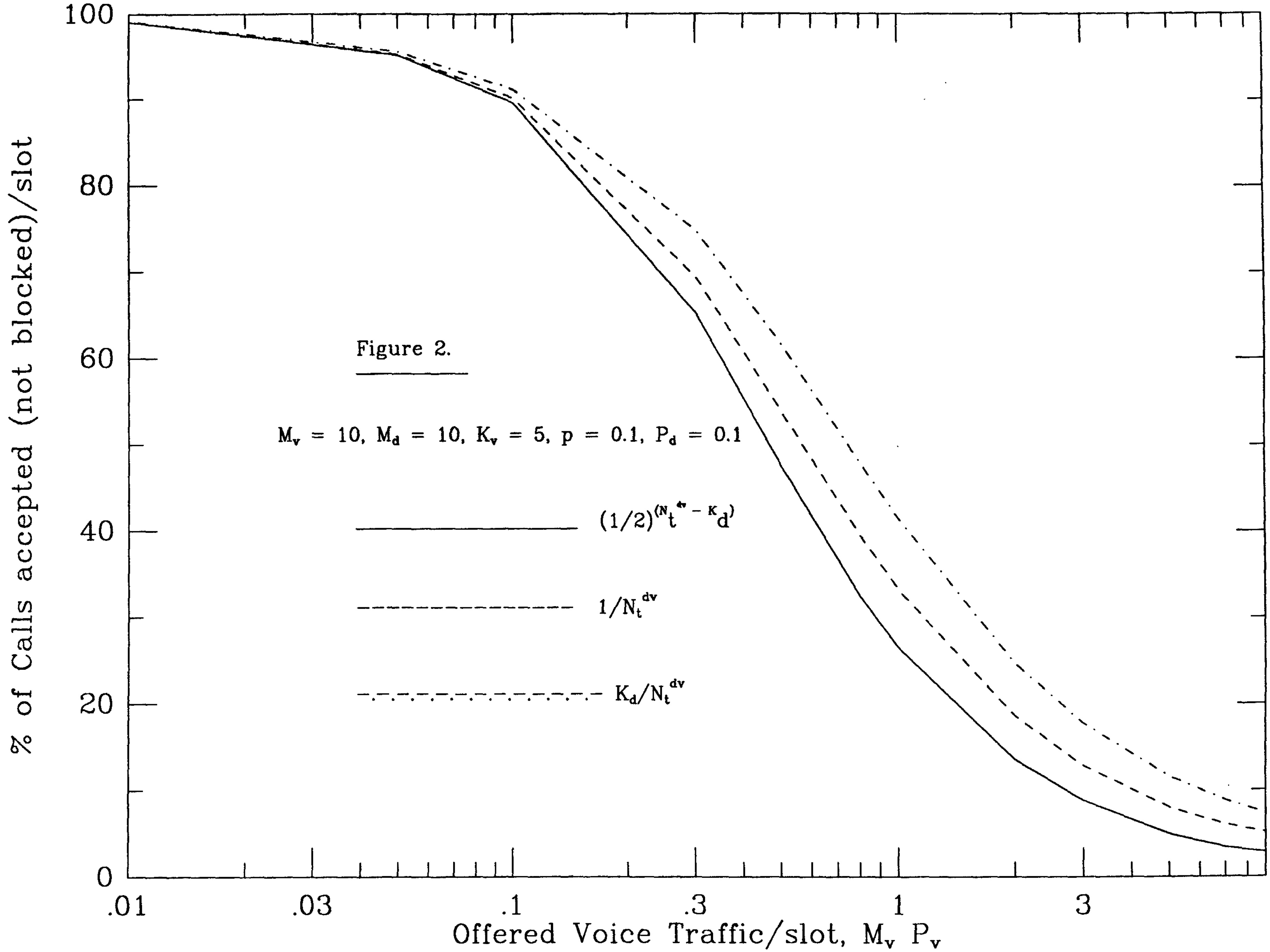
Figure 2.
M_v = 10, M_d = 10, K_v = 5, p = 0.1, P_d = 0.1
(1/2)^(N_t^dv − K_d)
1/N_t^dv
K_d/N_t^dv
100
80
60
40
20
0
% of Calls accepted (not blocked)/slot
.01
.03
.1
.3
1
3
Offered Voice Traffic/slot, M_v P_v

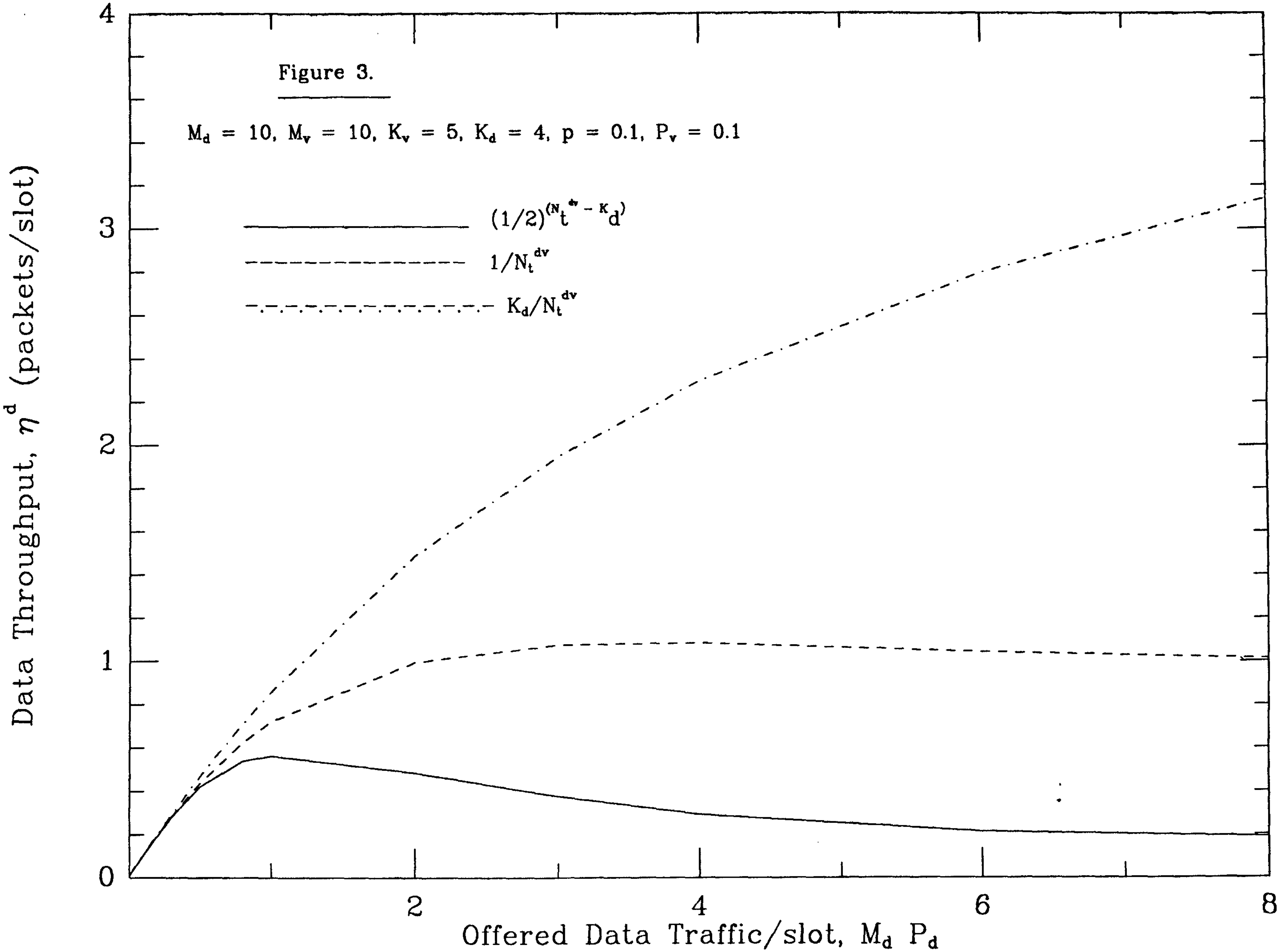
Figure 3.
$M_d = 10, M_v = 10, K_v = 5, K_d = 4, p = 0.1, P_v = 0.1$
$(1/2)^{(N_t^{dv} - K_d)}$
$1/N_t^{dv}$
K_d/N_t^{dv}
Data Throughput, η^d (packets/slot)
Offered Data Traffic/slot, M_d P_d
0
1
2
3
4
2
4
6
8

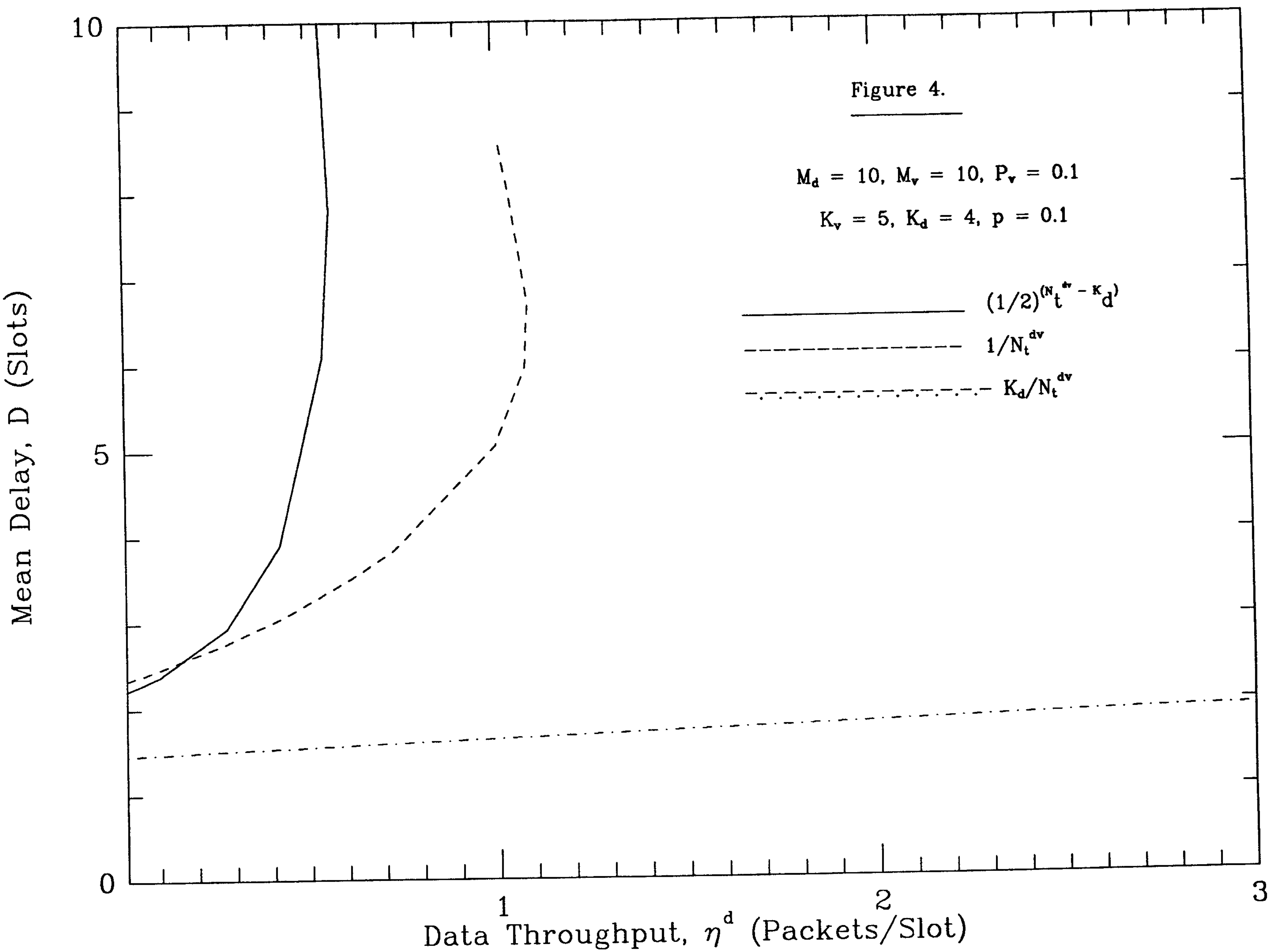

Figure 4.
$M_d = 10, M_v = 10, P_v = 0.1$
$K_v = 5, K_d = 4, p = 0.1$
$(1/2)^{(N_t^{dv} - K_d)}$
$1/N_t^{dv}$
K_d/N_t^{dv}
10
5
0
1
2
3
Mean Delay, D (Slots)
Data Throughput, η^d (Packets/Slot)

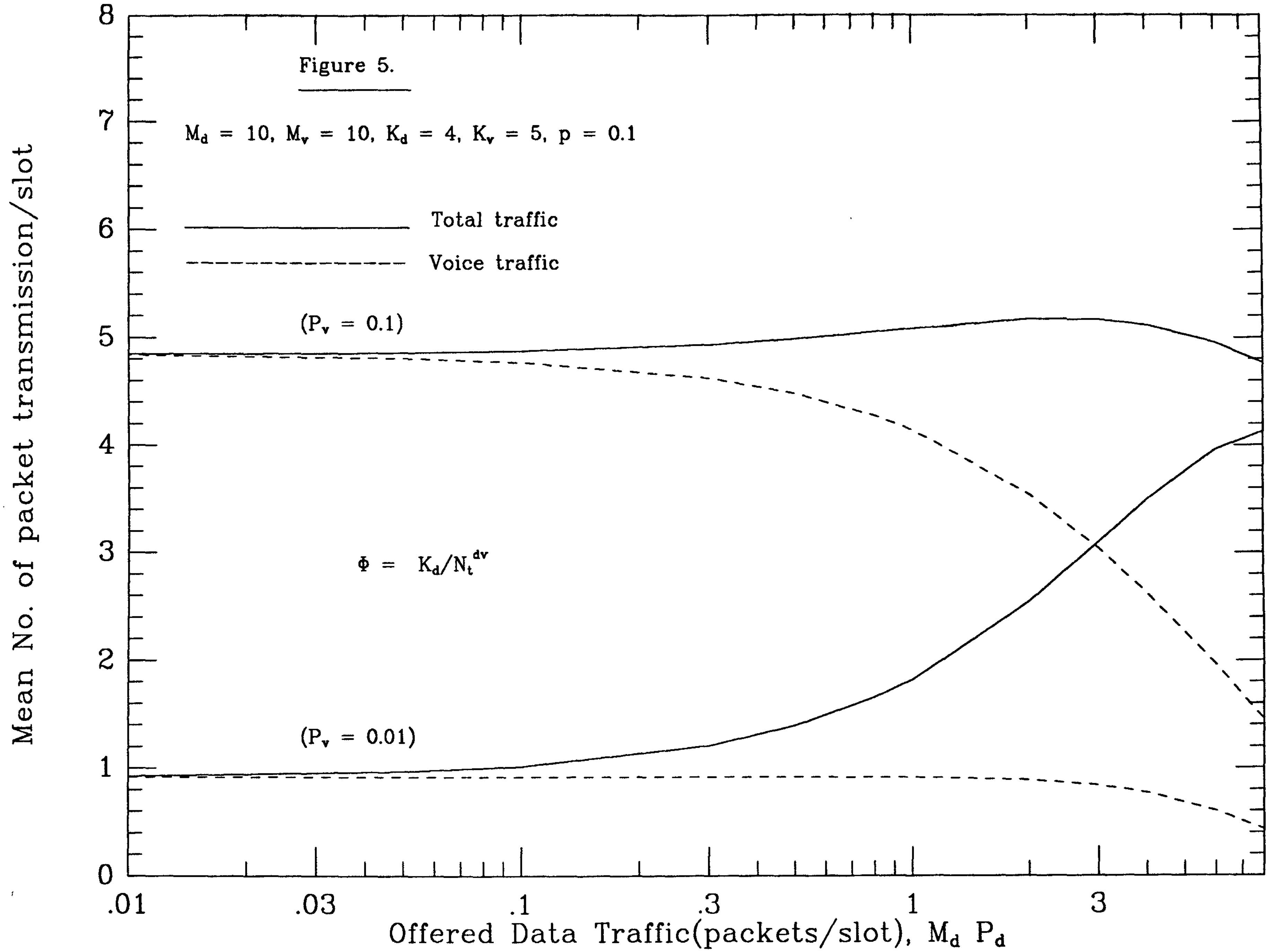

Figure 5.
$M_d = 10$, $M_v = 10$, $K_d = 4$, $K_v = 5$, $p = 0.1$
Total traffic
Voice traffic
($P_v = 0.1$)
$\Phi = K_d/N_t^{dv}$
($P_v = 0.01$)
Mean No. of packet transmission/slot
Offered Data Traffic(packets/slot), M_d P_d
8
7
6
5
4
3
2
1
0
.01
.03
.1
.3
1
3

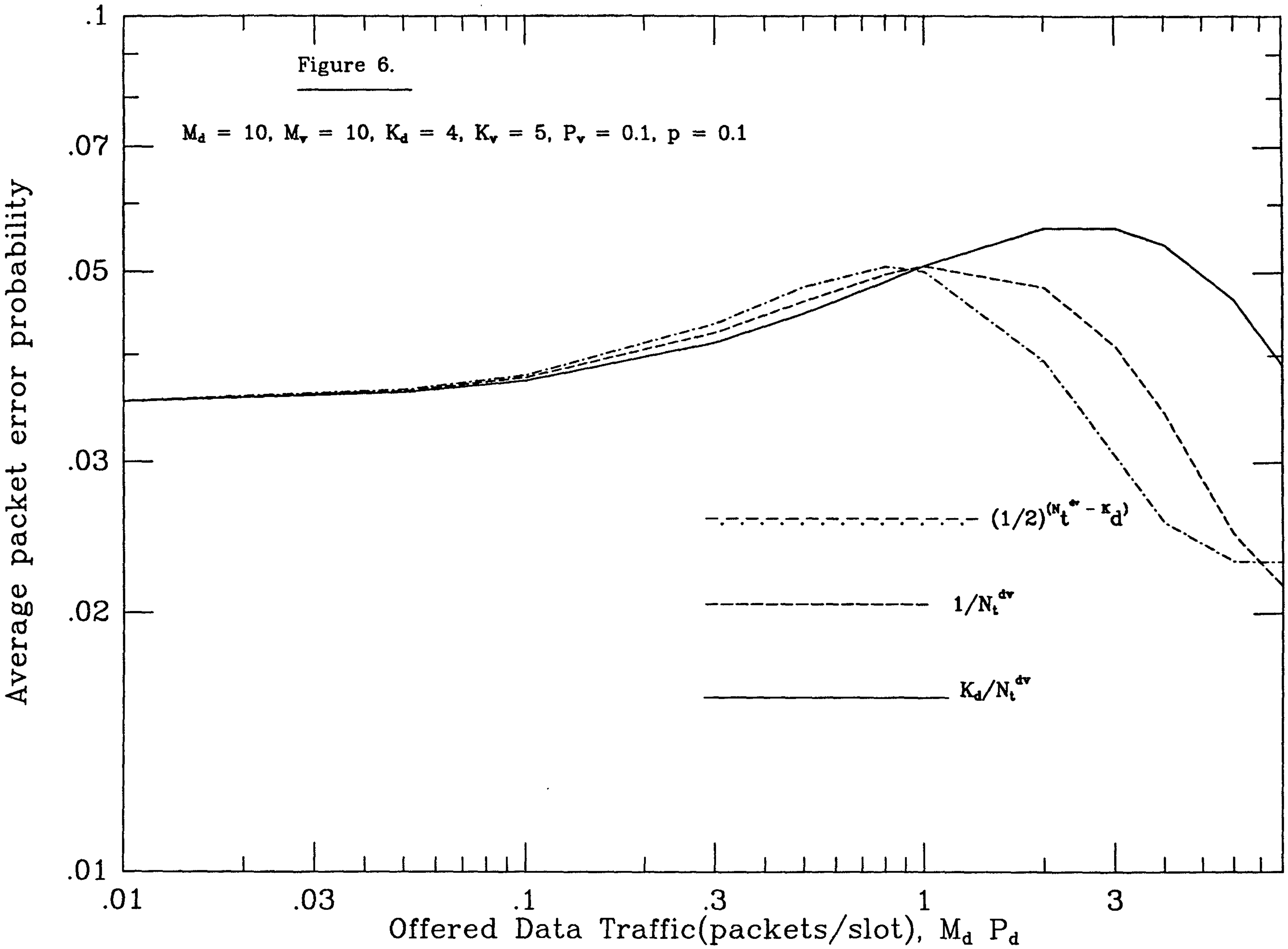

Figure 6.
$M_d = 10, M_v = 10, K_d = 4, K_v = 5, P_v = 0.1, p = 0.1$
Average packet error probability
.1
.07
.05
.03
.02
.01
.01
.03
.1
.3
1
3
Offered Data Traffic(packets/slot), M_d P_d
$(1/2)^{(N_t^{dv} - K_d)}$
$1/N_t^{dv}$
K_d/N_t^{dv}