Application-Level Jitter Reduction Scheme for Multimedia Communication over ATM-ABR Service

Naotoshi ADACHI[†a)], **Shoji KASAHARA**[†]**,** *and* **Yutaka TAKAHASHI**[††]**,** *Regular Members*

SUMMARY   The ATM-ABR service category provides minimum cell rate (MCR) guarantees and robust connections even with insufficient network resources. Recently proposed rate-management algorithms for supporting multimedia applications over ABR mainly aim at minimizing the cell loss and delay. However, jitter is also an important element of QoS for multimedia applications. In this paper, we focus our attention on the arrival point of the critical cell corresponding to the end of data packet and propose a simple cell scheduling scheme for source node to reduce the jitter on application level over the ATM-ABR service class. In our proposed method, critical cells are delayed intentionally and the packet stream at application level becomes smooth. We verify the effectiveness of our proposed algorithm by an analytical model and simulation. From those results, we find that our proposed scheduling algorithm is effective in reducing the application level jitter even when the tagged cell stream is transmitted along the path with multiple nodes.
key words: *ATM-ABR, video application, jitter reduction algorithm, multimedia communication*

1. Introduction

The Available Bit Rate (ABR) service class in ATM is designed for data transmission [4] and hence ABR mainly supports the cell loss ratio (CLR) guarantee. The mechanism for supporting CLR in ABR service category is based on the feedback control where the allowed cell rate (ACR) of source node is dynamically adjusted by the resource management (RM) cells which are feedback signal with the information of congestion state of ATM network. However, the ABR does not provide any other QoS guarantees such as cell transmission delay (CTD) and cell delay variation (CDV). Therefore the ordinary ABR service class is not sufficient for multimedia communication such as real-time video.

On the other hand, [11] and [12] proposed a design method with queue control function which is used for calculating bandwidth allocated to a source node. Using this queue control function, it is possible to control transmission delay and to achieve low CLR by adjusting the allowed cell rate (ACR) according to the queue length of the bottleneck switch along a path. The queue control function algorithm proposed in [11] and [12] is quite attractive since the ABR service category can support multimedia communica-

tion such as real-time video transmission with small delay. However, CDV, or equivalently, jitter is not taken into consideration in their algorithm. The jitter is also important element for the real-time video transmission where the jitter affects the quality of decoded video at destination node.

Recently, some dynamic encoding rate adjustment schemes have been proposed and studied [1]–[3]. In these schemes, the encoding rate of source node is dynamically adjusted according to the congestion state of the network. Therefore it is expected that the dynamic encoding-rate adjustment scheme is useful for ATM-ABR if ACR is used for the notification of congestion to the source node. Throughout the paper, we assume that the source node adjusts the data transmission rate dynamically according to the ACR. Note that this dynamic adjustment does not provide the perfect guarantee of small jitter at application level.

Here we consider how the jitter affects the QoS of multimedia application. The application data packet is segmented into ATM cells at ATM adaptation layer (AAL) of a source node and then those are sent into the network. The cells arriving at the destination node are reassembled into the application data at AAL. The QoS at the application level is largely affected by the time to finish reassembling the data packet from ATM cells. This finishing time is determined by arriving points of cells corresponding to the last part of data packet. Throughout the paper, we call the cells corresponding to the end part of the original data packet the critical cells, and define the application-level jitter as the standard deviation of inter-arrival time of critical cells at destination node. For the QoS guarantee at the application level, it is crucial to keep the application-level jitter as small as possible.

Note that, in general, time scale at application level is larger than that at cell level. For example, time scale of video frame processing is in the order of millisecond while cell transmission time is in the order of microsecond. Therefore if critical cells are synchronized, the resulting inter-transmission time of critical cells is constant and in the order of larger time scale than that of cell transmission time. This large time is likely to absorb the transient network congestion at cell level and this is expected to result in the jitter reduction at application level.

In this paper we propose a scheduling scheme at source node to reduce the jitter at application level over the ATM-ABR service class. In our proposed scheme, we focus on the departure points of critical cells. The critical cell is intentionally delayed until the next data packet generation and

transmitted at the beginning of the next cycle of packet generation. According to this algorithm, the departure points of critical cells at source node are like CBR traffic and therefore the reduction of jitter at application level is expected. Since the departure points of critical cells from source node are intentionally delayed, we call our proposed scheme intentionally delayed transmission (IDT). The strong point of IDT scheme is that we need not change the existing ATM facilities except the source node.

As for the previous researches of the jitter behavior in ATM networks, [5], [6] and [7] considered cell-level jitter. In [5] and [7], the authors considered the two types of traffic, tagged stream and background one. They analyzed the jitter process of tagged renewal stream in the case of single node. [6] analyzed the jitter process in multiple nodes case using results of [5] and [7]. We apply the jitter model in [5], [6] and [7] to our case, and analyze the jitter process of critical cells. We also verify the effectiveness of our proposed method by simulation.

This paper is organized as follows. In Sect. 2, we describe our proposed scheme in detail. In Sect. 3, we show the analytical model proposed in [5], [6] and [7] and apply it to our proposed scheme. In Sect. 4, we describe the simulation models for our proposed scheme and present some numerical results of analytical model and simulation. Finally, we conclude our paper in Sect. 5.

2. IDT Scheme

In this section, we describe the IDT scheme in detail. First we suppose that the application layer of a source node generates data packets for multimedia communication and that the interarrival time of consecutive packets is constant equal to T (Fig. 1). The period T is regarded as a cycle of packet generation. In addition, we assume that the application program generates at least one cell during each period T and that the number of cells generated within the period T is bounded according to the ACR as stated in Introduction. Finally we suppose that the maximum number of cells segmented from a data packet is bounded in terms of T. For example, if T cells can be transmitted during the period T, the maximum number of cells segmented from a data packet is T.

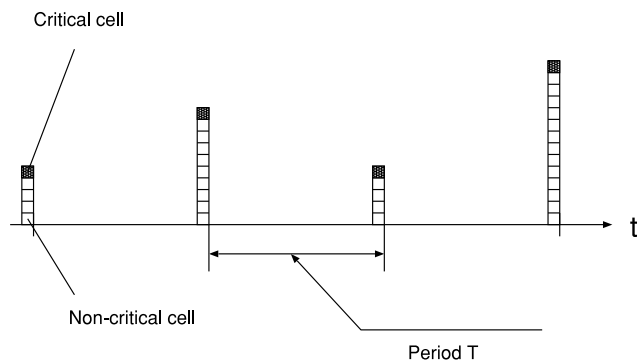In the case without IDT algorithm, a source node sends cells according to the transmission rate calculated with allocated ACR. Since the packet size at application level is variable, interdeparture time of critical cells varies depending on the packet size.

Our strategy of jitter reduction is as follows. The time to complete reassembling segmented cells into an original packet is determined by the arrival of the critical cell. Therefore we focus our attention on the departure points of critical cells at source node. In our proposed method, the critical cell is delayed until next data packet generation and transmitted at the beginning of the next cycle of packet generation. By this operation, the interdeparture time of critical cells is constant with period T and it is expected that the resulting interarrival time of critical cells at destination node varies less than that of ordinary ABR service (Fig. 2).

The procedure of IDT scheme is as follows.

1. The application data is segmented into ATM cells at AAL.
2. AAL tags critical cells.
3. All ATM cells are sent to the ATM Switch Layer through the ATM Layer.
4. In the ATM switch layer;
   a. Non-critical cells are transmitted to the network according to the ACR.
   b. The critical cell is not sent until next data packet generation. This critical cell is sent into the network when cells generated at the next packet generation enter the ATM switch layer.

There are two drawbacks in the IDT scheme. One is layer violation between AAL and ATM switch layer and the other is the additional delay critical cells suffer from at the ATM switch layer.

The layer violation is caused by the implementation requirement where the AAL tags critical cells and notifies to the ATM switch layer which cell is critical. However, this modification is required only at source nodes and we need not change other existing ATM facilities. It seems to be inevitable for supporting the QoS of real-time application such as audio and video over ATM-ABR service class.

As for the additional delay, note that the worst delay a critical cell suffers from at source node is at most T, which is the interarrival time of consecutive data packets. We can set T equal to the interval of video frames. In this case,
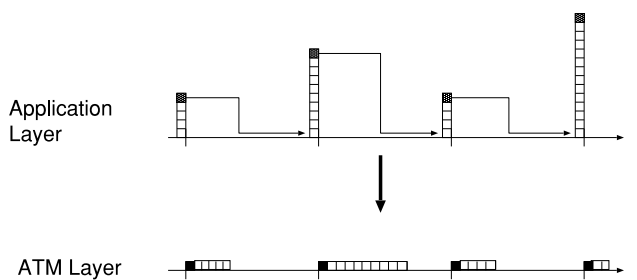

Fig. 1   Departure process at source node.

Fig. 2   IDT scheme.

the video is played with one frame delay at destination if cells does not suffer from any jitter in the network. Therefore the IDT scheme may not be appropriate for real-time applications such as remote control and high quality video conference, which require stringent QoS guarantee. However, this delay value is enough to support one-way session applications such as broadcast services [8].

Comparing the IDT with the system without IDT where the destination has a buffer large enough to absorb the jitter of critical cells, the later is more efficient than the former under the small jitter case since there is no additional delay in the system without IDT. However, as the jitter becomes larger, more buffer capacity is needed and we have to send more cells to absorb the large jitter with frame buffer. This results in the delay for playing video at destination. From the application point of view, application data experiences additional delay at the source node under IDT while this occurs at the destination node in the system without IDT. In Sect. 4.6, we will compare the mean delay of IDT system with that in the system where the jitter is absorbed in a destination buffer.

3. Jitter Process of Critical Cells

In this section, we summarize the analysis of jitter process studied in [5], [6] and [7], and apply their results to our proposed scheme.

We consider a discrete-time single-server queueing system with infinite buffer (Fig. 3). The time axis is segmented into a sequence of slots and one cell transmission time is equal to a slot. The cells are served according to FIFO discipline. Here we are interested in the jitter of critical cells at destination node. At source node, data packets are generated periodically with cycle T. Data packets are segmented into ATM cells which consist of a critical cell and non-critical cells. Let U denote the number of non-critical cells generated at the beginning of the cycle. We assume that the maximum number of cells generated from one packet is equal to T. Hence $0 \le U \le T - 1$. Let $u(k)$ $(0 \le k \le T - 1)$ denote the probability distribution function (pdf) of U and $U(z)$ the probability generating function (pgf) of $u(k)$.

Now we consider nth transmission cycle of a data packet. As described in the previous section, there are T slots between the consecutive packet generation points. The first slot of n-th transmission cycle is used for the critical cell which is the last cell of $n - 1$st data packet. Then $T - 1$
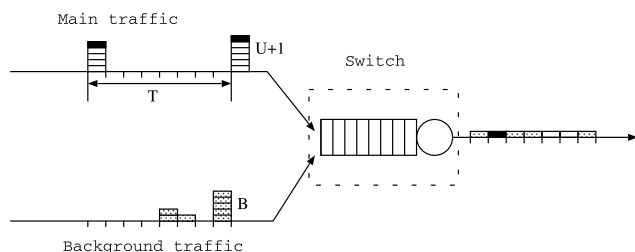

Fig. 3   Queueing model.

slots are used for non-critical cells and consecutive departure points conform to the allocated ACR. However, we assume for analytical simplicity that a non-critical cell is transmitted within a slot with probability p. Let $V(z)$ denote the pgf of the number of non-critical cells transmitted within a slot. Then we obtain

$$V(z) = 1 - p + pz. \tag{1}$$

Therefore $U(z)$ becomes

$$U(z) = V(z)^{T-1} = (1 - p + pz)^{T-1}. \tag{2}$$

Here we analyze the single node case where there is a switch node between source and destination. At the ATM switch, the tagged packet stream is multiplexed with background traffic. Let B denote the batch size of background traffic within a slot. We assume that B is independent and identically distributed (i.i.d.) and that its pdf and pgf are $b(k)$ $(k \ge 0)$ and $B(z)$, respectively.

[5] and [6] analyzed the jitter process in the case where the queue accepts two classes of cells, the GI class and the B class. GI represents the tagged cell stream of interest while B stands for background traffic. The GI class cells arrive at the queue with interarrival time I which is distributed according to a renewal process. It is assumed that the interarrival time I has a finite upper bound G_{max}. We denote pgf of the integer-valued random variable I by $G(z)$.

Let Q and $Q(z)$ denote the queue length at the arrival points of GI class cells and the pgf of Q, respectively. From [5], we obtain

$$Q(z) = G'(1)(1 - \rho_t)\frac{(1 - z^{-1})(B(z)/z)^K}{1 - zG(B(z)/z)}$$
$$\times \frac{\Pi_{k=1}^{K}[(z/B(z)) - (r_k/B(r_k))]}{\Pi_{k=1}^{K}[1 - (r_k/B(r_k))]}, \tag{3}$$

where $K = G_{max} - 2$ and

$$\rho_t = B'(1) + \frac{1}{G'(1)} \stackrel{\text{def}}{=} \rho + \rho_{GI}, \tag{4}$$

is the total offered load. It is also shown that if $\rho_t < 1$, the equation

$$1 - zG\left(\frac{B(z)}{z}\right) = 0, \tag{5}$$

has K roots inside the unit circle excluding 1. We denote these roots by $r_1, r_2, \cdots, r_K$. We define J as the interdeparture time of two successive GI-cells. We have

$$J \stackrel{\text{dst}}{=} Q_2 - Q_1 + I, \tag{6}$$

where Q_1 and Q_2 are the queue sizes seen by two consecutive GI class cells. From [5],

$$J(z) = \sum_{i=G_{min}}^{G_{max}} g_i J_i(z), \tag{7}$$

where

$$J_i(z) = z(B(z))^i + (B(z))^{i-1}(z-1)$$
$$\times \sum_{k=1}^{i-1} (z^{-1}B(z))^{-k}\Phi(z^{-1};k), \tag{8}$$

and

$$\Phi(z;k) = \sum_{l=0}^{k-1} z^l \pi_k^B(0;l)Pr(Q=l), \quad 1 \leq k \leq T-1. \tag{9}$$

The $\pi_k^B(0;l)$'s are obtained by the recursive algorithm described in [5] and the $Pr(Q=l)$'s are obtained by inverting (3).

In our IDT scheme, the interarrival time of GI class cells corresponding to critical cells is constant and equal to T. Hence

$$g_i = \begin{cases} 1, & i = T, \\ 0, & i \neq T. \end{cases} \tag{10}$$

The B class cells consist of non-critical cells and background traffic. Therefore the pgf of the number of B class cells within a slot is given by $V(z)B(z)$. Replacing $B(z)$ in (8) with $V(z)B(z)$ and using (10), the pgf of the interdeparture time of critical cells is given by

$$J(z) = z(V(z)B(z))^T + (V(z)B(z))^{T-1}(z-1)$$
$$\times \sum_{k=1}^{T-1} (z^{-1}V(z)B(z))^{-k}\Phi(z^{-1};k). \tag{11}$$

In the heavy traffic case, the total utilization ρ_t becomes close to 1. From [5] we obtain that

$$J(z) \rightarrow z(V(z)B(z))^T, \quad \text{when } \rho_t \rightarrow 1, \tag{12}$$

since $\Phi(.;.) \rightarrow 0$ as $\rho_t \rightarrow 1$.

On the other hand, from [7], as $T \rightarrow \infty$ the system behaves like a $Geo^{[X]}/D/1$ queue, that is

$$Q(z) \rightarrow (1-\rho_t)\frac{z-1}{z-V(z)B(z)}, \quad \text{as } T \rightarrow \infty. \tag{13}$$

This approximation yields

$$J(z) = E(z^{Q_2-Q_1+I}) = z^T Q(z)Q(z^{-1}). \tag{14}$$

We will show numerical examples calculated from (12) and (14) and compare the results with simulation one.

4. Performance Evaluation of IDT Scheme

In this section, we investigate the performance of our proposed scheme by both analytical results and simulation. First we present the simulation models and then show the numerical results in both single and multiple nodes cases.

4.1 Simulation Model

In the simulation experiment, we use OPNET version6.0 [14]. Figure 4 shows the block diagram of the source node in OPNET. We modified the AAL and ATM_switch blocks
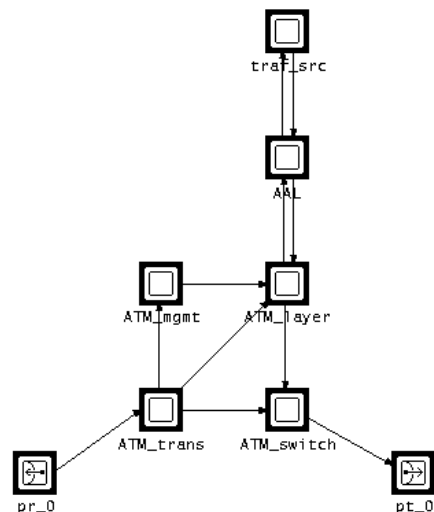

Fig. 4 Block diagram of source node in OPNET.

of the source node for implementing the IDT scheme. When the AAL block receives data packet from the traf_src block, the AAL block segments the packet into cells and marks the cell corresponding to the end of the packet (i.e. critical cell). The ATM_switch block receives the ATM cells through the ATM_layer block and adjusts the departure points of critical cells.

In this simulation model, the capacity of all links is equal to 155 Mbps and all connections are established over ABR service category. Since the multimedia application of interest is the real-time video application, we assume that the time between the consecutive points of packet generation is 1/30 sec. The number of slots corresponding to this interval is

$$T = \frac{155,000,000\,\text{(bps)}}{53\,\text{(byte)} \times 8\,\text{(bit)}} \times \frac{1}{30}\,\text{(sec)}$$
$$\simeq 12186\,\text{(slot)}. \tag{15}$$

We also assume that the bitrate of application data is 7.2 Mbps, which is a typical value of MPEG2 encoder [13]. From the assumption of (2), the number of cells for application packet length is distributed according to binomial distribution. The mean size of the application packet is given by

$$U'(z)|_{z=1} + 1 = (T-1)p + 1$$
$$= \frac{7,200,000\,\text{(bps)}}{53\,\text{(byte)} \times 8\,\text{(bit)} \times 30\,\text{(sec)}}$$
$$= 566\,\text{(cells)}. \tag{16}$$

Therefore p in (2) is set to $565/(12186-1) \simeq 0.04637$.

The number of cells for background traffic generated within a slot is distributed according to geometric distribution where its mean is set to 0.35 cell/slot (corresponding to 55 Mbps) 0.50 (77), 0.57 (88), 0.65 (100) and 0.71 (110), respectively.
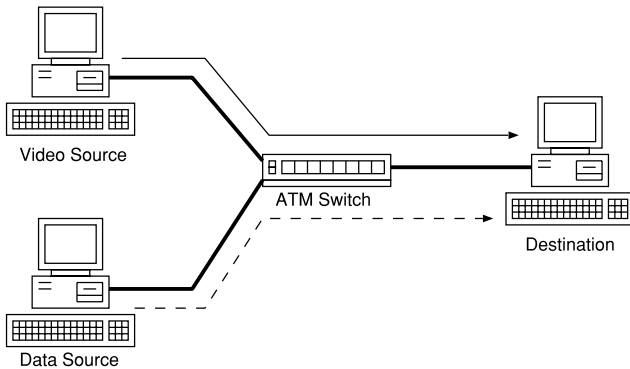
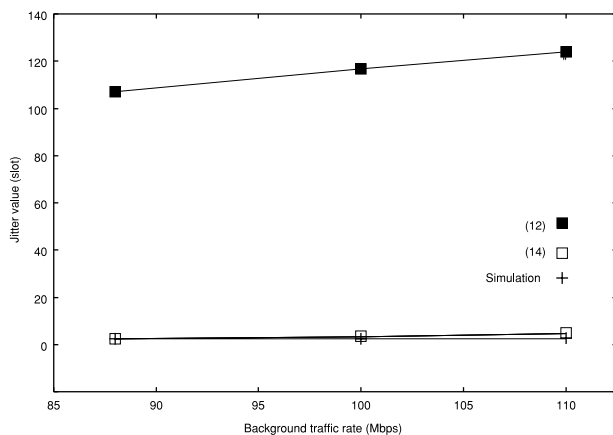**Fig. 5**   Simulation model for single node case.

Fig. 6   Jitter in single node case.

Table 1   Jitter values of simulation and approximation (14).

	Background Traffic (Mbps)		
	88	100	110
Simulation	0.580	0.728	0.739
Approximation (14)	2.37	3.36	4.51

Fig. 7   Jitter in single node case.

4.2   Single Node Case

We consider the network topology shown in Fig. 5 to investigate the jitter process in single node case. The critical and non-critical cells are generated at the Video Source as shown in Fig. 1 and are transmitted to the Destination. The background traffic cells are generated at Data Source and transmitted to the same destination. The IDT scheme is implemented at Video Source. In this case, these two streams are multiplexed at the output buffer of the ATM Switch and share the link between ATM Switch and Destination. In the simulation, we record the interarrival times of critical cells for Video Source at Destination and calculate the jitter, the standard deviation of the interarrival time. We also calculate the jitter values using approximations (12) and (14).

Figure 6 shows the jitter values calculated from (12), (14) and simulation. In Fig. 6, the horizontal axis represents the bitrate of background traffic and the vertical axis means the jitter value (*slot*). We observe that the jitter values of (14) and simulation are almost same while the result calculated from (12) is quite different. This implies that T is too large to calculate the jitter from (12). That is, T is large enough to consider the system as $Geo^{[X]}/D/1$ and hence (14) is more suitable than (12). In Table 1, we show the jitter values of (14) and simulation. Though the simulation re-
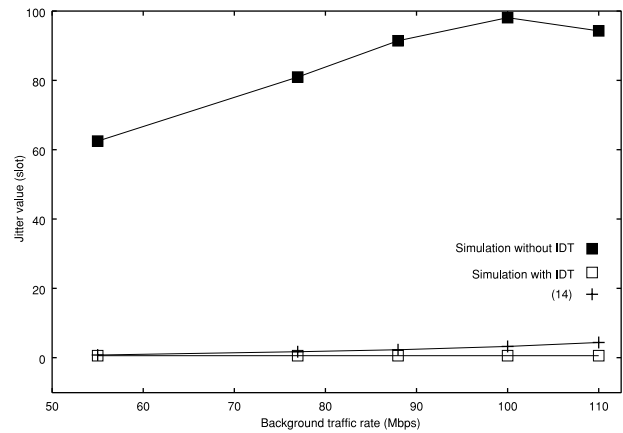
sults are not strictly equal to those of (14), the tendency to increase is same in both cases.

Figure 6 also shows that the jitter is small even when the rate of background traffic is large. To investigate the efficiency of the IDT scheme, we also plot the jitter values without IDT in Fig. 7. From this figure, we observe that the jitter value with IDT algorithm is smaller than that without IDT irrespective of the rate of background traffic. Intuitively, the number of cells between consecutive critical cells becomes large as background traffic increases. In the case without IDT, the interarrival time of critical cells at ATM switch varies according to the packet size at application level of the source node and this causes the large jitter at destination. However, in the case with IDT, the interarrival time of critical cells at ATM switch is constant and this makes the amount of background traffic less variable. Note that the mean interarrival time of critical cells at destination becomes large in both cases when background traffic increases.

One more important characteristic observed from Fig. 7 is that the jitter with IDT scheme is quite small and insensitive to background traffic. This implies that the output process of the ATM switch is also constant regardless of background traffic. Therefore we can expect that the jitter with IDT is small even in the multiple nodes case. We investigate this in the next subsection.

4.3   Multiple Nodes Case

In order to investigate the jitter behavior in the multiple nodes case, we consider the network topology shown in Fig. 8. In this case, there are three switch nodes in the network. The critical and non-critical cells are generated at the Video Source and are transmitted to the Destination. Data
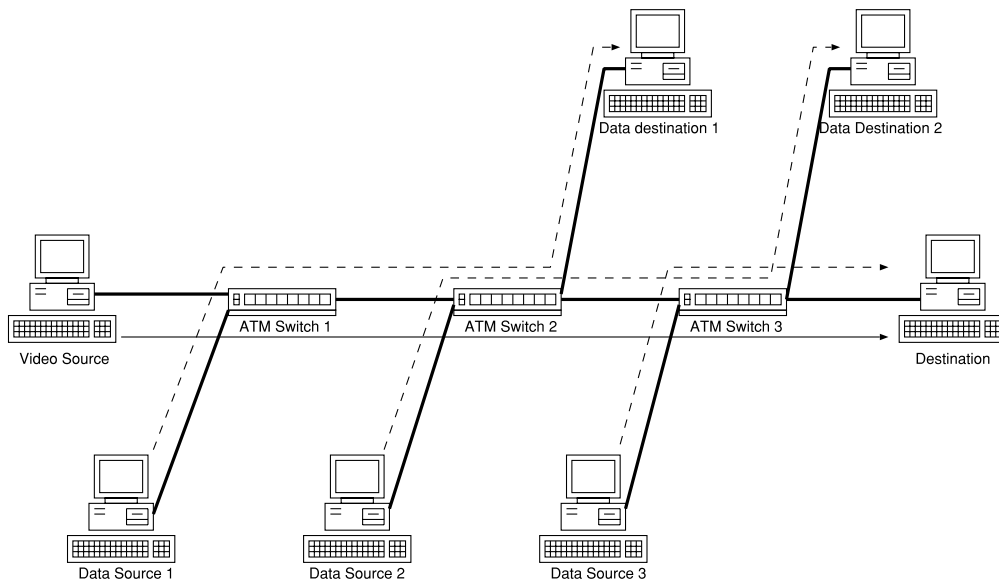
Fig. 8   Simulation model for multiple nodes case.

Table 2   Destination of each source in multiple nodes case.

source	Video Source	Data Source 1	Data Source 2	Data Source 3
destination	Destination	Data Destination 1	Data Destination 2	Destination

Table 3   Destination of each source in multiple video sources case.

source	Video Source 1	Video Source 2	Video Source 3	Data Source 1	Data Source 2
destination	Destination 1	Destination 2	Destination 3	Data Destination 1	Data Destination 2

Source 1, Data Source 2 and Data Source 3 generate background traffic and Data Destination 1, Data Destination 2 and Destination are the corresponding destinations, respectively (Table 2). The Destination is for Video Source and Data Source 3. In this case, traffic from Video Source is multiplexed with background traffic from Data Source 1 at the output buffer of ATM Switch 1. Then aggregated traffic is transmitted to ATM Switch 2 and background traffic from Data Source 1 is switched to Data Destination 1 while traffic from Video Source is multiplexed with another background traffic from Data Source 2. The same situation as the ATM Switch 2 occurs at ATM Switch 3 and finally aggregated traffic from Video Source and Data Source 3 is transmitted to Destination. We summarize the source-destination pairs in Table 3. As for the parameter of Video Source, we use the same values as the single node case and investigate the jitter value of the Video Source traffic when the cell-generation rate of background traffic is set to 50 and 100 (Mbps), respectively. Note that MPEG stream suffers from large jitter in this network model since MPEG stream is multiplexed with background traffic at each ATM switch.

As we stated in the previous subsection, we cannot use (12) due to large T. Though [6] provides the jitter analysis in the multiple nodes case, the main results are derived with (12) and hence we cannot use the results in [6]. Therefore we investigate the jitter behavior in multiple nodes by
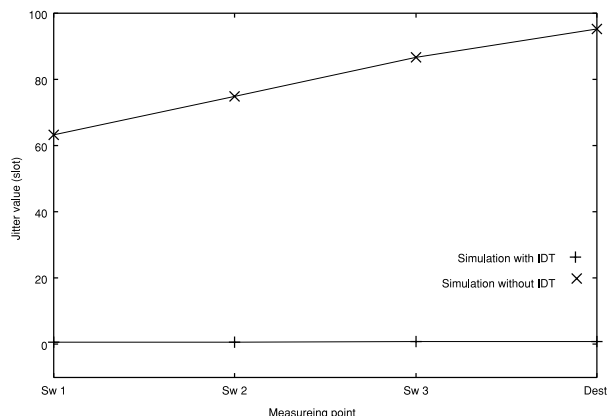

Fig. 9   Jitter in multiple nodes case with 50 Mbps background traffic.

simulation.

Figure 9 shows the simulation results with and without IDT scheme under 50 Mbps background traffic. In Fig. 9, the horizontal axis represents the measuring point of the jitter for critical cells, and vertical axis means the jitter value. From Fig. 9, we observe that the jitter value without IDT becomes large as the number of ATM switches increases and that it is always larger than that with IDT. We also observe that the jitter value with IDT algorithm is almost same even when the number of intermediate ATM switches increases.
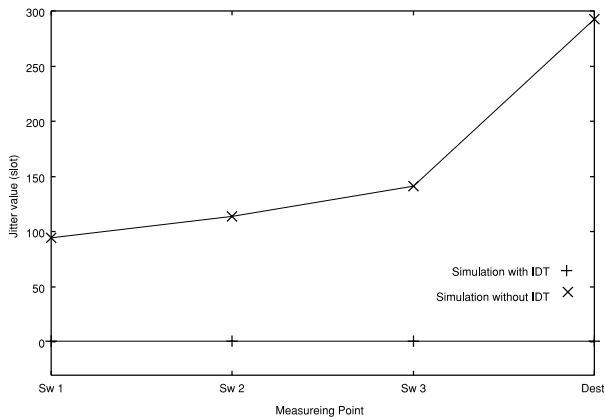
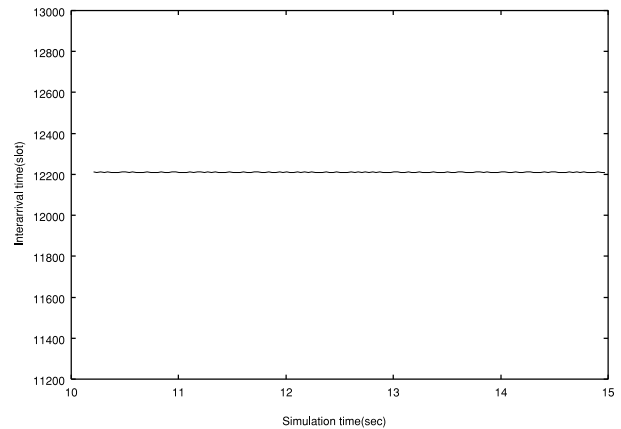**Fig. 10** Jitter in multiple nodes case with 100 Mbps background traffic.

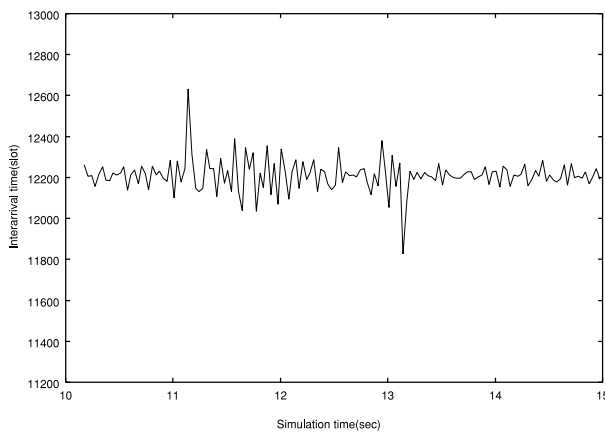**Fig. 12** Interarrival time with IDT in single node case.

Fig. 11 Interarrival time without IDT in single node case.

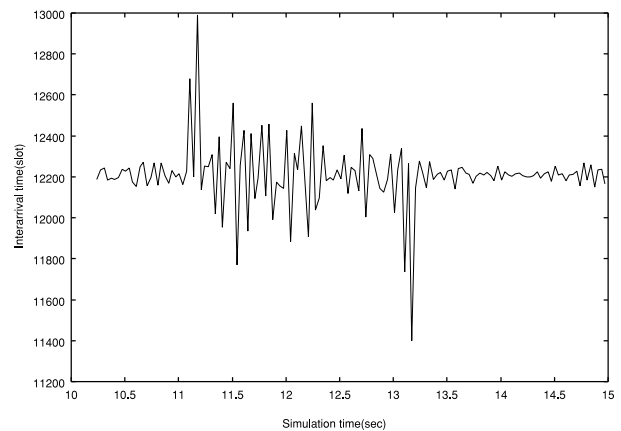**Fig. 13** Interarrival time without IDT in multiple nodes case.

Fig. 14 Interarrival time with IDT in multiple nodes case.

This is just what we expected in the previous subsection.

Figure 10 shows the simulation results with 100 Mbps background traffic. From this figure, we observe that the jitter value has the same tendency as the case with 50 Mbps background traffic. Note that the jitter values are quite larger than those under 50 Mbps case. In the case without IDT, the jitter rapidly increases at ATM switch 3. This implies that the variation of the interdeparture time between critical cells causes the large variation of the interdeparture time at next ATM switch. Therefore keeping the interdeparture time of critical cells constant is effective for reducing the jitter at destination. From this reason, the IDT scheme is efficient for assuring the QoS at application level.

4.4 Robustness of IDT Scheme

In order to investigate the robustness of IDT scheme against background traffic, we focus on the dynamics of the inter-arrival time of critical cells at destination. Figures 11 to 14 show the simulation results with and without IDT scheme in the following case: The video traffic is transmitted to the destination during the simulation time from 10 to 15, and the data source nodes start to transmission of 100 Mbps background traffic at 11 and end at 13. In these figures, the
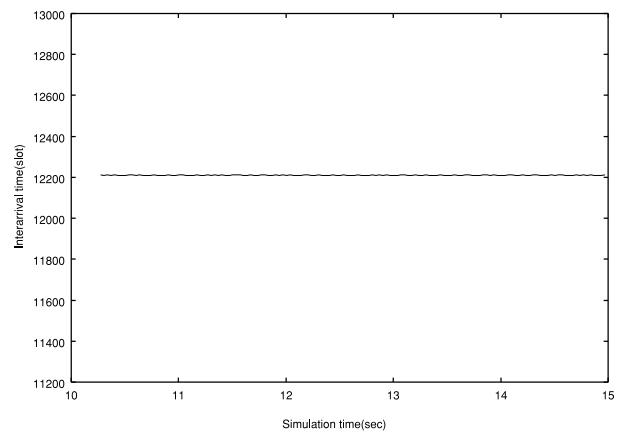
horizontal axis represents the simulation time and the vertical axis means the interarrival time of critical cells at destination. Figures 11 and 12 are the single node case while Figs. 13 and 14 are the multiple nodes case where the number of ATM switches is three.

From Fig. 11, we observe that the interarrival times vary largely when background traffic is multiplexed. We
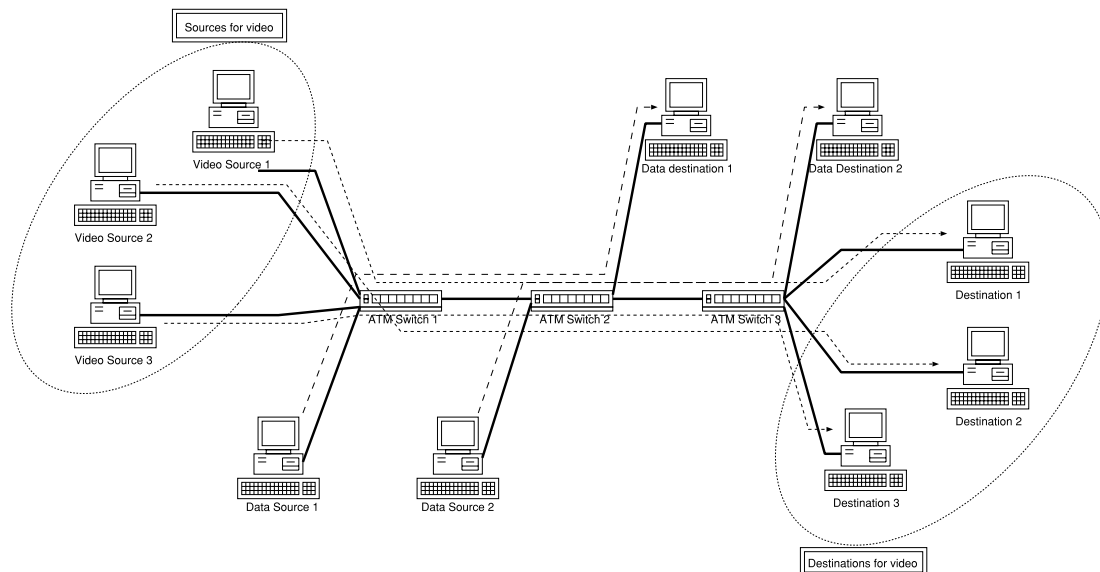
Fig. 15 Simulation model for multiple source case.

also observe that the interarrival times still vary even when there is no background traffic. This is because the packet size at application level is variable. On the other hand, from Fig. 12, the interarrival time is almost constant even when background traffic multiplexed.

Figure 13 shows the simulation result of the multiple nodes case without IDT scheme, and we find the same tendency as Fig. 11. Note that the degree of variation of interarrival time is larger than that of Fig. 11. Figure 14 is the case with IDT. We observe that the interarrival time is almost constant insensitive to background traffic. In addition, the number of intermediate nodes does not affect the variation of interarrival time so much.

From these results, we conclude that the IDT scheme is robust against the impact of background traffic.

4.5 Fairness of IDT Scheme

In this subsection, we investigate the performance of IDT scheme under multiple video sources case. We consider the network topology shown in Fig. 15. In this topology, Video sources 1, 2 and 3 send the data to Destinations 1, 2 and 3, respectively. At the same time, Data source 1 (2) sends 50 Mbps background traffic to Data Destination 1 (2). Figure 16 illustrates the simulation result of jitter for each video sources.

In Fig. 16, the horizontal axis represents the measuring point of the jitter for critical cells and vertical axis means the jitter value. From this figure, we find that the jitter value with IDT scheme is almost same while that without IDT is different from each other and becomes large as the number of ATM switches which critical cells pass through increases. This implies that the IDT scheme provides the fair jitter reduction among video sources.
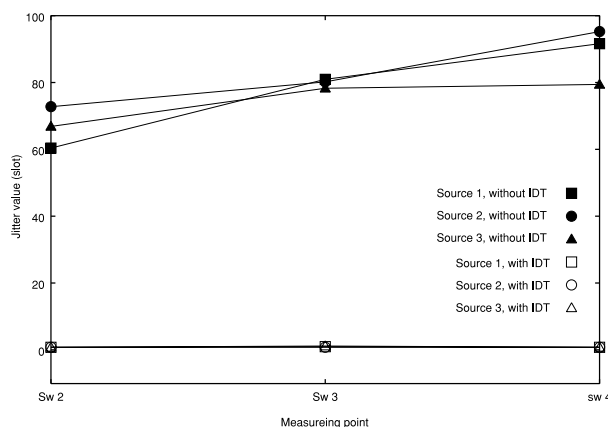

Fig. 16 Jitter in multiple sources case with 50 Mbps background traffic.

4.6 Mean Delay

Finally we investigate the mean end-to-end delay of data packet with IDT scheme in the network topology shown in Fig. 8. As a counterpart of the system with IDT, we also investigate the system without IDT where the destination has a frame buffer to absorb the jitter at application level. The end-to-end delay of critical cell is defined as the time from the point just before segmentation of data packet at source node to the point just after reassembling into the packet at destination. Note that the end-to-end delay includes queueing delay at destination for both systems with and without IDT and IDT processing time for IDT system.

In the case without IDT, the system avoids the jitter by introducing a frame buffer at destination. The frame buffer size is set to zero (no frame buffer) and two [9], respectively. Furthermore, we consider the case where the first packet arriving at destination is delayed in the frame buffer for the
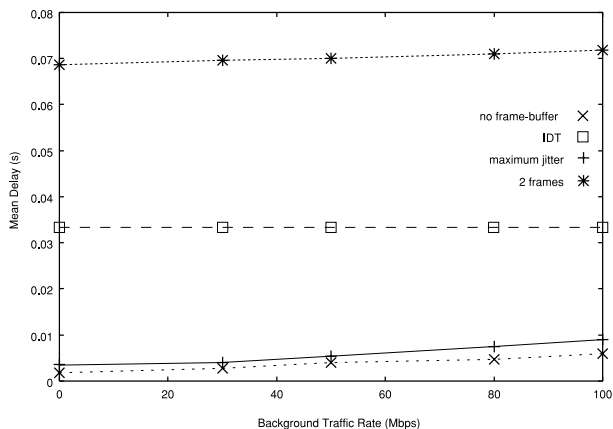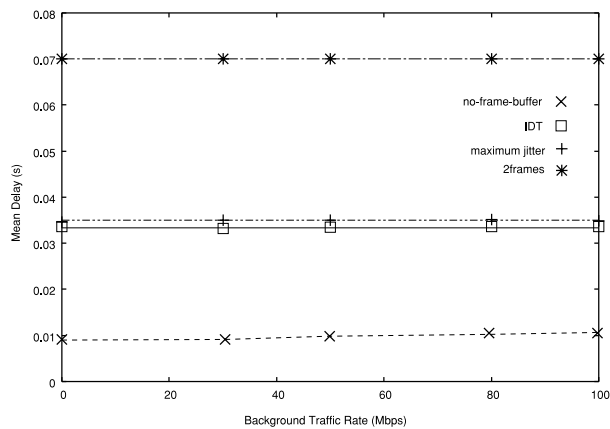
Fig. 17   Mean delay (Binominal).

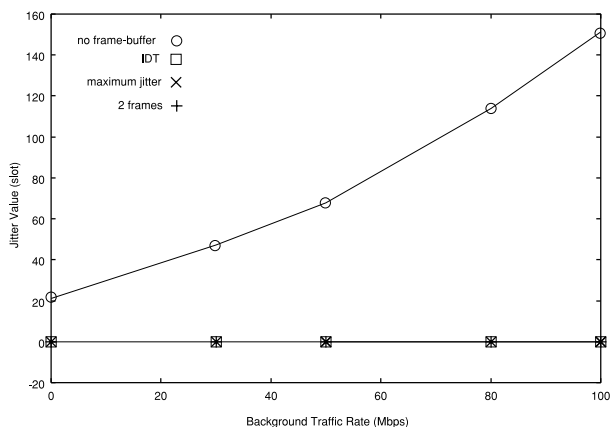
Fig. 19   Mean delay (MPEG movie).

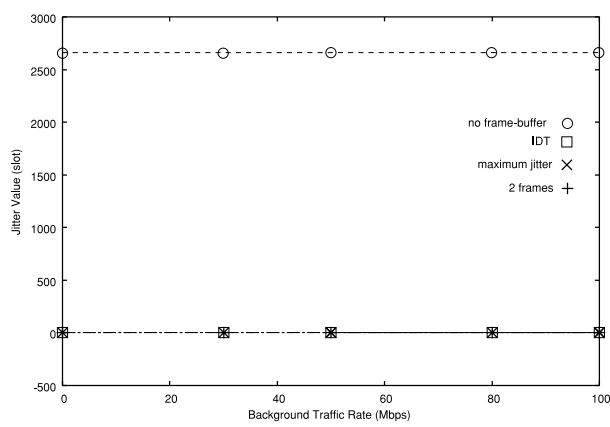**Fig. 18**   Jitter in cases with and without IDT (Binominal).

Fig. 20   Jitter in cases with and without IDT (MPEG movie).

maximum jitter which is determined by the maximum difference between end-to-end delays of data packets experienced through the network [10]. We obtain the maximum jitter value from the simulation result in the no frame-buffer case.

Figure 17 shows simulation results of the mean delay at application level in cases with and without IDT. In Fig. 17, the horizontal axis represents the background traffic rate and the vertical axis means the end-to-end transmission delay. We also present the simulation results of jitter value with and without IDT scheme in Fig. 18. In Fig. 18, the horizontal axis represents the background traffic rate and the vertical axis means the jitter value.

From Fig. 18, we find that when there are no background traffic in the network, the jitter value without IDT is quite small and therefore we need not delay playing the video stream in the frame buffer. On the other hand, when background traffic is transmitted in the network, the frame buffer is needed for absorbing the jitter because the jitter value in the no frame buffer case increases. Figure 18 also shows that two frame-buffer size and maximum jitter cases are enough to absorb the jitter and the jitter in both cases are the same as the jitter in the case with IDT.

As for the delay performance, we observe from Fig. 17

that the maximum jitter case achieves smaller delay than IDT. In the simulation, we assumed that the number of cells for packet length is distributed according to binomial distribution. Under this assumption, the variation of packet size is not large and it results in the small delay for the maximum jitter case.

Since the packet size of actual MPEG2 data fluctuates according to the frame types such as I, B and P frames, it is considered that this fluctuation is one of the factors for the large jitter at destination node. Next, we show simulation results by using the actual MPEG2 data. The MPEG2 data which we use in the simulation is generated by VBR encoding for TV program with about 8 Mbps.

Figure 19 shows simulation results of the mean delay at application level in cases with and without IDT using MPEG2 data. In Fig. 19, the horizontal axis represents the background traffic rate and the vertical axis means the end-to-end transmission delay. We also present the simulation results of jitter value with and without IDT scheme using MPEG2 data in Fig. 20. In Fig. 20, the horizontal axis represents the background traffic rate and the vertical axis means the jitter value.

From Figs. 19 and 20, we observe that in the case of no-frame buffer, the large jitter occurs at destination and the

additional delay required to absorb this large jitter is about 0.035(s). In this case, the total delay of the maximum jitter scheme is larger than that of IDT. Therefore the IDT is useful when the variation of data packet size is large.

In addtion, in the maximum jitter case, it is required to estimate the maximum jitter before transmitting the video data packets. In general it is difficult to precisely estimate the maximum jitter before transmission and when if the network condition changes after the estimation, it may be required to re-estimate the maximum jitter value during the transmission. In the case with IDT scheme, we do not suffer from these problems.

5. Conclusion

In this paper, we focused our attention on departure points of last cells for data packets and proposed IDT scheme to reduce the application level jitter. The advantage of our IDT scheme is that we need not change the existing ATM facilities except the source node. We investigated the jitter process with IDT scheme by analysis and simulation. We compared the IDT scheme with original ABR system and investigated the robustness of IDT scheme against the interruption of the background traffic. We also examined fairness and mean delay and showed how the IDT scheme works well even when the bitrate of background traffic is large.

As we see in the numerical examples, the variation of the interdeparture time of a node causes further variation of the interdeparture time of the next node. Therefore it is important for the source node to make the departure process of critical cells less variable. From this point, the IDT scheme is quite efficient.

In general, it is difficult to design the appropriate buffer sizes of data packets at destination node in advance. As we showed in Sect. 4, the IDT is quite efficient to reduce the jitter even in the multiple nodes case. One of the best solutions to guarantee the QoS of real-time application over ATM-ABR is the system where the IDT is implemented at source node and the buffer capacity of destination node is moderate.

In practice, other service categories such as CBR and VBR, which have higher priority than ABR, are used for communication at the same time. In such cases, ACR for ABR service category is dynamically varied according to the congestion state of the network. Even in this situation, it seems possible to reduce the jitter value if the ATM switch adopts the queue control function proposed in [11], [12]. Further research is needed for the mutual effects between the transmission algorithm of the source node and the rate adjusting scheme within the ATM switch.

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Naotoshi Adachi received his B.Eng. degree from the Department of Electrical and Electoronic Engineering, Ritsumeikan University, Shiga, Japan, in 1996 and M.Eng. and Dr.Eng. from the Department of Information Systems, Graduate School of Information Science, Nara Institute of Science and Technology, Japan, in 1998 and 2001, respectively. He is currently Assistant Professor at the Department of Information Systems, Graduate School of Information Science, Nara Institute of Science and Technology, Ikoma, Nara, Japan. His research interests include queueing theory and performance analysis of multimedia communication systems.

Shoji Kasahara received the B.Eng. degree from the Department of Applied Mathematics and Physics, Faculty of Engineering, Kyoto University, Kyoto, Japan, in 1989. He received the M.Eng. and Dr.Eng. degrees from the Division of Applied Systems Science, Faculty of Engineering, Kyoto University, Kyoto, Japan, in 1991 and 1996, respectively. He was with the Educational Center for Information Processing, Kyoto University from 1993 to 1997, as an Assistant Professor. In 1996, he was a visiting scholar of University of North Carolina at Chapel Hill, USA. In 1997, he moved to the Department of Information Systems, Graduate School of Information Science, Nara Institute of Science and Technology, Ikoma, Nara, Japan, where he has been an Associate Professor since August 1998. His research interests include queueing theory and performance analysis of computer and communication systems. Dr. Kasahara is a member of the IEEE, IPSJ and the Operations Research Society of Japan.

Yutaka Takahashi received the B.Eng., M.Eng., and Dr.Eng. degrees from Kyoto University, Kyoto, Japan, in 1975, 1977, and 1982, respectively. He was with the Department of Applied Mathematics and Physics, Faculty of Engineering, Kyoto University from 1980 to 1995 and with the Department of Applied Systems Science at the same faculty from 1995 to Jun 1996. He was with the Graduate School of Information Science, Nara Institute of Science and Technology (NAIST), from July 1996 to 1999. Since April 1999, he has been with the Graduate School of Informatics, Kyoto University, where he is currently a Professor supervising Information systems Lab. From 1983 to 1984 he was with the Institut National de Recherche en Informatique et en Automatique (INRIA), France, as an Invited Professor. He is now a Cochairman of IFIP TC6 WG6.3 on Performance of Communication Systems and an Elected Full Member of IFIP TC6 WG7.3 on Computer Performance Evaluation. He is also an Associate Editor of Telecommunication Systems, an Area Editor of Mobile Networks & Applications, and an International Advisory Committee Member of NIS (Networking and Information Systems) Journal. His research interests include queueing theory and its application to performance analysis of computer communication systems as well as database systems. Dr. Takahashi is a member of the Information Processing Society of Japan, the Operations Research Society of Japan, and the Institute of Systems, Control, and Information Engineers.