論 文

음성과 데이타가 집적된 Cut-Through 교환망의 성능 분석

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Performance Analysis of an Integranted Voice / DataCut-Through Switching Network

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要 約 본 논문에서는 cut-through 스위칭 방식으로 음성과 데이타를 집적하였을때, 그 성능을 분석하고 패킷 스위칭 방식과 비교하였다. 우선 음성과 데이타 패킷이 네트웍 내의 한 node를 cut-through 할 확률을 유도하고, 각각의 네트웍 delay의 Laplace transform을 구했다. 분석결과로 부터 cut-through 스위칭 방식이 패킷 스위칭 방식보다 성능이 우수함을 알 수 있다.

ABSTRACT In this paper, the performance of an integrated voice / data cut-through switching network is studied. We first derive cut-through probabilities of voice and data packets at intermediate nodes. Then, the Laplace transform for the network delay is obtained. According to numerical results, the performance of cut-through switching is superior to that of packet switching for integrated voice / data networks.

I. Introduction

The integration of packet voice with data in a common packet-switched network has received considerable attention due to its economic and technical benefits [1][2]. But, a serious problem of voice communication on a store-and-forward packet network is large end-to-end variable network delay [3][4]. This delay can be reduced by applying simplified protocols to packet voice and giving a priority service to voice packets over data. Nevertheless, it is still too large in multi-hop transmission. To reduce the end-to-end network delay, cut-through switching (CTS) was proposed which appears to be a promising switching method. In a cut-through switching system,

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when the output channel is free, packets are transmitted as soon as their headers arrive. Hence, CTS will singificantly reduce end-to-end delay in multi-node conections [5][7]. So far, the CTS has been studied only for data packets.

In this paper, the performance of an integrated packet voice / data cut-through switching network is investigated when nonpreemptive priority is given to voice packets. First, cut-through probabilities of voice and data packets are derived by a new approach. Then, the Laplace transform for the network delay is obtained. In addition, the average network delays of voice and data packets in the CTS network are determined, and compared with those in the classical packet-switched network.

II. Analysis of cut-through probabilities

The header of a packet normally contains the information on routing and addressing. Hence, if the output channel is not busy, the incoming packet may be transmitted toward the next node, although only a small portion of the packet has been received.

In cut-through switching, there are two modes of cuts, full cut and partial cut. A full cut occurs when the output channel is idle at the arrival of a packet header. On the other hand, even though the output channel is not idle at the arrival of a packet header, the packet is transmitted to the next node before it is received completely, if no other packets exist in the queueing buffer. When this happens, the packet is said to have made a partial cut.

To derive cut-through probabilities of voice and data packets at intermediate nodes, we make the following assumptions:

i) Voice packets have non-preemptive priority over data.

ii) The arrival process of voice and data packets at each node is modeled by the Poisson process.

iii) The transmission time of voice and data packets, b_V and b_d, are fixed in the network.*

iv) The length of a data pcket is longer than that of a voice packet and shorter than two times that of a voice packet.

From the above assumptions, we can use the head-of-the line(HOL) M / D / 1 queue as our analytic model.

In order to obtain the prabability of making cut-through, we must know the queue size of the node and the remaining service time of the packet being served at the arrival of the packet header. For analysis, we assume that the queue size and the remaining service time are independent. Now, let us define P_{kl} as the probability that the node has k voice packets and l data packets including the packet in the service facility (output channel) when the header of a pcket arrives. Then, P_{kl} can be found by using the probability generating function $\Pi(z_V, z_d)$ for HOL M / D / 1 queues [8]:

$$\Pi(z_v, z_d) = \sum_{k=0}^{\infty} \sum_{l=0}^{\infty} P_{kl} z_v^k z_d^l,$$

$$P_{kl} = \frac{1}{k!\,l!} \frac{\partial^{k+l}}{\partial z_v^k \partial z_d^l} \Pi(z_v, z_d) \mid z_v = z_d = 0 \tag{1}$$

In addition, from the busy period analysis, the Laplace transforms of waiting time of voice and data packets are given, respectively, as

$$W_v(s) = \frac{(1-\rho_v)s + \lambda_d(1-e^{-b_d s})}{s - \lambda_v + \lambda_v e^{-b_v s}} \tag{2}$$

and

$$W_d(s) = (1-p)\frac{s+\lambda_v-\lambda_v B_v(s)}{s-\lambda_d+\lambda_d B_d(s)} \qquad (3)$$

where λ_v and λ_d are arrival rates of voice and data packets, respectively, $p(=p_v+p_d)$ is the total traffic intensity and

$$B_v(s)=\exp\{-b_v[s+\lambda_v(1-B_v(s))]\},$$

$$B_d(s)=\exp\{-b_d[s+\lambda_v(1-B_v(s))]\}.$$

Let V_{fc} be defined as the probability that full cuts occur when voice packets arrive. We note that this is the same as the probability that the output channel is idle. Hence, we have

$$V_{fc}=P_{00}=1-p \qquad (4)$$

The probability of a partial cut for a voice packet, V_p, can be divided into two events which are mutually exclusive:

A: a partial cut of a voice packet after the departure of a voice packet

B: a partial cut of a voice packet after the departure of a data packet.

Each of those events is illustrated in Fig. 1 by using timing diagrams. Form Fig. 1(a), one can see that if the voice packet v_m makes a partial cut, v_m must arrive after v_{m-1} enters the server and v_{m-1} must depart before v_m is received completely. Thus, we have

$$V_{pv}=\Pr\{A\}$$
$$=\Pr\{\text{no arrival while } v_{m-1} \text{ waits for}\}\cdot\Pr\{\alpha_v b_v \le t_v \le b_v\}$$
$$=W_v(\lambda_v)\frac{e^{p_v(1-\alpha_v)}-1}{e^{p_v}} \qquad (5)$$

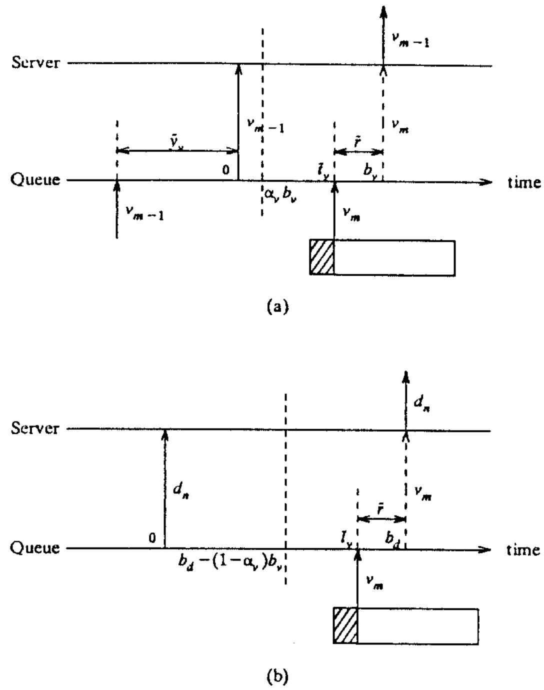

Fig. 1 Timing diagram for partial cut of voice packet.
(a) Partial cut of the *m* th arrived voice packet after the departure of the *m* −1 th arrived voice packet
(b) Partial cut of the *m* th arrived voice packet after the departure of the *n* th arrived data packet.

where t_v is the arrival time of v_m and $\alpha_v$is the relatiue header size of a voice packet. On the other hand, the probability of event B can be derived in a different procedure. In Fig. 1(b), Pr{B} is the probability that a cut is made by voice packet v_m which has arrived while a data packet is being served. Thus, we can express it as

$$V_{pd}=\Pr\{B\}$$
$$=\sum_{l=1}^{\infty}P_{0l}\cdot\Pr\{b_d-(1-\alpha_v)b_v\le t_v\le b_d \mid v_m$$
$$\text{arrived at interval } [0, b_d]\} \qquad (6)$$

Consequently, the cut-through probability of voice packet, V_c, is given by

$$V_c = V_{fc} + V_{pv} + V_{pd}$$

$$= (1-p)\,e^{\rho_v(1-\alpha_v)} + \frac{\lambda_d}{\lambda_v}\{e^{\rho_v(1-\alpha_v)} - 1\} \quad (7)$$

The cut-through probability of data packet, D_c, can be derived similarly. However, the procedure is more complicated since the length of a data packet is longer than that of a voice packet and the priority is given to voice packets. Let us define two events, C and D, as the following:

C: a partial cut of a data packet after the departure of a data packet

D: a partial cut of a data packet after the departure of a voice packet.

Then, we have

$$D_{pd} = \Pr\{C\} = W_d(\lambda_d)\frac{e^{\rho_d(1-\alpha_d)} - 1}{e^{(\lambda_v+\lambda_d)b_d}} \quad (8)$$

where α_d is the relative header size of a data packet. Because the length of a data packet is longer than that of a voice packet, the event D can be divided into the following three subevents as D_1, D_2 and D_3:

D_1: a partial cut of a data packet after the departure of one voice packet

D_2: a partial cut of a data packet after the departure of two voice packets

D_3: a partial cut of a data packet after the departure of one data and one voice packet

Consequently, we have

$$D_{pv} = \Pr\{D\} = \Pr\{D_1\} + \Pr\{D_2\} + \Pr\{D_3\} \quad (9)$$

Hence, the cut-through probability of data packet is given by

$$D_c = (1-p) + D_{pv} + D_{pd} \quad (10)$$

Ⅲ. Analysis of network delay

Although voice and data can be represented in compatible formats, their performance requirements are very different. Voice is normally tolerant of some channel errors because of the redundancy in speech. Hence, error control is not normally required. For data communication, however, error-free transmission is essentially required.

The error control in a packet-switched network is very simple because data are transmitted by means of store-and-forwarding. But, in a cut-through switching network, it is very complicated. The reason is that erroneous packets can be transmitted to the next node before error is detected.

In cut-through switching, two error-checking schemes can be considered. One is that a packet error is checked only at store-and-forward nodes. In the other scheme, the error control is the same as in the classical packet-switched network. In other words, if a packet error is detected at any node, a retransmission request signal is sent backward to the previous node. These have been studied by Shin and Un[7].

To analyze the performance of an integrated packet voice / data cut-through switching network, we consider a tandem network. It has been reported that the analysis of a tandem network is not easy since the assumption of Poisson arrival at intermediate nodes is not valid. However, if the network topology is quite complex, the independence assumption can be generally used. Hence, in this paper we use the independence assumption, and the packet arrival process is assumed to be Poisson at every intermediate node. Furthermore, we shall assume that the network is balanced, that is, the traffic intensity for each channel is almost the same.

A. Effective arrival rate of data packets

Before analysis, let us define $D_{cc}(n,k)$ as the probability that a data packet makes $k-1$ continuous cuts and is finally stored at the k th node when there are n tandem nodes to the destination. Then, it is given by

$$D_{cc}(n,k)=\begin{cases}(D_c')^{k-1}(1-D_c') & \text{if } 1\leq k\leq n-1\\ (D_c')^{k-1} & \text{if } k=n\end{cases} \tag{11}$$

where D'_c is the cut-through probability of a data packet when the effective arrival rate is used. Note that D'_c is recursively obtained by the effective arrival rate of data packets.

Error checking at store-and-forward node: Let $N_s^{(n)}$ be the average number of nodes for a data packet to have traversed when the actual number of nodes is n. Then, we have

$$N_s^{(n)}=\frac{\sum_{k=1}^{n}D_{cc}(n,k)\,[k+(1-P_e)^k\cdot N_s(n-k)]}{\sum_{k=1}^{n}D_{cc}(n,k)\,(1-P_e)^k}$$

Consequently, the effective arrival rate of data packets, $\lambda'_{d,s}$, is given by

$$\lambda'_{d,s}=\lambda_d\cdot\frac{N_s^{(n)}}{n}. \tag{12}$$

Error checking at every node: When errors of data packets are checked at checked at every node, the average number of nodes visited by a data packet, $N_e^{(n)}$, is given by

$$N_e^{(n)}=n+\frac{P_e}{1-P_e}\sum_{k=1}^{n}k[n-(k-1)]D_{cc}(n,k).$$

Hence, the effective arrival rate of data packets, $\lambda'_{d,e}$, is given by

$$\lambda'_{d,e}=\lambda_d\cdot\frac{N_e^{(n)}}{n}. \tag{13}$$

B. Network delay of voice packet

The Laplace transform for network delay of a vocie packet in the packet switching network, $T_v^{(n)}(s)$, is given by

$$T_v^{(n)}(s)=[W_v(s)e^{-b_v s}]^n \tag{14}$$

where $W_v(s)$ is the Laplace transform of waiting time of voice packet. In the cut-through switching network, the arrival of a header can be regarded as the arrival of a packet. Thus, the transmission time of the packet is effectively the same as that of a header. Hence, the Laplace transform for network delay is

$$T_{v,c}^{(n)}(s)=[W_v(s)e^{-\alpha_v b_v s}]^n e^{-(1-\alpha_v)b_v s} \tag{15}$$

Also, the average network delays of a voice packet in packet switching and cut-through switching networks, $T_v^{(n)}$ and $T_v^{(n)}{}_{,c}$, are given, respectively, by

$$T_v^{(n)} = -\frac{d}{ds} T_v^{(n)}(s) \Big|_{s=0}, \tag{16a}$$

$$T_{v,c}^{(n)} = -\frac{d}{ds} T_{v,c}^{(n)}(s) \Big|_{s=0}. \tag{16b}$$

C. Network delay of data packet

In a packet switching network, a packet error is checked at every node. Thus, the effective arrival rate is given by

$$\lambda'_d = \frac{\lambda_d}{1-P_e}. \tag{17}$$

Let $T_d^{(n)}(s)$ be defined as the Laplace transform of network delay of a data packet in n tandem queues. Then, we have

$$T_d^{(n)}(s) = \frac{(1-P_e)T_d^{(n-1)}(s)\,W_d(s)\,e^{-b_d s}}{1-P_e T_b(s)\,W_b(s)\,e^{-b_d s}} \tag{18}$$

where $T_b(s)$ is the Laplace transform of transmission time of negative acknowledgement signal and $W_b(s)$ is the Laplace transform for waiting time at a node before retransmission. Also, the average network delay is given by

$$T_d^{(n)} = -\frac{d}{ds} T_d^{(n)}(s) \Big|_{s=0} \tag{19}$$

Error checking at store-and-forward node: Let $T_{sl}^{(n)}(s)$ be the Laplace transform of time delay form the received instant of a header when a data packet is transmitted from the intermediate stroe-and-forward node. Then, we get the following relations:

$$T_{si}^{(n)}(s \mid n_{cc}=k-1 \text{ with no error}) = T_{si}^{(n-k)}(s)\,T_{dnc}(s)\,[T_{dc}(s)]^{k-1}$$

$$T_{si}^{(n)}(s \mid n_{cc}=k-1 \text{ with error}) = T_{si}^{(n)}(s)\,[T_{dc}(s)]^{k-1}\,[T_b(s)]_k\,W_b(s)e^{-b_d s}$$

where n_{cc} is the number of continuous cuts made and

$$T_{dc}(s) = W_d(s \mid \text{cut})\, e^{-a_d b_d s},$$

$$T_{dnc}(s) = W_d(s \mid \text{no cut})\, e^{-a_d b_d s} = W_{dc}(s)\, e^{-a_d b_d s}$$

Combining the above equations, we have

$$T_{si}^{(n)}(s) = \frac{\sum_{k=1}^{n} D_{cc}(n,k)\,(1-P_e)^k [T_{dc}(s)]^{k-1} T_{dnc}}{1-\sum_{k=1}^{n} D_{cc}(n,k)[1-(1-P_e)^k]\,[T_{dc}(s)]} \frac{(s)T_{si}^{(n-k)}(s)}{^{k-1}[T_b(s)]^k W_b(s)e^{-b_d s}}$$

Let $T_{ss}^{(n)}(s)$ be the Laplace transform of time delay when a data packet is transmitted from the source node. Then, we have

$$T_{ss}^{(n)}(s) = T_{si}^{(n)}(s)\,W_d(s)/W_{dnc}(s).$$

Finally, the Laplace transform of network delay can be obtained by including the time duration required for receiving the body of the packet at the destination node:

$$T_s^{(n)}(s) = T_{ss}^{(n)}(s)e^{-(1-a_d)b_d s} \tag{20}$$

And the average network delay is given by

$$T_s^{(n)} = -\frac{d}{ds} T_s^{(n)}(s) \Big|_{s=0} \tag{21}$$

Error checking at every node: In this case, the procedure is similar to that of a packet switched network. Let $T_{es}^{(n)}(s)$ be the Laplace transform of time delay form the source node. Then, we obtain

$$T_{es}^{(n)}(s\,|\,\text{no error}) = T_{es}^{(n-1)}(s)\,W_d(s)\,e^{-\alpha_d b_d s},$$

$$T_{es}^{(n)}(s\,|\,\text{error}) = T_{es}^{(n)}(s)\,T_b(s)\,W_b(s)\,e^{-b_d s}$$

Hence, we have

$$T_{es}^{(n)}(s) = \frac{(1-P_e)\,W_d(s)\,e^{-\alpha_d b_d s}\,T_{es}^{(n-1)}(s)}{1-P_e T_b(s)\,W_b(s)\,e^{-b_d s}}$$

Therfore, the Laplace transform of network delay is

$$T_e^{(n)}(s) = T_{es}^{(n)}(s)\,e^{-(1-\alpha_d)\,b_d s} \qquad (22)$$

Also, the average network delay is given by

$$T_e^{(n)} = -\frac{d}{ds}\,T_e^{(n)}(s)\,|_{s=0} \qquad (23)$$

Ⅳ. Simulation and Numerical results

To get simulation and numerical results, we chose parameter values as the following:

- Vl_p (voice packet length): 500bits
- Vl_h (vocie header length): 50bits
- Dl_p (data packet length): 1000bits
- Dl_h (data header length): 100bits
- P_e (probability of channel errors): 0, 0.01, 0.1

Fig. 2 shows cut-through probabilities of voice packets. It is noted from this figure that the analytic result is in good agreement with simulation. The cut-through probability of data packets is shown in Fig.3. In this case, the result is slightly less accurate, but the discrepancy is within 5%.

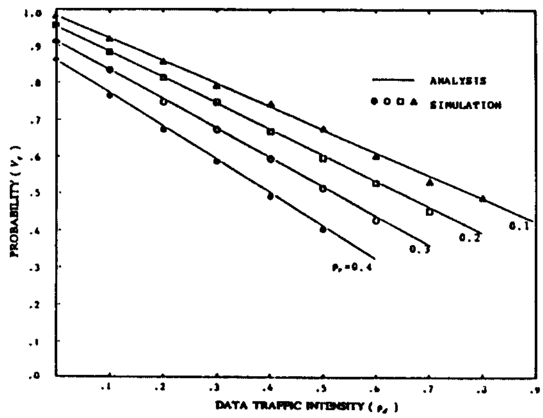

Fig. 2 Cut-through probability of voice packet. (Vl_p=500bits, Vl_h=50bits, Dl_p=1000bits)

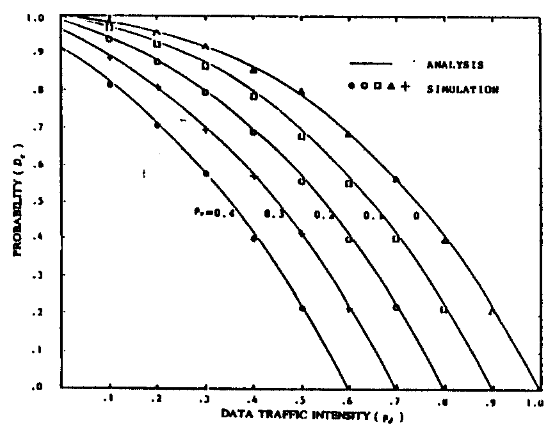

Fig. 3 Cut-through probability of data packet (Vl_p=500bits, Dl_p=1000bits, Dl_h=100bits)

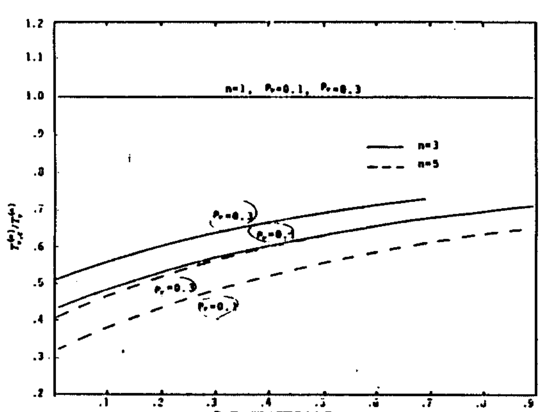

Fig. 4 Ratio of network delay of voice packets in cut-through switching and that in packet switching

Fig. 4 shows ratio of average network delay of voice packets in a cut-through switching network and in a packet switching network with n and p_v as parameter values. It is seen that, for the same number of nodes n, this ratio increases as the voice traffic intensity increacses, and for the same voice traffic intensity, the ratio decreases as the number of nodes increases. Note that, except for the case of $n=1$, this ratio is always smaller than unity. Hence, it is seen that the cut-through switching method is superior in delay characteristics to the classical packet switching method. Fig. 5 shows the ratio of average network delay of data packets in a cut-through switching network with error checking at store-and -forward nodes and in a packet switching network for the case of $n=3$. It is

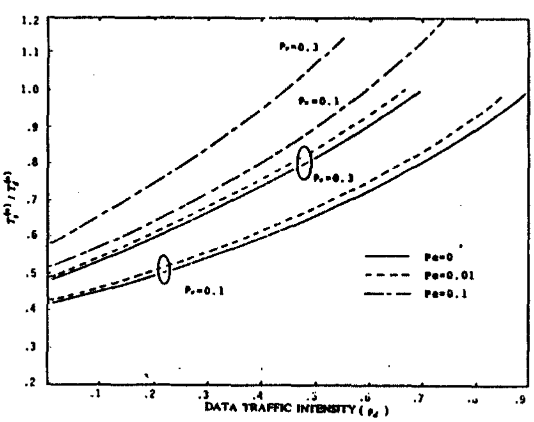

Fig. 5 Ratio of network delay of data packets in cut-through switching with error checking at store-and-forward nodes and that in packet switching

seen from the figure that the ratio increases as P_e and p_v increase.

Finally, Fig. 6 shows delay performance when a packet error is checked at every node. In this figure, the result is similar to that of error checking at store-and-forward nodes. But, in this schem, the effect of channel errors on the performance is smaller than that in the scheme in which error is checked at store-and-forward nodes.

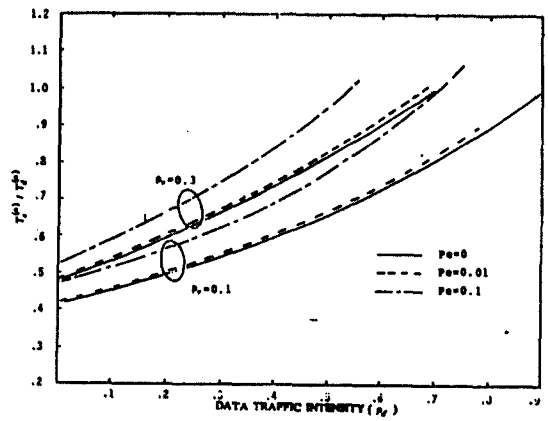

Fig. 6 Ratio of network delay of data packets in cut-through switching with error checking at every node and that in packet switching

V. Conclusions

In this paper, we analyzed the performance of an integrated voice/data cut-through switching network when nonpreemptive priority is given to voice packets, and compared it with that of a conventional packet switching network. First, we derived cut-through probabilities of voice and data packets at intermediate nodes, respectively. For the case of voice packets, the analytic result appears to be in good agreement with simulation, and for the case of deta packets, it is slightly less accurate, but the discrepancy is very small, i.e., less than 5%.

Second, We investigated the performance of an integrated vocie/data cut-through switching network. We then determined the average network delay of data packet, and compared with the classical packet switching through numerical computaions. Numerical results have been presented in terms of various parameter values. These results have shown that the introduction of voice traffic in a cut-through switching results in the worsening of delay performance of data traffic. Nevertheless, the performance of cut-through switching appears to be superior to that of packet switching except when high error rate occurs. Furthermore, the result shows that voice communication in a cut-through switching network in much superior to that is a packet switching network.

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