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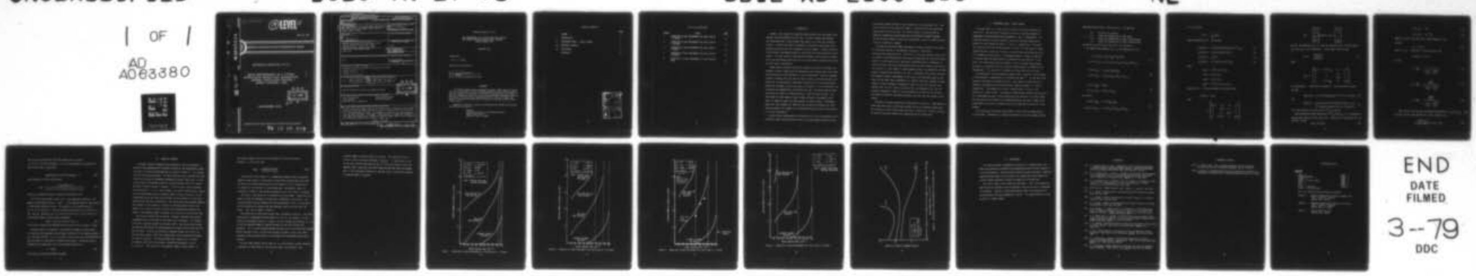

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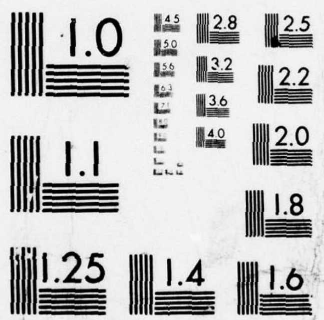

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DEFENSE COMMUNICATIONS ENGINEERING CENTER

TECHNICAL NOTE NO. 27-78

DATA PERFORMANCE IN A SYSTEM WHERE DATA PACKETS ARE TRANSMITTED DURING VOICE SILENT PERIODS - SINGLE CHANNEL CASE

NOVEMBER 1978

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Transmitting data packets over voice channels during silences in voice conversations has recently been proposed as a way of increasing effective capacity of a telecommunication system. A mathematical model for the single channel case has been developed. The model is described and a comparison between the case where data packets are allowed and the case where they are not allowed to be transmitted during voice silences is made.

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NOVEMBER 1978

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FOREWORD

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I. INTRODUCTION

Recently, the concept of transmitting data packets over the unused voice channels has been studies [1], [2], [3]. In such a system, data packets are transmitted over a voice channel when there is no voice call presently using it. Coviello and Vena [1] presented a statistical multiplexer which allowed data and voice to be integrated in such a manner. Fischer and Harris [2] developed a performance model for the Coviello and Vena statistical multiplexer. A further analysis (see Weinstrin, Malpass, and Fischer [3]) showed that large data queues could build up if no flow control procedures were used for this type of multiplexer.

These studies simulated the work of Castello [4] to see if the data performance of such a system could be improved by also allowing data to be transmitted during the silent periods of a voice conversation. Castello extended the work of Brady, [5], [6], and [7], Kekre, Sexena, and Srivasiava [8], and Sexena [9], by developing a three-state formulation of a voice conversation (talkspurts followed to one of two types of silences). In his formulation, the talkspurts are exponentially distributed with mean .05 second; once the talkspurt ends two types of silences may occur - gaps and long silences. The distribution of the length of each of these silences is exponential with mean .01 second for gaps and 2.846 seconds for long silences. Furthermore, gaps occur with probability .9685 and long silences with probability .0315. The data packets are allowed to be transmitted during the long silent periods of a voice conversation.

Castello then incorporated this formulation of a voice conversation into a simulation model, which predicted various links performance measures such as the average number and delay of data packets over a multichannel link. The basic structure of this simulation model is consistent with the multiplexer advanced by Coviello and Vena. The Castello model is now being used to quantify transmission savings which may be obtained by transmitting data during the long silences in the voice conversation, as well as, when no voice calls are present on a channel.

Although the simulation model developed by Castello certainly can be used for the quantification of the potential savings, it lacks some other desirable characteristics. First of all, since the average holding time to transmit a data packet may be four orders of magnitude less than the holding time for a voice conversation, many events representing what is happening to the data traffic have to be characterized during one voice holding time period. This results in a vast number of random numbers having to be generated, and thus, greatly increasing the simulation run time. Furthermore, the current structure and computer resource requirements of the single link simulation model do not make the link model easily extendable to a network algorithm. Because of these reasons, we have developed a mathematical model for the single channel case. Sherman [10], Hsu [11], Kekre, and Saxena [12], and Georganas [13], also developed a mathematical model for data performance, but they did not consider the random fluctuations representing whether the voice call was present on the channel.

The model including assumptions are described in section II. Some numerical comparisons with cases in which data packets are given a separate channel and not allowed to use the voice pauses are given in section III. Finally, section IV contains concluding remarks and recommendations for future work.

II. PERFORMANCE MODEL - SINGLE CHANNEL

We assume there is a single channel serving arriving data packets and voice calls. Let the arrival process of voice calls and data packets be independent Poisson processes with parameters λ_1 and λ_2, respectively. We assume the length of time to transmit a voice call and data packet be exponentially distributed with means μ_1^{-1} and μ_2^{-1}, respectively. The voice calls have priority over the data packets, in that if a data packet is being transmitted and a voice call arrives or a voice silent period ends the voice call preempts the data packet. The preempted packet is placed at the head of the data queue and data packets are selected from the queue on a first-come, first-served basis when the channel is free to transmit data packets. No queue is allowed for voice calls and an infinite queue is allowed for the data packets. If a voice call is present, arriving voice calls are lost.

We assume that a voice conversation begins with a talkspurt, and the length of the talkspurt is exponentially distributed with mean α^{-1}. At the end of the talkspurt two types of silences may occur. Each type of silence has an exponential distribution which describes their length and means, β_1^{-1} and β_2^{-1}, respectively. The probability a type 1 silence occurs is Π and $1-\Pi$ the probability a type 2 occurs. We assume all these random variables are statistically independent and that a voice conversation may end even in a silent period. The data packets are allowed to be transmitted during the type 2 (long) silent period when a voice call is present, and at all times when a voice call is not present.

Let Q_V and Q_D be the steady state number of voice calls and data packets in the system. Furthermore, in steady state define V to be the random variable representing the status of the voice call. We then have

$$V = \begin{cases} 0 & \text{no voice call present} \\ 1 & \text{voice call present and in talk spurt} \\ 2 & \text{voice call present and in type 1 silent period} \\ 3 & \text{voice call present and in type 2 silent period} \end{cases}$$

and define $P_{i,j} \equiv Pr\{V=i, Q_D=j\}$ for $i=0,1,2,3$, and $j=0,1,\ldots$.

The steady state equations for $P_{i,j}$ now become for $j=1,2,\ldots$

$$(\lambda_1+\lambda_2)P_{0,0} = \mu_1 P_{1,0}+\mu_1 P_{2,0}+\mu_1 P_{3,0}+\mu_2 P_{0,1}$$

$$(\lambda_1+\lambda_2+\mu_2)P_{0,j} = \mu_1 P_{1,j}+\mu_1 P_{2,j}+\mu_1 P_{3,j}+\mu_2 P_{0,j+1}+\lambda_2 P_{0,j-1}; \tag{1}$$

$$(\lambda_2+\alpha+\mu_1)P_{1,0} = \lambda_1 P_{0,0}+\beta_1 P_{2,0}+\beta_2 P_{3,0}$$

$$(\lambda_2+\alpha+\mu_1)P_{1,j} = \lambda_1 P_{0,j}+\beta_1 P_{2,j}+\beta_2 P_{3,j}+\lambda_2 P_{1,j-1}; \tag{2}$$

$$(\lambda_2+\beta_1+\mu_1)P_{2,0} = \alpha\Pi P_{1,0}$$

$$(\lambda_2+\beta_1+\mu_1)P_{2,j} = \alpha\Pi P_{1,j}+\lambda_2 P_{2,j-1}; \tag{3}$$

and finally,

$$(\lambda_2+\beta_2+\mu_1)P_{3,0} = \alpha(1-\Pi)P_{1,0}+\mu_2 P_{3,1}$$

$$(\lambda_2+\beta_2+\mu_1+\mu_2)P_{3,j} = \alpha(1-\Pi)P_{1,j}+\lambda_2 P_{3,j-1}+\mu_2 P_{3,j+1}. \tag{4}$$

For $|z| \leq 1$ define

$$P_i(z) = \sum_{j=0}^{\infty} P_{ij} z^j.$$

Then, from equations (1) - (4) we have

$$a_0(z)P_0(z) = \mu_1 z[P_1(z)+P_2(z)+P_3(z)]+\mu_2(z-1)P_{0,0} \quad (5)$$

$$a_1(z)P_1(z) = \lambda_1 P_0(z)+\beta_1 P_2(z)+\beta_2 P_3(z) \quad (6)$$

$$a_2(z)P_2(z) = \alpha \Pi P_1(z) \quad (7)$$

$$a_3(z)P_3(z) = \alpha(1-\Pi)zP_1(z)+\mu_2(z-1)P_{30}; \quad (8)$$

where

$$a_0(z) = -\lambda_2 z^2+(\lambda_1+\lambda_2+\mu_2)z-\mu_2$$

$$a_1(z) = -\lambda_2 z+\lambda_2+\alpha+\mu_1$$

$$a_2(z) = -\lambda_2 z+\lambda_2+\beta_1+\mu_1$$

$$a_3(z) = -\lambda_2 z^2+(\lambda_2+\beta_2+\mu_1+\mu_2)z-\mu_2$$

Equations (5) - (9) can be written in the matrix form

$$A(z)P(z) = B(z)$$

with

$$A = \begin{bmatrix} a_0(z) & -\mu_1 z & -\mu_1 z & -\mu_1 z \\ -\lambda_1 & a_1(z) & -\beta_1 & -\beta_2 \\ 0 & -\alpha\Pi & a_2(z) & 0 \\ 0 & -\alpha(1-\Pi)z & 0 & a_3(z) \end{bmatrix}$$

$$P(z) = \begin{bmatrix} P_0(z) \\ P_1(z) \\ P_2(z) \\ P_3(z) \end{bmatrix} \text{ and } B(z) = \mu_2(z-1) \begin{bmatrix} P_{0,0} \\ 0 \\ 0 \\ P_{30} \end{bmatrix}$$

We note from equations (6), (7), and (8) that once $P_1(z)$ is found, $P_0(z)$, $P_2(z)$, and $P_3(z)$ can be determined. Using Cramer's Rule we have

$$P_1(z) = \frac{\det(A_1(z))}{\det(A(z))} \tag{9}$$

where

$$A_1(z) = \begin{bmatrix} a_0(z) & \mu_2(z-1)P_{0,0} & -\mu_1 z & -\mu_1 z \\ -\lambda_1 & 0 & -\beta_1 & -\beta_2 \\ 0 & 0 & a_2(z) & 0 \\ 0 & \mu_2(z-1)P_{30} & 0 & a_3(z) \end{bmatrix}.$$

The determinents $\det(A_1(z))$ and $\det(A(z))$ can be evaluated directly, one gets

$$\det(A_1(z)) = \mu_2(z-1)a_2(z)[P_{30}(\beta_2 a_0(z)+\lambda_1\mu_1 z)+\lambda_1 a_3(z)P_{0,0}] \tag{10}$$

and

$$\det(A(z)) = a_0(z)a_1(z)a_2(z)a_3(z)-\alpha\Pi a_3(z)(\beta_1 a_0(z)+\lambda_1\mu_1 z) -\alpha(1-\Pi)za_2(z)(\beta_2 a_0(z)+\lambda_1\mu_1 z)-\lambda_1\mu_1 a_2(z)a_3(z)z. \tag{11}$$

Thus, the problem is solved once we find $P_{0,0}$ and P_{30}.

Before developing some expressions for $P_{0,0}$ and $P_{3,0}$, it is necessary to develop some results for the voice calls. Putting z=1 in equations (5), (7), and (8), we have

$$\lambda_1 P_0 = \mu_1(P_1+P_2+P_3) \tag{12}$$

$$(\beta_1+\mu_1)P_2 = \alpha\Pi P_1 \tag{13}$$

$$(\beta_2+\mu_1)P_3 = \alpha(1-\Pi)P_1, \tag{14}$$

where $P_i = \Pr\{V=i\}$ for i=0,1,2,3. Since $P_1+P_2+P_3 = 1-P_0$,

we have

$$P_0 = 1/(1+\rho_1) \tag{15}$$

where $\rho_1=\lambda_1/\mu_1$. Using (13), (14) and the fact that

$$P_1+P_2+P_3 = \rho_1/(1+\rho_1)$$

we have

$$P_1 = \frac{\rho_1}{1+\rho_1} \frac{1}{1+\frac{\alpha\Pi}{\beta_1+\mu_1}+\frac{\alpha(1-\Pi)}{\beta_2+\mu_1}} \tag{16}$$

$$P_2 = \frac{\rho_1}{1+\rho_1} \frac{\frac{\alpha\Pi}{\beta_1+\mu_1}}{1+\frac{\alpha\Pi}{\beta_1+\mu_1}+\frac{\alpha(1-\Pi)}{\beta_2+\mu_1}} \tag{17}$$

$$P_3 = \frac{\rho_1}{1+\rho_1} \frac{\frac{\alpha(1-\Pi)}{\beta_2+\mu_1}}{1+\frac{\alpha\Pi}{\beta_1+\mu_1}+\frac{\alpha(1-\Pi)}{\beta_2+\mu_1}} . \tag{18}$$

These results shall be used to develop two equations in $P_{0,0}$ and $P_{3,0}$. Since $P_1(1)\equiv P_1$, we have from equation (9), using l'Hopital rule,

$$\frac{P_1 \det A'(1)}{\lambda_1\mu_2(\beta_2+\mu_1)(\beta_1+\mu_1)} = P_{0,0} + P_{3,0} . \tag{19}$$

One can show (see [14] and [15]) that there exists a unique z_0, in (0,1) such that $\det(A(z_0)) = 0$; since the numerator of equation (9) must also be zero at z_0, we have

$$P_{30}(\beta_2 a_0(z_0)+\lambda_1\mu_1 z_0)+\lambda_1 a_3(z_0)P_{0,0} = 0 \ . \tag{20}$$

Using equations (19) and (20), we have

$$P_{30} = \frac{P_1 a_3(z_0)\det A'(1)}{\mu_2(\beta_2+\mu_1)(\beta_1+\mu_1)(\lambda_1 a_3(z_0)-\beta_2 a_0(z_0)-\lambda_1\mu_1 z_0)} \tag{21}$$

and $P_{0,0}$ is determined from this equation and equation (19).

This solves the problem, since $P_1(z)$ is now completely specified. One measure of performance that we seek is the expected number of data packets in the system, denoted by $E(Q_D)$. This quantity can be found from $P_0'(1)$, $P_1'(1)$, $P_2'(1)$, and $P_3'(1)$. One can differentiate equation (9) to find $P_1'(1)$ and then use equations (6), (7), and (8) to find $P_0'(1)$, $P_2'(1)$, and $P_3'(1)$. The expected number of data packets now becomes

$$E(Q_D) = P_0'(1)+P_1'(1)+P_2'(1)+P_3'(1) \tag{22}$$

Using Little's Formula the expected waiting time for data is $E(W_D)=\lambda_2^{-1}E(Q_D)$.

Another quantity of interest is the maximum throughput of data packets. This quantity also represents the existence condition for the random variable, Q_D, to be well defined. If $\rho_2=\lambda_2/\mu_2$ then ρ_2 must be less than the proportion of time the channel is available to transmit data packets. The quantity P_0+P_3 represents this proportion, and so the existence condition is

$$\rho_2 < P_0+P_3 \tag{23}$$

with $(P_0+P_3)\mu_2$ being the maximum throughput.

III. NUMERICAL EXAMPLES

A computer program implementing the mathematical results presented in section II was developed and is currently running on the IBM 370/155 at DCEC. Five sets of curves were developed and are shown in Figure 1-5 for comparative and illustrative purposes. For each figure the voice conversation formulation and the specific parameters developed by Castello were used. A voice holding time of 100 seconds as well as the data holding time of .01 second were held fixed for Figures 1 through 4. The first four figures represent voice loadings of .1, .5, 1, and 5 erlangs. For each of these figures comparisons are made between giving a separate channel to data, allowing it to be transmitted during the long silent periods, and not allowing it to be transmitted during the voice conversation. Each figure shows the expected number of data packets in the system as a function of the data packet arrival rate. Figure 5 is a plot of a family of curves (each representing different data loads) of the expected number of packets as a function of voice holding time.

The vertical dashed lines on Figures 1 through 4 represent the maximum throughputs that are achieved for the three cases considered. For instance, if we give a separate channel to the data traffic, then the maximum throughput would be 100 packets per second because the channel would always be available for data. If we do not give data a separate channel but allow it to use the long silences in the voice conversation, then the maximum throughput is given by $\mu_2(P_0+P_1)$. For the case where data transmission is allowed only when no voice calls are present, the maximum throughput is given by $\mu_2/(1+\rho_1)$. The result for the expected number of data packets in the system, $E(Q_D)$, for this rule was developed in [3], and for ease of reference is also given here:

$$E(Q_D) = \frac{\rho_2(1+\rho_1)^2 \rho_1\lambda_2/\mu_1}{(1+\rho_1)(1-\rho_1\rho_2-\rho_2)} . \tag{24}$$

From the first four figures it is immediately apparent that the expected number of data packets in the system is significantly reduced when one allows the data to be transmitted during the long silences of a voice conversation, as compared to not allowing it to be transmitted. Furthermore, there is a substantial increase in the maximum throughput as the voice load is increased.

On Figure 3, the results of a simulation are also plotted. These plots are added to check the mathematical formulation, assumptions, and solution. The simulation model used was adapted from the one described in [3]. Its structure and philosophy tend to ensure that the simulated results are lower than the actual. This fact is shown in Figure 3.

The sensitivity to the voice holding time is examined in Figure 5. The voice holding time is decreased from 100 seconds (μ_1=.01) to .001 seconds (μ_1=1-00) while the load remains fixed at 1 erlang. For this loading the $P\{V=0\}\equiv P_0=.5$; i.e., (λ_2=40), the expected number of packets decrease as the voice holding time decreases. This is to be expected because the voice calls are holding the channels shorter periods of time. As the voice holding time goes to zero, the system converges to the case where data is not allowed to be transmitted during voice silences.

For data loads greater than or equal to .5 (λ_2=50 and 60), a similar decrease is observed for some values of voice holding times, but eventually the expected number of packets starts to increase. The reason for this increase is that the maximum throughput, $(P_0+P_3)\mu_2$, is decreasing to $\mu_2 P_0$. As the voice holding times goes to zero ($\mu_1^{-1}\to 0$), P_3 also goes to zero (see equation (18)). Since $P_0=.5$ for this figure and the data loads are greater than .5, the saturation condition on the data loads is causing the increase in expected number of packets.

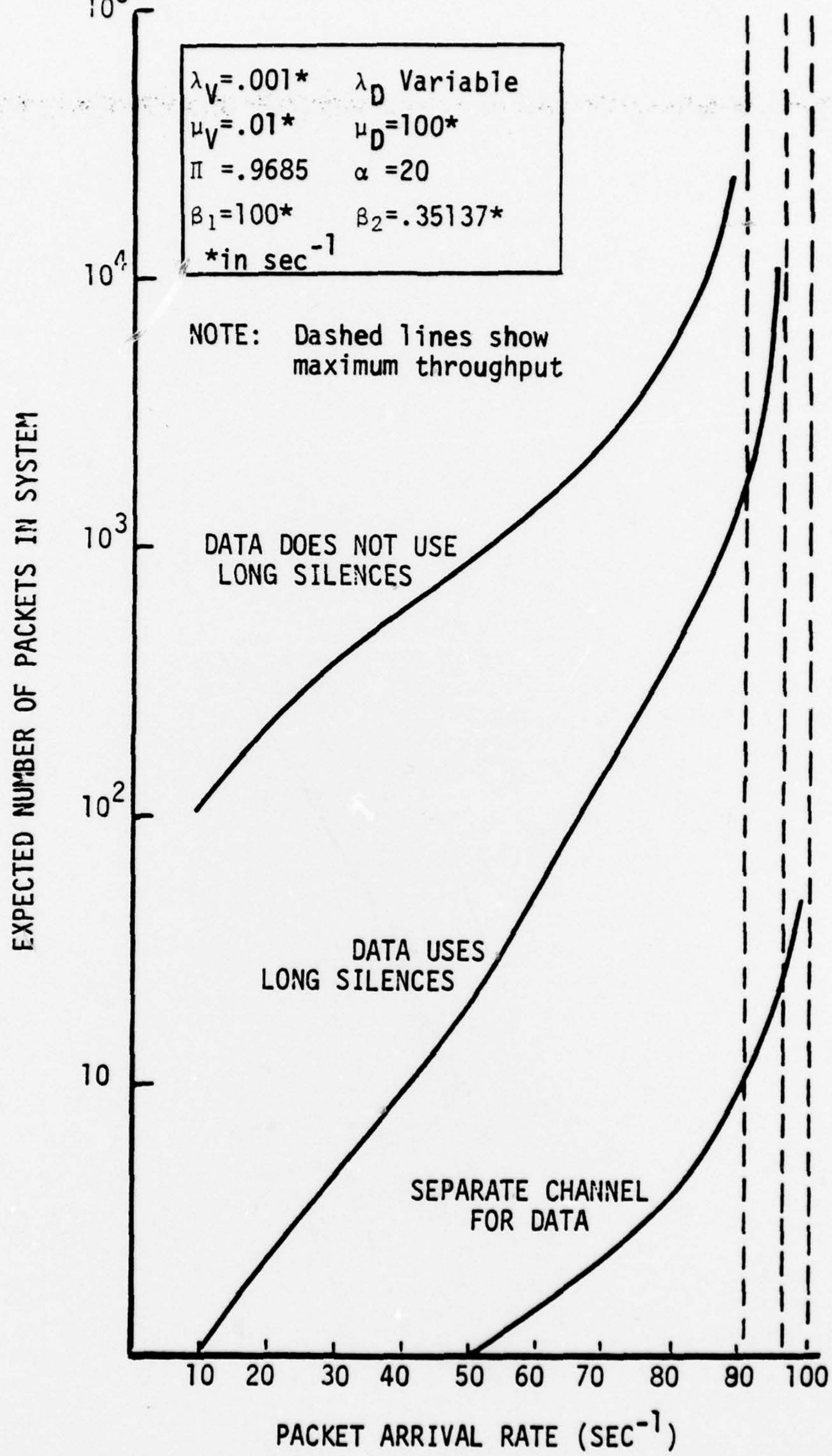

Figure 1. Comparison of Data Performance for Voice Load of .1 Erlangs

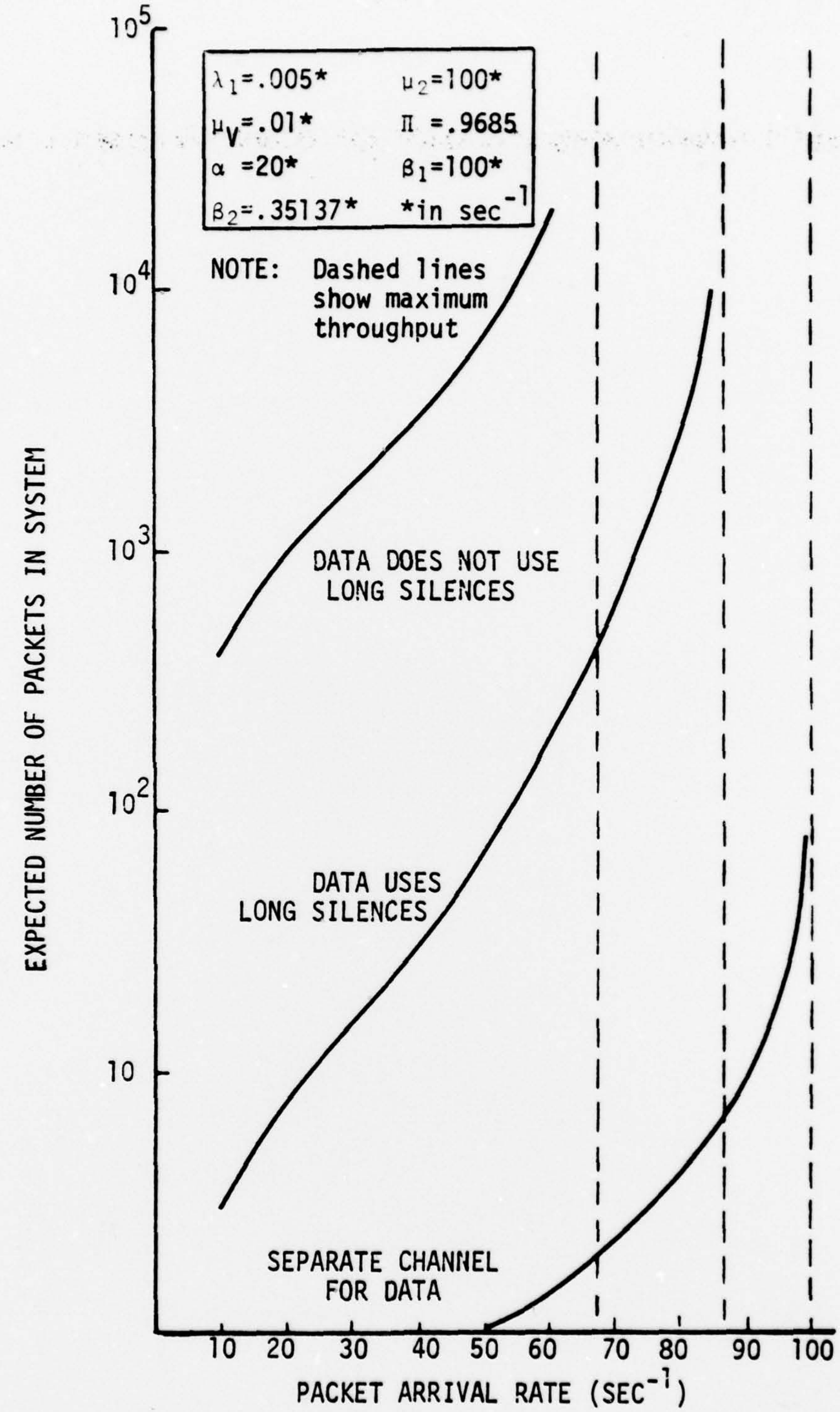

Figure 2. Comparison of Data Performance for Voice Load of .5 Erlangs

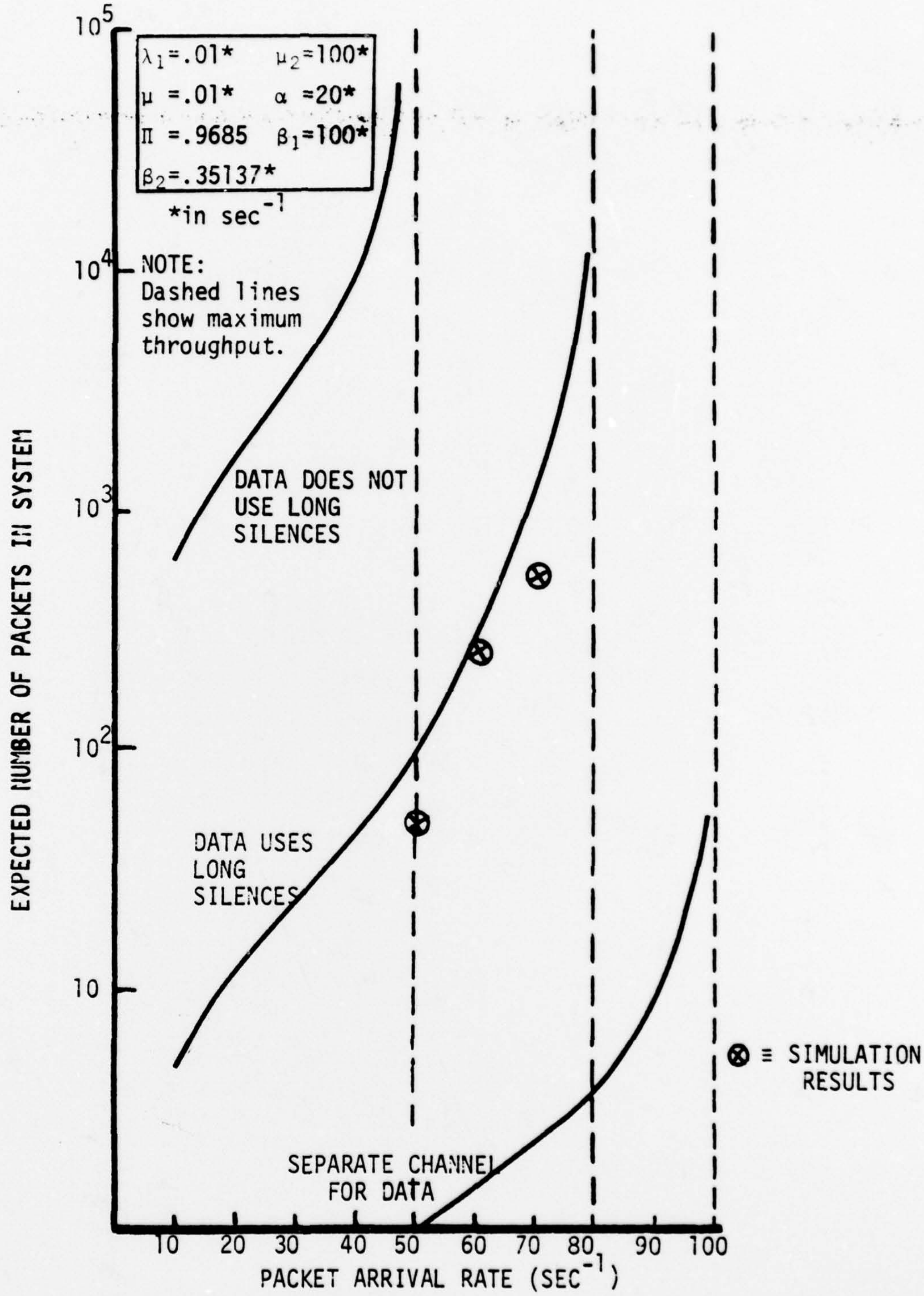

Figure 3. Comparison of Data Performance for Voice Load of 1 Erlang

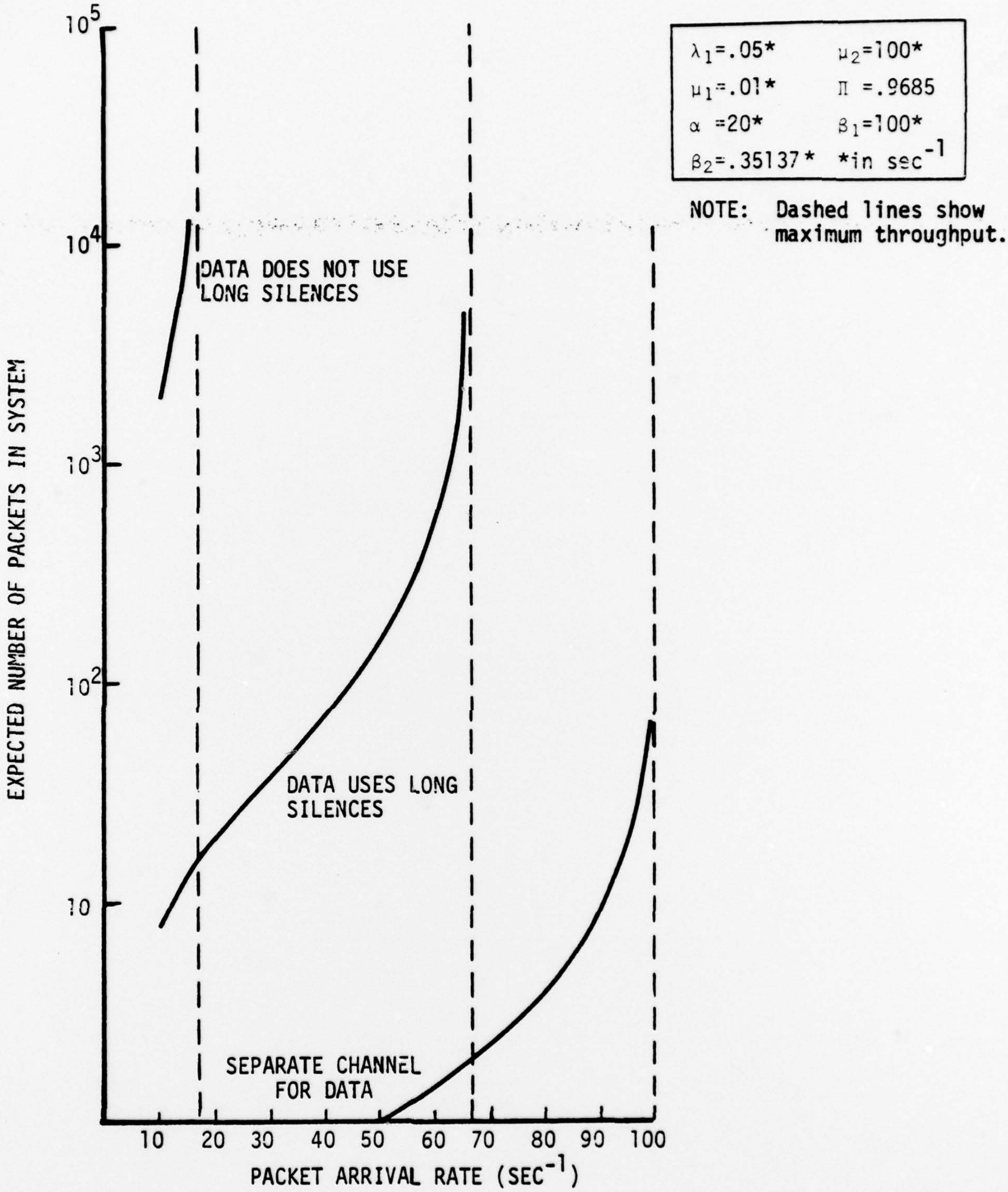

Figure 4. Comparison of Data Performance for Voice Load of 5 Erlangs

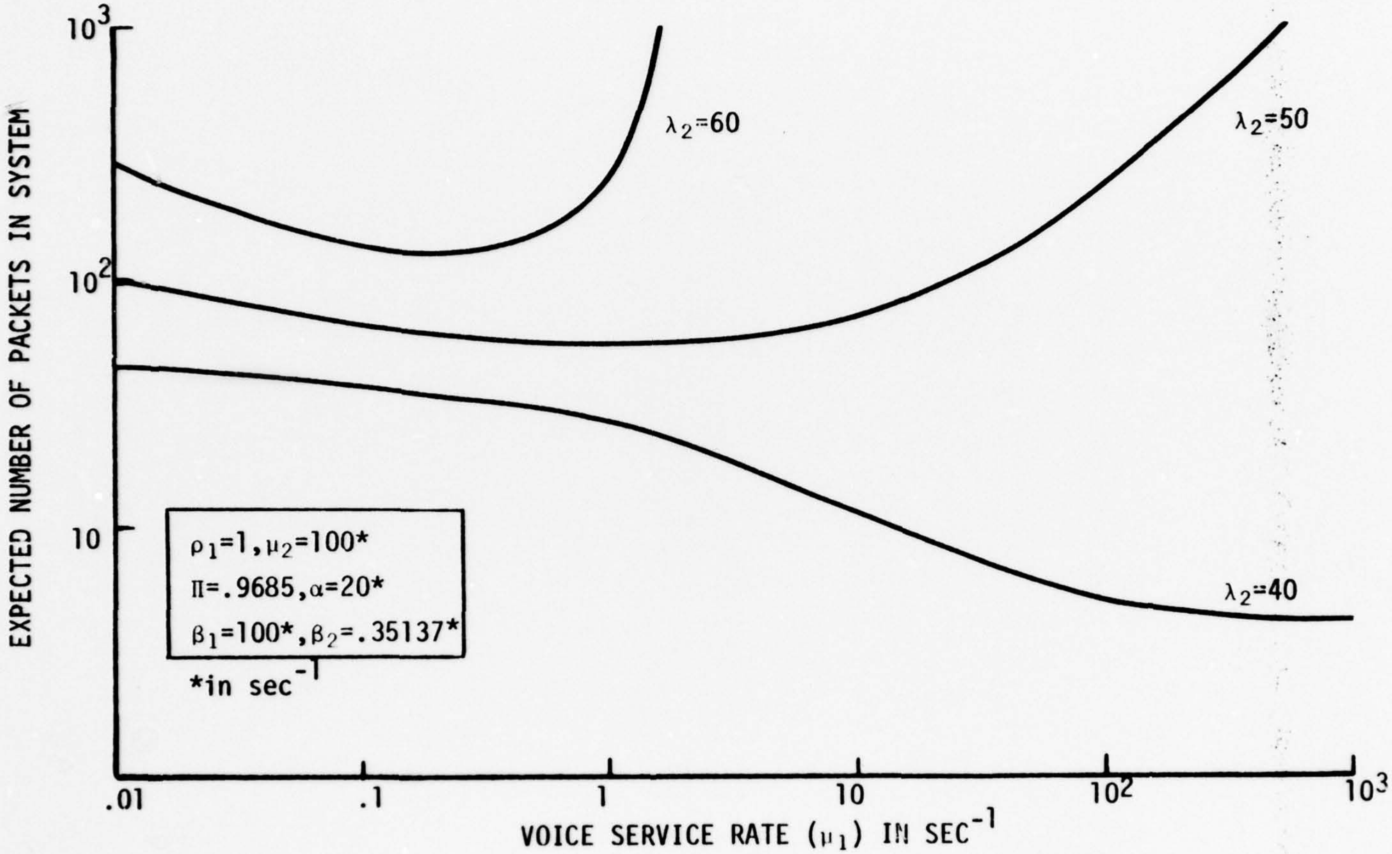

Figure 5. Sensitivity of Data Performance to Voice Holding Times

IV. CONCLUSIONS

This report presents a mathematical analysis of a communications link where data packets are allowed to be transmitted during the long silences in a voice conversation. Only the single channel case was considered. Numerical comparisons where data is not allowed to be transmitted during these long silences showed substantial savings and benefits can be obtained.

Two future pieces of work should be undertaken. First, the mathematical model whould be extended to the multichannel case and the result implemented on a computer. Second, a single approximation needs to be developed and checked against the exact mathematical results. This approximation can then be used in a network model.

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