Scheme for Estimating Upper-Bound of Link Utilization Based on RTT Measurements with Consideration of Processing Delay Jitter and Practical Timer Granularity

Kiyofumi Igai and Eiji Oki

Dept. of Communication Engineering and Informatics, Graduate School of Informatics and Engineering,
The University of Electro-Communications, Tokyo, Japan

***Abstract*— This paper proposes a link utilization estimation scheme based on Round-Trip Time (RTT) measurement with consideration of the processing delay jitter and practical timer granularity. In a previous work, RTT measurements are used to estimate link utilization from a probability distribution of the minimum RTTs on a link where the RTT variation is considered to be caused by only queuing delay. However, the original RTT-based scheme has problems with the two factors of processing delay jitter and actual timer granularity. The first problem is that processing delay jitter is only one cause of RTT variation. The observer cannot differentiate the impact of queuing from that of processing. The second problem is that measuring the minimum RTT is difficult if the timing accuracy is of the order of micro-seconds because an observer cannot determine whether an RTT value is minimum or not. The proposed scheme, which extends the previous scheme, identifies and eliminates the effects of processing delay jitter and estimates the probability of the minimum RTT with micro-second order timer granularity. The former is achieved with division by an appearance probability value of the processing delays at the ingress and egress nodes of a targeted link from the estimated RTTs on the target link. To realize the latter, the proposed scheme sets a suitable range from estimated RTTs on a target link. Experiments show that the proposed scheme effectively eliminates the jitter effects under low load with a 0.2 reduction in deviation from the true value.**

***Index Terms*—link utilization, round-trip time, active measurement, processing delay jitter, timer granularity, timer jitter.**

I. Introduction

LINK utilization, which is the ratio of total traffic passing through the link to the link capacity, is a key indicator for detecting network congestion, or judging that link capacity is sufficient to ensure that the communication quality will meet the customer's requirements [1], [2]. The network operators should by observe link utilization continuously or at very short intervals to control communication quality. If they find that link utilization of a targeted link is higher than a specified value, they should reroute traffic flows, or increase link capacity, to avoid network congestion. There are two approaches to estimating link utilization: passive measurement and active measurement.

The operators do not transmit any probe packets in the passive measurement approach, so traffic passing though the link is not affected by the measurement. Traffic information is directly collected from each node on the link by using, for example, the Simple Network Management Protocol (SNMP) [3].This approach measures the link utilization in an accurate manner and frequently or constantly measures it. However, it is not always possible for observers to collect the information because of administrative restrictions.

In the active measurement approach, network performance is estimated by sending probe packets from a source host to other nodes on the targeted path. Active measurement techniques include Train Of Packet Pair (TOPP) and Self Loading Periodic Stream (SLoPS) [4]-[9]. TOPP sends sequential packet pairs on the path at increasing rates [4]-[6],. Based on the relationship between the sending and receiving rates of the different packet pairs, the observer can estimate the available bandwidth of a bottleneck link, i.e. the smallest available bandwidth on a path.

In SLoPS, the source host sends periodic streams over the path at several constant rates [7]-[9]. If the rate is higher than the available bandwidth, the one-way delay variation of received packets tends to increase. The available bandwidth of the bottleneck link is estimated by analyzing the interval of probing stream packets received at the observer host. In addition, these techniques are not to estimate bandwidth, but to estimate available bandwidth. In order to determine link utilization, it is necessary to estimate path bandwidth in advance [10]. The advantage of the active measurement approach is that the observer can estimate link utilization

Manuscript received March xx, 2013.

This work was supported in part by the Strategic Information and Communications R&D Promotion Programme of the Ministry of Internal Affairs and Communications, Japan.

Kiyofumi Igai and Eiji Oki are with Dept. of Communication Engineering and Informatics, Graduate School of Informatics and Engineering, The University of Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo, Japan.

without accessing nodes. In other words, observers do not face any administrative restriction. However, both the TOPP method and SLoPS send probing packets over the path, at rates temporarily greater than the available bandwidth. This causes overloading making continual or frequent measurements infeasible.

We presented the first scheme to estimate link utilization by measuring RTT in [11], [12]. In our previous work, link utilization is obtained from the probability of the minimum RTTs on a link where RTT variation is caused by only queuing delay. This scheme measures two RTTs from a source host to both end nodes of a targeted link to obtain their probability of minimum RTTs. The link utilization rate is estimated by the measured RTT values. The advantage of this scheme is that measurements do not cause any overload because the probe packets can be small. This scheme assumes that queuing delay is variable but the other delay factors such as processing delay are fixed. In addition, timer granularity must be of the nano-second order to determine the minimum RTTs.

However, in applying the scheme to real networks, an observer faces problems in trying to offset the effects of processing delay jitter and large timer granularity. The former causes RTT variation and the observer cannot separate the delay caused by queuing from that by processing. The latter complicates the measurement of minimum RTTs. Required timer granularity is 256 [nsec] when the minimum Internet Protocl (IP) packet size is 32 [bytes] (Link capacity is 1 [Gbit/s]). To estimate link utilization in giga-bit networks, nano-second order timer granularity is required to determine whether RTT is minimum or not. Thus, measurement applications running on the current kernel have great difficulty in measuring the minimum RTTs accurately.

This paper is an enhanced version of our previous work in [11], [12]. This paper proposes a link utilization estimation scheme that takes into consideration the processing delay jitter and available timer granularity that is practical given current networks and equipment. The proposed scheme eliminates the effects of processing delay jitter and estimates the probability of the minimum RTT with micro-second order timer granularity. The effects of processing delay jitter are eliminated through the use of an appearance probability value of the processing delays at the ingress and egress nodes of the targeted link from the estimated RTTs on the target link. To estimate the actual minimum RTTs with micro-second order timer granularity, the proposed scheme sets a suitable range from estimated RTTs on the target link. Experiments show that the proposed scheme is effective under low loads with reduction of 0.2in deviation from the true value.

The remainder of this paper is as follows: Section II explains link utilization without processing delay jitter. Section III discusses how the proposed scheme eliminates effects of the jitter. Section IV discusses timer granularity problem in current networks. Section V describes our experiments the proposed scheme and the results in the experiment network. Finally, Section VI summarizes this paper.

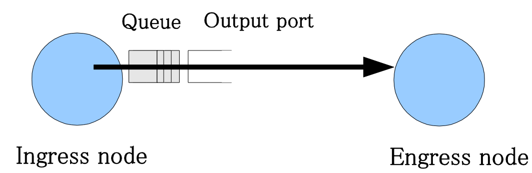

Fig. 1. Network model for one-way delay.

II. LINK UTILIZATION WITHOUT PROCESSING DELAY JITTER

When a packet passes through a node, it is delayed. The one-way delay from one node to the next node on the link under test (LuT) consists of fixed and variable components, see Figure 1. Delay $D(i)$ for packet *i* is expressed by,

$$D(i) = T_f + T_q(i). \quad (1).$$

T_f includes fixed delays for forwarding and switching, serialization/de-serialization, and propagation. T_f is a constant value, which is independent of *i*, as long as the route of each packet is not changed. $T_q\,(i)$ is caused by queuing at the ingress node, where a packet has to wait before being transmitted to the output port.

Link utilization is estimated by using the delay probability distribution for LuT from the ingress node to the next node; note that the queuing delay at the neighboring node is not included.

Queuing delay is measured by sending probe packets from the ingress to the neighbor hop node through the link. When the queue at the ingress node is empty, there is no queuing delay, i.e., $T_q\,(i) = 0$. In this case, the observer measures the minimum delay, which is defined as $D_{min} = \min D(i)$. If the observer focuses on the instantaneous time, where $T_q\,(i) = 0$, the link is not utilized. On the other hand, if the queue is not empty, the delay for a packet passing through the link is varied by queuing. The measured delay is larger than $Dmin$, where ($T_q\,(i) > 0$). At this moment, the link is utilized. Let $p(x)$ be the probability distribution function with

$$x = D(i) - Dmin. \quad (2)$$

p_{min}, the probability of minimum RTT, is defined by

$$p_{min} = \int_0^{\delta T} p(x)\,dx. \quad (3)$$

In this equation, ΔT is a small insignificant value.

The link utilization, which is denoted as U is the probability that the queue is occupied by at least one packet, in other words, $x > 0$. Let the number of probe packets whose delay is $Dmin$ be $N_{x=0}$, and the number of probe packets whose delay is larger than D_{min} be $N_{x>0}$. U is obtained by,

$$U = 1 - p_{min} = 1 - \frac{N_{x=0}}{N_{x>0} + N_{x=0}}. \quad (4)$$

A. Network Models

To estimate the link utilization expressed in Eq.(4), conventional scheme measures the RTTs of many probe packets, as shown in Figure 2, instead of measuring one-way delay [11], [12]. This is because the measurement of one-way delay requires the strict time and clock synchronization of the two nodes. In general, queuing occurs at both ingress and egress nodes, as shown in Figure 2. The network model of Figure 2, employs the ping mechanism as specified in the Internet Control Message Protocol (ICMP) [13].

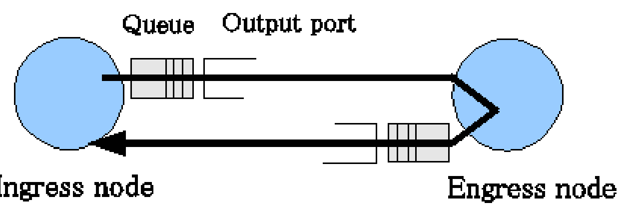

Fig. 2. Network model for RTT.

B. Procedure

The scheme proceeds in three steps; measurement of RTT, estimation of the probability of minimum RTT, and estimation of link utilization [11], [12].

RTT Measurement

An observer at the source host measures the RTTs to both end nodes of the LuT at the same time and gathers enough RTT-data to allow the probability distribution functions of RTT from the source host to both end nodes to be determined with sufficient accuracy.

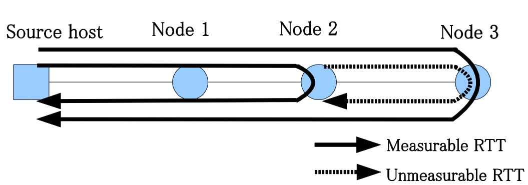

Fig. 3. Various RTTs on the path

As shown Figure 3, the observer obtains $f(x)$ and $h(x)$ as the probability distribution functions of RTTs from the source host to node 2 and node 3, respectively $g(x)$ is the probability distribution function of RTT, x, from node 2 (ingress node), to node 3 (egress node), of the LuT, where $g(x)$ can not be measured directly. g_{min} is the probability that the RTT from the ingress node to the egress node on the LuT takes the minimum delay.

Estimation of probability distribution function

$g(x)$ is estimated by $f(x)$ and $h(x)$ which are directly measured. h_{min} is the product of f_{min} and g_{min}, where h_{min} is expressed by,

$$h_{min} = f_{min}\, g_{min}. \quad (5)$$

h_{min} in Eq.(5) is the probability that the RTT measured from a source host to an egress node of the target link takes the minimum value. f_{min} in Eq.(5) is the probability that the RTT measured from a source host to an ingress node of the link takes the minimum value. g_{min} in Eq.(5) is the probability that the RTT measured from an ingress node to an egress node of the link takes the minimum value.

By using Eq.(5), g_{min} which is not directly measurable, is obtained by,

$$g_{min} = \frac{h_{min}}{f_{min}} \quad (f_{min} \neq 0). \quad (6)$$

Estimation of probability distribution function

The observer is able to estimate the link utilization by using g_{min} . Let $p(x)$ and $q(x)$ be the probability distribution functions of the one-way delay in the forward and backward directions, respectively (Figure 4). g_{min} is given by p_{min} and q_{min},

$$g_{min} = p_{min}\, q_{min}. \quad (7)$$

Because of queueing delay in the backward direction, $q(0) \leq 1$. So, $g(0) \leq p(0)$. By using Eq. (3) and $g(0) \leq p(0)$, the upper bound of U is obtained as,

$$U \leq 1 - g_{min}. \quad (8)$$

If there is no queueing delay in the backward direction, $q(0) = 1$. Thus, $g(0) = p(0)$.

By using Eq. (3) and $g(0) = p(0)$, U is obtained by,

$$U = 1 - g_{min}. \quad (9)$$

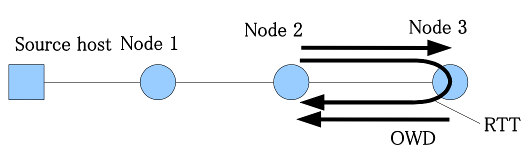

Fig. 4. OWDs and RTT on the LuT

III. PROPOSED SCHEME WITH REVISION

Real networks exhibit processing delay jitter; in other words, the processing delay at network nodes is not constant. Figure 5 shows measured RTT under no load. No load means that the RTT delay variation is caused only by processing delay. The fluctuation in measured RTT is about 0.1 [msec]. If there is no processing delay jitter, RTT is constant value under no load. The jitter impacts the estimation accuracy because the observer on the source host can not determine whether the RTT delay is due to processing or queuing. Thus, a procedure to eliminate the jitter is required.

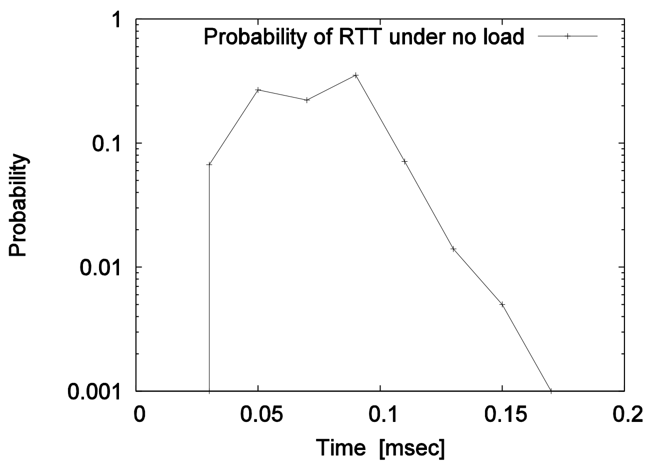

Fig. 5. Histogram of RTT under no load.

The basic procedure is the same as that of the previous work; measure RTT, estimate the probability of minimum RTT, and estimate link utilization. In estimating the link utilization, the processing delay jitter is revised from estimated RTTs on the target link.

Due to the jitter, the observer cannot determine the actual probability of minimum RTT. Accordingly, the proposed scheme requires integration of the RTT's delay distribution function in a certain range size is required. In short, we obtain the cumulative frequency of the probability in each range. The apparent probability of minimum RTT, g_{min}^{m}, is defined by estimated RTT $g_m(x)$ ($g_m(x)$ is estimated by $f_m(x)$ and $h_m(x)$.

$$g_{min}^{m} = \int_0^{\delta T} g_m\,(x)dx. \quad (10)$$

Revised Procedure for Processing Delay

g_{min}^{m} is the product of g_{min}^{proc} and g_{min}^{que}. Figure 6 shows the link model with processing delay. g_{min}^{proc} is the appearance probability of minimum value of processing delay on the two nodes. g_{min}^{que} is the appearance probability of the minimum value of the queueing delay. Thus measured probability of minimum RTT, g_{min}^{m} is expressed by,

$$g_{min}^{m} = g_{min}^{proc}\, g_{min}^{que}\ . \quad (11)$$

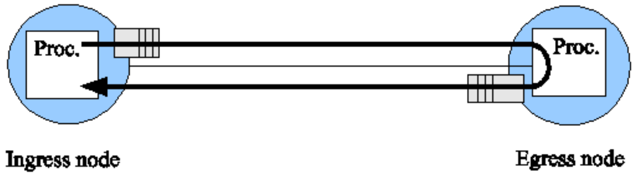

Fig. 6. Link model with processing delay.

Revised utilization U_{re} is expressed by,

$$U_{re} = 1 - g_{min}^{que}. \quad (12)$$

From Eq.(11), $g_{min}^{m} = g_{min}^{que} / g_{min}^{proc}$, so U_{re} is rewritten as,

$$U_{re} = 1 - \frac{g_{min}^{m}}{g_{min}^{proc}}. \quad (13)$$

By using Eq.~(5), g_{min}^{m} is estimated from measured h_{min}^{m} and f_{min}^{m},

$$U_{re} = 1 - \frac{h_{min}^{m}}{f_{min}^{m}\, g_{min}^{proc}}. \quad (14)$$

Thus, revised utilization U_{re} is obtained by g_{min}^{proc}.

Measuring for Processing Delay

To measure just the processing delay at network nodes, the no-load condition, i.e. queuing delay is zero, is needed. In the proposed scheme, the probability of the minimum processing delay is required to revise the processing delay jitter. The processing delay jitter can be measured by an experiment. An observer prepares two routers and measures RTT from an ingress router to an egress router. The observer takes the RTT to represent processing delay jitter. To achieve high revision accuracy, the observer should repeat the measurements on several different routers.

IV. TIMER JITTER AND TIMER GRANULARITY

The packet arrived time can not to be measured accurately due to two issues: timer jitter and timer granularity at the observer host.

First, we explain the issue of timer jitter. Timer jitter is triggered when a NIC (Network Interface Card) transfers packets to the operating system (OS) and when the OS passes a process right to a timer process. Therefore, the delay variation measured by the timer process is not always equal to that of the actual arrival time at an NIC. Suppose that the packet arrival time at an NIC is 800 [nsec], and the delay jitter range is from 10 to 100 [nsec]. The recorded time range runs from 810 [nsec] to 900 [nsec]. In the case that there are multiple packets that have 800 nano-second delay, the recorded histogram might look like that in Figure 6 (a). The number of minimum-delay packets measured by the timer process is smaller than the number of minimum-delay packets arriving at the NIC.

The other problem is the timer granularity at the observer host. The larger the timer granularity is, the less accurately the arrival time can be measured by the timer. Consider two packets at an NIC. One packet has 1000 [nsec] delay. The other has 1400 [nsec] delay. However, if the timer granularity is of the order of 1 micro-second, the measured time of each packet becomes 1 [msec]. Therefore, as shown in Figure 6 (b), packets that arrive at different times have the same recorded arrival time.

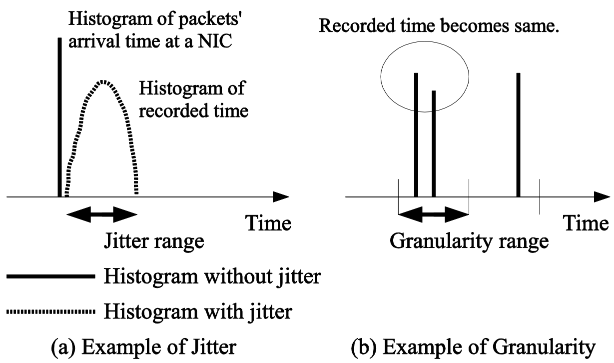

Fig. 6. An example of timer jitter and timer granularity

The issues of timer jitter and timer granularity are independent, but both affect the measurement accuracy (see Figure 7). Thus, due to the jitter and the granularity, the packet arrived time cannot be measured accurately.

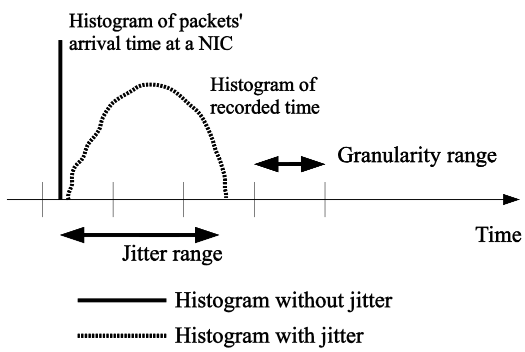

Fig. 7. Effects of timer jitter and timer granularity

To determine the minimum RTT accurately, the timer granularity is desired to be of the order of nano-seconds. Note:Effects of the timer jitter at the observer host can be roughly eliminated (see Section III). Complete removal of the jitters effects is difficult because of the impact of timer granularity on measurement accuracy. Figure 8 shows an example of measured RTTs with ideal timer granularity. The gap between minimum RTT and non-minimum RTT equals the queuing time of the probe packet. The required granularity is 256 [nsec] when probe packet size is 32 [bytes] and link capacity is 1 [Gbit/s].

In the experiments, however, timer granularity is of the order of micro-seconds, so the observer cannot exactly determine the bunned probability of the minimum RTTs. Figure 9 shows an example of measured RTTs with practical timer granularity. The minimum RTTs overlap the non-minimum RTTs because the minimum RTTs are perturbed by the large timer granularity. A small range cannot pick up all values of the minimum RTTs. In contrast, a large range picks up not only values of the minimum RTTs but also values of the non-minimum RTTs. We want to find the effective range in Figure 9 and determine the desired range δT so that the accuracy can be maximized considering possible parameters. The highest accuracy is realized in the desired range. According to Eq.(10), non-exact values of minimum RTTs (due to excessively small or large ranges) degrade the accuracy.

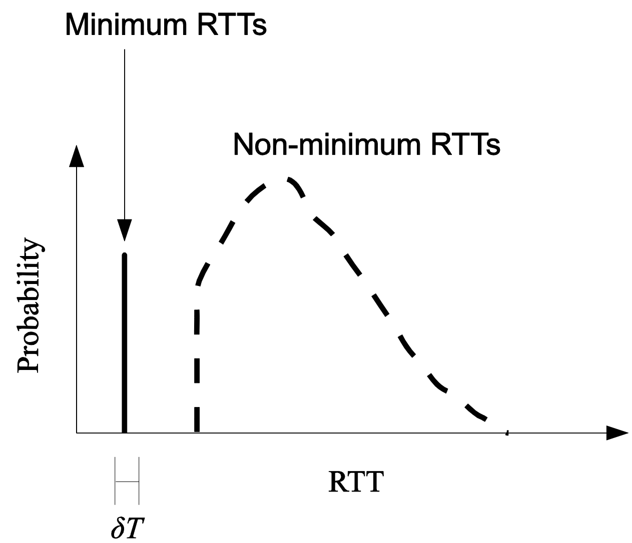

Fig. 8. An example of measured RTTs under ideal timer granularity

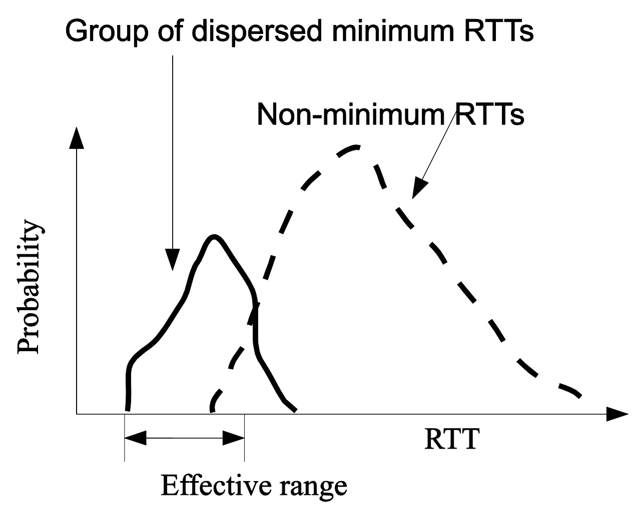

Fig. 9. An example of measured RTTs under practical timer granularity

V. EVALUATION

We evaluated the estimation accuracy of link utilization on the network as shown in Figure 10 and Figure 11. Table I shows the details of each node. All Network Interface Cards (NICs) use Transmission Control Protocol (TCP) segmentation offloading, along with IP, TCP and User Datagram Protocol (UDP) checksum offloading. In checksum offloading, NIC hardware calculates the checksum of packet headers to reduce the loads imposed on the operating system.

In Figures 10 and 11, among the two links through which cross traffic passes, link 3-4 was set to be the bottleneck, the one with the smallest available bandwidth on the path, its cross-traffic load was set to 0.5, 0.6, 0.7, 0.8 or 0.9. Link 2-3 was set to the non-bottleneck link with fixed load of 0.3. Load is ratio of traffic rate to link capacity. Node 1 was set to the source host and the observer sent probe packets to node 3 and node 4. Request and reply packets were 32 [bytes] long. Packets of cross traffic were generated by Iperf [14]. One cross traffic stream rate is 50 [Mbit/s]; packets are generated at a constant interval. The stream is UDP and datagram size is 1470 [bytes]. In Figure 11, the total data rate of cross traffic streams to node 4 was 100 [Mbit/s], and the total data rate of non-cross traffic streams to the node was set to 400, 500, 600, 700, and 800 [Mbit/s].

Sending interval of request packets was set to 10 [msec]. In this experiment, backward traffic was set to zero to evaluate the basic characteristics of the proposed scheme, i.e. link utilization is fully specified by Eq. (14).

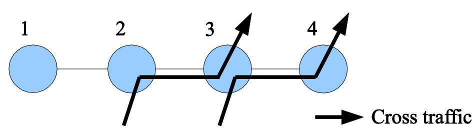

Fig. 10. Topology with traffic pattern 1

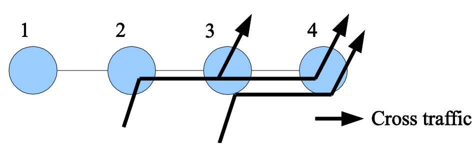

Fig. 11. Topology with traffic pattern 2

Table I. Detail of nodes.

OS (Kernel)	Linux Ubuntu Server 10.04 (2.6.32-24-generic-pae)
CPU	AMD Phenom(TM) II X6 1090T Processor 3.2 GHz
NIC	Intel(R) Gigabit ET2 Dual Port Server Adapter

Apparent probability of minimum processing

The apparent probability of processing delay is obtained by measuring RTT under the no load condition. Figure 5 plots the probability of processing delay. Eq.(10) and Figure 5 indicate that g_{min}^{proc} changes with range δT. Table II shows the values of g_{min}^{proc} in each range in Figure 5.

Table II. Values of g_{min}^{proc} in each range.

Range δT	Values of g_{min}^{proc}
0.02	0.067
0.05	0.386
0.10	0.991
0.20	1.000

Estimation accuracy of link utilization

Let U_{act} be the actual link utilization. Let U_{est} be the link utilization estimated by the proposed scheme or the original scheme. The deviation between U_{act} and U_{est}, Δ, is defined by,

$$\Delta = |U_{act} - U_{est}|. \quad (15)$$

Figure 12 shows Δs with the revised and original schemes under traffic pattern 1. δT was set to 0.05 [msec]. The revision in the proposed scheme is effective at low loads since Δ was reduced by 0.2. In contrast, the revision is not effective at high loads. This is because the processing delay jitter significantly impacts the probability of minimum RTTs at low loads. The average Δ at high load ($U = 0.8, 0.9$) is smaller than that at low load. At high loads, Δ is under 0.1. This is because g_{min}^{m} becomes small value due to the strong effect of queuing delay at high loads.

Figure 13 shows Δs with the revised and original schemes under traffic pattern 2. It shows the same results as Figure 12.

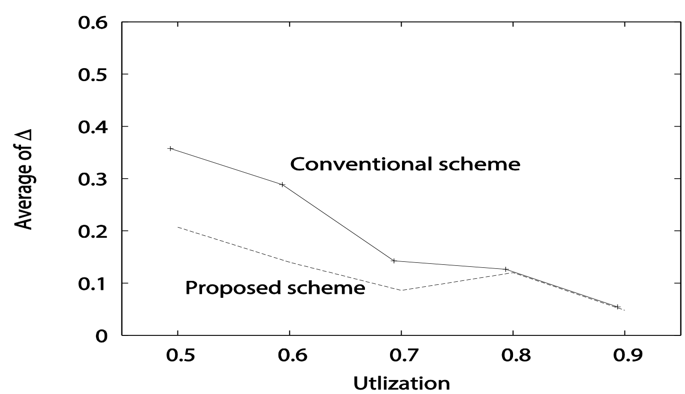

Fig. 12. Estimation accuracy under traffic pattern 1 (δT is 0.05 [msec]).

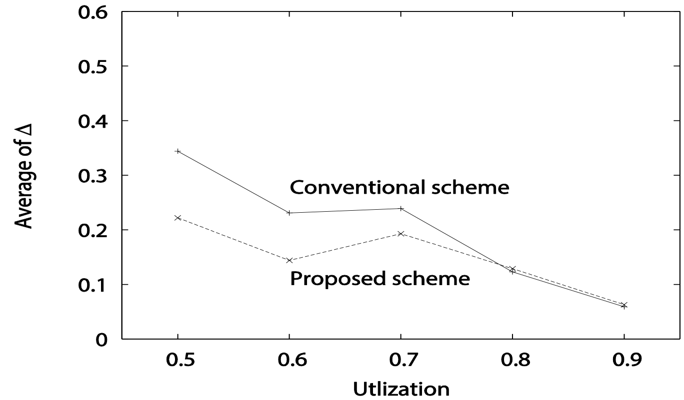

Fig. 13. Estimation accuracy under traffic pattern 1 (δT is 0.05 [msec]).

Figure 14 shows the relationship between Δ with the proposed scheme and range size δT under traffic pattern 1. In this relationship, we want to find the effective range in Figure 9. The desired range δT determined so that the the accuracy to estimate the probability of the minimum RTT can be maximized considering possible parameters. The highest accuracy is realized in the desired range. In most cases, the average of Δ is smallest when δT lies in the range of 0.05 to 0.2 [msec]. Δ shows worse result when range size exceeds 0.3 [msec]. This is because large range sizes pick up many minimum RTTs and g^m_{min} is no longer exact yielding inaccurate U_{re} estimations. Also, Δ degrades when range size is 0.02 [msec]. The reason is as follows: small δT can not pick up all minimum RTTs and g^m_{min} is no longer exact yielding inaccurate U_{re} estimations. The deviation between revised utilization and actual utilization worsens as the accuracy degrades.

Figure 15 plots the relationship between Δ with the proposed scheme and each range δT under traffic pattern 2. This result is similar to that in Figure 10. In most cases, the average of Δ is smallest when δT lies in the range 0.05 to 0.1 [msec]. Δ degrades when range size exceeds 0.3 [msec] or is under 0.02 [msec].

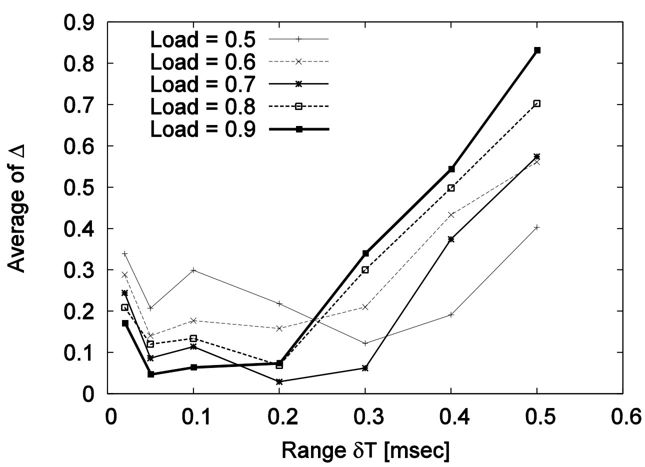

Fig. 14. Relationship between Δ and δT under traffic pattern 1.

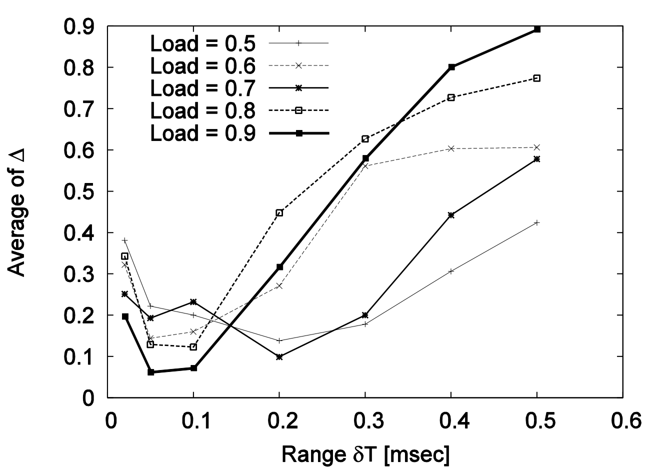

Fig. 15. Relationship between Δ and δT under traffic pattern 2.

These results show that the proposed scheme estimates link utilization with an accuracy of 10\% at high loads and the revision made to handle low loads is effective with 0.2 reduction in deviation from the actual value. In addition, the size of δT effects estimation accuracy and results show that suitable δT values run from 0.05 to 0.1 [msec].

VI. CONCLUSION

This paper has proposed a link utilization estimation scheme based on Round-Trip Time (RTT) measurements with consideration of processing delay jitter and practical timer granularity.

Link utilization is obtained by determining the probability of the minimum RTTs on a link when the RTT variation is caused by only queuing delay. The original scheme measures two RTTs from a source host to both end nodes of the targeted link to obtain their probabilities of minimum RTTs. The link utilization rate is estimated by the measured RTT values. This scheme assumes that queuing delay is variable while the other delay factors such as processing delay are fixed; timer granularity is nano-second order to determine the minimum RTTs.

However, real networks exhibit processing delay jitter and large timer granularity. Processing delay jitter is another source of RTT variation, and the observer cannot determine whether delay is caused by queuing or processing. Measuring the minimum RTTs is difficult with the current kernel because its timer granularity, of the order of micro-seconds, does not allow accurate RTT classification as minimum or non-minimum delay values.

The proposed scheme eliminates the processing delay jitter and estimates probability of the actual minimum RTT even though timer granularity is of the order of micro-seconds. The proposed scheme eliminates processing delay jitter with division by an appearance probability value of the processing delays at the ingress and egress nodes of the targeted link from the estimated RTTs on the target link. To estimate the minimum RTTs with timer granularity of micro-second order, the proposed scheme sets a suitable range from estimated RTTs on the target link. Experiments have shown that the proposed scheme effectively eliminates the jitter at low loads with micro-second order timer granularity..

REFERENCES

[1] E. Oki, ``On Optimization of Load-Balanced IP Routing for Hose Model,'' Progress in Informatics, no. 8, pp. 115-123, Mar. 2011.

[2] N. Kitsuwan, P. Pavarangkoon, and E. Oki, ``Design and Implementation of A Scheme based on Link-Load Ranking to Reduce Advertisements in OSPF-TE Networks,'' IEICE Communications Express, vol. 2, no. 6, pp. 251-256, Jun. 2013.

[3] J. Case, J. M. Fedor, M. Schoffstall, and J. Davin. ``A Simple Network Management Protocol (SNMP)'', RFC1157, May 1990.

[4] B. Melander, M. Bjorkman, and P. Gunningberg, ``A New End-to-End Probing and Analysis Method for Estimating Bandwidth Bottlenecks,'' Proc. GLOBECOM'00, 2000, San Francisco, USA, pp. 415-420.

[5] N. Hu, and P. Steenkiste, ``Estimating Available Bandwidth Using Packet Pair Probing," Computer Science Technical Reports, Carnegie Mellon Univ., 2002
[6] X. Liu, K. Ravindran, and D. Loguinov ``Multi-Hop Probing Asymptotics in Available Bandwidth Estimation: Stochastic Analysis," Proc. the 5th ACM SIGCOMM, California, USA, 2005, pp. 15-28.
[7] M. Jain, and C. Dovrolis, ``End-to-End Available Bandwidth: Measurement Methodology, Dynamics, and Relation with TCP Through-put," Proc. the 2002 SIGCOMM, Pennsylvania, USA, 2002, pp. 295-308.
[8] V. Ribeiro, R. Riedi, R. Baraniuk, J. Navratil, and L. Cottrell, ``pathChirp: Efficient Available Bandwidth Estimation for Network Paths," Passive and Active Measurement Workshop, 2003.
[9] M. Jain, and C. Dovrolis, ``End-to-end Estimation of the Available Bandwidth Variation Range," Proc. the 2005 ACM SIGMETRICS, 2005, Banff, Alberta, Canada, pp.265-276.
[10] C. Dorolis, ``Packet-Dispersion Techniques and a Capacity-Estimation Methodology," IEEE/ACM Trans., 2004, 12, (6)., pp. 963-977
[11] K. Igai, and E. Oki, ``A Simple Link-Utilization Estimation Scheme Based on RTT Measurement." Proc. IEEE-ISAS 2011, Yokohama, Japan, 2011, pp.266-270.
[12] K. Igai, and E. Oki, ``A Simple Estimation Scheme for Upper Bound of Link Utilization Based on RTT Measurement." Cyber Journals: Journal of Selected Areas in Telecommunications (JSAT)., August 2011 Edition, 2011, pp.10-16.
[13] J. Postel, ``Internet Control Message Protocol," RFC792, Sept. 1981.
[14] ``Iperf" http://sourceforge.net/projects/iperf/ , accessed August 2011.

Kiyofumi Igai received the B.E. and M.E. degrees from the University of Electro-Communications, Tokyo, Japan, 2011 and 2013, respectively. His research interests are network measurement and network tomography in particular estimation schemes for available bandwidth or link utilization.

Eiji Oki is a Professor at the University of Electro-Communications, Tokyo, Japan. He received the B.E. and M.E. degrees in instrumentation engineering and a Ph.D. degree in electrical engineering from Keio University, Yokohama, Japan, in 1991, 1993, and 1999, respectively. In 1993, he joined Nippon Telegraph and Telephone Corporation (NTT) Communication Switching Laboratories, Tokyo, Japan. He has been researching network design and control, traffic-control methods, and high-speed switching systems. From 2000 to 2001, he was a Visiting Scholar at the Polytechnic Institute of New York University, Brooklyn, New York, where he was involved in designing terabit switch/router systems. He was engaged in researching and developing high-speed optical IP backbone networks with NTT Laboratories. He joined the University of Electro-Communications, Tokyo, Japan, in July 2008. He has been active in standardization of path computation element (PCE) and GMPLS in the IETF. He wrote more than ten IETF RFCs and drafts. Prof. Oki was the recipient of the 1998 Switching System Research Award and the 1999 Excellent Paper Award presented by IEICE, the 2001 Asia-Pacific Outstanding Young Researcher Award presented by IEEE Communications Society for his contribution to broadband network, ATM, and optical IP technologies, and the 2010 Telecom System Technology Prize by the Telecommunications Advanced Foundation. He has authored/co-authored four books, Broadband Packet Switching Technologies, published by John Wiley, New York, in 2001, GMPLS Technologies, published by CRC Press, Boca Raton, FL, in 2005, Advanced Internet Protocols, Services, and Applications, published by Wiley, New York, in 2012, and Linear Programming and Algorithms for Communication Networks, CRC Press, Boca Raton, FL, in 2012. He is an IEEE Fellow.