Evaluation of Congestion Control Method using Multiple-Constant Bit Rate Streams over XCAST6

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Abstract

Small Group Multicast (SGM) is an effective solution for conducting a large number of simultaneous small-sized group communications. Sender Initiated Congestion Control (SICC) has been proposed as a congestion control method. It is intended to provide TCP Fairness and Intra Session Fairness for real-time streaming applications based on SGM. In SICC, multiple constant transmission rates are predefined for a given session, and each rate is associated with a different SGM group containing receivers with similar acceptable sending rates. The acceptable sending rate of each receiver is estimated at the sender using TCP-friendly Rate Control (TFRC) in response to feedback generated by the receiver.

We apply SICC on one of the most typical SGM mechanism called XCAST6 and evaluate the performance of SICC using network simulator. Since XCAST6 allows packets to be forwarded among receivers even if there is no XCAST aware router along the forwarding path, the performance of mechanisms based on XCAST6 can be significantly affected by the arrangement of XCAST aware router on the networks. To the best of our knowledge, we are the first to evaluate how XCAST6 packet forwarding scheme influences the characteristics of the transport layer protocol (i.e., SICC) for real-time streaming applications.

***Keywords*:** Small Group Multicast, XCAST6, TFRC, TCP Fairness, Congestion Control, Intra Session Fairness

1. Introduction

We have previously proposed a congestion control method called Sender Initiated Congestion Control (SICC) [18] that is intended to provide TCP Fairness, Fast Congestion Avoidance and Intra Session Fairness for applications based on Small Group Multicast (SGM) [1, 2, 3, 14]. SGM is a suitable method to transmit a packet to a group containing from 10 to 100 participants.

In SGM, the sender specifies receivers' addresses in a packet explicitly. XCAST6 [1] is the most typical mechanism based on SGM in IPv6 network. The XCAST aware router copies and forwards packets using the unicast routing table thus that no control packet is needed to exchange multicast routing entries. In addition, XCAST6 has a tunneling mechanism which can be used to provide connectivity between XCAST aware nodes (i.e., router or receiver) and to pass over the non-XCAST aware routers. As a result, it is possible to gradually deploy XCAST6 over the Internet. Since XCAST6 allows packets to be forwarded among receivers even if there is no XCAST aware router along the forwarding path, the performance of mechanisms based on XCAST6 are significantly affected by the arrangement of XCAST aware router on the networks.

In this paper, we evaluate the performance of SICC, which is currently integrated into XCAST6, using the network simulator ns-2 [4]. In addition, to show how different XCAST aware router XCAST6 packet forwarding scheme influences the characteristics of SICC, we conduct a quantitative throughput evaluation.

In Section 2, we give an overview of SICC, and discuss the features of SICC and related work. Section 3 describes the XCAST6 mechanism and evaluates how the XCAST6 forwarding scheme influences the characteristics of SICC using ns-2. Finally, we conclude our work and discuss future work in Section 4.

2. SICC Protocol

2.1. Target Application

The target Application of SICC is real-time streaming which consists of a selectable sequence of frames (e.g. Motion-JPEG and DV).

2.2. SICC feature

SICC integrates the following functions into SGM.

- **TCP Fairness and Fast Congestion Control**

  In the Internet, it is important to avoid congestion and maintain fairness with other competing best-effort traffic. A transport protocol should avoid any congestion which arises on the path from a sender to receivers [5, 13]. When we design such a transport protocol, it is sufficient to consider fairness of the TCP traffic (TCP Fairness) which is the major traffic in the current Internet [5, 7, 8].

  Because a large number of group communication sessions will be held simultaneously competing with other best-effort traffic, TCP Fairness and Congestion Control are very important functions.

- **Intra Session Fairness**

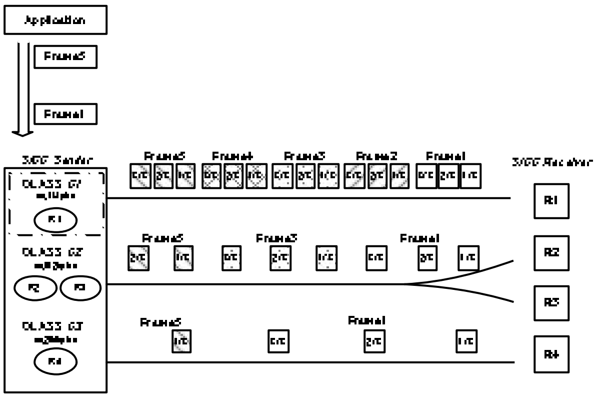
  It is preferable that a sender can transmit to each receiver in various environments at the allowed maximum transmitting rate in the multicast group communication [6]. This is called Intra Session Fairness which is a requirement specific to multicast transport protocol. We should consider the Intra Session Fairness in designing a multicast congestion control method.

2.3. Protocol Overview

2.3.1. SICC CLASS and Transmitting SGM Packet

The SICC sender has several CLASSes which transmits packets at a constant bit rate. The sender classifies receivers into the suitable CLASS based on estimation of acceptable sending rate, and each CLASS transmits SGM packet that specifies the addresses of the classified receivers.

The number of CLASSes is statically fixed, that is, the user of SICC configures the number of CLASSes when he opens the SICC socket at the sender node.



adopted the TCP-friendly Rate Control protocol (TFRC) [7] approach. TFRC is an equation base congestion control method for unicast applications which provides smooth rate control while maintaining fairness with other TCP traffic competing with the flow.

SICC sender estimates the acceptable sending rate of receivers based on the Equation (1) with the feedback from each receivers.

$$Xcal = \frac{8s}{R(\sqrt{2p/3} + t\_RTO \times \sqrt{3p/8} \times p \times (1+32p^2))} \quad ...(1)$$

In the Equation (1), s [byte] is the packet size, R [sec] is the estimated RTT tempered with past history by weighted average, p is the loss event rate and t_RTO [sec] is the TCP retransmission timeout value (usually 4*R).

After the estimation, the SICC sender classifies receivers into the suitable CLASS. In SICC, the sender has several CLASSes: Ci which transmits packets to accommodated receivers using SGM at constant bit rate Bi [bps]. The sender classifies the receiver to the CLASS: Cx so that the following Equation (2) is satisfied.

$$B_{x+1} < B_x \leq Xcal < B_{x-1} \quad ...(2)$$

Therefore, SICC can control the rate while maintaining fairness with the TCP traffic in the routes to the receivers.

2.4. Related Works

There are many multicast congestion control methods proposed for traditional IP Multicast. However, these methods are not necessarily widespread to the Internet. We think the deployment of SGM and the SICC congestion control method will enable large-scale use of multicast group communication all over the Internet in the future.

In this subsection we describe the congestion control methods based on IP Multicast. And, we show that SICC based on the SGM is a more applicable technique compared with existing methods based on traditional IP Multicast.

2.4.1. Receiver-driven protocol

In this method each receiver controls its transmitting rate according to the reception condition when joining or leaving some groups which the sender prepared.

RLM (Receiver-driven Layered Multicast) [9] and RLC (Receiver-driven Layered Congestion control) [10] are enumerated as the typical methods. In these methods, each receiver has initiative to control transmitting rate by itself, and the Intra Session Fairness can be achieved. However, the problem has been identified that over shooting traffic still remains in the network after the receiver issues a leave message from the group, because of the IGMP leave confirmation delay [15]. This means that it takes too much time to avoid congestion.

2.4.2. Sender-driven protocol

In this method the sender controls the transmitting rate by using feedback from receivers or routers when congestion arises. In general, the sender controls the transmitting rate on a single-group (single-rate) without an IGMP leave message, so there is no IGMP leave confirmation delay problem. Fast Congestion Control is also possible. TFMCC (TCP-Friendly Multicast Congestion Control) [8], PGMCC (Pragmatic General Multicast Congestion Control) [12], and ARE/NCA (Active Error Recovery / Nominee Congestion Avoidance) [11] are enumerated as typical methods.

TFMCC is a method that adopts the TFRC approach as well as SICC, and TCP Fairness can be achieved in TFMCC. On the other that PGMCC and ARE/NCA are methods which achieve TCP Fairness to control the transmitting rate based on the AIMD (Additive Increase Multiplicative Decrease) approach.

However, the transmitting rate is decided according to the receiver whose reception condition is worst in the group, so in general it can't achieve Intra Session Fairness.

2.4.3. Comparison with SICC

Receiver-driven congestion control methods have problems of the performance of congestion avoidance because of the IGMP leave confirmation delay. On the other hand, sender-driven congestion control methods achieve TCP Fairness and Fast Congestion Avoidance. But these methods don't achieve Intra Session Fairness because these methods adjust transmitting rate with a single-rate.

As we have already described, SICC doesn't use IGMP, and immediately classifies each receiver to the suitable CLASS satisfying the evaluation of TFRC according to the receiver's feedback. Therefore, SICC achieves the TCP Fairness, Fast Congestion Avoidance and Intra Session Fairness requirements.

2.5. Confirming Basic Protocol Feature

We have evaluated TCP Fairness and Intra Session Fairness which are the main characteristics of SICC. This section shows the results of our evaluation using the network simulator (ns-2).

2.5.1. TCP Fairness

To achieve TCP Fairness, SICC adopts TFRC approach while estimating the sending rate. We confirm whether SICC applied on XCAST6 is able to share the bandwidth with the TCP flows fairly on the bottleneck link.

First of all, we use the well known single-bottleneck link topology (Figure 2) to evaluate whether each flow maintains fairness with other flows when both the SICC flows and TCP flows increase synchronously.

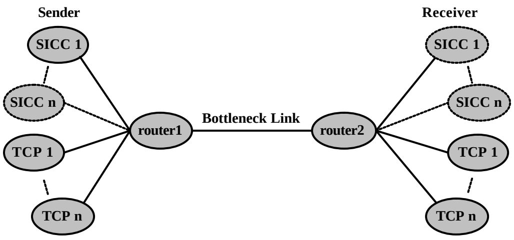


Fig 2 Single-bottleneck topology

Figure 3 and Figure 4 show the throughput of a SICC flow and a TCP (Reno) flow competing with SICC. In this simulation, the number of CLASSes is five, and the transmitting rate of each CLASS is {C1: 1Mbps, C2: 512kbps, C3: 256kbps, C4: 128kbps, C5: 64kbps}. In the bottleneck link, bandwidth is 1Mbps, delay is 64ms, and DropTail or RED queue. These results show that SICC can perform rate control keeping fairness with TCP flows.

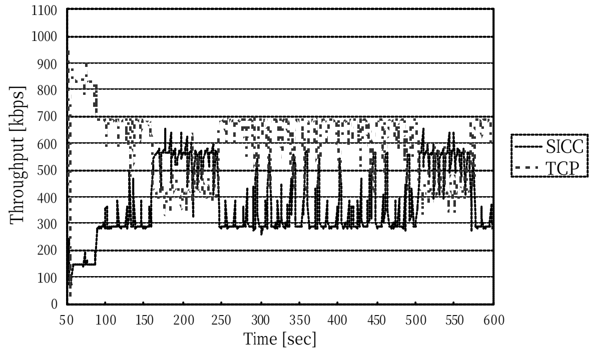


Fig 3 SICC and TCP throughput (DropTail)

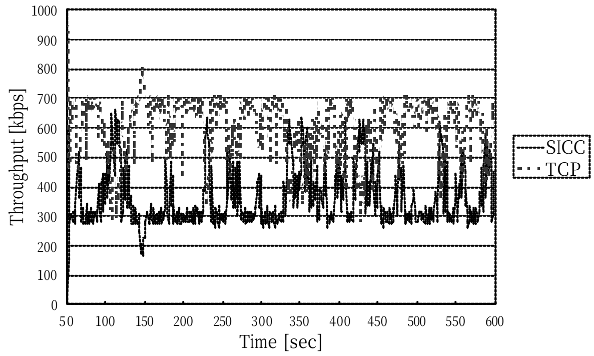


Fig 4 SICC and TCP throughput (RED)

We introduce the index: *F* that shows whether SICC and co-existing TCP flows can fairly share the bandwidth on the bottleneck (Equation 3). In this equation, *linkbw* [bps] is the bandwidth of bottleneck link, *i* is the number of flows on the bottleneck link, *thput_avg* [bps] is the average throughput of each flow. *F* becomes 1 when the flow fairly shares bandwidth of the link.

$$F = \frac{thput\_avg}{\frac{linkbw}{i}} \quad ...(3)$$

Figure 5 illustrates the fairness of each flow, when n is changed {1, 2, 4, 8, 16, 32, 64, 128} on the simulator. The simulation time is 600 seconds, the delay is 64ms, the queue is RED, bandwidth and queue length of the bottleneck link are increased in proportion to the number of flows. The CLASS transmitting rate of SICC is same as Figure 3 and 4.

It shows the fairness (*F*) of each SICC flow indicates the value of about 0.75 to 1, the one of each TCP flow indicate the value from 0.9 to 1.2. According to the above results, even if the number of both flows increases, the average throughput of all the competing flows is within twice in each other. Therefore it is confirmed each SICC flow can share the bandwidth with other SICC flows and TCP flows fairly on a bottleneck link.

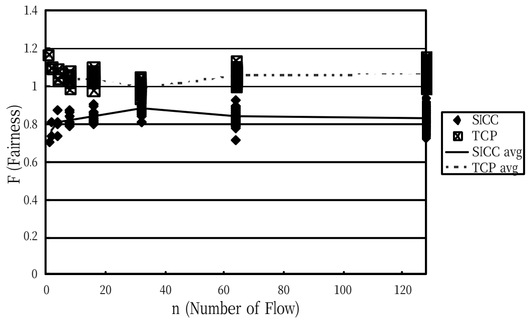


Fig 5 TCP Fairness with n SICC and n TCP flows

Next, we evaluate whether a SICC flow and a TCP flow maintains fairness with each other on a bottleneck link when the SICC sender transmits XCAST6 packets to ten receivers in the topology shown in Figure 6.

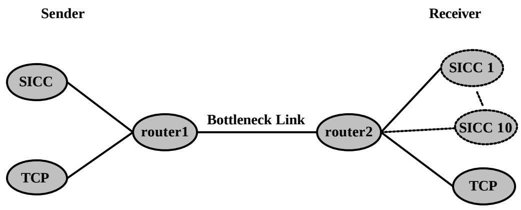


Fig 6 Single-bottleneck topology2

Figure 7 illustrates the fairness of each flow, when delay is changed {4, 8, 16, 32, 64, 128} on the simulator. The simulation time is 600 seconds, the bandwidth is 1Mbps. The CLASS transmitting rate of SICC is same as Figure 3 and 4.

It shows the fairness (*F*) of a SICC flow indicates the value of about 0.9 to 1.1, the one of a TCP flow indicate the value from 0.8 to 0.9. Therefore it is confirmed multicast communication on SICC can share the bandwidth with TCP flow fairly on a bottleneck link.

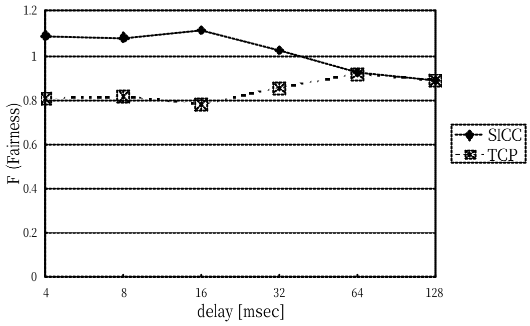


Fig 7 TCP Fairness with SICC multicast and TCP

2.5.2. Intra Session Fairness

To confirm Intra Session Fairness of SICC, we conduct an evaluation using a simple multicast topology as shown in Figure 8, where there are ten receivers and one sender. The bandwidth of the link between the router and each receiver is generated randomly from a uniform distribution between 64kbps to 1Mbps, and the link delay is generated randomly from a uniform distribution between 4ms to 128ms. In addition, we set the number of the CLASSes is five, the transmitting rate of each CLASS is {C1: 1Mbps, C2: 512kbps, C3: 256kbps, C4: 128kbos, C5: 64kbps} and the simulation time is 600 seconds.

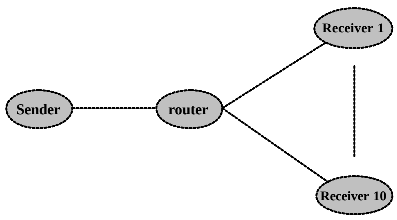


Fig 8 Simple Multicast topology

Figure 9 shows the relative bandwidth utilization of each receiver to various types of available uplink bandwidths. We observe in Figure 9 that each receiver is capable of receiving packets at the rate corresponding to available uplink bandwidth. This shows that SICC achieves its objective of Intra Session Fairness.

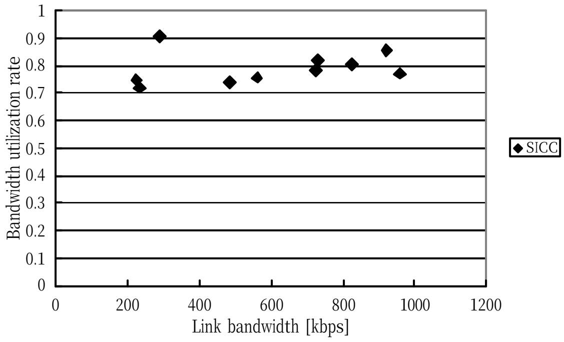


Fig 9 Bandwidth Using Rate

3. Discussion of applying on XCAST6

In this section, we give an overview of the XCAST6 packet forwarding scheme, and discuss how the XCAST6 forwarding scheme affects the characteristics of SICC.

3.1. XCAST6 Forwarding scheme

XCAST6 is a new form of multicast mechanism where the sender explicitly specifies the list of receiver's addresses. The list is stored in an IPv6 routing option header with a bitmap that shows the status of “undelivered” or “delivered” for each destination listed in the header.

When an XCAST router receives a packet, it lookups the unicast routing table to determine the next hop for each receiver with the status of “undelivered” shown in the bitmap. Comparing the result of routing lookup, the XCAST router can analyze whether or not the listed receivers have common path to the next hop and update the status of bitmap related to those receivers. After that, it duplicates the datagram and branches it for the other path if needed. Note that non-XCAST aware router treats and processes the XCAST6 packet as normal unicast packet. Each receiver behaves as an XCAST router duplicates and forwards the packet to the remaining destinations if there exists receivers with the status of “undelivered”.

As a result, Multicast-like functions can be achieved in XCAST6 by simply encoding all the destinations into the packet header.

3.2. Effects of XCAST6 Packet Forwarding

An XCAST6 receiver forwards a packet to the other receivers if the status of “undelivered” remains up in the packet. In addition, the delivery order depends on the order of the list of destinations in the packet.

An SICC sender transmits a send timestamp to SICC receivers with every data packet. Upon the receipt of the data packet, the SICC receiver sends the timestamp back to the sender if reporting is promoted. After that, this timestamp is used by the sender to estimate the value of round-trip time (RTT) between the SICC sender and the SICC receiver. With the estimated RTT, the SICC sender can further estimate the acceptable sending rate of the SICC receiver.

As mentioned above, the delivery order of XCAST6 packet depends on the order of the list of destinations in the packet. That is, the lower the order of a receiver listed in the packet, the longer the measured RTT will likely be. This inevitably leads to an inaccurate estimation on the acceptable sending rate (Equation 1), and decreases the throughput of the receiver.

3.3. SICC Performance Analysis on XCAST6

In this section, we evaluate how the daisy chain delivery of XCAST6 affects the performance of SICC using the network simulator (ns-2). First, we evaluate the performance of SICC when no XCAST aware routers are located in the simulation environment.

As shown in Figure 10, we consider a tree topology due to its similarity to a general multicast communication environment and measure the throughput of each receiver. In addition, we compare two models with different receiver capabilities on the same topology. In Model A as shown in Table 1, we assume the reception capability of each receiver is uniform. That is, we set the available bandwidth of all the links between router and receivers is set to 540kbps. In Model B as shown in Table 2, we assume each receiver has various reception capabilities. That is the available bandwidth of the link between each router and each receiver are generated randomly from a uniform distribution between 64kbps to 1Mbps. In addition, the eight receivers which join in a SICC session are chosen at random on the leaf of the tree topology; the total number of SICC CLASSes is five; the simulation time is 600 seconds; and the transmitting rate of each

CLASS is {C1: 1Mbps, C2: 512kbps, C3: 256kbps, C4: 128kbps, C5: 64kbps} respectively.

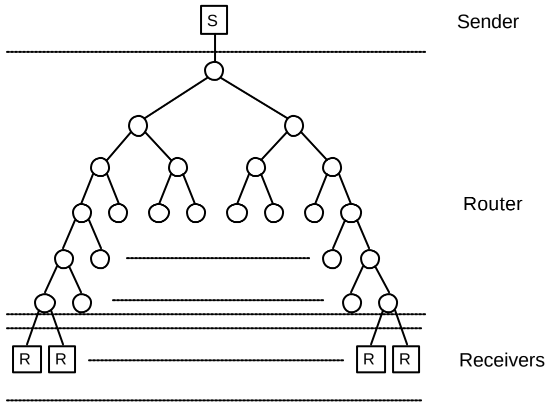


Fig 10 Tree Topology

Table 1 Model A

Link	Bandwidth	Delay
Sender – Router	100Mbps	10ms
Router – Router	100Mbps	10ms
Router — Receiver	540kbps	10ms

Table 2 Model B

Link	Bandwidth	Delay
Sender – Router	100Mbps	10ms
Router – Router	100Mbps	10ms
Router — Receiver	Uniform distribution (64kbps - 1Mbps)	10ms

Table 3 shows the bandwidth utilization between each receiver and its uplink router and the average of these bandwidth utilizations obtained in both Model A and B by one simulation run. Table4 shows the average of the bandwidth utilizations obtained from one hundred simulations with different random seeds. We observe that the bandwidth utilization in Model A is about 18 % lower than in Model B. This is because the SICC sender can classify receivers into different CLASSes based on their reception ability. In Model A, all the receivers have the same reception capability, that is, all the receivers tend to be classified into a single CLASS for a session thus that the length of the list of destinations in the XCAST6 packet becomes longer. As mentioned previously, since the lower the order in the list of destinations, the longer the measured RTT in the environment. As a result, the average of the measured performance becomes worse.

Table 3 Bandwidth utilization rate

Receiver	Model A	Model B
R1	0.47	0.63
R2	0.19	0.70
R3	0.19	0.49
R4	0.52	0.20
R5	0.23	0.31
R6	0.26	0.29
R7	0.32	0.15
R8	0.21	0.49
Session Average	**0.30**	**0.41**

Table 4 Bandwidth utilization rate

Receiver	Model A	Model B
Session Average (100 times)	0.30	0.48

Figure 11 is distribution chart that the bandwidth utilization of each receiver, and the average own order of each receiver in the list of destinations in all the transmitted XCAST6 packets obtained in Model A from one simulation run. We observe that the lower the order of receivers in the list of destinations, the less the bandwidth utilization of the receivers.

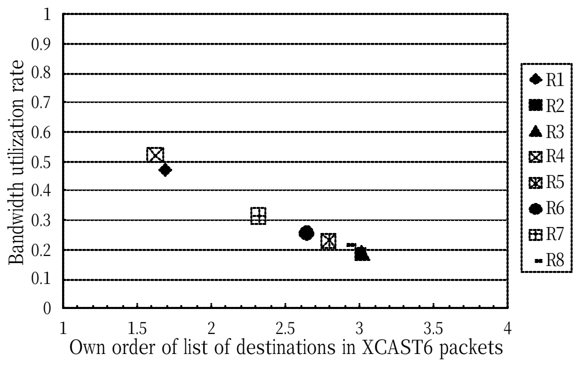


Fig 11 Bandwidth utilization rate at the order of the list of destinations

In another simulation, Model B, we set the reception capability of each receiver is widely distributed. Therefore, the receivers tend to be classified into several different CLASSes for a session, that is, the length of the list of destinations in the XCAST6 packet is shorter than that of Model A. Table 5 shows the average length of the list of destinations in all the transmitted XCAST6 packets and the average bandwidth utilization of the session obtained in both Model A and B from one simulation run. We observe that the average length of the list of destinations in Model B is shorter than that of Model A, and Model B achieves better average bandwidth utilization than Model A. This is because that Model B has a shorter length of the list of destinations thus that the occurrence of the daisy chain delivery delay can be avoided and the problem caused by unexpected RTT increase is alleviated.

Table 5 XCAST6 Packet list length and Bandwidth Using Rate

Result	Model A	Model B
Average length of the list of destinations	2.51	2.08
Bandwidth utilization rate of one session	0.30	0.41

Next, we evaluate the bandwidth utilization between each receiver and its uplink router for both Model A and B when the XCAST routers are partially arranged on the tree topology as shown in Figure 10.

Figure 12 shows the average bandwidth utilization between each receiver and its uplink router in a session where the XCAST aware routers are randomly selected at each probability. The results are obtained from one hundred simulations with different random seeds. We observe that the more the XCAST aware routers are deployed in the topology, the higher the average throughput of the session obtained in each Model.

We also observe that Model A achieves lower bandwidth utilization than Model B, when the ratio of the total number of the XCAST aware routers is set to 50% or lower. This is because the length of the list of destinations in the XCAST6 packet influences the overall throughput of the session. On the other hand, when the ratio of the total number of the XCAST aware routers is set to 75% or higher, similar results from both Models are observed. This is because that the occurrence of the daisy chain delivery is low thus that the length of the list of destinations in the XCAST6 packet does not affect the performance of SICC.

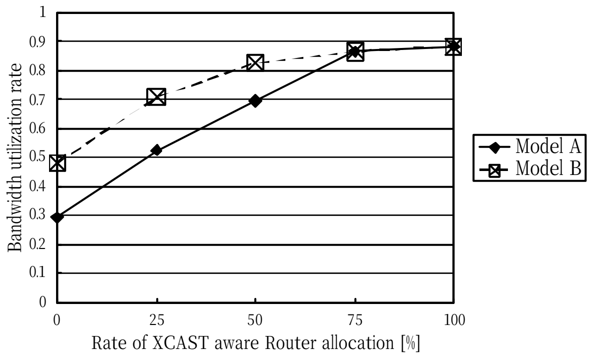


Fig 12 Bandwidth utilization rate

4. Conclusion

SICC is a congestion control method that integrates the following functions into SGM.

i. TCP Fairness and Fast Congestion Avoidance
ii. Intra Session Fairness

We applied SICC on XCAST6 which is one of the most typical IPv6-based SGM mechanisms. In addition, we evaluated the performance of SICC using the network simulator. In this paper, we have focused on the discussion of how the XCAST6 forwarding scheme affects the characteristics of SICC. That is, the process which the daisy chain delivery of XCAST6 affects the throughput of SICC sessions.

In the XCAST6 packet forwarding scheme, the daisy chain delivery among receivers occurs when the XCAST aware routers are not located on the delivery path. In addition, the sequence of delivery depends on the order of the list of destinations in the XCAST6 packet. In SICC, the lower the order of a receiver listed in the packet, the longer the measured RTT, which is a result of the delay caused by the daisy chain delivery, and thus decreasing the throughput of the receiver. In this simulation, we measured the throughput of eight receivers which were chosen randomly on the tree topology. We considered two models for this topology, due to its similarity to a general multicast communication environment. In Model A, we assumed the reception capability of all receivers was uniform, and in Model B, each receiver had different reception capability. In the case that no XCAST aware routers are located in the topology, the result is the average of bandwidth utilization between each receiver and its uplink router in Model A is about 20% lower than Model B. This is because the average length of the list of destinations in the packet is longer in Model A than in Model B. In addition, since the receiver at the lower order of the list of destinations in the XCAST6 packet has a lower throughput because of the frequent occurrence of the daisy chain delivery, the average throughput of the whole SICC session in Model B is better than that of Model A.

Next, we evaluated the bandwidth utilization between each receiver and its uplink router for both Model A and B when the XCAST aware routers were partially located on the same tree topology. The result is the average of bandwidth utilization of each receiver in Model A is lower than Model B, when the ratio of the number of XCAST aware routers was set to 50% or lower. This is because the length of the list of destinations in the XCAST6 packet influenced the throughput of the SICC session. However, when the ratio of the number of XCAST aware routers was set to 75% or higher, we obtained similar results from both Models. This is because that the occurrence of the daisy chain delivery is low thus that the length of the list of destinations in the XCAST6 packet does not affect the performance of SICC.

Our future work includes developing and evaluating the mechanism that efficiently avoids the inherent problems caused by the daisy chain delivery delay in XCAST6.

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