論 文

통계적 패킷 음성/데이타 다중화기의 성능 해석

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Performance Analysis of a Statistical Packet Voice/Data Multiplexer

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要 約 본 논문에서는 통계적 패킷 음성/데이타 다중화기의 성능을 연구하였다. 성능해석은 음성과 데이타가서로 분리된 한정된 queue를 사용하고, 전송에 있어서 음성이 데이터보다 우선권을 갖는 것을 가정하고, 다중화기의 출력 link를 시간 slot 단위로 나누고 음성은 (M+1)-state의 Markov Process로, 데이타는 Poisson process로 modeling 하여 수행하였다. 전송시 음성신호가 데이타 신호보다 우선권을 가지므로 음성의 queueing behavior는 data에 거의 영향을 받지 않는다. 따라서 본 연구에서는 음성의 queueing behavior를 먼저 해석한 다음 data의 queueing behavior를 해석하였다. 패킷 음성 다중화기의 성능해석은 입력상태와 buffer의 점유를 2차원의 Markov chain을 가지고 formulation 하였고, 집적된 음성/data의 다중화기는 data를 추가한 3차원 Markov chain으로 하였다. 이러한 model을 사용하여 Gauss-Seidel 방법으로 결과를 얻고 simulation으로 입증하였다. 이들 결과로 부터 음성 가입자의 수, 출력 link 용량, 음성의 queue 크기, 음성의 overflow 확률에서는 서로 trade-off 가 있고 data에서도 비슷한 tradeoff가 있음을 알았다. 또한 입력 traffic 량과 link의 용량에 따라서 음성과 데이타간의 성능에서 서로 tradeoff가 있고, TASI의 이득이 2이상이고 음성가입자의 수가 적을 경우 데이타의 평균 지연시간은 buffer의 최대길이 보다 길음을 알아내었다.

ABSTRACT In this paper, the performance of a statistical packet voice/data multiplexer is studied. In this study we assume that in the packet voice/data multiplexer two separate finite queues are used for voice and data traffics, and that voice traffic gets priority over data. For the performance analysis we divide the output link of the multiplexer into a sequence of time slots. The voice signal is modeled as an (M+1)-state Markov process, M being the packet generation period in slots. As for the data traffic, it is modeled by a simple Poisson process. In our discrete time domain analysis, the queueing behavior of voice traffic is little affected by the data traffic since voice signal has priority over data. Therefore, we first analyze the queueing behavior of voice traffic, and then using the result, we study the queueing behavior of data traffic. For the packet voice multiplexer, both input state and voice buffer occupancy are formulated by a two-dimensional Markov chain. For the integrated voice/data multiplexer we use a three-dimensional Markov chain that represents the input voice state and the buffer occupancies of voice and data. With these models, the numerical results for the performance have been obtained by the Gauss-Seidel iteration method. The analytical results have been verified by computer simulation. From the results we have found that there exist tradeoffs among the number of voice users, output link capacity, voice queue size and overflow probabi-

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lity for the voice traffic, and also exist tradeoffs among traffic load, data queue size and overflow probability for the data traffic. Also, there exists a tradeoff between the performance of voice and data traffics for given input traffics and link capacity. In addition, it has been found that the average queueing delay of data traffic is longer than the maximum buffer size, when the gain of time assignment speech interpolation (TASI) is more than two and the number of voice users is small.

I. INTRODUCTION

As a first step toward the integrated services digital network (ISDN), extensive research effort is being given for the development of an integrated voice and data network. For voice/data integration, many researchers have studied some form of the hybrid switching technique in which circuit switching is used for voice traffic, and packet switching is used for data traffic. According to the study of Gitman and Frank, however, it is known that packet switching is superior to other switching techniques for voice/data integration.[1] In spite of the advantage of the packet switching technique for votice/data integration, the queueing behavior of voice/data transmission systems in a packet-switched network has not been studied extensively.

So far, the queueing behavior of a packet-switching system for voice and data transmission has been studied by seveal researchers.[2-6] Also, the buffer behavior of data traffic in a synchronous packet switching system has been investigated. Considering the voice signals as the main source of random interrupts, Heins,[7] Shanthikumar,[8] Kekre et al.[9] and Sriram et al.[10] analyzed queueing models in an integrated voice and data network. Modeling the voice traffic as a Markov source, Shanthikumar,[8] and Kekre et al.[9] showed that the behavior of the data buffer with Markovian interruptions is considerably different from the case when the source is interrupted from service by random interruptions. Recently, Lee ana Un analyzed the performance of statistical voice/data multiplexing systems with voice storage.[11] In their study, the performance of four statistical voice/data multiplexing schemes based on newly proposed queueing, frame management and integrated flow control methods have been analyzed.

In this work, we are concerned with the performance analysis of a statistical packet voice/data multiplexer. In a frame-based multiplexing system, it is normally assumed that all the input voice traffics are generated only at the start of a frame. But, this assumption is far from the reality because input voice traffics are independent from each other. Moreover, the data arriving in the middle of a frame must wait to be transmitted until the start of the next frane even if the output link is free for transmission at the instant of arrival. Therefore, it is desirable to have a new queueing model for voice by which voice packets can be generated without interrelationship among voice users. For this purpose we propose the slotted synchronous packet switching technique.

In our study we assume that in the packet voice/data multiplexer, two separate finite queues are used for voice and data traffics, and that each queue is served based on the first-in first-out (FIFO) rule. Also, we assume that voice traffic gets priority over data, and that the output channel is divided into a sequence of time slots.

For the voice traffic, we formulate a joint probability density function of the input state and the queue size of the voice buffer. This enables to determine the queueing behavior of the voice buffer. To satisfy the high channel utilization and short time delay for voice traffic, we introduce a variable rate coding scheme as a means of flow control for the voice traffic. In this scheme the packet generation period of input voice is changed according to the queueing behavior of voice traffic. As for the data traffic, we formulate a three dimensional joint probability density function for the input voice state, voice queue and data queue, which leads to finding the queueing behavior of the data buffer.

Following this introduction, in Section II statistical characteristics and modeling of voice signals are studied. Then, the queueing behavior of a packet voice multiplexer with fixed rate coding is analyzed in Section III, and the behavior of the same system with flow control is investigated in Section IV. In Section V, the queueing behavior of a packet data multiplexer under voice interrupt is analyzed. Finally, in Section VI, the procedure of computer simulation is discussed, and its results are compared with those of analysis to verify the analytical results.

II. STATISTICAL CHARACTERISTICS AND MODELING OF VOICE SIGNALS

We first consider the statistical characteristics of voice signals for the performance analysis of a packet voice multiplexer. It is known that talkspurts have approximately negative exponential distribution, and pauses have constant-plus-negative exponential distribution.[12,13] Here we approximate the speech pattern as a two-state Markov process with the negative exponential density functions. Then, we can write distributions of talkspurt and pause lengths, $f_T(t)$ and $f_P(t)$, respectively, as

$$f_T(t)=\frac{1}{L_T}\exp(-t/L_T), \quad t\geq 0,$$
$$f_P(t)=\frac{1}{L_P}\exp(-t/L_P), \quad t\geq 0, \tag{1}$$

where L_T and L_P are average talkspurt and pause lengths, respectively.

As a first step to analyze the performance of a statistical packet voice multiplexer with voice buffer, we divide the output channel into a sequence of time slots. Also, we choose a two-state Markov model for a single voice signal as shown in Fig. 1, since both the distributions of talkspurt and pause lengths can be approximated by exponential distributions.

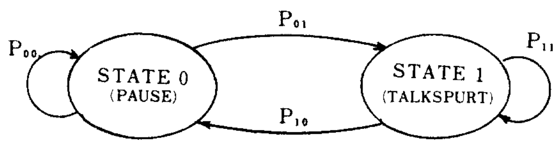


Fig. 1 Two-state markov chain model of speech activity.

Let the state 0 represent a pause state and the state 1 a talkspurt state in Fig. 1. Then, the transition probabilities representing the two-state Markov process between two adjacent time slots are given by

$$\begin{aligned} P_{00}&=1-T_S/L_P,\\ P_{01}&=T_S/L_P,\\ P_{10}&=T_S/L_T,\\ P_{11}&=1-T_S/L_T, \end{aligned} \tag{2}$$

where T_S is one slot interval time. Eq.(2) indicates that the probability that there exists a talkspurt in a slot time depends on its immediately preceding state. With two-state modeling of the Markov chain, the probabilities that there exist pause and talkspurt in the steady state are given, respectively, by

$$P_0=\frac{L_P}{L_P+L_T}$$
and
$$P_1=\frac{L_T}{L_P+L_T}. \tag{3}$$

Here we assume that one voice packet is transmitted in a slot time. We note that there are a fixed number of slots between successive packet generations from a voice source. We define a frame to be the interval between two successive packet generations from a single talkspurt of a voice signal.

The slot time at which talkspurt is packetized is in general different for each user. In a new talkspurt period which comes after a pause period, the timing of packet generation is not necessarily synchronized at the multiples of a frame time with that of the preceding talkspurt even for the same voice source. Furthermore, the timing of voice packet generation is in general not synchronized between different voice sources. But, voice packets are generated periodically in one talkspurt segment.

To analyze the performance of the packet voice multiplexer, therefore, we propose here to use an (M+1)-state Markov process (M is a packet generation period in slots) as a model for a single voice source. A state transition diagram of the Markov process is shown in Fig. 2. In this figure the state M reflects the fact that the talkspurt remains continuously for the M slot periods, and one voice packet will be generated at this state. In other words, the state 0 represents pause, and the state M is used to differentiate the packet generation time from the other (M-1) states which generate no packets during the talkspurt period. We assume that a talkspurt that is shorter than one packet interval and the trailing

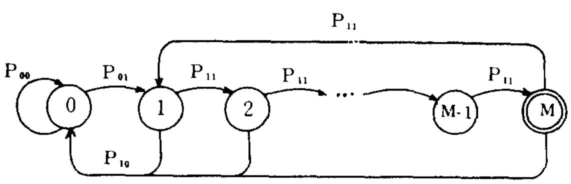


Fig. 2 State transition diagram of voice states of a single user in a packet voice multiplexer (M is the number of packet generation state and state 0 indicates pause).

portion of a long talkspurt whose length is shorter than one frame period are discarded in our model.

The steady state probability U_i(i = 0,1, ..., M) that voice remains in the state i can be obtained from the state transition diagram of Fig. 2 as follows:

$$U_o = \frac{P_{10}}{P_{01} + P_{10}}$$
$$U_k = \frac{P_{01} P_{10}}{(1 - P_{11}^{M})(P_{10} + P_{01})} P_{11}^{k-1}, \qquad (4)$$
$$k = 1, \cdots, M.$$

If we assum that there are N voice users, the probability P[x=i] that i voice packets are generated in a time slot in the steady state is equal to the probability that there are i voice users at the state M. That is,

$$P[X=i] = {}_N C_i U_M^i (1 - U_M)^{N-i}$$
$$\text{where} \quad {}_N C_i = \frac{N!}{(N-i)!\,i!}. \qquad (5)$$

The basic configuration of a statistical packet voice/data multiplexer which is being studied in this work is shown in Fig. 3, and its queueing model is given in Fig. 4. In Fig. 4, N is the number of input voice users. Speech detectors are used to determine active voice users. Each voice terminal generates a voice packet in every M-th slot as seen from Fig. 2. In this case, the time assignment speech interpolation (TASI) gain is given from its definition by

$$G = N / M. \qquad (6)$$

The state of each voice terminal is in one of (M+1) states, and therefore, the N voice terminals are distributed over (M+1) states. Let g_i be the number of voice sources which remain at the

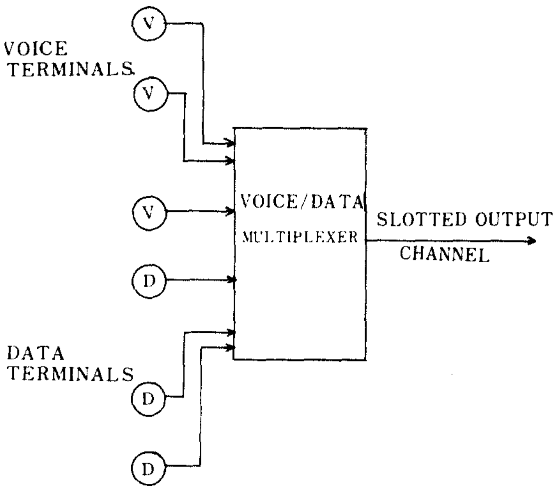


Fig. 3 Statistical multiplexing of voice / data signals.

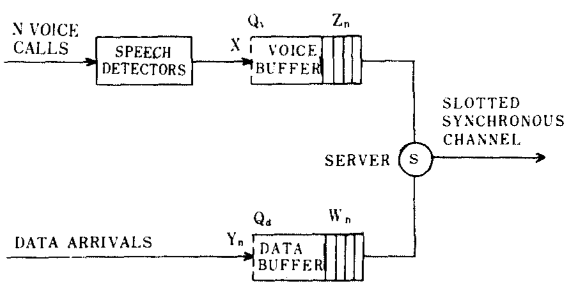


Fig. 4 Queueing model for a voice / data multiplexer.

state i at some slot time. Then, we have

$$\sum_{i=0}^{M} g_i = N. \tag{7}$$

Here, we assign a state for voice input according to the distribution of N voice users over (M+1) states. Therefore, the set $(g_0, g_1, \ldots, g_M)$ represents a state, each state being different from others. Each voice source generates a packet when it is in the state M. Therefore, the number of voice packets generated in a slot time is the same as the number of voice users remaining at the state M for the given slot time. Let us denote the input state number S defined with a given set of $(g_0, g_1, \ldots, g_M)$ as $S \triangleq f(g_0, g_1, \ldots, g_M)$, and the number of packets generated at the given state S by X(S). Then, the following relation holds:

$$X(S) = g_M. \tag{8}$$

The total number of states, S_{max}, is given by

$$S_{max} = {}_{(M+1)}H_N = {}_{M+N}C_N \tag{9}$$

where ${}_{M+N}C_N$ is the number of ways in choosing N elements from (M+N) elements.

In the next section we investigate the queueing behavior of a packet voice multiplexer with fixed rate coding.

III. ANALYSIS OF A PACKET VOICE MILTIPLEXER WITH FIXED RATE CODING

There can be many ways in assigning a state S(S=1,2,..., S_{max}) to a set of user distribution. The method we chose here is as follows. The number of voice sources in the higher-numbered state becomes larger as the state number S increases. Table 1 shows an example showing the relationship between the state number S and the distribution of voice users over (M+1)-states for the case of N=4 and M=3. One can note that there is one-to-one correspondence between the input state number and the user distribution. It is convenient to asign the voice user distribution as shown in Table 1 in analyzing the performance of a fixed rate coder and in considering variable rate coding which will be discussed in the next section.

Since the state number of an input state, S, can be identified by the user distribution, $(g_0, g_1, \ldots, g_M)$, we can write the state number from the set $(g_0, g_1, \ldots, g_M)$ as

$$S = \sum_{j=0}^{M-2} \sum_{k=0}^{g_{M-j}-1} {}_{(M-j)}H_N - \sum_{l=1}^{j} g_{M-l+1} + g_1 + 1. \tag{10}$$

For example, let us find the input state number when we have the distribution of users over (M+1) -states as $(g_0=2, g_1=1, g_2=1, g_3=0)$, and M=3 and N=4. From (10), the state number S=f(2,1, 1,0) can be calculated as

$$S=\sum_{j=0}^{1}\sum_{k=0}^{g_{3-j}-1}{}_{(3-j)}H_4-\sum_{l=1}^{j}g_{3-l+1}+1+1=7. \tag{11}$$

Note that this result corresponds to the seventh row shown in Table 1.

To find the queueing behavior in voice multiplexing, it is necessary to find the transition probability between two arbitrary states. Let $P_t(A,B)$ be the probability that a state changes from A to B. Here the states A and B are represented by the distributions of users, $(g_0, g_1, \cdots, g_M)$ and $(h_0, h_1, \cdots, h_M)$, respectively. And the total number of voice users in the system is N, that is, $\sum_{i=0}^{M} g_i = \sum_{i=0}^{M} h_i = N$. $P_t(A, B)$ can be determined using the binomial distribution as follows:

$$\begin{aligned} P_t(A,B) &= {}_{g_1}C_{h_2}P_{11}^{h_2}P_{10}^{g_1-h_2}\cdot{}_{g_2}C_{h_3}P_{11}^{h_3}P_{10}^{g_2-h_3}\cdots \\ &\cdot{}_{g_{M-1}}C_{h_M}P_{11}^{h_M}P_{10}^{g_{M-1}-h_M} \\ &\cdot\Big(\sum_{V=0}^{V_{max}}{}_{g_M}C_V P_{11}^{V}P_{10}^{g_M-V}\cdot \\ &\quad {}_{g_0}C_{h_1-V}\,P_{01}^{h_1-V}\,P_{00}^{g_0-h_1+V}\Big) \\ &\text{with } g_i\geq h_{i+1},\ 1\leq i\leq M-1, \end{aligned} \tag{12}$$

where V_{max} is given by

$$V_{max}=\min(g_M, h_1)\ . \tag{13}$$

Note that v_{max} is equal to the number of voice users which changes from the state M to the state 1, while talkspurt remains continuously after packet generation. Also, we note that the sum of state transition probabilities from the state A to all states including the state A itself is one. That is,

$$\sum_{B=1}^{S_{max}} P_t(A, B)=1. \tag{14}$$

In the steady state, we can find the probability of each state as

$$\begin{aligned} P[A] &= P[A(g_0, g_1, \cdots, g_M)] \\ &= \frac{N!}{g_0!\,g_1!\cdots g_M!}U_0^{g_0}U_1^{g_1}\cdots U_M^{g_M} \end{aligned} \tag{15}$$

and $\sum_{A=1}^{S_{max}} P[A]=1$.

Now, we analyze the finite buffer behavior for voice traffic. To investigate the queueing behavior, it is necessary to find a joint probability density function of the input state and the queue size of the voice buffer. Before considering the system performance, let us define the following notations:

- N Number of input voice calls,
- Q_V Voice buffer length (in packets),
- Z_n Occupancy of the voice queue at the beginning of n-th slot time (in packets),
- $X(S_n)$ Number of voice packets generated in the slot n when the number of input voice states is $S_n(g_0, g_1, \cdots, g_M)$. $X(S_n)$ is equal to g_M.

For the buffer size Z_n, the following recurrence relationship holds for two consecutive time slots:

$$\begin{aligned} Z_{n+1} &= (Z_n-1)^+ + X(S_n) \\ &= (Z_n-1)^+ + g_M(S_n) \end{aligned} \tag{16}$$

where $(Z_n-1)^+ \triangleq \max(Z_n-1, 0)$.

Here the sequences $\{Z_n\}$ and $\{S_n\}$ form a two-cimensional Markov chain, and their joint probability is given by

$$P[Z_{n+1}=l',\ S_{n+1}=m']$$

$$= \sum_{l=0}^{Q_V} \sum_{m=1}^{S_{max}} P[Z_{n+1}=l',\ S_{n+1}=m' \mid Z_n=l,\ S_n=m] \cdot P_{old}[Z_n=l,\ S_n=m] \quad (17)$$

$$\text{for} \quad 0 \leq l' \leq Q_V, \quad 1 \leq m' \leq S_{max}$$

where

$$P[Z_{n+1}=l',\ S_{n+1}=m' \mid Z_n=l,\ S_n=m] = \alpha_{l'lm} \cdot P_t(m, m'),$$

$$\alpha_{l'lm} = P[Z_{n+1}=l' \mid Z_n=l,\ S_n=m], \quad (18)$$

$$P_t(m, m') = P[S_{n+1}=m' \mid S_n=m].$$

Here the probaility $P_t(m, m')$ is given by (12), and $\alpha_{l'lm}$ is equal to zero or one depending on the values Z_{n+1} and Z_n. That is,

$$\alpha_{l'lm} = 1, \text{ if } l' = \min[Q_V, (l-1) + g_M(S_n=m)] \quad (19)$$
$$= 0, \quad \text{otherwise.}$$

Combining (17) through (19),we can obtain the following set of equations in the steady state:

$$P[l', m'] = \sum_{m=1}^{S_{max}} P_t(m, m') \sum_{l=0}^{Q_V} \alpha_{l'lm} P_{old}[l, m] \quad (20)$$

$$\text{for} \quad 0 \leq l' \leq Q_V \text{ and } 1 \leq m' \leq S_{max},$$

$$\sum_{l'=0}^{Q_V} \sum_{m'=1}^{S_{max}} P[l', m'] = 1.$$

In the above equation, $P[l, m]$ represents the steady state joint probability of the voice buffer and the input voice state. We can solve the set of state equations using the Gauss-Seidel iteration method with the initial value chosen as

$$P_{old}[l, m] = \frac{P[m]}{Q_V + 1} \quad (21)$$

where P[m] is given by (15) with A replaced by m. $P[l, m]$ is adjusted at every iteration such that

$$P_{new}[l, m] = P[l, m] \frac{P[m]}{\sum_{l=0}^{Q_V} P[l, m]} \quad (22)$$

where $P_{new}[l, m]$ represents the joint pdf adjusted after one iteration. The iterative numerical calculation is repeated until a convergence criterion is met. The marginal pdf of $P[l, m]$ gives an indication of the buffer behavior in the steady state, and can be obtained as

$$Z[l] = \sum_{m=1}^{S_{max}} P_{new}[l, m] \quad (23)$$

And the carried voice load α_V which indicates the utilization of the output channel is given simply by

$$\alpha_V = 1 - Z[l=0]. \quad (24)$$

In addition, the offered voice traffic load β_V is obtained as

$$\beta_V = \sum_{m=1}^{S_{max}} P[S=m]\ X(m) \quad (25)$$

where X(m) is the number of packets generated in the state m. β_V can also be obtained by multiplying the voice user number N by the steady state probability that a single voice user remains in the state M:

$$\beta_V = N \cdot U_M \quad (26)$$

where U_M is given in[4]. Then, the overflow probability of a voice packet is given by

$$P_{ofv} = 1 - \alpha_V / \beta_V. \tag{27}$$

The average queueing delay T_{qv} in slot duration is given by

$$T_{qv} = \sum_{l=0}^{Q_V} l \cdot Z(l) / \alpha_V \cdot \tag{28}$$

IV. ANALYSIS OF A PACKET VOICE MULTIPLEXER WITH FLOW CONTROL

In a statistical packet voice multiplexer, the fluctuation of the occupancy of the voice queue is inevitable because of the inherent burstiness of voice signals, but this variation of buffer occupancy is not desirable. When the occupancy of the voice queue becomes large, it results in long delay, and there might be some overflow. On the contrary, when the occupancy of the voice queue becomes too small, there can be underflow of the voice queue, thus wasting resources. In this case, it is desirable to make the input traffic to increase so that the resources such as the queueing buffer and the output channel can be effectively utilized. To satisfy such requirements, we introduce here a variable rate coding scheme as a means of flow control of voice traffic.[14,15] In our slotted packet switching system,we assume that the packet size is fixed, and that a packet generation interval from each voice terminal is a multiple of one slot time. To control the coding rate of the input signal according to the buffer status, we divide the buffer status into several classes according to the amount of buffer occupied by voice. The packet generation period in slots M of input voice sources is determined at the end of every slot depending on the current buffer occupany Z_n as follows:

$$\begin{aligned} M(Z_n) &= M_1 \quad \text{for} \quad TH_0 \leq Z_n < TH_1 \\ &= M_2 \quad \text{for} \quad TH_1 \leq Z_n < TH_2 \\ &\vdots \\ &= M_L \quad \text{for} \quad TH_{L-1} \leq Z_n \leq TH_L = Q_V \end{aligned} \tag{29}$$

where Q_V is the voice buffer size; $(TH_0, TH_1, TH_2, \cdots, TH_{L-1}, TH_L)$ is a sequence of L+1 integers in ascending order representing threshold values of the voice buffer; and $M_1, M_2, \cdots, M_L$ are the packet generation intervals in slots in ascending order for each corresponding buffer status.

The state transition diagram of each voice terminal for different buffer occupancies in a packet voice multiplexer with flow control is shown in Fig. 5. As the buffer occupancy increases, the state transition moves toward G_L to reduce the overflow probability by decreasing the coding rate. And, as the buffer occupany decreases, it moves toward G_1 to utilize fully the given channel capacity by increasing the coding rate. The recursive relation describing the queueing behavior of the voice buffer is given by

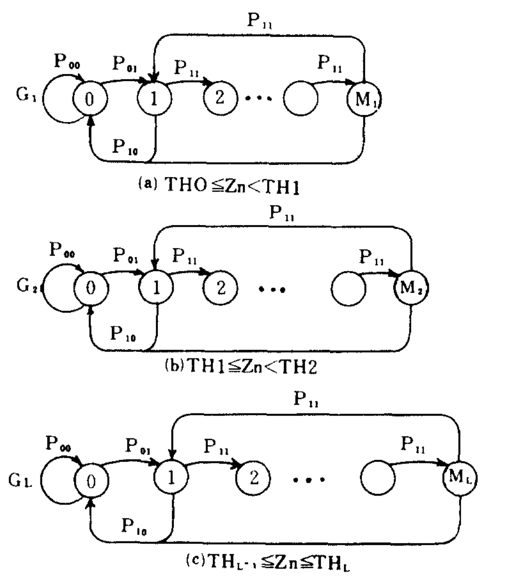


Fig. 5 State transition diagrams of each voice termial for different buffer occupancies in a packet voice multiplexer with flow control ($M_1 \leq M_2 \leq \cdots \leq M_L$).

$$Z_{n+1}=\min[Q_v,\ (Z_n-1)^{+}+X(A)]$$
$$=\min[Q_v,\ (Z_n-1)^{+}+g_{M(Z_n)}(A)] \quad (30)$$

where $g_{M(Z_n)}(A)$ is the number of voice packets generated when the number of input states is A, and the packet generation period $M(Z_n)$ in slots is dependent on the buffer size Z_n. And the number of states $S_{max}(Z_n)$ for a buffer level Z_n is given by

$$S_{max}(Z_n)=S(M(Z_n))=$$
$$={}_{(M(Z_n)+1)}H_N={}_{(M(Z_n)+N)}C_N \cdot \quad (31)$$

Table 1. Mapping table between the input state number and user distribution for the case of N=4 and M=3.

State No. S	$g_0(S)$	$g_1(S)$	$g_2(S)$	$g_3(S)$
1	4	0	0	0
2	3	1	0	0
3	2	2	0	0
4	1	3	0	0
5	0	4	0	0
6	3	0	1	0
7	2	1	1	0
8	1	2	1	0
9	0	3	1	0
10	2	0	2	0
11	1	1	2	0
12	0	2	2	4
13	1	0	3	0
14	0	1	3	0
15	0	0	4	0
16	3	0	0	1
17	2	1	0	1
18	1	2	0	1
19	0	3	0	1
20	2	0	1	1
21	1	1	1	1
22	0	2	1	1
23	1	0	2	1
24	0	1	2	1
25	0	0	3	1
26	2	0	0	2
27	1	1	0	2
28	0	2	0	2
29	1	0	1	2
30	0	1	1	2
31	0	0	2	2
32	1	0	0	0
33	0	1	0	3
34	0	0	1	3
35	0	0	0	4

Note that the number of states is a function of the voice buffer occupancy Z_n.

The two-dimensional diagram showing the buffer size occupied by input voice packets and the corresponding number of states which is determined by the buffer occupancy as well as the number N of voice users is shown in Fig. 6. The buffer between two threshold levels TH_i and TH_{i+1} forms the area A_i. Each state in an area describes the distribution of input voice users over the given states for the given packet generation period in slots M from (29). An example of user distribution for each input state has been given in Table 1 for N=4 and M=3. Note that each state in the same horizontal line across the threshold value of the buffer occupancy does not describe the same distribution of input voice user, and that there are no direct relationships among the states in the same horizontal line. The joint pdf between the buffer occupancy and the input voice state satisfies the following recursive equation:

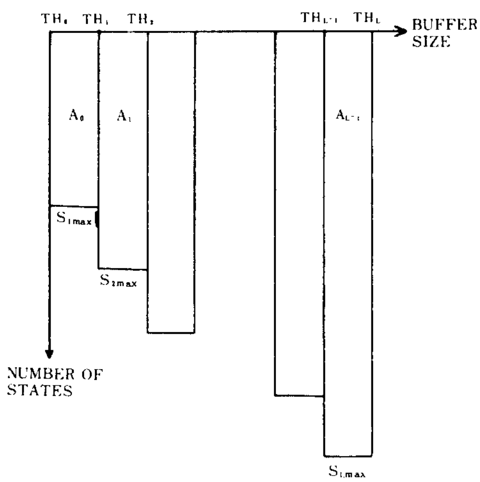


Fig. 6 Two-dimensional diagram showing the buffer size occupied by input voice packets and the corresponding number of states.
$(S_{1max} = {}_{(N+M(A0))}C_N,\ S_{2max} = {}_{(N+M(A1))}C_N, \ldots, S_{L\,max} = {}_{(N+M(AL-1))}C_N)$

$$P[Z_{n+1}=l',\ S_{n+1}=m'']$$
$$=\sum_{i=0}^{L-2}\sum_{l=TH_i}^{TH_{i+1}-1}\sum_{m=1}^{S_{max}(l)} P[Z_{n+1}=l',\ S_{n+1}=m' \mid Z_n=l,\ S_n=m]$$
$$\cdot P_{old}[Z_n=l,\ S_n=m]+\sum_{l=TH_{L-1}}^{TH_L}\sum_{m=1}^{S_{max}(l)} P[Z_{n+1}=l',\ S_{n+1}=m' \mid Z_n=l,\ S_n=m]$$
$$\cdot P_{old}[Z_n=l,\ S_n=m], \qquad (32)$$

for $Z_n \in A_i$ and $Z_{n+1} \in A_q$

where m'' is equal to m' if the buffer occupancies l and l' are in the same buffer area. If the buffer occupancies l and l' are not in the same area, a new destination state m'' corresponding to m' should be assigned. Let l and l' be in the buffer area A_i and A_q, respectively. There are two cases when Z_n and Z_{n+1} are not in the sme buffer area: (1) $i > q$ and (2) $i < q$. The first case means that the buffer occupancy decreases, and the second case indicates that the buffer occupancy increases. Fig. 7 shows an example of state change for each voice user between M=3 and M=5. In the first case, we choose m'' as the destination state which satisfies the following equations:

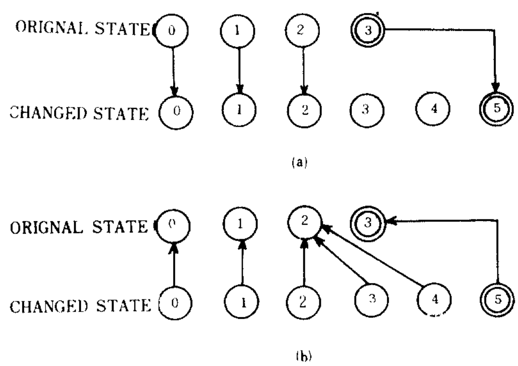


Fig. 7 An example of state change for each voice user between M=3 and M=5.
(a) When buffer occupancy increases (i. e., packet generation interval becomes longer).
(b) When buffer occupancy decreases (i. e., packet generation interval becomes shorter).

$$g_j^{l'}(m'') = g_j^{l}(m') \quad \text{for } 0 \leq j \leq M(l') - 2$$

$$g_j^{l'}(m'') = \sum_{i=M(l')-1}^{M(l)-1} g_j^{l}(m') \quad \text{for } j = M(l') - 1 \qquad (33)$$

$$g_j^{l'}(m'') = g_{M(l)}^{l}(m') \quad \text{for } j = M(l'),$$

where $g_j^{l'}(m'')$ is the number of voice users in the state j when one packet generation interval in slots is $M(l')$ and the input state number is m''. And $g_j^{l}(m')$ is the number of voice users in the state j when the packet generation period in slots is $M(l)$ and the input state number is m'.

In the second case, we choose m'' as the destination state which satisfies the following equations:

$$g_j^{l'}(m'') = g_j^{l}(m') \quad \text{for } 0 \leq j \leq M(l) - 1$$

$$g_j^{l'}(m'') = 0 \quad \text{for } M(l) \leq j \leq M(l') - 1 \qquad (34)$$

$$g_j^{l'}(m'') = g_{M(l)}^{l}(m') \quad \text{for } j = M(l').$$

Now, we have the following set of equilibrium equations in the steady state:

$$P[l', m''] = \sum_{i=0}^{L-2} \sum_{l=TH_i}^{TH_{i+1}-1} \sum_{m=1}^{S_{max}(l)} P_t(m, m') \, \alpha_{l'lm} P_{old}[l, m] + \sum_{l=TH_{L-1}}^{TH_L} \sum_{m=1}^{S_{max}(l)} P_t(m, m') \, \alpha_{l'lm} P_{old}[l, m] \qquad (35)$$

where $\alpha_{l'lm} = 1, \text{ if } l' = \min[Q_v, (l-1)^+ + g_M(m)],$
$0, \text{ otherwise.}$

Note that $P[l', m'']$ in (35) is the steady state joint probabilities of voice buffer contents and the input state number. From (35) we can find the steady state behavior of the joint pdf of the input state and buffer occupany. We use the Gauss-Seidel iteration method to determine the steady state behavior of a two-dimensional Markov chain with the initial value chosen as

$$P[l, m] = \frac{P[m]}{TH_{i+1} - TH_i} \quad \text{for } l \in A_i \text{ and } 0 \leq i \leq L-2 \qquad (36)$$
$$= \frac{P[m]}{TH_{i+1} - TH_i + 1} \quad \text{for } l \in A_i \text{ and } i = L-1,$$

where P[m] is the steady state probability that the input voice will remain in the state m when the total number of input states is decided according to the buffer region A_i. We note that the denominator on the right side of (36) is the size of the buffer area A_i. The joint pdf for the next slot time can be obtained from (35), and is adjusted as

$$P_{new}[l', m''] = P[l', m''] \cdot \frac{1}{\sum_{l'=0}^{Q_v} \sum_{m''=1}^{S_{max}(l')} P[l', m'']} \cdot \qquad (37)$$

This recursive iteration continues until a convergence condition is met.

The buffer behavior in the steady state can be obtained as follows:

$$Z[l] = \sum_{m=1}^{S_{max}(l)} P_{new}[l, m] \qquad (38)$$

where the number of input states, $S_{max}(l)$ is determined depending on the buffer level. The carried voice load or channel utilization α_V is given in the same form as in (24). We note that the buffer behavior is determined by the input voice traffic which is affected not only by its own statistical properties but also by the buffer occupancy. Therefore, the offered voice traffic load β_v of a variable rate coder can be written as

$$\beta_V = \sum_{i=0}^{L-2} \sum_{l=TH_i}^{TH_{i+1}-1} \sum_{m=1}^{S_{max}(l)} P[l,m] \cdot X^l(m) + \sum_{l=TH_{L-1}}^{TH_L} \sum_{m=1}^{S_{max}(l)} P[l,m] \cdot X^l(m) \qquad (39)$$

where $TH_0=0$, $TH_L=Q_V$, $\sum_{l=TH_i}^{TH_{i+1}-1} P[l,m]$ is the marginal probability for the input state when the buffer occupancy is in the area A_i and $X^l(m)$ is the number of voice packets generated at the given state m. Then, the overflow probability of the voice buffer can be obtained by (27), and the average queueing delay of voice packets is given by (28).

V. ANALYSIS OF A PACKET DATA MULTIPLEXER UNDER VOICE INTERRUPT

In this section, we analyze the queueing behavior of data traffic in an integrate voice/data multiplexer shown in Fig. 3. Since the voice traffic gets priority over the data traffic, the queueing behavior of voice traffic is little affected by the data traffic in our slotted synchronous transmission system. Therefore, here we can use the results for the queueing behavior of voice traffic presented in the preceding sections.

The queueing model of an integrated voice/data multiplexing system is shown in Fig. 4. Before analysis, let us define additional notations as the following:

Q_d Data buffer length (in packets)

W_n Occupancy of the data queue at the beginning of n-th slot time (in packets)

Y_n Number of input data packets arrived during the n-th slot time (in packets)

In a simple Poisson process, the size of an incoming data packet is fixed and the number of data packets arrived during a slot time Y is Poisson-distributed with the arrival rate of λ_s per slot interval. The discrete density function of Y is

$$P_Y(d) = \lambda_s^d \exp(-\lambda_s)/d!, \quad d=0, 1, 2, \cdots. \qquad (40)$$

The offered data traffic load β_d for the simple Poisson process is given by

$$\beta_d = \lambda_s. \qquad (41)$$

We first consider the behavior of the data buffer in the integrated voice/data multiplexing system without flow control, and then study it when flow control is done for the voice traffic. When no flow control is done, the queueing behavior of the voice/data multiplexing system of Fig. 4 can be represented mathematically in a recursive form as

$$\begin{aligned} Z_{n+1} &= \min[(Z_n-1)^+ + X(S_n), Q_v], \\ W_{n+1} &= \min[W_n+Y_n, Q_d], \quad \text{if } Z_n \neq 0, \\ &= \min[(W_n-1)^+ + Y_n, Q_d], \quad \text{if } Z_n = 0, \end{aligned} \qquad (42)$$

where $X(S_n)$ is the number of input voice packets generated at the n-th time slot according to the input state S_n. $X(S_n)$ is equal to the number of input voice users at the state M, g_M, which is determined by the input state S_n [see Fig. 2(a)]. Then, the sequence of three random variables (i.e., Z_n, S_n and W_n) forms a three-dimensional Markov chain. The joint pdf of those variables is given by

$$P[l', m', k'] = \sum_{l=0}^{Q_v} \sum_{m=1}^{S_{max}} \sum_{k=0}^{Q_d} \alpha_{l'lm} \beta_{k'lk} P_t(m, m') \cdot P[l, m, k] \qquad (43)$$

where

$\alpha_{l' lk} = P[l' | l, m] = 1$

if $l' = \min[Q_V, (l-1)^+ + g_M(S=m)]$ S_n

$= 0$ otherwise

$\beta_{l' lk} = 0$ if $j < 0$

$= P_Y(j)$ if $0 \leq j \leq Q_d$

where

$j = k' - k$ if $l \neq 0$

$= k' - (k-1)$ if $l = 0$.

$P_Y(j)$ in (43) is the probability that j data packets will arrive in time slot n. The above recurrence equation can be solved using the Gauss-Seidel iteration method with the following initial values:

$$P[l, m, k] = P[l, m] / (Q_d + 1), \quad (44)$$

where $P[l, m]$ is the two-dimensional steady state joint probability of the voice buffer and the input voice state. When $P[l', m', k']$ is calculated from the iterative equation (43), it is adjusted such that the numerical errors occurring during the iteration process may be minimized. Once we determine the converged value of $P[l, m, k]$, the queueing behavior of the data buffer can easily be found.

The average carried data traffic load can be obtained as

$$\alpha_d = \sum_{m=1}^{S_{max}} \sum_{k=1}^{Q_d} P[0, m, k]. \quad (45)$$

Note that α_d is the sum of probability distribution over the range where the voice queue size is zero, but the data queue size is not. The overflow probability of data packets is then

$$P_{ofd} = 1 - \alpha_d / \beta_d, \quad (46)$$

where β_d is the input load of data traffic given by (41). The average queueing time in slots of data packets can be obtained using α_d as

$$T_{qd} = \{ \sum_{k=0}^{Q_d} k (\sum_{l=0}^{Q_V} \sum_{m=1}^{S_{max}} P[l, m, k]) \} / \alpha_d + 2. \quad (47)$$

The term $\sum_{l=0}^{Q_V} \sum_{m=1}^{S_{max}} P[l, m, k]$ is the probability that the data queue size is k in the steady state. The total input traffic load is equal to the sum of each traffic load, and is given by

$$\beta_t = N \cdot U_M + \beta_d \cdot \quad (48)$$

And the total carried traffic load is equal to the sum of each carried traffic load, that is,

$$\alpha_t = \sum_{l=1}^{Q_V} \sum_{m=1}^{S_{max}} \sum_{k=0}^{Q_d} P[l, m, k] + \sum_{m=1}^{S_{max}} \sum_{k=1}^{Q_d} P[0, m, k]. \quad (49)$$

The overflow probability of total input traffics is then obtained from the total input load and carried load as

$$P_{of} = 1 - \alpha_t / \beta_t. \quad (50)$$

When a variable rate coding scheme is introduced as a means of flow control on the input voice traffic, the joint probability of having transition from (l, m) to (l', m'') is given by (32) where m'' satisfies (33) or (34) if l and l' are not in the same buffer area. Based on the formulation of (33) and (34), we have a three-dimensional Markov chain with the sequence $\{(Z_n, S_n, W_n)\}$ whose joint pdf is given by

$$P[l', m'', k'] = \sum_{l=0}^{L-2} \sum_{l=TH_l}^{TH_{l+1}-1} \sum_{m=1}^{S_{max}(l)} \sum_{k=0}^{Q_d} \alpha_{l'lm} \beta_{k'lk} \cdot P_t(m, m') P_{old}[l, m, k] + \sum_{l=TH_{L-1}}^{TH_L} \sum_{m=1}^{S_{max}(l)} \sum_{k=0}^{Q_d} \alpha_{l'lm} \beta_{k'lk} P_t(m, m') P_{old}[l, m, k]$$

where $\alpha_{l'lm} = P[l' | l, m]$, (51)

$\beta_{k'lk} = P[k' | l, k]$,

for $0 \le l' \le Q_v$, $1 \le m' \le S_{max}(l)$,

$0 \le k' \le Q_d$.

This recurrence equation can be solved by the Gauss-Seidel iteration method following the same procedure as we did for the voice/data multiplexing system without flow control on the voice traffic. Once we obtain the converged value, we can find the queueing behavior of the data buffer. The carried load α_d of the data traffic is given by

$$\alpha_d = \sum_{m=1}^{S_{max}(l)} \sum_{k=1}^{Q_d} P[0, m, k]. \quad (52)$$

The overflow probability of the data buffer is exactly the same as the case without flow control [see Eq. (46)]. The average queueing delay in slots of data packets is then obtained by

$$T_{qd} = \{ \sum_{k=0}^{Q_d} k (\sum_{l=0}^{L-2} \sum_{l=TH_l}^{TH_{l+1}-1} \sum_{m=1}^{S_{max}(l)} P[l, m, k] + \sum_{l=TH_{L-1}}^{TH_L} \sum_{m=1}^{S_{max}(l)} P[l, m, k]) \} / \alpha_d + \frac{1}{2}. \quad (53)$$

Also, the total carried traffic load is

$$\alpha_t = \sum_{l=0}^{L-2} \sum_{l=TH_l}^{TH_{l+1}-1} \sum_{m=1}^{S_{max}(l)} \sum_{k=0}^{Q_d} P[l, m, k] + \sum_{l=TH_{L-1}}^{TH_L} \sum_{m=1}^{S_{max}(l)} \sum_{k=0}^{Q_d} P[l, m, k] \quad (54)$$

$$+ \sum_{m=1}^{S_{max}(0)} \sum_{k=1}^{Q_d} P[0, m, k].$$

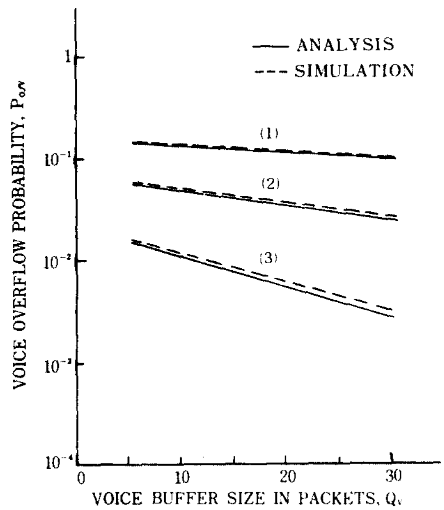
In addition, the equation of overflow probability for total input traffics is exactly the same as the case without flow control, which is given by (52).

VI. COMPUTER SIMULATION AND NUMBERICAL RESULTS

In this section, we discuss the procedure of computer simulation and compare its results to those of analysis. For the simulation of a packet voice multiplexer, we generated on-off speech patterns and voice packets using the random numbers, RAN, with uniform distribution in the range of 0 to 1. We performed the computer simulation up to ten million slots. Consequently, we could obtain the steady state queueing behavior by averaging the accumulated system parameters over the total number of slots.

For the case of variable rate coding, all the procedures were the same as for the case of fixed rate coding except that the buffer occupancy was checked at the end of every time slot. If the buffer occupancy exceeded a certain threshold, and was changed to other buffer region, the state number for each user was decided again by reallocating the input voices to new states using the method shown in Fig. 7. The method of assigning a new state was discussed in Section IV [see Eqs. (33) and (34)].

For numerical analysis, the Gauss-Seidal iteration method was programmed in FORTRAN and run on Data General MV-8000 supermini-computer. In computation for the steady state queueing behavior of the two- and three- dimensional Markov process, the parametric values were chosen as follows: L_T = 1.366 s (average talkspurt length), L_P=1.802 s (average pause length), T_s = 10 ms (time slot size), $T_f = T_s \cdot M$ (frame time). The number (N) of input users used in this study ranged from three to six. And the packet generation interval (M), which is counted in slots, was assumed to range from two to four. With N and M, the TASI gain may be obtained by N/M as given in (6). With these parameter values, we



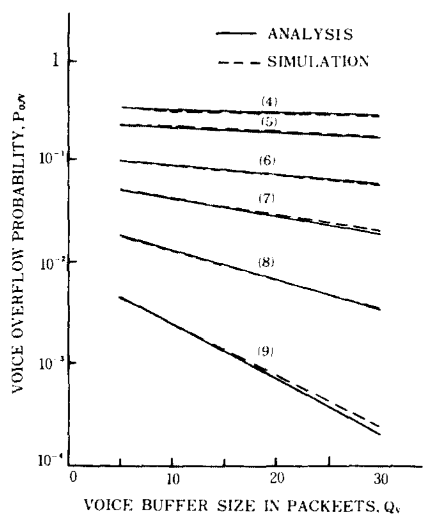


Fig. 8 Voice buffer overflow probability versus voice buffer size (Q_v)) with N and M as parameters when no flow control is done ($T_s = 10$ms).
(a) (1) N=4, M=2, G=2 (2) N=3, M=2, G=1.5
(3) N=4, M=3, G=1.33
(b) (4) N=6, M=2, G=3 (5) N=5, M=2, G=2.5
(6) N=6, M=3, G=2 (7) N=5, M=3, G=1.7
(8) N=6, M=4, G=1.5 (9) N=5, M=4, G=1.25

obtained overflow probability. This is shown in Fig. 8. As one can expect, it is seen in the figure that as the buffer size increases, the overflow probability decreases. The decreasing rate becomes smaller as the TASI gain increases, and vice versa. Also, it is seen that for the same TASI gain (e.g., (N=3, M=2) and (N=6, M=4)), the decreasing rate becomes larger when the input traffic is larger. It is known that, for satisfactory voice communication, the loss rate of voice packets should be less than 1 percent. For the speech loss to be in the tolerable range (i.e., for the overflow probability to be less than one percent), the following conditions should be satisfied. For N=4, M should be equal to or larger than three, and the buffer size must be larger than 150 ms in time. In this case, the TASI gain would be about 1.33. For M=4, the buffer size should be larger than 150 ms. For N=5, M should be at least four so that TASI gain is 1.25. In this case, the overflow probability is always less than one percent regardless of the buffer size. Also, for N=6, M should be equal to or larger than four, and the buffer size should be equal to or larger than 150 ms.

For the packet voice multiplexer with flow control, the packet generation interval in slots was decided by the following decision rule:

$$\begin{aligned} M &= 2 \quad \text{if } TH_0 \leqq Z_n \leqq TH_1, \\ M &= 3 \quad \text{if } TH_1 \leqq Z_n < TH_2, \\ M &= 4 \quad \text{if } TH_2 \leqq Z_n \leqq TH_3. \end{aligned} \tag{55}$$

Fig. 9 shows the overflow probability when flow control is applied to the input voice traffic according to the buffer status. For the case of N=5, M=3,4, the threshold TH_2 was chosen to be $Q_v - 2$ in packets, so that we have m=3 for $Z_n \leq TH_2$ and M=4 for $TH_2 < Z_n \leq Q_v$. We can see that the overflow probability with this flow control scheme lies between the two overflow probability curves for the cases of (M=5, M=3) and (M=5, M 4)

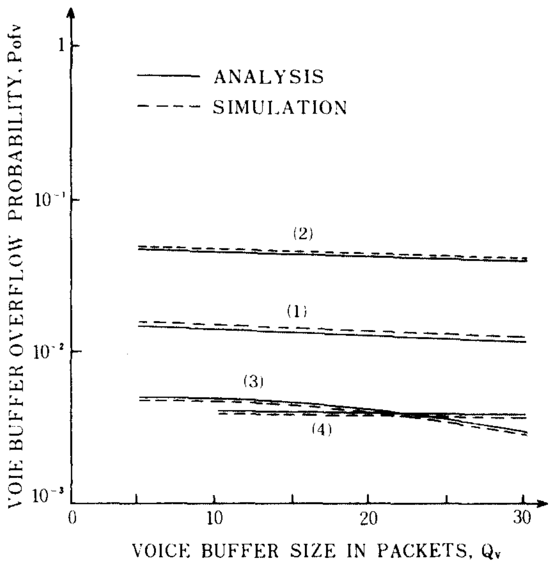


Fig. 9 Voice buffer overflow probability versus voice buffer size (Q_v) with N, M and threshold as parameters in packet voice multiplexer when flow control is done (T_s = 10ms).
(1) N = 4, M = 2,3, $TH_1 = Q_v - 2$, $TH_2 = Q_v$, G = 2 ~1.3
(2) N = 5, M = 2,3, $TH_2 = Q_v - 2$, $TH_2 = Q_v$, G = 2.5~1.7
(3) N = 5, M = 2,3,4, $TH_1 = Q_v - 5$, $TH_2 = Q_v - 3$, $TH_3 = Q_v$, G = 2.5~1.25
(4) N = 5, M = 3, 4, $TH_2 = Q_v - 2$, $TH_3 = Q_v$, G = 1.7~1.25

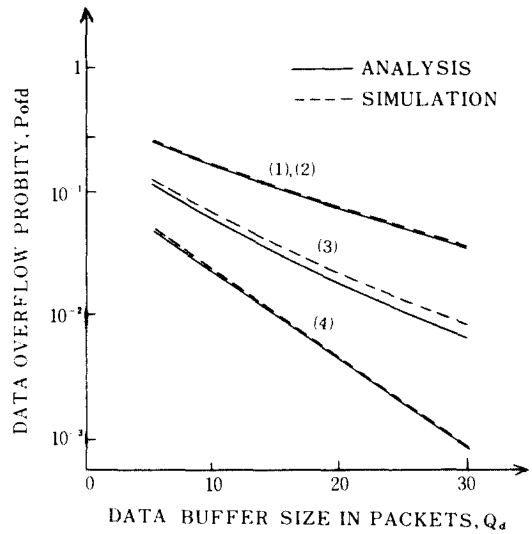


Fig 10 Data buffer overflow probability versus data buffer size with the number of voice users and packet generation interval of voice as parameters. (T_s = 10ms, B_D = 0.1 and Q_v = 5).
(1) N = 3, M = 2 (2) N = 5, M = 3
(3) N = 4, M = 3 (4) N = 5, M = 4

[see Fig. 8] without flow control. And overflow probability is changed slightly as the allowable buffer size increases.

From the above discussion, we can conclude that the performance of the packet voice multiplexer with (N=5 and M=3) for $Z_n \leq TH_2$ and M=4 for $Z_n > TH_2$ is close to that of (N=5, M=4) in overflow probability, thus indicating that the packet voice multiplexer with flow control is superior to that without flow control.

Let us now discuss the computer simulation and numerical results for the performances of the packet data multiplexer under voice interrupt. First, let us discuss the generation of data packets. We note that the probability that d data packets arrive in a slot time is given by (40), and the sum of $P_Y(d)$ for all d's is one. We can calculate $P_Y(d)$ using (40) and then find the cdf of $P_Y(d)$.

Fig. 10 shows the data overflow probability as a function of data buffer size with the number of voice users, the packet generation interval of voice and the voice buffer size as parameters. Compraring the curves (2) and (3), we observe that for the same output link capacity the overflow probability of data traffic decreases as the number of voice user decreases. Comparing now the curves (2) and (4), we see that for the same number of voice users the data overflow probability decreases as the link capacity increases.

The average queueing delay of data packets as a function of data buffer size with the traffice load of $\beta_d = 0.1$ is shown in Fig. 11. Note that the average queueing delay of data traffic is longer than the maximum data buffer size when the voice parameter values are N=3, M=2 and Q_V=5. This is due to the fact that the incoming data packets should wait before getting the channel service until all the voice packets in the voice queue are

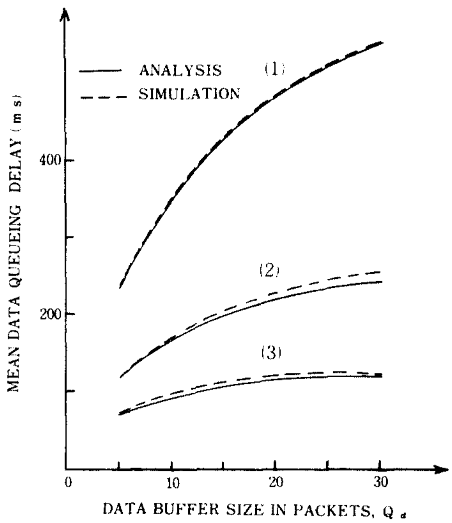


Fig. 11 Queueing delay of data packet versus data buffer size Q_d with the number of voice users and packet generation interval of voice as paramete s when no flow control is done (T_s = 10 ms, β_d = 0.1 and Q_v = 5).
(1) N = 3, M = 2 (2) N = 4, M = 3 (3) N = 5, M = 4

transmitted out. We observe in Fig. 10 that the overflow probability is about 10^{-3} for the case of Q_d = 30, N = 5, M = 4, Q_v = 5 and β_d = 0.1 In this case, the average queueing delay of data traffic is about 100 ms.

7. CONCLUSIONS

In this paper, extensive studies have been done on the performance of a packet voice/data multiplexer. We analyzed its performance in discrete time domain. For analysis we divided the output link into a sequence of time slots. We modeled the on-off pattern of voice with a two-state Markov process, and the packet generation interval with an (M+1)-state Markov process. Then, we formulated a recursive equation for a joint pdf of the input state and the queue size using the two-dimensional Markov chain model, and solved it for the pdf numerically by the Gauss-Seidel iteration method. With the joint pdf obtained, we investigated the steady state queueing behavior of the packet voice multiplexer. It has been found that the use of flow control for the input traffic is much desirable to utilize the given channel bandwidth efficiently and to transmit voice in relatively good quality with low overflow probability.

We next studied the queueing behavior of the packet data multiplexer under voice interrupt. We analyzed the queueing behavior of the multiplexer with a three-dimensional Markov chain model using the Gauss-Seidel iteration method.

To prove the validity of analysis, we compared the analytical results with those obtained by simulation. We obtained the numerical results for overflow probability. These results show that there exist tradeoffs among the number of users, TASI gain, overflow probability for voice traffic, and tradeoffs among traffic load, overflow probability and delay for data traffic.

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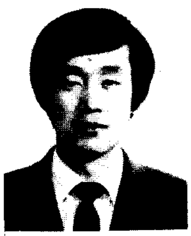

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