A Fair Scalable Inter-Domain TCP Marker for Multiple Domain DiffServ Networks

Kyeong Hur and Doo-Seop Eom

Abstract: **The differentiated services (DiffServ) is proposed to provide packet level service differentiations in a scalable manner. To provide an end-to-end service differentiation to users having a connection over multiple domains, as well as a flow marker, an intermediate marker is necessary at the edge routers, and it should not be operated at a flow level due to a scalability problem. Due to this operation requirement, the intermediate marker has a fairness problem among the transmission control protocol (TCP) flows since TCP flows have intrinsically unfair throughputs due to the TCP's congestion control algorithm. Moreover, it is very difficult to resolve this problem without individual flow state information such as round trip time (RTT) and sending rate of each flow. In this paper, to resolve this TCP fairness problem of an intermediate marker, we propose a fair scalable marker (FSM) as an intermediate marker which works with a source flow three color marker (sf-TCM) operating as a host source marker. The proposed fair scalable marker improves the fairness among the TCP flows with different RTTs without per-flow management. Through the simulations, we show that the FSM can improve TCP fairness as well as link utilization in multiple domain DiffServ networks.**

Index Terms: **Assured services, demotion, fairness, promotion.**

I. INTRODUCTION

Due to the increase of multimedia and real time applications, a wide range of quality of service (QoS) is needed on IP based networks [1]. The Internet Engineering Task Force (IETF) has proposed two major service models; integrated services (IntServ) and differentiated services (DiffServ) [2]–[5]. DiffServ provides simple and predefined per-hop behavior (PHB) level service differentiation, and scalability is achieved by moving complicated functionalities such as per-flow marking, aggregate marking, shaping, and policing to the edge routers and leaving the core routers with simple functionality. The IETF has defined one class for expedited forwarding (EF) PHB and four classes for assured forwarding (AF) PHB [6], [7]. EF PHB was proposed as premium services. It is adequate for real time services such as IP telephony and video conferences. AF PHB was proposed to use RIO and it allows an Internet service provider (ISP) to provide different levels of forwarding assurances according to the user profile [8].

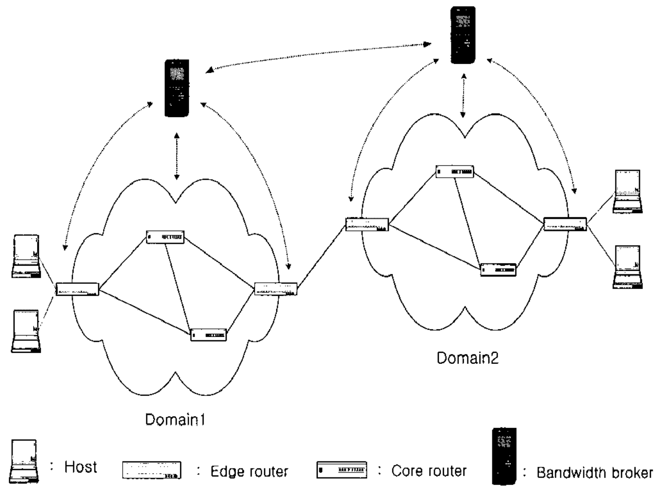

Fig. 1. DiffServ architecture.

Fig. 1 shows the typical DiffServ architecture. In assured services, the packets of a flow that obey the service profile are marked as in-profile (*green*) and the packets that are beyond the service profile are marked as out-of-profile (*red*). *Green* packets that have the lower drop precedence than *red* packets are properly protected at the time of congestion. Recent works on DiffServ mostly deal with per-flow management at edge routers to provide a QoS guarantee to users and those works require costly packet classifications, as well as complicated flow table maintenances to monitor the states of each flow. And some works have been studied to provide service differentiations to users with different classes of AF [9]–[11].

The Internet is comprised of multiple interconnected autonomous domains [12] and thus a connection can span through a path involving one or more domains. The DiffServ code point (DSCP) of a packet may be changed when it crosses the boundary of two domains. For example, in Fig. 1, a *green* packet in Domain 1 may be a *red* packet in Domain 2 if it disobeys the service profile at the ingressive edge router in Domain 2. So, if we want to guarantee an end-to-end minimum throughput of the connection, we have to make sure that the aggregate traffic along the path does not exceed any of the interdomain negotiated service level agreements (e.g., the traffic rate). However, this is very difficult to ensure since the interdomain service level agreement (SLA) is not renegotiated at the initiation of each new connection. For assured services, the interdomain contract rates are usually negotiated based on statistical estimation or updated periodically to avoid signaling overhead and the scalability problem [13], [14]. So, the instantaneous aggregate *green* traffic rate may be higher or lower than the negotiated rate determined by

Manuscript received September 28, 2005; approved for publication by Dongman Lee, Division III Editor, February 15, 2008.

This work was supported in part by the MKE (Ministry of Knowledge Economy), Korea, under the ITRC (Information Technology Research Center) support program supervised by the IITA (Institute for Information Technology Advancement) and in part by the Korea Research Foundation Grant funded by the Korean Government(MOEHRD) (KRF-2007-331-D00283).

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the interdomain SLA. If the aggregate *green* traffic rate is higher than the negotiated rate, the phase effect [14], [15] may occur and it could bring about unfairness problem among the flows. For example, suppose the packets from the flows (1 and 2) are interleaved in the following pattern $(1, 2, 1, 2, 1, 2, \cdots)$ and the remaining tokens in the token bucket are all consumed. In this situation, when the incoming aggregate *green* traffic rate from the two flows is twice the negotiated rate, a specific flow may consume all tokens and all packets from the other flow may be re-marked to *red* packets. On the other hand, if the aggregate *green* traffic rate is lower than the negotiated rate, the bandwidth may be wasted.

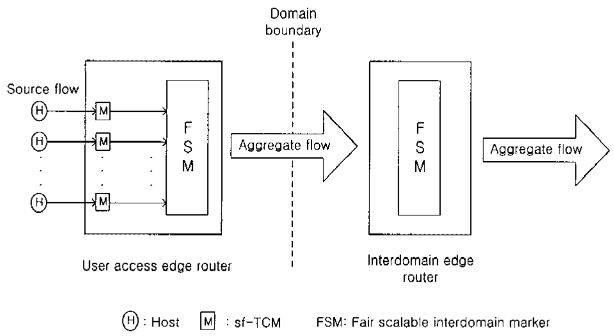
Concerning this problem, the random early demotion promotion (REDP) was proposed as an aggregate flow management [14]. It is an intermediate marker operated at user access (ingress) edge routers and interdomain edge routers. The token bucket of the REDP marker is divided into the demotion, balanced and promotion regions. When the incoming aggregate *green* traffic rate is higher than the negotiated rate, (i.e., the token consumption rate is higher than the token generation rate), the token level will be in the *demotion* region and *green* packets are randomly demoted to *yellow* packets to prevent the phase effect. If it is lower than the negotiated rate, the token level will be in the *promotion* region and *yellow* packets that were previously demoted are randomly promoted to *green* packets for increasing of link utilization. On the other hand, if the two rates are matched, the token level will stay in the *balanced* region and each packet is forwarded without changing the color.

We think that the main contribution of the REDP is the introduction of the packet demotion and promotion concept in DiffServ networks. That is, in the demotion situation, the REDP demotes *green* packets instead of dropping them so that they can be promoted in the following domains, to improve throughput, when the bandwidth is available to them. The REDP achieves very good user datagram protocol (UDP) fairness in demotion and promotion although it works at the aggregate flow level. However, it fails to give good fairness to the transmission control protocol (TCP) flows. In the case of the TCP flows, unlike the UDP flows, their transmission rates highly depend on the round trip time (RTT) due to the TCP's congestion control algorithm. This means that the TCP flow with a shorter RTT generates more *green* and *yellow* packets than the TCP flow with a longer RTT, which brings about the unfairness of throughput among the TCP flows with different RTTs as well as the lowering of link utilization. Note that it is very difficult to resolve this problem at the aggregate flow level since we must know the individual state information of each TCP flow such as RTT and TCP sending rate for improving TCP fairness, but such information is not available at the aggregate flow level. Basically, this is the reason why the REDP fails to give good fairness to the TCP flows. In the REDP, the probabilities for demotion and promotion are simply decided based on the token level; it is a function of remaining tokens. Therefore, all TCP flows belonging to the same aggregate flow have the same demotion and promotion probabilities.

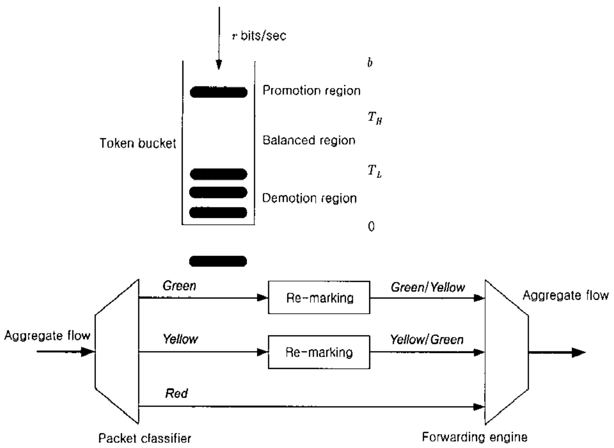
In [19], to resolve this TCP fairness problem of the REDP marker, the fair early drop (FED) interdomain marker was proposed. In the FED marker, instead of unfair packet droppings at the RIO buffers in the core routers, some of *green* packets of each TCP flow, which are likely to be demoted and dropped in the core routers, are fairly dropped based on current aggregate flow rates information. But, the performance of FED marker is very sensitive to the dropping probability so that it shows unstable performances. Furthermore, it performs this fair early droppings only at the ingressive edge router directly connected to the source host of a TCP flow. So, there still exists unfairness problem at domain boundaries where a large number of aggregate flows pass. Therefore, we need a more accurate and stable solution that is not an unstable packet dropping solution, to resolve the TCP fairness problem of the REDP marker at the aggregate flow level.

The key point of DiffServ is to provide fair throughputs among the DiffServ users by aggregated flow level handling mechanisms. Thus, if the network condition is exact provisioning, DiffServ network guarantees user profile rates. The other hands, if network condition is under-provisioning, it guarantees fair throughputs among the same profile users by sharing network bandwidth, irrespective of their RTTs. For the same reason, in over-provisioning networks, it is desirable for users to get their throughputs by sharing network bandwidth fairly. But as is explained before, TCP fairness problem is very difficult problem to resolve at aggregated level.

So, in this paper, to resolve the TCP fairness problem of an intermediate marker such as the REDP, we propose an fair scalable marker (FSM) as an intermediate marker. Also, it works with a source flow three color marker (sf-TCM) proposed as a per-flow marker for a source flow. Since the FSM is an intermediate marker, unlike a per-flow marker for a source flow, it is required to operate at the aggregate flow level without per-flow state information. The fundamental assumption of the proposed FSM intermediate marker is that the relative transmission rates among TCP flows do not remain constant at the intermediate marker. So, at certain times, the TCP flow with a longer RTT sends relatively more packets than the TCP flow with a shorter RTT. In that case, if we decrease the demotion probability and increase the promotion probability at those times, the TCP flow with a longer RTT gets more advantages from demotion and promotion than the TCP flow with a shorter RTT, which means the fairness among the TCP flows is improved. Similarly, to improve TCP fairness, we increase the demotion probability and decrease the promotion probability when the TCP flow with a shorter RTT sends relatively more packets than the TCP flow with a longer RTT. Then, the problem is how to determine the demotion and promotion probabilities for improving TCP fairness using only the aggregate flow state information. For this purpose, we measure the *green*, *yellow*, and *red*, and total rates at the edge router, and we infer the individual TCP flow states from the measurement results. Finally, using the inferred individual TCP flow state information, the demotion and promotion probabilities are determined to improve the fairness among the TCP flows with different RTTs without per-flow managements. But, it is hard to know that individual TCP flow sate at aggregate flow level. Because of environments of RTT and sending widow state for each TCP are not same at the aggregation level. Our proposed algorithm and marker aims to find the probabilistic behaviors of each TCP flow at the aggregated flow level by



(a)



(b)

Fig. 2. The structure of the proposed scheme at (a) edge router and (b) REDP marker.

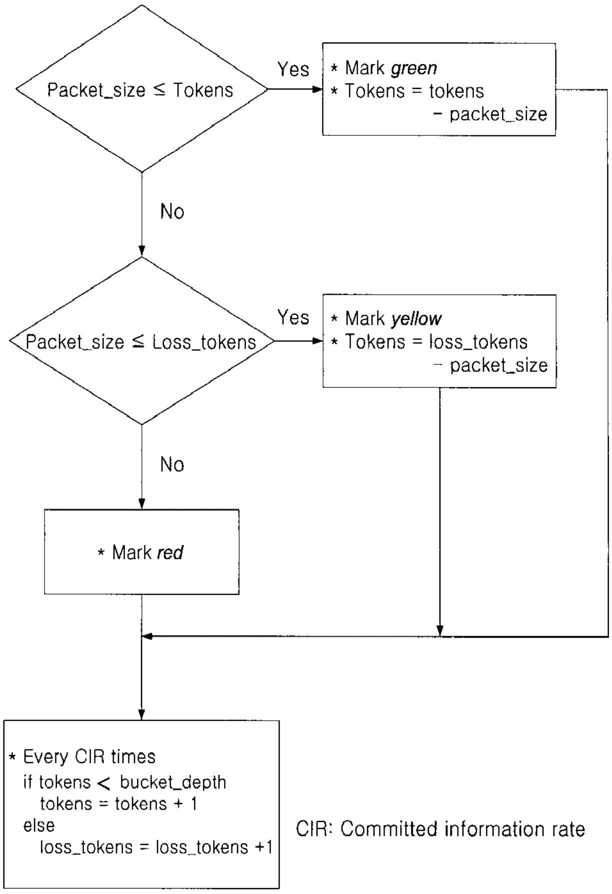


Fig. 3. The proposed sf-TCM algorithm.

observing packet arrival rates of total, *red*, *yellow*, and *green* packets. "It infers the individual TCP flow states from the measurement results" does not mean "must be" but "may be" with some high probability.

Also, in this paper, we use an sf-TCM. This sf-TCM marker can express the burst characteristics of greedy TCP flows by generating *yellow* packets whenever a TCP source generates burst packets. Then, the proposed FSM interdomain marker controls the demotion and promotion probabilities at all domain boundaries using more accurate aggregate TCP flow information from the sf-TCM markers. The rest of the paper is organized as follows. In Section II, we explain our proposed scheme. In Section III, we evaluate the performance of proposed scheme under the various network environments. Finally, Section IV concludes the paper.

II. THE PROPOSED FSM SCHEME

Our proposed scheme is composed of an sf-TCM and an FSM as shown in Fig. 2. A flow marker is necessary for monitoring a source flow and it initially marks the packets from a source flow according to its traffic profile. Such a flow marker should be as simple as possible because the number of flows to be handled is large. The proposed sf-TCM is a flow marker that monitors a source flow and marks the packets from the flow as *green*, *yellow*, or red. It is a token bucket based marker. The proposed FSM is an intermediate marker monitoring the aggregate traffic. According to incoming traffic situation, it fairly performs packet demotions and promotions. We accept the idea in the REDP scheme that a *yellow* packet is the temporally demoted *green* packet, but a *yellow* packet is also the packet that consumes the loss tokens of the proposed sf-TCM. A *yellow* packet is never demoted to red and a *red* packet is never promoted to *yellow*. In our scheme, to use the RIO buffer management, a *yellow* packet has the same drop precedence as a *red* packet at core routers because it disobeys user traffic profile. However, an FSM monitors not each flow profile but an aggregate traffic profile. Thus, if there are enough tokens to promote *yellow* packets to *green* packets, a *yellow* packet can consume the token in token bucket of FSM, and it thus obeys the aggregate profile. An sf-TCM and an FSM are operated independently as shown in Fig. 2 and they have a simple structure, so there are no scalability problems and no signaling overhead in multiple domain environments as well as in a single domain environment.

A. Source Flow Three Color Marker (sf-TCM)

For assured services, in general, the packets of each source flow are initially marked as *green* or *red* according to their profile obedience with ISP. A simple method for monitoring the traffic profile obedience of the flow is to use a token bucket marker. In the token bucket marker, the token filling (generation) rate is set to be the user contract rate and if there are enough tokens in the token bucket corresponding to the packet size, the packet from a source flow is marked as *green*. Otherwise, it is regarded as disobeying the traffic profile and packets are marked as red. In this way, we can properly protect the *green* packets and drop the *red* packets at the time of congestion at core routers. Currently, the most used transport protocols in the Internet are UDP and TCP. A UDP flow has the property of a constant traffic rate; the instant traffic rate is the same as the average traffic rate. Thus, there is no token loss as long as the traffic rate of a flow is higher than or equal to the contract rate. In the case of TCP, it sends packets in a burst within an RTT interval; the instant traffic rate is not same as the average traffic rate. Thus, there are some time intervals during which no traffic is generated from a user source. During these time intervals, a token loss is likely to occur from the token bucket; thus, it is hard to guarantee a minimum throughput to a user [16]. In order to improve this situation, we use a three color marking process instead of the normal two color marking process at each source flow marker. For this purpose, we propose a sf-TCM based on a simple token bucket algorithm. The operation algorithm of the proposed sf-TCM is shown in Fig. 3.

A packet is marked as *green* if there are enough tokens in the token bucket. It is marked as *yellow* if there are not enough tokens in the token bucket but there are enough loss tokens. Otherwise it is marked as red. If traffic is not generated from a flow source and thus the token bucket becomes full, further generated tokens are counted in the number of loss tokens as shown in Fig. 3. Thus, loss tokens are reused to mark the packet that cannot consume the tokens, as *yellow*. Note that if we set the token bucket size of sf-TCM as one packet and a token size corresponds to a packet size, *green* packets are generated when the instant traffic rate (short term average traffic rate) obeys the user traffic profile and *yellow* packets are generated when the instant traffic rate disobeys the user traffic profile but the long term average traffic rate obeys the user traffic profile. Given that 'r' tokens are generated per second, i.e., the contracted average traffic rate of source flow is 'r' (packets/s) and there are 'n' loss tokens, if 'n' burst packets are generated from a source flow source, then they can consume the 'n' loss tokens. Therefore, the long term average traffic rate, corresponding to the sum of *green* and *yellow* packets generated per second, is calculated as follows:

We can consider that there are some times during which no traffic is generated from the source flow source, and the total period of these times is equal to 'n/r' (s) because the generation time of a loss token is '$1/r$' (s). Also, we can consider that in the other times, the traffic rate of the source flow source is equal to or greater than the contracted average traffic rate, but 'r' *green* packets are generated per second during these times. However, the 'n' burst packets consume the 'n' loss tokens, and thus they generate 'n' *yellow* packets. This means that the long term average traffic rate is 'r' (packets/s) because we can consider that 'n' *yellow* packets are generated during 'n/r' (s).

Note that if a packet is marked as *yellow* in this way, the throughput of *green* packets of a flow cannot grow beyond the contracted rate even when all of the *yellow* packets are promoted to *green* packets in the following domains. This is because *yellow* packets can consume only the loss tokens. That is, the sf-TCM ensures that the packets from a flow never disobey its traffic profile which is agreed upon between a user and an ISP. However, the throughput of a TCP flow can be improved because *yellow* packets have a chance to be promoted to *green* in the following domains.

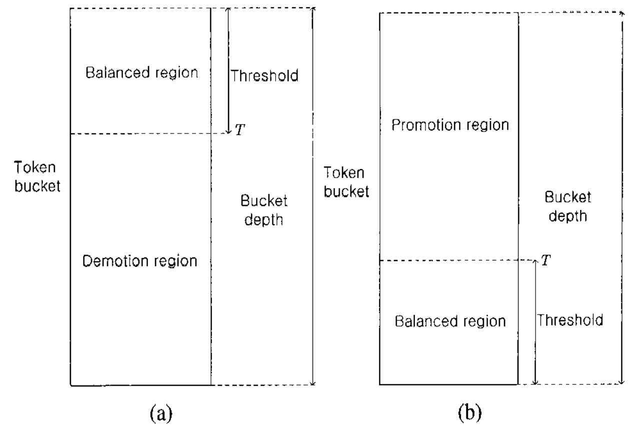


Fig. 4. The operation modes of FSM mode: (a) Demotion mode and (b) promotion mode.

B. Fair Scalable Marker (FSM)

The proposed FSM monitors aggregate flows at the edge routers and performs packet demotions and promotions according to the incoming traffic situation. If the rate of *green* packets is beyond the aggregate contract rate or negotiated rate between two domains, it enters the demotion mode. That is, if the token level of bucket is within the *demotion* region as shown in Fig. 4, the FSM fairly demotes *green* packets to *yellow* to prevent the phase effect. On the other hand, if the *green* rate is under the contract rate, it enters the promotion region. That is, if the token level is within the promotion region, it fairly promotes *yellow* packets to *green* to increase link utilization. The *balanced* region is needed to prevent unnecessary packet demotions and promotions [14]. That is, even in the demotion situation, if the token level of the token bucket is within the balanced region, the FSM does not perform the packet demotion to prevent lowering of link utilization and throughput. Similarly, even in the promotion situation, if the token level is within the balanced region, it does not perform a packet promotion to enable burst *green* packets consuming the remaining tokens in the token bucket.

The reason of using two boxes instead one box in Fig. 4 is that; in case of the demotion situation, if we demote all *green* packets violating the contract rate between two domains, many packets are dropped in a congested router, which situation will decrease TCP transmission rates and link utilization. In the demotion region, a *green* packet must be demoted, but if we use some demotion protection region that is a *balanced* region of a small size threshold, the problem of decrease of link utilization is solved in a probability. The other hands, if we promote all *green* packets, TCP transmission rates will be increased in case of the promotion situation. But this situation, many packets are dropped in a congested router because TCP connections having the promoted packets are increased in a probability. It results in an oscillation of TCP throughputs. This is another problem of decrease of link utilization. If we use some promotion protection region that is a *balanced* region of a small size threshold, the problem of oscillation of TCP throughputs is solved in a probability. In the demotion situation, tokens in the token bucket are decreasing so that the *balanced* region is in top of the token

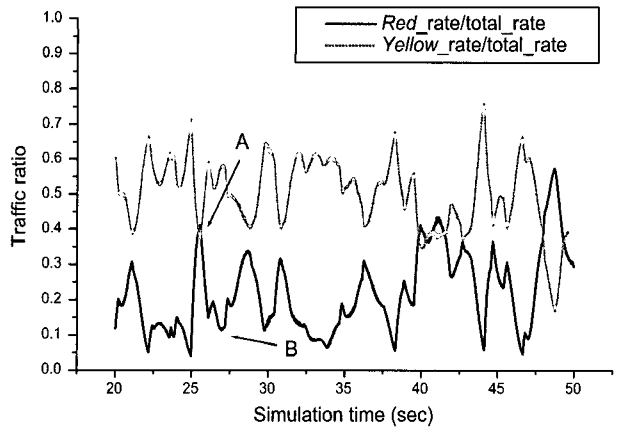


Fig. 5. Traffic rate ratios at an edge router under an over-provisioning case.

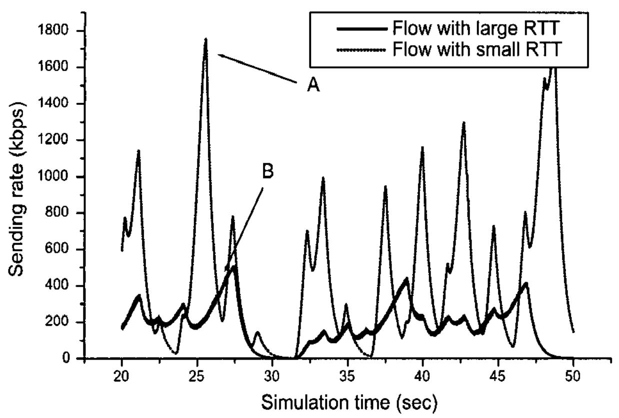


Fig. 6. Sending rates of each flow under an over-provisioning case.

bucket. The other hand, in the promotion situation, tokens in the token bucket is increasing so that the *balanced* region is in bottom of the token bucket. This is why we use two boxes instead of one.

As explained in Section I, the FSM measures *green*, *yellow*, and *red*, and total rates at an edge router, then it infers the individual TCP flow states from the measurement results and determines the demotion and promotion probabilities using the inferred individual TCP flow state information. This is done to improve TCP fairness using only the aggregate flow state information. In the following, we explain the reason why this works well.

C. FSM Mode in an Over-Provisioning Case

We first consider the over-provisioning case, i.e., the sum of the user contract rates is lower than the aggregate contract rate. In this case, a promotion situation is dominant. Fig. 5 shows a ratio of the incoming *red* traffic rate to the total incoming traffic rate and a ratio of the incoming *yellow* traffic rate to the total incoming traffic rate under an over-provisioning situation at an edge router. Fig. 6 shows the sending rates of a TCP flow with a shorter RTT and a TCP flow with a longer RTT in Fig. 5 situation, respectively. It is well known that the throughput of TCP is oscillating due to its congestion control algorithm. Thus, the ratio of the incoming *red* traffic rate to the total incoming traffic rate at an edge router is also oscillated. If the ratio is close to the maximum value 'A' as shown in Fig. 5, we can infer that the probability that the sending rates of the TCP flows with a shorter RTT are also close to the maximum value 'A' as shown in Fig. 6. That is, the TCP flows with a shorter RTT determine the dynamics of the aggregate flow. Therefore, if we increase the promotion probability in this case, it may bring about unfairness because of the increased probability that the TCP flows with a shorter RTT will receive more promoted packets than the TCP flows with a longer RTT. This means that a packet promotion is not desirable. Therefore, it is desirable to lower the promotion probability in this case.

On the other hand, if the ratio is close to the minimum value 'B' as shown in Fig. 5, we can infer that the probability that the sending rates of the TCP flows with a shorter RTT are also close to the minimum value 'B' as shown in Fig. 6. This is because when the sending rates of the TCP flows with a shorter RTT reach maximum values, the packet dropping probability of the TCP flows with a shorter RTT becomes much higher than that of the TCP flows with a longer RTT, and thus the sending rates of the TCP flows with a shorter RTT become lower. But by this reduction of the sending rates, the available bandwidth for the TCP flows with a longer RTT increases, so that their sending rates become higher until the time corresponding to the minimum value 'B.' After that, the ratio increases again since the TCP flows with a shorter RTT begin recovering their previous transmission window. From the above observation, if we increase the promotion probability when a TCP flow with a longer RTT send relatively more packets than a TCP flow with a shorter RTT, a TCP flow with a longer RTT get more advantages from promotion than a TCP flow with a shorter RTT, which means that TCP fairness is improved. Similarly, we decrease the promotion probability when a TCP flow with a shorter RTT sends relatively more packets than a TCP flow with a longer RTT. Therefore, we can improve TCP fairness by using the ratio of the incoming *red* traffic rate to the total incoming traffic rate without a per-flow management. Unlike UDP flows, it is well known that TCP transmission rates are not constant but oscillates due to their congestion control algorithm. So, their transmission rates highly depend on the RTT. TCP transmission interval of a small RTT flow is shorter than that of a large RTT flow. It means that tokens in the token bucket of a TCP flow with a smaller RTT have a lower probability of token loss than a TCP flow with a larger RTT in our proposed sf-TCM mechanism. That is, a TCP flow with a smaller RTT have generated *yellow* packets more than a TCP flow with a larger RTT. In Fig. 5, the situation of a *red* rate becoming higher than the *yellow* rate reflects that TCP flows with a smaller RTT is more dominant than those with a larger RTT in the aggregated flow level. In Fig. 6, in the time of a *red* rate becoming higher than the *yellow* rate of Fig. 5, TCP sending rates with a smaller RTT is much higher than those with a larger RTT. For the over-provisioning case, the packet promotion probability P_{promo} is determined as follows:

$$P_{\text{promo}} = 1 - \frac{red\_\text{rate}}{\text{total\_rate}} \cdot \left(1 - \frac{\text{number\_of\_tokens} - \text{threshold}}{\text{bucket\_depth} - \text{threshold}}\right). \quad (1)$$

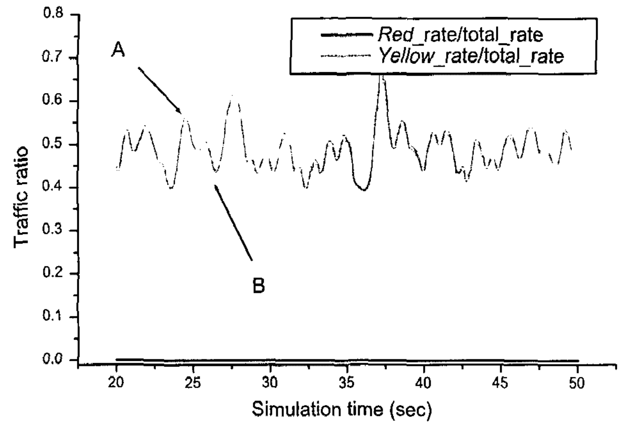


Fig. 7. Traffic rate ratios at edge router under under-provisioning.

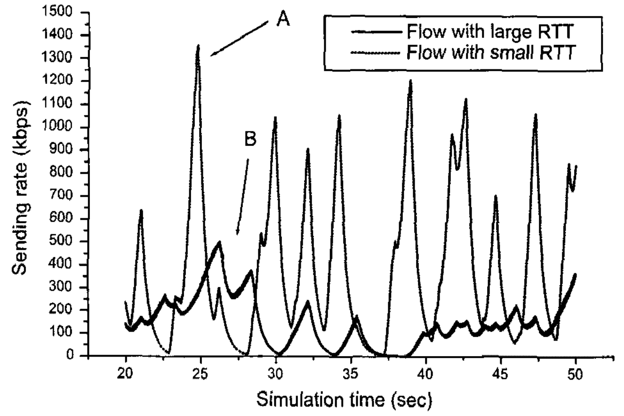


Fig. 8. Sending rates of each flow under under-provisioning.

D. FSM Mode in an Under-Provisioning Case

We next consider the under-provisioning case, i.e., the sum of the user contract rates is higher than the aggregate contract rate. Note that even in the under-provisioning case, due to the TCP congestion control algorithm, the incoming *green* traffic rate is hard to become beyond the aggregate contract rate at an edge router. Therefore, a promotion situation is dominant. Fig. 7 shows a ratio of the incoming *red* traffic rate to the total incoming traffic rate and a ratio of the incoming *yellow* traffic rate to the total incoming rate under an under-provisioning situation at an edge router. Fig. 8 shows the sending rates of a TCP flow with a shorter RTT and a TCP flow with a longer RTT in Fig. 7 situation. If the network is under-provisioned, the packets from a TCP flow cannot consume all of the loss tokens in most cases and thus TCP flows can hardly generate any *red* traffic, as shown in Fig. 7. However, the TCP flows with a shorter RTT generate more *yellow* traffic than the TCP flows with a longer RTT. Therefore, unlike the over-provisioning case, the ratio of the incoming *yellow* traffic rate to the total incoming traffic rate should be used for determining the promotion probability when the network is under-provisioned. Similar to the over-provisioning case, it is desirable to lower the promotion probability when the ratio of the *yellow* traffic rate to the total traffic rate becomes higher. For the under-provisioning case, the packet promotion probability P_{promo} is determined as follows:

$$P_{\text{promo}} = 1 - \frac{yellow\_\text{rate}}{\text{total\_rate}} \cdot \left(1 - \frac{\text{number\_of\_tokens} - \text{threshold}}{\text{bucket\_depth} - \text{threshold}}\right). \quad (2)$$

In the expression of probability P_{promo}, we consider the aggregate flow rate as well as tokens in the token bucket. In the expression of parenthesis, the ratio reflects the token level in the promotion region linearly. And the expression of ratio, i.e., a *red* rate to total rate in the under-provisioning case or a *yellow* rate to total rate in the over-provisioning case, reflects an inferred individual TCP sending state. In the under-provisioning case, if the *yellow* rate to total rate is increased more, a TCP flow with a smaller RTT generates the packets more than a TCP flow with a larger RTT. Therefore, we decrease the promotion probability more if the ratio of *yellow* rate to total rate is increased more and the portion of the tokens in the *promotion* region is decreased more compared to tokens in the token bucket. For the same reason, if the ratio of *yellow* rate to total rate is decreased more and the portion of the tokens in the *promotion* region is increased more, we increase the promotion probability more. In the over-provisioning case, the promotion probability mechanism is the same as the under-provisioning case. Finally, we consider the case that the incoming *green* traffic rate is beyond the negotiated contract rate at an interdomain edge router. In our proposed scheme, *yellow* packets of source flows are promoted to *green* packets within the aggregate contract rate at a user access (ingressive) edge router. When their connections are extended over multiple domains, if the incoming *green* traffic rate is beyond the negotiated contract rate at interdomain edge routers, *green* packets should be demoted to *yellow* because of the phase effect problem as explained before. In the demotion situation, we don't need to separately consider the over-provisioning case and the under-provisioning case because the ratio of the incoming *green* traffic rate to the aggregate contract rate at an interdomain edge router also oscillates, according to input traffic conditions. In a demotion mode, the higher the incoming rate of *green* packets is, the more the demotion probability must be increased. For both provisioning cases, the packet demotion probability P_{demo} is determined as follows:

$$P_{\text{demo}} = \left(1 - \frac{\text{contract\_rate}}{green\_\text{rate}}\right) \cdot \left(1 - \frac{\text{number\_of\_tokens}}{\text{bucket\_depth} - \text{threshold}}\right). \quad (3)$$

In the expression of probability P_{demo}, we consider the aggregate flow rate as well as tokens in the token bucket. In the expression of the first parenthesis, the ratio reflects the inferred individual TCP sending state. And in the expression of the second parenthesis, the ratio reflects the token level in the *demotion* region linearly. If the ratio of contract rate to *green* rate is decreased more, a TCP flow with a smaller RTT generates the *green* packets more than a TCP flow with a larger RTT. Thus, we decrease the demotion probability more if the ratio of contract rate to *green* rate is increased more and if the portion of the tokens in the *demotion* region is increased more compared to

if contract_rate $\leq$ *green*_rate
  enter Promotion mode
  if threshold $\geq$ number_of_tokens
    enter balanced region
    $P_{\text{promo}} = 0$
  elseif threshold $<$ number_of_tokens
    enter promotion region
    if *red*_rate/total_rate > 0.01
      decision over-provisioning
      $R_{\text{promo}} = red\text{\_rate/total\_rate}$
    elseif red_rate/total_rate ≤ 0.01
      decision over – provisioning
    $$P_{\text{promo}} = 1 - P_{\text{promo}}\left(1 - \frac{\text{number\_of\_tokens} - \text{threshold}}{\text{bucket\_depth} - \text{threshold}}\right)$$
elseif contact_rate $<$ *green*_tokens
  enter Demotion mode
  if threshold $\geq$ token_depth $-$ number_of_tokens
    enter balanced region
    $P_{\text{demo}} = 0$
  elseif threshold $<$ number_of_tokens
    enter demotion region
    $R_{\text{demo}} = 1 - \text{contact\_rate}/green\text{\_rate}$
    $$P_{\text{demo}} = R_{\text{demo}}\left(1 - \frac{\text{number\_of\_tokens}}{\text{bucket\_depth} - \text{threshold}}\right)$$

Fig. 9. The proposed FSM algorithm.

tokens in the token bucket. For the same reason, we increase the demotion probability more if the ratio of contract rate to *green* rate is decreased more and if the portion of the tokens in the *demotion* region is decreased more. Note that the promotion and demotion probabilities are determined by using not only the traffic ratios but also the token level of the token bucket. This is because the token level reflects the difference between the incoming *green* traffic rate and the contract rate. For example, if there are enough tokens in the bucket, by making the promotion probability higher, we can increase link utilization in a promotion situation. On the other hand, by making the demotion probability lower, we can prevent a decrease of link utilization in a demotion situation.

E. FSM Operation Algorithms and Structure

Fig. 9 shows the re-marking probability decision pseudo codes for the proposed FSM. The operation mode is determined by a comparison between the contract rate and the rate of *green* packets. When the ratio of the incoming *red* traffic rate to the total incoming traffic rate at an edge router is under '0.01' in a promotion situation, as shown in the proposed FSM algorithm, we assume that it is an under-provisioning case. This is because there are some possibilities that TCP flows with a shorter RTT may send some *red* packets in spite of under-provisioning situations. The FSM uses a time sliding window (TSW) [8] to estimate the incoming traffic rates at edge routers. Whenever a packet is arrived at an edge router, it updates *green*, *yellow*, and *red* rates and total traffic rate simultaneously. That is because, when a *green* packet arrives, if we update only *green* rate and total rate, the update times of the others are relatively longer, which can bring about incorrect decisions of probabilities. In the FSM, if a *green* packet arrives, as well as *green* rate and total traffic rate, *yellow*, and *red* rates are also updated as if a null packet arrives. Fig. 10 shows the functional structure of FSM.

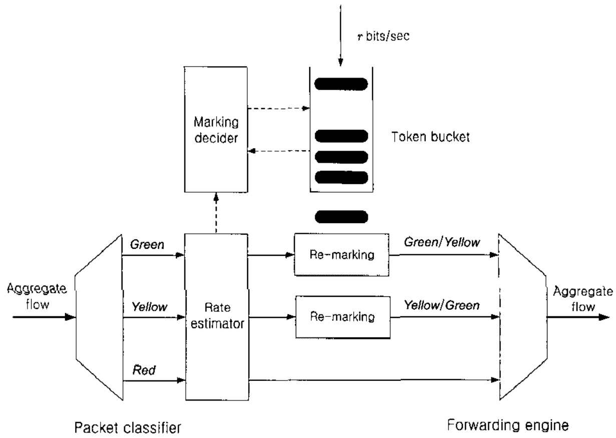


Fig. 10. The functional structure of FSM.

III. PERFORMANCE STUDY

In this section, we analyze the performance of the proposed scheme, in comparison with the REDP combined with a two color token bucket marker acting as a flow marker, and the REDP combined with the proposed sf-TCM. For a multiple domain environment and a single domain environment, we simulate the above three schemes under the over-provisioning, exact-provisioning and the under-provisioning cases, and we compare throughput, link utilization and fairness index [17] of three schemes

$$\text{Fairness index} = \left(\sum_{i=1}^{n} T_i\right)^2 \Bigg/ \left(N \sum_{i=1}^{n} {T_i}^2\right). \tag{4}$$

The fairness index is defined as the above expression and it is ranged from 0 to 1. N is the number of flows and T_i is the throughput of i flow. According to this definition, the closer the fairness index is to 1, the fairer is the throughput distribution of flows that have the same contract rate. We implemented a simple RIO queue [8] with parameters $(q_{\min}/q_{\max}/p_{\max}) = (45/60/0.02)$ for 'in' packets and $(20/40/0.2)$ for 'out' packets in ns2 simulator [18]. In all our simulations, we set the bucket depth as one packet for an source flow marker and we set that as 60 packets for an intermediate marker. In case of the REDP schemes, T_L is set to 15 packets, T_H is set to 45 packets, MAX_{demo} is set to 0.5, and $\text{MAX}_{\text{promo}}$ is set to 0.5 [14]. In case of the FSM scheme, we set the range of the *balanced* region as 10 packets and the sliding window size for measuring traffic rates is set to 0.1 sec to smooth out the incoming traffic rates.

A. Single Domain Scenario

Fig. 11 shows the single domain model where the aggregate contract rate is 4 Mbps at E1 router and the bottleneck link bandwidth is 4 Mbps. There are 10 TCP hosts and each host contracts 267 kbps for the over-provisioning case (150% provisioning), 400 kbps for the exact-provisioning case (100% provisioning)

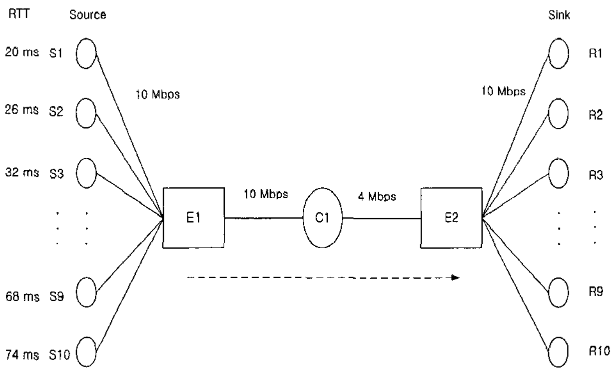


Fig. 11. The simulation model for the single domain case.

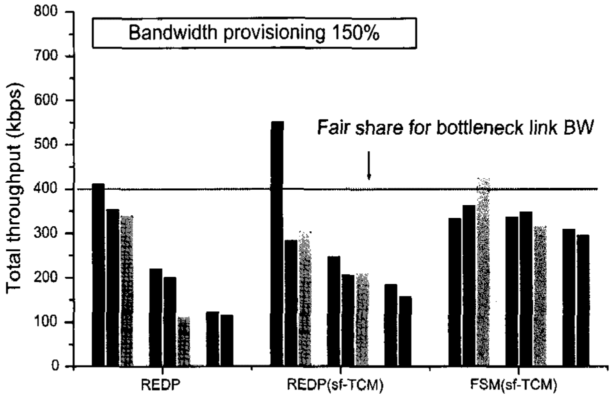


Fig. 12. Over-provisioning case in a single domain.

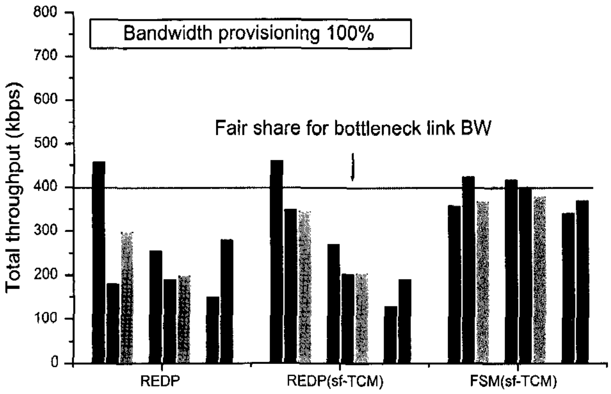


Fig. 13. Exact-provisioning case in a single domain.

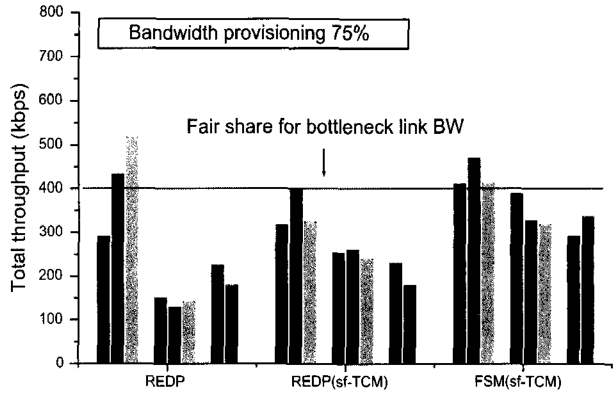


Fig. 14. Under-provisioning case in a single domain.

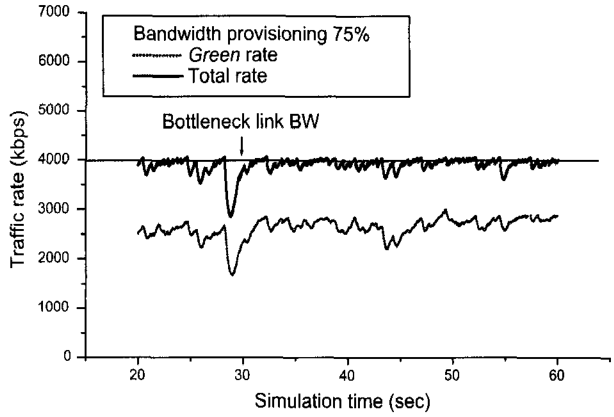


Fig. 15. Incoming traffic rate at E1 edge router for under-provisioning case.

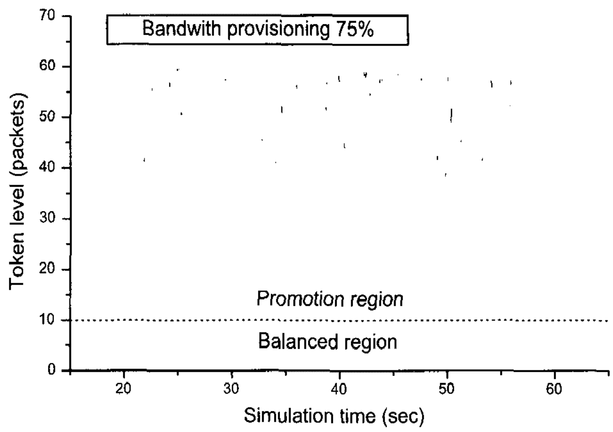


Fig. 16. Token level variations in token bucket in FSM at E1 edge router.

and 533 kbps for the under-provisioning case (75% provisioning). The RTT of each flow is ranged from 20 ms to 74 ms with 6 ms differences.

Figs. 12–14 show the average total throughput of each host for the three cases. As shown in Figs. 12–14, our proposed scheme shows the best results for all the cases. The REDP combined with the two color token bucket marker, which corresponds to the original REDP [14], cannot increase link utilization and TCP fairness because each flow marker is based on a simple two color token bucket. However, in case of the REDP combined with the proposed sf-TCM, compared with the above original REDP scheme, as well as link utilization, TCP fairness is improved to some degree because *yellow* packet is generated from each flow source by the sf-TCM and it has a chance to be promoted to *green* packet at an edge router. This means that the throughput of each flow can be improved. However, compared with our proposed scheme, the performance improvement is restricted, due to the simple probability decision of the REDP marker that depends only on the token level as explained before.

Fig. 15 shows the total incoming traffic rate and the incoming *green* traffic rate at E1 edge router for the under-provisioning case. Due to the TCP congestion control algorithm, the total incoming traffic rate is adjusted to the bottleneck link bandwidth. Thus, although the sum of the user contract rates is higher than the aggregate contract rate (under-provisioning), the incoming

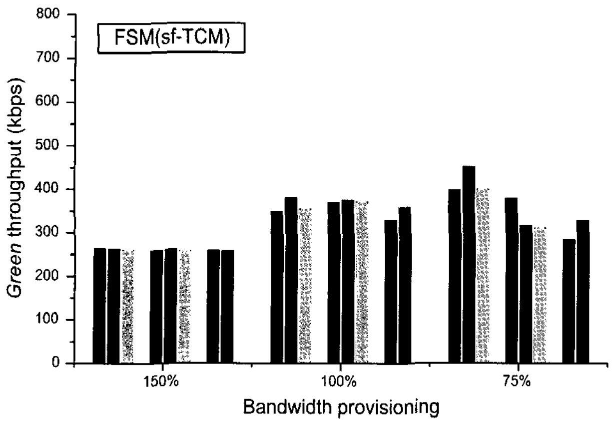


Fig. 17. Average *green* throughput of proposed scheme for each provisioning case.

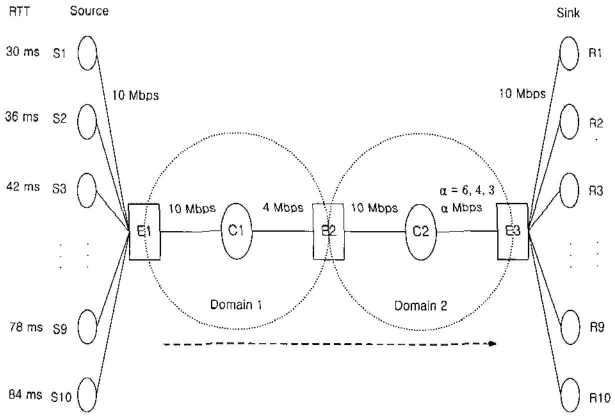


Fig. 18. The simulation model for the multiple domain case.

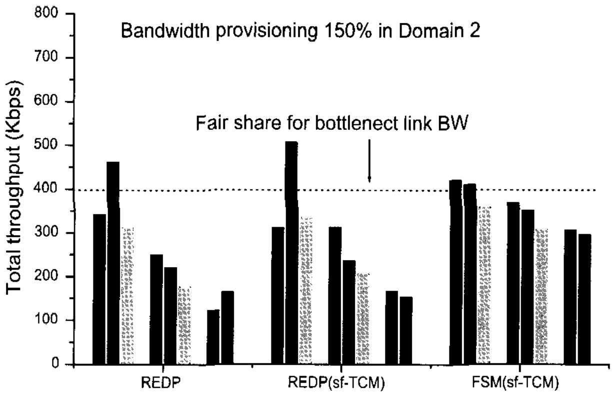


Fig. 19. Over-provisioning case in multiple domains.

green traffic rate is hardly beyond the aggregate contract rate at edge router. Therefore, a promotion situation is dominant at the user access edge router. In the promotion situation, our goal is for *green* packet to be promoted properly by consuming tokens in promotion region. That does not mean a full consuming of tokens in the promotion region. We focus that tokens are consumed between top and bottom of the promotion region. Fig. 16 shows token level variations of token bucket in the FSM at E1 edge router for the under-provisioning case. The token level is shown as a white line. Token level variations in the token bucket reflects the link utilization because the more packets are marked as *green*, the more packets are protected at the congestion time. Thus, it is important to consume remaining tokens in the token bucket properly. As shown in Fig. 16, in our proposed scheme, the remaining tokens are properly consumed in the *promotion* region and thus the FSM can improve link utilization as well as fairness of TCP flows.

Fig. 17 shows the average *green* throughput of the proposed scheme for each provisioning case. The fairness of *green* throughputs is good for the over-provisioning case and the other provisioning cases are properly fair. In our simulation model of Fig. 11, the bottleneck link bandwidth is 4 Mbps. Thus, in 150% provisioning case, 267 kbps *green* rates are generated at E1 edge router for each host. In 100% provisioning case, 400 kbps *green* rates are generated at E1 edge router for each host. And in 75% provisioning case, 533 kbps *green* rates are generated at E1 edge router for each host. In 150% provisioning case, it is hard to demote some *green* packet to *yellow* because of sum of *green* packet rates are 2.67 Mbps that is under bottleneck link bandwidth. Thus, in Fig 17, the *green* throughput is lower than other cases. In Figs. 12–14, we achieved that total average throughputs of each case are nearly same. In a single domain simulation model, our simulation results show that the fairness indexes of REDP/REDP(sf-TCM)/FSM(sf-TCM) are 0.837/0.87/0.99 for the over-provisioning case, 0.879/0.884/0.99 for the exact-provisioning case, and 0.816/0.957/0.98 for the under-provisioning case, respectively.

The link utilizations of REDP/REDP(sf-TCM)/FSM(sf-TCM) are 57.67%/67.38%/85.28% for the over-provisioning case, 59.46%/65.82%/94.4% for the exact-provisioning case and 65.04%/70.11%/94.74% for the under-provisioning case. The REDP marker fails to give good fairness to the TCP flows because it only considers the token level state in token buckets, which means that the REDP algorithm infers a TCP flow state from the only aggregate TCP flow state so that it brings about the unfairness of throughputs among TCP flows with different RTTs, as well as the decrease of link utilization. On the other hand, in our FSM algorithm, we consider individual TCP flow state as well as the aggregate flow level condition by measuring incoming traffic rate ratios. Thus, by reflecting a factor of the inferred TCP sending states into our FSM scheme, we achieved better results than the REDP algorithm in the over- provisioning case, as well as the exact-provisioning and the under-provisioning cases.

B. Multiple Domain Scenario

Fig. 18 shows the simulation model for the multiple domain case. There are 10 TCP hosts and each host contracts 400 kbps. The RTT of each flow is ranged from 30 ms to 84 ms with 6 ms differences and each flow connection spans through Domains 1 and 2. The contract rate of the aggregate flows is 4 Mbps in Domain 1. So, the exact-provisioning case occurs in Domain 1. In Domain 2, the interdomain negotiated contract rate is Mbps as shown in Fig. 18 and we set as 6 Mbps for the over-provisioning case (150% provisioning), and set as 4 Mbps for the exact-provisioning case (100% provisioning), and set as 3 Mbps for the under-provisioning case (75% provisioning), respectively.

As shown in Figs. 19–21, the FSM(sf-TCM) scheme also

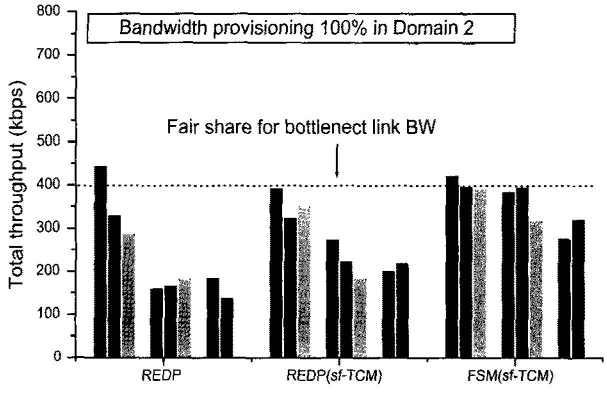


Fig. 20. Exact-provisioning case in multiple domains.

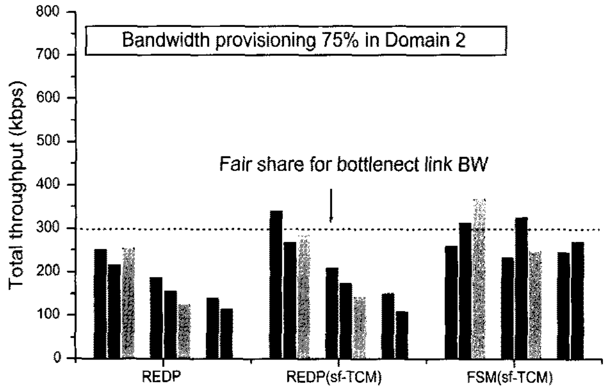


Fig. 21. Under-provisioning case in multiple domains.

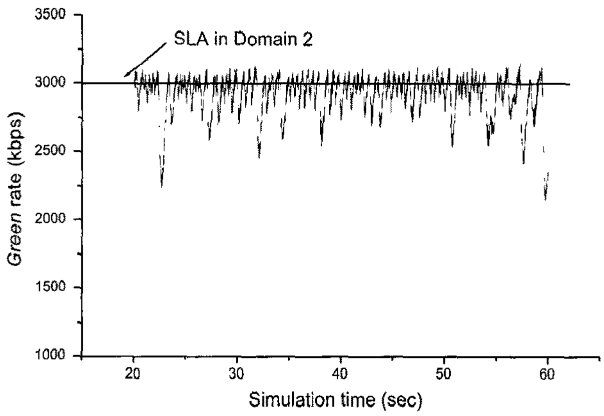


Fig. 22. Incoming *green* rate at E2 edge router in the under-provisioning case in Domain 2.

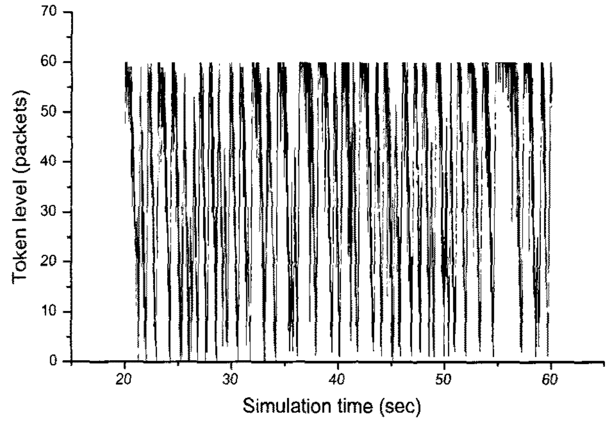


Fig. 23. Token level of token bucket in FSM at E2 edge router without demotion process in the under-provisioning case.

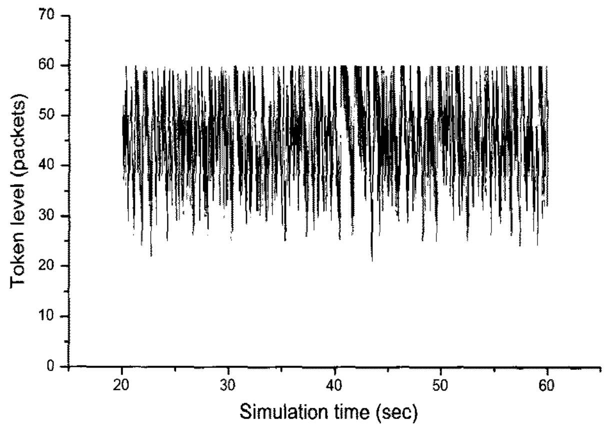


Fig. 24. Token level of token bucket in FSM at E2 edge router with demotion process in the under-provisioning case.

shows the best TCP fairness performance for each provisioning case in multiple domain environment. In case of the REDP combined with the sf-TCM, link utilization is increased because in this scheme *yellow* packets are also generated from the packets consuming the loss tokens in the sf-TCM, as well as from the demoted *green* packets in the REDP scheme. Thus, the processing range of demotion and promotion are wider than the original REDP and thus link utilization can be improved.

In cases of exact-provisioning and over-provisioning, the incoming *green* traffic rate from the previous Domain 1 hardly disobeys the negotiated contract rate at the E2 edge router because the aggregate contract rate is 4 Mbps in Domain 1 and *yellow* packets of source flows are promoted to *green* packets within the aggregate contract rate at the El edge router in Domain 1. Therefore, the incoming *green* traffic rate is same as the negotiated contract rate or lower than the negotiated rate, thus the promotion situation is dominant at the E2 edge router in Domain 2. However, in case of the under-provisioning, there exists some possibility that the incoming *green* rate at the E2 edge router is beyond the negotiated contract rate as shown in Fig. 22 because the arrival rate of *green* packets is within the aggregate contract rate (4 Mbps) of previous Domain 1 and the negotiated contract rate is 3 Mbps between two domains. Therefore, the phase effect may occur and it could bring about the unfairness problem as explained in Section I. In this case, a packet demotion process is necessary at the E2 edge router. Fig. 23 shows the token level variation of the FSM without the packet demotion process in a demotion situation. As shown in Fig. 23, when we do not perform a packet demotion in a demotion situation at the E2 edge router, the token level in token bucket is oscillating from 0 to 60 and there are some time intervals that the tokens are not properly consumed. Thus, we can know that the incoming *green* packet rate also oscillates. We can explain this phenomenon as follows. If the incoming *green* traffic rate is higher than the negotiated contract rate, the remaining tokens are gradually consumed and finally, no token remains in the token bucket. In this time, some flows of aggregate flows may consume the generated tokens and

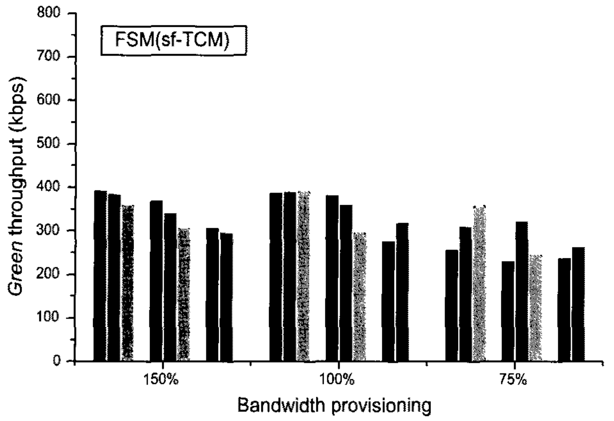


Fig. 25. Average *green* throughput of proposed scheme for each provisioning case.

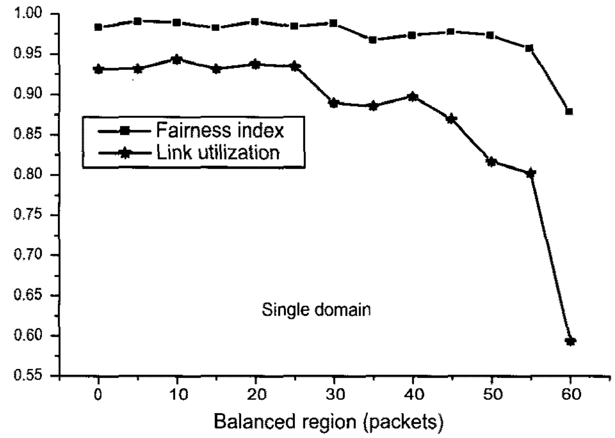


Fig. 26. Fairness index and link utilization in single domain case.

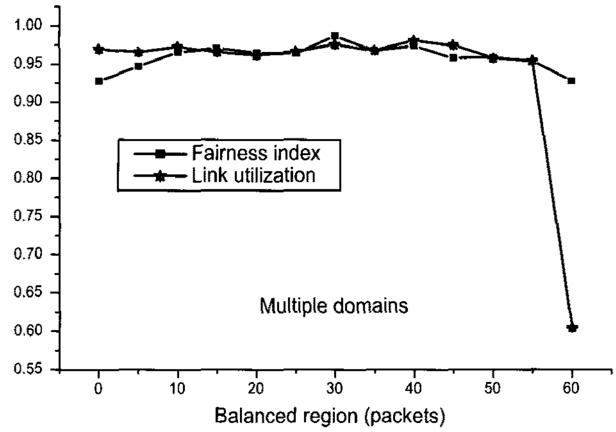


Fig. 27. Fairness index and link utilization in multiple domain case.

the other flows of aggregate flow may not consume the generated tokens, thus TCP flows that cannot consume the generated tokens decrease their sending rates since their packet dropping probability is high. After that, the incoming *green* traffic rate is lower than the negotiated contract rate and the remaining tokens in bucket are increased until the flows that cannot consume the tokens recover their previous sending rates. Again the incoming *green* traffic rate becomes higher than the negotiated contract rate. The above phenomenon results from the phase effect and it could bring about TCP unfairness as well as decrease of link utilization. Therefore if we do not remove the phase effect, this situation is continued as shown in Fig. 23.

Packet demotions are needed in a demotion situation so that it can prevent the phase effect. Our proposed demotion process early demotes some *green* packets to *yellow* packets with fairness before the remaining tokens in bucket will be zero. As shown in Fig. 24, tokens in the token bucket are properly consumed by the proposed FSM with demotion process and thus it prevents the phase effect. By doing this, we can improve fairness and link utilization in a demotion situation as well as in a promotion situation.

As shown in Fig. 25, in the proposed method, the average *green* throughputs of hosts are relatively fair for each provisioning case in spite of performing at the aggregate flow level.For the multiple domain simulation model, our simulation results show that the fairness indexes of REDP/REDP(sf-TCM) /FSM(sf-TCM) are 0.874/0.897/0.988 for the over-provisioning case, 0.848/0.943/0.983 for the exact-provisioning case, and 0.928/0.892/0.976 for the under-provisioning case, respectively. The link utilizations of REDP/REDP(sf-TCM)/FSM(sf-TCM) are 64.50%/71.19%/88.5% for the over-provisioning case, 57.53%/67.97%/89.4% for the exact-provisioning case, and 60.46%/69.39%/96.04% for the under-provisioning case, respectively.

C. Decision of Balanced Region

In this subsection, we investigate how the fairness index and link utilization vary according to the range of the balanced region. For this purpose, we use the simulation models in Figs. 11 and 18 for the single and multiple domains, respectively. The *balanced* region of the proposed FSM prevents unnecessary packet demotions and promotions. If source flows have a connection over multiple domains, packets are likely to be alternated between *green* and *yellow* at each domain edge router. This means that the throughput of each flow is likely to be unstable since the drop precedence of *yellow* packet is higher than that of *green* packet at the time of congestion. Note that in a single domain environment, there is little chance for such packet alternation since a promotion situation is dominant in a single domain case. However, if we set the range of the *balanced* region as a large value, i.e., the range of the *promotion* region has a small value, *yellow* packets have the low chance to be promoted to *green* packets. Therefore, we could not expect improvements of link utilization and TCP fairness as much as we expected. Fig. 26 shows how the TCP fairness and link utilization vary in the single domain case of Fig. 11 when we increase the range of the *balanced* region from 0 to 60 packets with 5 packets difference. When the *balanced* region is set to 0 packet, packet demotion and promotions are performed through all of the range in token bucket. On the other hand, when the *balanced* region is set to 60 packets, packet demotions and promotions are not performed. As shown in Fig. 26, when the range of the *balanced* region is between 0 and 30 packets, the TCP fairness and link utilization show good results. However, when the range of the *balanced* region is more than 30 packets, the fairness and link utilization become lower. Especially, the degradation of link utilization is larger than that of the TCP fairness. In a multiple domain environment, if we set the *balanced* region as a large value, we could not expect a prevention of the phase effect when a demotion situation occurs because the *demotion* region has relatively a small value so that the FSM cannot perform the packet demotion process properly. On the other hand, if we set the *balanced* region as a small value, the FSM could perform unnecessary packet demotions even when there are many remaining tokens in the token bucket, and thus it could bring about the decrease of throughput of each flow. Fig. 27 shows how the TCP fairness and link utilization vary in the multiple domain environment of Fig. 18. To analyze packet demotion and promotion effects simultaneously, we consider the situation in which the packet promotion is dominant in Domain 1 and the packet demotion is dominant in Domain 2. We set the range of the *balanced* region as a same value for both of the FSM markers at E1 and E2 routers. As shown in Fig. 27, when the range is between 10 and 40 packets, TCP fairness and link utilizations show good performances. When the range is less than 10 packets, we cannot see the decrease of link utilization in our simulation results but we can see degradations of TCP fairness. If the *balanced* region is too small, packets are likely to be alternated between *green* and *yellow* at each domain edge router, and it thus could bring about the unstable state of TCP throughput as explained before. Therefore, the *balanced* region can prevent the unstable state of each flow throughput. On the other hand, when the range of *balanced* region is more than 40 packets, the fairness index and link utilizations become lower. From the results of the Figs. 26 and 27, we can know that when the range of *balanced* region is between 10 and 30 packets, the TCP fairness and link utilizations show good performances for both of single and multiple domains.

D. Recommended Parameters

In our proposed algorithm, sf-TCM must be operated in per packet level because of classifications of packet colors such as *green, yellow,* and *red* packets. But markers of edge and intermediate routers are not needed to calculate each packet arrival rate at per packet level because of overhead problem. Thus, measurements of each packet arrival rate at edge and intermediate markers are calculated in a long term average rate level. In our simulation model, we consider this problem and get the results by using a long term average measurement for packet arrival rates. As like our simulation model parameters, we recommend a simple RIO queue [8] with parameters $(q_{\min}/q_{\max}/p_{\max}) = (45/60/0.02)$ for 'in' packets and $(20/40/0.2)$ for 'out' packets and the bucket depth as one packet for sf-TCM marker. It is not easy to recommend the bucket depth of FSM marker because we must consider the real network link bandwidth capacity, but we recommend the range of the *balanced* region as 17 percentage of a bucket depth at least. Finally, we recommend the sliding window size for measuring traffic rates as 0.1 sec to 1.0 sec by considering a router's capacity and overheads.

IV. CONCLUSION

To provide an end-to-end service differentiation for the assured services in DiffServ, as well as a flow marker that initially marks source flow, an intermediate marker is necessary. In this paper, to resolve the TCP fairness problem of the intermediate REDP marker, we have proposed an FSM as an intermediate marker which works with an sf-TCM operating as a flow marker for a source flow. The proposed FSM improves the fairness among the TCP flows with different RTTs without per-flow managements. Through the simulations, we have shown that the proposed fair scalable marker can improve TCP fairness and link utilization under all of the over-provisioning, exact-provisioning and under-provisioning network environments in multiple DiffServ domains, as well as in a single DiffServ domain.

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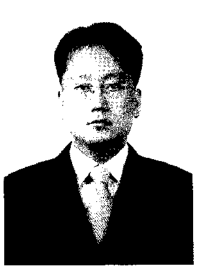

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