Session Initiation Protocol (SIP) Server Overload Control: Design and Evaluation

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Abstract. A Session Initiation Protocol (SIP) server may be overloaded by emergency-induced call volume, "American Idol" style flash crowd effects or denial of service attacks. The SIP server overload problem is interesting especially because the costs of serving or rejecting a SIP session can be similar. For this reason, the built-in SIP overload control mechanism based on generating rejection messages cannot prevent the server from entering congestion collapse under heavy load. The SIP overload problem calls for a pushback control solution in which the potentially overloaded receiving server may notify its upstream sending servers to have them send only the amount of load within the receiving server's processing capacity. The pushback framework can be achieved by either a rate-based feedback or a window-based feedback. The centerpiece of the feedback mechanism is the algorithm used to generate load regulation information. We propose three new window-based feedback algorithms and evaluate them together with two existing rate-based feedback algorithms. We compare the different algorithms in terms of the number of tuning parameters and performance under both steady and variable load. Furthermore, we identify two categories of fairness requirements for SIP overload control, namely, user-centric and provider-centric fairness. With the introduction of a new double-feed SIP overload control architecture, we show how the algorithms can meet those fairness criteria.

1 Introduction

The Session Initiation Protocol [1] (SIP) is a signaling protocol standardized by IETF for creating, modifying, and terminating sessions in the Internet. It has been used for many session-oriented applications, such as calls, multimedia distributions, video conferencing, presence service and instant messaging. Major standards bodies including 3GPP, ITU-I, and ETSI have all adopted SIP as the core signaling protocol for Next Generation Networks predominately based on the Internet Multimedia Subsystem (IMS) architecture.

The widespread popularity of SIP has raised attention to its readiness of handling overload [2]. A SIP server can be overloaded for many reasons, such as emergency-induced call volume, flash crowds generated by TV programs (e.g.,

American Idol), special events such as "free tickets to third caller", or even denial of service attacks. Although server overload is by no means a new problem for the Internet, the key observation that distinguishes the SIP overload problem from others is that the cost of rejecting a SIP session usually cannot be ignored compared to the cost of serving a session. Consequently, when a SIP server has to reject a large amount of arriving sessions, its performance collapses. This explains why using the built-in SIP overload control mechanism based on generating a rejection response messages does not solve the problem. If, as is often recommended, the rejected sessions are sent to a load-sharing SIP server, the alternative server will soon also be generating nothing but rejection responses, leading to a cascading failure. Another important aspect of overload in SIP is related to SIP's multi-hop server architecture with name-based application level routing. This aspect creates the so-called "server to server" overload problem that is generally not comparable to overload in other servers such as web server.

To avoid the overloaded server ending up at a state spending all its resources rejecting sessions, Hilt et al. [3] outlined a SIP overload control framework based on feedback from the receiving server to its upstream sending servers. The feedback can be in terms of a rate or a load limiting window size. However, the exact algorithms that may be applied in this framework and the potential performance implications are not obvious. In particular, to our best knowledge there has been no published work on specific window-based algorithms for SIP overload control, or comprehensive performance evaluation of rate-based feedback algorithms that also discusses dynamic load conditions and overload control fairness issues.

In this paper, we introduce a new dynamic session estimation scheme which plays an essential role in applying selected control algorithms to the SIP overload environment. We then propose three new window-based algorithms for SIP overload. We also apply two existing load adaption algorithms for rate-based overload control. We thus cover all three types of feedback control mechanisms in [3]: the absolute rate feedback, relative rate feedback and window feedback. Our simulation evaluation results show that although the algorithms differ in their tuning parameters, most of them are able to achieve theoretical maximum performance under steady state load conditions. The results under dynamic load conditions with source arrival and departure are also encouraging. Furthermore, we look at the fairness issue in the context of SIP overload and propose the notion of user-centric fairness vs. service provider-centric fairness. We show how different algorithms may achieve the desired type of fairness. In particular, we found that the user-centric fairness is difficult to achieve in the absolute rate or window-based feedback mechanisms. We solve this problem by introducing a new double-feed SIP overload control architecture.

The rest of this paper is organized as follows: Section 2 presents background on the SIP overload problem, and discusses related work. In Section 3 we propose three window-based SIP overload control algorithms and describe two existing load adaptation algorithm to be applied for rate-based SIP overload control. Then we present the simulation model and basic SIP overload results without feedback control in Section 4. The steady load performance evaluation of the

control algorithms are presented in Section 5, followed by dynamic load performance with fairness consideration in Section 6. Finally Section 7 concludes the paper and discusses future work.

2 Background and Related Work

2.1 SIP Overview

SIP is a message based protocol for managing sessions. There are two basic SIP entities, SIP User Agents (UAs), and SIP servers. SIP servers can be further grouped into proxy servers for session routing and registration servers for UA registration. In this paper we focus primarily on proxy servers. In the remainder of this document, when referring to SIP servers, we mean proxy server unless explicitly mentioned otherwise. One of the most popular session types that SIP is used for is call session. This is also the type of session we will consider in this paper. In a typical SIP call session, the caller and callee have UA functionalities, and they set up the session through the help of SIP servers along the path between them. Figure 1 shows the SIP message flow establishing a SIP call session. The caller starts with sending an INVITE request message towards the SIP proxy server, which replies with a 100 Trying message and forwards the request to the next hop determined by name-based application level routing. In Figure 1 the next hop for the only SIP server is the callee, but in reality it could well be another SIP server along the path. Once the INVITE request finally arrives at the callee, the callee replies with a 180 Ringing message indicating receipt of the call request by the callee UA, and sends a 200 OK message when the callee picks up the phone. The 200 OK message makes its way back to the caller, who will send an ACK message to the callee to conclude the call setup. Afterwards, media may flow between the caller and callee without the intervention of the SIP server. When one party wants to tear down the call, the corresponding UA sends a BYE message to the other party, who will reply with a 200 OK message to confirm the call hang-up. Therefore, a typical SIP call session entails processing of five incoming messages for call setup and two incoming messages for call teardown, a total of seven messages for the whole session.

SIP is an application level protocol on top of the transport layer. It can run over any common transport layer protocol, such as UDP and TCP. A particular aspect of SIP related to the overload problem is its timer mechanism. SIP defines a large number of retransmission timers to cope with message loss, especially when the unreliable UDP transport is used. As examples, we illustrate three of the timers which are commonly seen causing problems under overload. The first is timer A that causes an INVITE retransmission upon each of its expirations. With an initial value of $T_1 = 500 \ ms$, timer A increases exponentially until its total timeout period exceeds 32 s. The second timer of interest is the timer that controls the retransmission of 200 OK message as a response to an INVITE request. The timer for 200 OK also starts with T_1 , and its value doubles until it reaches $T_2 = 4 \ s$. At that time the timer value remains at T_2 until the total timeout period exceeds 32 s. The third timer of interest is timer E, which controls

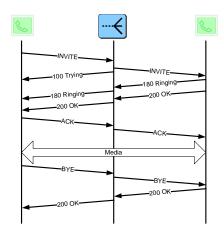


Fig. 1. SIP call session message flow

the BYE request retransmission. Timer E follows a timeout pattern similar to the 200 OK timer. Note that the receipt of corresponding messages triggered by each of the original messages will quench the retransmission timer. They are the 100 Trying for INVITE, ACK for 200 OK, and 200 OK for BYE. From this description, we know that for example, if an INVITE message for some reason is dropped or stays in the server queue longer than 500 ms without generating the 100 Trying, the upstream SIP entity will retransmit the original INVITE. Similarly, if the round trip time of the system is longer than 500 ms, then the 200 OK timer and the BYE timer will fire, causing retransmission of these messages. Under ideal network conditions without link delay and loss, retransmissions are purely wasted messages that should be avoided.

2.2 Types of SIP Server Overload

There are many causes to SIP overload, but the resulting SIP overload cases can usually be grouped into either of the two types: server to server overload or client to server overload.

A typical server to server overload topology is illustrated in Figure 2. In this figure the overloaded server (the Receiving Entity or RE) is connected with a relatively small number of upstream servers (the Sending Entities or SEs). One example of server to server overload is a special event such as "free tickets to the third caller", also referred to as flash crowds. Suppose RE is the Service Provider (SP) for a hotline N. SE_1 , SE_2 and SE_3 are three SPs that reach the hotline through RE. When the hotline is activated, RE is expected to receive a large call volume to the hotline from SE_1 , SE_2 and SE_3 that far exceeds its usual call volume, potentially putting RE into a severe overload. The second type of overload, known as client-to-server overload is when a number of clients overload the next hop server directly. An example is avalanche restart, which happens when power is restored after a mass power failure in a large metropolitan area.

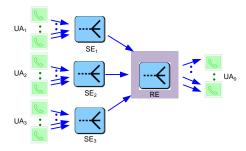


Fig. 2. Server to server overload

At the time the power is restored, a very large number of SIP devices boot up and send out SIP registration requests almost simultaneously, which could easily overload the corresponding SIP registration server. This paper only discusses the server-to-server overload problem. The client-to-server overload problem may require different solutions and is out of scope of this paper.

2.3 Existing SIP Overload Control Mechanisms

Without overload control, messages that cannot be processed by the server are simply dropped. Simple drop causes the corresponding SIP timers to fire, and further amplifies the overload situation.

SIP has a 503 Service Unavailable response message used to reject a session request and cancel any related outstanding retransmission timers. However, because of the relatively high cost of generating this rejection, this message cannot solve the overload problem.

SIP also defines an optional parameter called "Retry-after" in the 503 Service Unavailable message. The "Retry-after" value specifies the amount of time that the receiving SE of the message should cease sending any requests to the RE. The 503 Service Unavailable with "Retry-after" represents basically an on/off overload control approach, which is known to be unable to fully prevent congestion collapse [2]. Another related technique is to allow the SE to fail over the rejected requests to an alternative load-sharing server. However, in many situations the load-sharing server could ultimately be overloaded as well, leading to cascading failure.

2.4 Feedback-based Overload Control

The key to solving the SIP server overload problem is to make sure the upstream SEs only send the amount of traffic that the RE is able to handle at all times. In this ideal situation, there will be no message retransmission due to timeout and no extra processing cost due to rejection. The server CPU power can be fully utilized to deliver its maximum session service capacity.

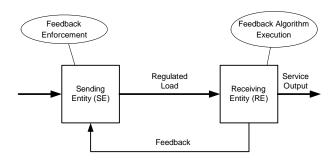


Fig. 3. Generalized feedback architecture

A feedback loop is a natural approach to achieve the ideal overload control goal. Through the loop, RE notifies SEs the amount of load that is acceptable.

To some extent the existing SIP 503 Service Unavailable mechanism with the "Retry-after" header is a basic form of the feedback mechanism. Unfortunately, its on/off control nature has proven to be problematic. Therefore, the IETF community has started looking at more sophisticated pushback mechanisms including both rate-based and window-based feedback. A generalized model of the feedback-based control model is shown in Figure 3. There are three main components in the model: feedback algorithm execution at RE, feedback communication from RE to SE, and feedback enforcement at the SE.

Feedback Algorithm Execution Absolute rate, relative rate and window feedback are three main SIP feedback control mechanisms. Each mechanism executes specific control algorithms to generate and adapt the feedback value.

In absolute rate-based feedback, the feedback generation entity RE needs to estimate its acceptable load and allocate it among the SEs. The feedback information is an absolute load value for the particular SE. The key element in absolute rate feedback is an algorithm for dynamic acceptable load estimation.

In relative rate-based feedback, the feedback generation entity RE computes an incoming load throttle percentage based on a target resource metric (e.g., CPU utilization). The feedback information is a dynamic percentage value indicating how much proportion of the load should be accepted or rejected relative to the original incoming load. The key element in relative rate feedback is the dynamic relative rate adjustment algorithm and the choosing of the target metric.

In window-based feedback, the feedback generation entity RE estimates a dynamic window size for each SE which specifies the number of acceptable sessions from that particular SE. The feedback information is the current window size. The key element in window-based feedback is a dynamic window adjustment algorithm.

The feedback generation could be either time-driven or event-driven. In timedriven mechanisms, the control is usually exercised every pre-scheduled control interval, while in event-driven mechanisms, the control is executed upon the occurrence of some event, such as a session service completion. We will examine both time-driven and event-driven algorithms in this paper.

Feedback Enforcement Mechanisms The SEs may choose among many well-known traffic regulation mechanisms to enforce feedback control, such as percentage throttle, leaky bucket and token bucket, automatic call gapping, and window throttle. Since our focus is on the feedback algorithms, throughout this paper we will use percentage throttle for rate-based feedback and window-throttle for window-based feedback mechanisms. In our percentage throttle implementation we probabilistically block a given percentage of the load arrival to make sure the actual output load conforms to the regulated load value. For window throttle implementation, we only forward a specific session arrival when there is window slot available.

Feedback Communication The feedback information for SIP signaling overload control can be communicated via an in-band or out-of-band channel. In this paper, we have chosen to use the in-band feedback communication approach. Specifically, any feedback information available is sent in the next immediate message that goes to the particular target SE. This approach has an advantage in server to server overload because there is generally no problem finding existing messages to carry feedback information under overload and it incurs minimal overhead.

2.5 Related Work

Signaling overload itself is a well studied topic. Many of the previous work on call signaling overload in general communication networks is believed to be usable by the SIP overload study. For instance, Hosein [4] presented an adaptive rate control algorithm based on estimation of message queuing delay; Cyr et al. [5] described the Occupancy Algorithm (OCC) for load balancing and overload control mechanism in distributed processing telecommunications systems based on server CPU occupancy; Kasera et al. [6] proposed an improved OCC algorithm call Acceptance-Rate Occupancy (ARO) by taking into consideration the call acceptance ratio, and a Signaling RED algorithm which is a RED variant for signaling overload control.

Specifically on SIP, Ohta [7] showed through simulation the congestion collapse of SIP server under heavy load and explored the approach of using a priority queuing and Bang-Bang type of overload control. Nahum *et al.* [8] reported empirical performance results of SIP server showing the congestion collapse behavior.

In addition, Whitehead [9] described a unified overload control framework called GOCAP for next generation networks, which is supposed to cover SIP as well. But there has been no performance results yet and it is not clear at

this time how the GOCAP framework may relate to the IETF SIP overload framework.

In the most closely related work to this paper, Noel and Johnson [10] presented initial results comparing a SIP network without overload control, with the built-in SIP overload control and with a rate-based overload control scheme. However, their paper does not discuss window-based control, or present performance results under dynamic load, and it does not address the overload fairness problem.

3 Feedback Algorithms for SIP Server Overload Control

The previous section has introduced the main components of SIP overload feedback control framework. In this section we investigate its key component - the feedback algorithm. We propose three window-based SIP overload control methods, namely win-disc, win-cont, and win-auto. We also apply two existing adaptive load control algorithms for rate-based control. Before discussing algorithm details, we first introduce a dynamic SIP session estimation method which plays an important role in applying selected rate-based or window-based algorithms to SIP overload control.

3.1 Dynamic SIP Session Estimation

Design of SIP overload control algorithm starts with determining the control granularity, i.e., the basic control unit. Although SIP is a message-based protocol, different types of SIP messages carry very different weights from admission control perspective. For instance, in a typical call session, admitting a new IN-VITE message starts a new call and implicitly accepts six additional messages for the rest of the session signaling. Therefore, it is more convenient to use a SIP session as the basic control unit.

A session oriented overload control algorithm frequently requires session related metrics as inputs such as the session service rate. In order to obtain session related metrics a straightforward approach is to do a *full session check*, i.e., to track the start and end message of all SIP signaling sessions. For example, the server may count how many sessions have been started and then completed within a measurement interval. In the case of a call signaling, the session is initiated by an INVITE request and terminated with a BYE request. The INVITE and BYE are usually separated by a random session holding time. However, SIP allows the BYE request to traverse a different server from the one for the original INVITE. In that case, some SIP server may only see the INVITE request while other servers only see the BYE request of a signaling session. There could also be other types of SIP signaling sessions traversing the SIP server. These factors make the applicability of the *full session check* approach complicated, if not impossible.

We use an alternative *start session check* approach to estimate SIP session service rate . The basic idea behind is that under normal working conditions,

the actual session acceptance rate is roughly equal to the session service rate. Therefore, we can estimate the session service rate based only on the session start messages. Specifically, the server counts the number of INVITE messages that it accepts per measurement interval T_m . The value of the session service rate is estimated to be $\mu = N_{inv}^{accepted}/T_m$. Standard smoothing functions can be applied to the periodically measured μ .

One other critical session parameter often needed in SIP overload control algorithms is the number of sessions remaining in the server system, assuming the server processor is preceded by a queue where jobs are waiting for service. It is very important to recognize that the number of remaining sessions is NOT equal to the number of INVITE messages in the queue, because the queue is shared by all types of messages, including those non-INVITE messages which represent sessions that had previously been accepted into the system. All messages should be counted for the current system backlog. Hence we propose to estimate the current number of sessions in the queue using Eq. 1:

$$N_{sess} = N_{inv} + \frac{N_{noninv}}{L_{sess} - 1} \tag{1}$$

where N_{inv} and N_{noninv} are current number of INVITE and non-INVITE messages in the queue, respectively. The parameter L_{sess} represents the average number of messages per-session. N_{inv} indicates the number of calls arrived at the server but yet to be processed; $N_{noninv}/(L_{sess}-1)$ is roughly the number of calls already in process by the server.

Eq. 1 holds for both the full session check and the simplified start session check estimation approaches. The difference is how the L_{sess} parameter is obtained. When the full session check approach is used, the length of each individual session will be counted by checking the start and end of each individual SIP sessions. With our simplified start session check approach, the session length can be obtained by counting the actual number of messages N_{msg}^{proc} , processed during the same period the session acceptance rate is observed. The session length is then estimated to be $L_{sess} = N_{msg}^{proc}/N_{inv}^{accepted}$.

3.2 Active Source Estimation

In some of the overload control mechanisms, the RE may wish to explicitly allocate its total capacity among multiple SEs. A simple approach is to get the number of current active SEs and divide the capacity equally. We do this by directly tracking the sources of incoming load and maintaining a table entry for each current active SE. Each entry has an expiration timer set to one second.

3.3 The win-disc Window Control Algorithm

A window feedback algorithm executed at the RE dynamically computes a feedback window value for the SE. SE will forward the load to RE only if window slots are currently available. Our first window based algorithm is win-disc, the short name for window-discrete. The main idea is that at the end of each discrete

control interval of period T_c , RE re-evaluate the number of new session requests it can accept for the next control interval, making sure the delays for processing sessions already in the server and upcoming sessions are bounded. Assuming the RE advertised window to SE_i at the k^{th} control interval T_c^k is w_i^k , and the total window size for all SEs at the end of the k^{th} control interval is w^{k+1} , the win-disc algorithm is described below:

```
\begin{array}{l} w_i^0 := W_0 \text{ where } W_0 > 0 \\ w_i^k := w_i^k - 1 \text{ for INVITE received from } SE_i \\ w_i^{k+1} := \mu^k T_c + \mu^k D_B - N_{sess}^k \text{ at the end of } T_c^k \\ w_i^{k+1} := round(w^{k+1}/N_{SE}^k) \end{array}
```

where μ^k is the current estimated session service rate. D_B is a parameter that reflects the allowed budget message queuing delay. N_{sess}^k is the estimated current number of sessions in the system at the end of T_c^k . $\mu^k T_c$ gives the estimated number of sessions the server is able to process in the T_c^{k+1} interval. $\mu^k D_B$ gives the average number of sessions that can remain in the server queue given the budget delay. This number has to exclude the number of sessions already backlogged in the server queue, which is N_{sess}^k . Therefore, w^{k+1} gives the estimated total number of sessions that the server is able to accept in the next T_c control interval giving delay budget D_B . Both μ^k and N_{sess}^k are obtained with our dynamic session estimation algorithm in Section 3.1. N_{SE}^k is the current number of active sources discussed in Section 3.2. Note that the initial value W_0 is not important as long as $W_0 > 0$. An example value could be $W_0 = \mu_{eng} T_c$ where μ_{eng} is the server's engineered session service rate.

3.4 The win-cont Window Control Algorithm

Our second window feedback algorithm is win-cont, the short name for windowcontinuous. Unlike the time-driven win-disc algorithm, win-cont is an event driven algorithm that continuously adjusts advertised window size when the server has room to accept new sessions. The main idea of this algorithm is to bound the number of sessions in the server at any time. The maximum number of sessions allowed in the server is obtained by $N_{sess}^{max} = \mu^t D_B$, where D_B is again the allowed message queuing delay budget and μ^{t} is the current service rate. At any time, the difference between the maximum allowed number of sessions in the server N_{sess}^{max} and the current number of sessions N_{sess} is the available window to be sent as feedback. Depending on the responsiveness requirements and computation ability, there are different design choices. First is how frequently N_{sess} should be checked. It could be after any message processing, or after an INVITE message processing, or other possibilities. The second is the threshold number of session slots to update the feedback. There are two such thresholds, the overall number of available slots W_{ovth} , and the per-SE individual number of available slots W_{indvth} . To make the algorithm simple, we choose per-message processing N_{sess} update and fix both W_{ovth} and W_{indvth} to 1. A general description of the win-cont algorithm is summarized as below:

```
\begin{aligned} w_i^0 &:= W_0 \text{ where } W_0 > 0 \\ w_i^t &:= w_i^t - 1 \text{ for INVITE received from } SE_i \\ w_{left}^t &:= N_{sess}^{max} - N_{sess} \text{ upon msg processing} \\ if(w_{left}^t \geq 1) \\ w_{share}^t &= w_{left}^t / N_{SE}^t \\ w_{i'}^t &:= w_{i'}^t + w_{share}^t \\ if(w_{i'}^t \geq 1) \\ w_{i'}^t &:= (int)w_{i'}^t \\ w_{i'}^t &:= (frac)w_{i'}^t \end{aligned}
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Note that since w_i^t may contain a decimal part, to improve the feedback window accuracy when w_i^t is small, we feedback the integer part of the current w_i^t and add its decimal part to the next feedback by using a temporary parameter $w_{i'}^t$. In the algorithm description, μ^t , N_{sess} and N_{SE} are obtained as discussed in Section 3.1 and Section 3.2. The initial value W_0 is not important and a reference value is $W_0 = \mu_{enq} T_c$ where μ_{enq} is the server's engineered session service rate.

3.5 The win-auto Window Control Algorithm

Our third window feedback algorithm, win-auto stands for window-autonomous. Like win-cont, win-auto is also an event driven algorithm. But as the term indicates, the win-auto algorithm is able to make window adjustment autonomously. The key design principal in the win-auto algorithm is to automatically keep the pace of window increase below the pace of window decrease, which makes sure the session arrival rate does not exceed the session service rate. The algorithm details are as follows:

```
\begin{array}{l} w_i^0 := W_0 \text{ where } W_0 > 0 \\ w_i^t := w_i^t - 1 \text{ for INVITE received from } SE_i \\ w_i^t := w_i^t + 1 \text{ after processing a } new \text{INVITE} \end{array}
```

The beauty of this algorithm is its extreme simplicity. The algorithm takes advantage of the fact that retransmission starts to occur as the network gets congested. Then the server automatically freezes its advertised window to allow processing of backlogged sessions until situation improves. The only check the server does is whether an INVITE message is a retransmitted one or a new one, which is just a piece of normal SIP parsing done by any existing SIP server. There could be many variations along the same line of thinking as this algorithm, but the one as described here appears to be one of the most natural options.

3.6 The rate-abs Rate Control Algorithm

We implemented an absolute rate feedback control by applying the adaptive load algorithm of Hosein [4], which is also used by Noel [10]. The main idea is to ensure the message queuing delay does not exceed the allowed budget value. The algorithm details are as follows.

During every control interval T_c , the RE notifies the SE of the new target load, which is expressed by Eq. 2.

$$\lambda^{k+1} = \mu^k (1 - \frac{(d_q^k - D_B)}{C}) \tag{2}$$

where μ^k is the current estimated service rate and d_q^k is the estimated server queuing delay at the end of the last measurement interval. It is obtained by $d_q^k = N_{sess}/\mu^k$, where N_{sess} is the number of sessions in the server. We use our dynamic session estimation in Section 3.1 to obtain N_{sess} , and we refer to this absolute rate control implementation as rate-abs in the rest of this document.

3.7 The rate-occ Rate Control Algorithm

Our candidates of existing algorithms for relative rate based feedback control are Occupancy Algorithm (OCC) [5], Acceptance-Rate Occupancy (ARO), and Signaling RED (SRED) [6]. We decided to implement the basic OCC algorithm because this mechanism already illustrates inherent properties with any occupancy based approach. On the other hand, tuning of RED based algorithm is known to be relatively complicated.

The OCC algorithm is based on a target processor occupancy, defined as the percentage of time the processor is busy processing messages within a measurement interval. So the target processor occupancy is the main parameter to be specified. The processor occupancy is measured every measurement interval T_m . Every control interval T_c the measured processor occupancy is compared with the target occupancy. If the measured value is larger than the target value, the incoming load should be reduced. Otherwise, the incoming load should be increased. The adjustment is reflected in a parameter f which indicates the acceptance ratio of the current incoming load. f is therefore the relative rate feedback information and is expressed by the Eq. 3:

$$f^{k+1} = \begin{cases} f_{min}, & \text{if } \phi^k f^k < f_{min} \\ 1, & \text{if } \phi^k f^k > 1 \\ \phi^k f^k, & \text{otherwise} \end{cases}$$
 (3)

where f_k is the current acceptance ratio and f_{k+1} is the estimated value for the next control interval. $\phi^k = min(\rho_B/\rho_t^k, \phi_{max})$. f_{min} exists to give none-zero minimal acceptance ratio, thus prevents the server from completely shutting off the SE. ϕ_{max} defines the maximum multiplicative increase factor of f in two consecutive control intervals. In this paper we choose the two OCC parameters ϕ and f_{min} to be 5 and 0.02, respectively in all our tests.

We will refer to this algorithm as rate-occ in the rest of this paper.

4 Simulation Model

4.1 Simulation Platform

We have built a SIP simulator on the popular OPNET modeler simulation platform [11]. Our SIP simulator captures both the INVITE and non-INVITE state

machines as defined in RFC3261. It is also one of the independent implementations in the IETF SIP server overload design team, and has been calibrated in the team under common simulation scenarios.

Our general SIP server model consists of a FIFO queue followed by a SIP processor. Depending on the control mechanisms, specific overload related prequeue or post-queue processing may be inserted, such as window increase and decrease mechanisms. The feedback information is included in a new *overload* header of each SIP messages, and are processed along with normal SIP message parsing. Processing of each SIP messages creates or updates transaction states as defined by RFC3261. The transport layer is UDP, and therefore all the various SIP timers are in effect.

Our UA model mimics an infinite number of users. Each UA may generate calls at any rate according to a specified distribution and may receive calls at any rate. The processing capacity of a UA is assumed to be infinite since we are interested in the server performance.

4.2 Simulation Topology and Configuration

We use the topology in Figure 2 for current evaluations. There are three UAs on the left, each of which represents an infinite number of callers. Each UA is connected to an SE. The three SEs all connect to the RE which is the potentially overloaded server. The queue size is 500 messages. The core RE connects to UA_0 which represents an infinite number of callees. Calls are generated with exponential interarrival times from the callers at the left to the callees on the right. Each call signaling contains seven messages as illustrated in Figure 1. The call holding time is assumed to be exponentially distributed with average of 30 seconds. The normal message processing rate and the processing rate for rejecting a call at the RE are 500 messages per second (mps) and 3000 mps, respectively.

Note that the server processor configuration, together with the call signaling pattern, results in a nominal system service capacity of 72 cps. All our load and goodput related values presented below are normalized to this system capacity. Our main result metric is goodput, which counts the number of calls with successful delivery of all five call setup messages from INVITE to ACK below 10 s.

For the purpose of this simulation, we also made the following assumptions. First, we do not consider any link transmission delay or loss. However, this does not mean feedback is instantaneous, because we assume the piggyback feedback mechanism. The feedback will only be sent upon the next available message to the particular next hop. Second, all the edge proxies are assumed to have infinite processing capacity. By removing the processing limit of the edge server, we avoid the conservative load pattern when the edge proxy server can itself be overloaded.

These simple yet classical network configuration and assumptions allow us to focus primarily on the algorithms themselves without being distracted by less important factors, which may be further explored in future work.

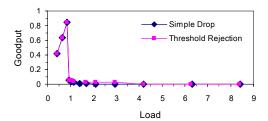


Fig. 4. SIP overload with no feedback control

4.3 SIP Overload Without Feedback Control

For comparison, we first look at SIP overload performance without any feedback control. Figure 4 shows the simulation results for two basic scenarios. In the "Simple Drop" scenario, any message arrived after the queue is full is simply dropped. In the "Threshold Rejection" scenario, the server compares its queue length with a high and a low threshold value. If the queue length reaches the high threshold, new INVITE requests are rejected but other messages are still processed. The processing of new INVITE requests will not be restored until the queue length falls below the low threshold. As we can see, the two result goodput curves almost overlap. Both cases display similar precipitous drops when the offered load approximates the server capacity, a clear sign of congestion collapse. However, the reasons for the steep collapse of the goodput are quite different in the two scenarios. In the "Simple Drop" case, there are still around one third of the INVITE messages arriving at the callee, but all the 180 RINGING messages are dropped, and most of the 200 OK messages are also dropped due to queue overflow. In the "Threshold Rejection" case, none of the INVITE messages reaches the callee, and the RE is only sending rejection messages.

5 Steady Load Performance

We summarize in Table 1 the parameters for all the rate-based and window-based overload control algorithms we discussed in Section 3. In essence, most of the algorithms have a *binding* parameter, three of them use the budget queuing delay D_B , and one uses the budget CPU occupancy ρ_B . All three discrete time control algorithms have a control interval parameter T_c .

There is also a server metric measurement interval T_m used by four of the five algorithms. T_m and T_c need to be separate only when T_c is relatively large compared to T_m . The choice of the T_m value depends on how volatile the target server metric is over time. For example, if the target metric is the server service rate, which is relatively stable, a value of 100 ms is usually more than sufficient. If on the other hand, the target metric is the current queue length, then smaller or larger T_m makes clear differences. In our study, when the specific algorithm

Table 1. Parameter sets for overload algorithms

Algorithm	Binding	Control	Measure	Additional
		Interval	Interval	
rate-abs	D_B	T_c	T_m	
rate-occ	$ ho_B$	T_c	T_m	f_{min} and ϕ
win- $disc$	D_B	T_c	T_m	
win- $cont$	D_B^*	N/A	T_m	
win- $auto$	N/A^{\dagger}	N/A	N/A	

 D_B : budget queuing delay

 ρ_B : CPU occupancy

 T_c : discrete time feedback control interval

 T_m : discrete time measurement interval for selected server metrics; $T_m \leq T_c$ where applicable

 f_{min} : minimal acceptance fraction

 ϕ : multiplicative factor

requires to measure the server service rate and CPU occupancy, we apply T_m ; when the algorithm requires information on the current number of packets in the queue, we always obtain the instant value. Our results show that $T_m = min(100 \ ms, T_c)$ is a reasonable assumption, by which we basically reduce the two interval parameters into one.

We looked at the sensitivity of D_B and T_c for each applicable algorithms. Figure 5 and Figure 6 show the results for win-disc. All the load and goodput values have been normalized upon the theoretical maximum capacity of the server.

We started with a T_c value of 200 ms and found that the server achieves the unit goodput when D_B is set to 200 ms. Other $0 < D_B < 200$ ms values also showed similar results. This is not surprising given that both the SIP caller INVITE and callee 200 OK timer starts at $T_1 = 500$ ms. If the queuing delay is smaller than 1/2 T_1 or 250 ms, then there should be no timeout either on the caller or callee side. A larger value of D_B triggers retransmission timeouts which reduce the server goodput. For example, Figure 5 shows that at $D_B = 500$ ms, the goodput has already degraded by 25%.

Letting D=200 ms, we then looked at the influence of T_c . As expected, the smaller the value of T_c the more accurate the control would be. In our scenario, we found that a T_c value smaller than 200 ms is sufficient to give the theatrical maximum goodput. A larger T_c quickly deteriorates the results as seen from Figure 6.

The effect of D_B for win-cont and rate-abs show largely the similar shape, with slightly different sensitivity. Generally speaking, a positive D_B value centered at around 200 ms provides a good outcome for all cases.

 $^{^*}$ D_B recommended for robustness, although a fixed binding window size can also be used

[†] Optionally D_B may be applied for corner cases

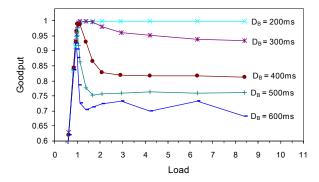


Fig. 5. win-disc goodput under different queuing delay budget

Figure 7 and Figure 8 compare the T_c parameter for win-disc, rate-abs and rate-occ with $D_B=200ms$. For the rate-occ binding parameter ρ_B , we used 85% for the tests in Figure 7 and Figure 8. We will explain why this value is chosen shortly. It can be seen that the performance of win-disc and rate-abs are very close to maximum theoretical value in all cases except for when $T_c=1s$ in the heavy load case. This shows win-disc is more sensitive to control interval than rate-abs, which could also be caused by the more busty nature of the traffic resulted from window throttle. It is clear that for both win-disc and rate-abs a shorter T_c improves the results, and a value below 200 ms is sufficient. Overall, rate-occ performs not as good as the other two. But what is interesting about rate-occ is that from 14 ms to 100 ms control interval, the goodput increases in light overload and decreases in heavy overload. This could be a result of rate adjustment parameters which may have cut the rate too much at the light overload.

To further understand the binding parameter ρ_B of rate-occ, we illustrate in Figure 9 the relationship between the goodput and the value of ρ_B under different load conditions. A ρ_B value higher than 95% easily degrades the performance under heavy overload, because the instantaneous server occupancy could still exceeds the healthy region and causes longer delays which result in SIP timer expiration and message retransmissions. A tradeoff ρ_B value with the highest and most stable performance across all load conditions in the given scenario is 85%, which is the reason we used it in Figure 7 and Figure 8.

Finally, for the *win-auto* algorithm, we have found in most cases with a reasonable initial window size in the order of 10, the output matches perfectly the theoretical maximum line. We also see some cases where the system could experience periods of suboptimal yet still stable performance. The most common case happens when the server is started with a large initial window and the offered load is a steep jump to a heavily loaded region. Our investigation reveals that, this suboptimal performance is caused by the difference in the stabilized queu-

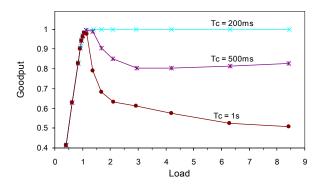


Fig. 6. win-disc goodput under different control interval T_c

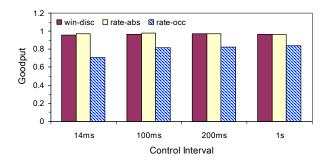


Fig. 7. Goodput vs. T_c at load 1

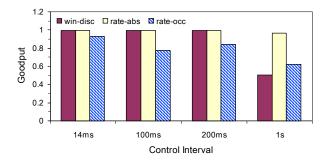


Fig. 8. Goodput vs. T_c at load 8.4

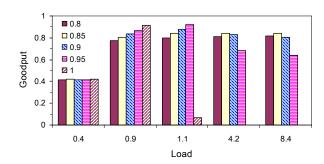


Fig. 9. Goodput vs. ρ_B at different loads

ing delay. In most of the normal cases, when the system reaches steady state, the queuing delay is smaller than half of the SIP timer T_1 value or 250 ms. In the suboptimal case, the system may become stable at a point where the queuing delay can exceed 250 ms. The round-trip delay then exceeds 500 ms, which triggers the 200 OK timer and the BYE timer, each of which uses 500 ms. The two timer expirations introduce three additional messages to the system, a retransmitted 200 OK, the ACK to the retransmitted 200 OK, and a retransmitted BYE. This change increases the session length from seven to ten and reduces the maximum server goodput by 28%. A cure to this situation is to introduce an extra queuing delay parameter to the window adjustment algorithm. Specifically, before the server increases the window size, it checks the current queuing delay. If the queuing delay value already exceeds the desired threshold, the window is not increased. However, we found that determining the optimal value of the queuing delay threshold parameter is not very straightforward and makes the algorithm much more complex. The small chance of the occurrence of the suboptimal performance in realistic situations may not justify the additional delay binding check.

Having looked at various parameters for all different algorithms, we now summarize the best goodput achieved by each algorithm in Figure 10. The specific parameters used for each algorithm is listed in Table 2.

Table 2. Parameters used for comparison

	$D_B(ms)$	$T_c(ms)$	$T_m(ms)$
rate-abs	200	200	100
$rate ext{-}occ1^\ddagger$		200	100
$rate-occ2^{\ddagger}$	N/A	14	14
win-disc	200	200	100
win-cont	200	N/A	100
win- $auto$	N/A	N/A	N/A

[†] in addition: $\rho_B = 0.85, \phi = 5, f_{min} = 0.02$

It is clear from Figure 10 that all algorithms except for rate-occ are able to reach the theoretical maximum goodput. The corresponding CPU occupancy also confirms the goodput behavior. What is important to understand is that the reason rate-occ does not operate at the maximum theoretical goodput like the others is not simply because of the artificial limit of setting the occupancy to 85%. This point can be confirmed by the earlier Figure 9. The inherent issue with an occupancy based heuristic is the fact that occupancy is not as direct a metric as queue length or queuing delay in solving the overload problem. Figure 10 shows one factor that really helps improve the rate-occ performance at heavy load seem to be using extremely small T_c . But updating the current CPU occupancy every 14 ms is not straightforward in all systems. Furthermore, when this short T_c is used, the actual server occupancy rose to 93%, which goes contrary to the original intention of setting the 85% budget server occupancy. Yet another issue with setting the extremely short T_c is its much poorer performance than other algorithms under light overload, which should be linked to the tuning of OCC's heuristic increase and decrease parameters.

The merits of all the algorithms achieving maximum theoretical goodput is that they ensure no retransmission ever happens, and thus the server is always busy processing messages, with each single message being part of a successful session.

Another metric of interest for comparison is the session setup delay, which we define as from the time the INVITE is sent until the ACK to 200 OK message is received. We found that the *rate-occ* algorithm has the lowest delay but this is not significant considering it operates at the suboptimal region in terms of goodput. *win-cont* comes next with a delay of around 3 ms. The *rate-abs* offers a delay close to that of *win-cont* at about 3.5 ms. The remaining two *win-disc* and *win-auto* have a delay of 5 ms and 6 ms respectively. In fact all these values are sufficiently small and are not likely making any difference.

From the steady state load analysis so far, we conclude that the occupancy based approach is less favorable than others because of its relatively more number of tuning parameters and not being able to adapt to the most efficient processing condition for the maximum goodput. win-disc and abs-rate are by definition quite similar and they also have the same number of parameters. Their performance is also very close, although rate-rate has shown a slight edge, possibly because of the smoother arrival pattern resulting from the percentage throttle. win-cont has less tuning parameter than win-disc and abs-rate, and offers equal or slightly better performance Finally, win-auto is an extremely simple algorithm yet achieves nearly perfect results in most situations.

6 Dynamic Load Performance and Fairness for Overload Control

Although steady load performance is a good starting point for evaluating the overload control algorithms, most of the regular overload scenarios are not persistent steady overload. Otherwise, The issue would become a poor capacity

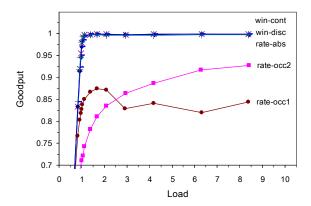


Fig. 10. Goodput performance for different algorithms

planning problem. The realistic server to server overload situations are more likely short periods of bulk loads, possibly accompanied by new sender arrivals or departures. Therefore, in this section we extend our evaluation to the dynamic behavior of overload control algorithms under load variations. Furthermore, we investigate the fairness property of each of the algorithms.

6.1 Fairness for SIP Overload Control

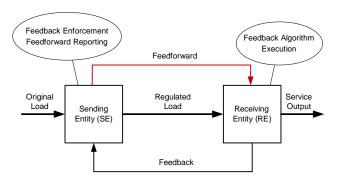


Fig. 11. The double feed architecture

Defining Fairness Under overload, the server may allocate its available capacity among all the upstream senders using criteria considered fair. Theoretically, fairness can be coupled with many other factors and could have an unlim-

ited number of definitions. However, we see two basic types of fairness criteria which may be applicable in most scenarios: service provider-centric and end user-centric.

If we consider the upstream servers representing service providers, a serviceprovider centric fairness means giving all the upstream servers the same aggregate success rate.

The user-centric fairness criteria aim to give each individual user who are using the overloaded server the same chance of call success, regardless of where the call originated from. Indeed, this end user-centric fairness may be preferred in regular overload situation. For example, in the TV hotline "free tickets to the third caller" case, user-centric fairness ensures that all users have equal winning probability to call in. Otherwise, a user with a service provider who happens to have a large call volume would have a clear disadvantage.

Achieving Fairness Technically, achieving the basic service provider-centric fairness is easy if the number of active sources are known, because the overloaded server simply needs to split its processing capacity equally in the feedback generated for all the active senders.

Achieving user-centric fairness means the overloaded server should split is capacity proportionally among the senders based on the senders original incoming load. For the various feedback mechanisms we have discussed, technically the receiver in both the absolute rate-based and window-based feedback mechanisms does not have the necessary information to do proportional capacity allocation when the feedback loop is activated. The receiver in the relative rate-based mechanism does have the ability to deduce the proportion of the original load among the senders.

To achieve user-centric fairness in absolute rate and window-based mechanisms, we introduce a new feedforward loop in the existing feedback architecture. The resulting double-feed architecture is shown in Figure 11. The feedforward information contains the sender measured value of the current incoming load. Like the feedback, all the feedforward information is naturally piggybacked in existing SIP messages since SIP messages by themselves travel in both directions. This way the feedforward introduces minimal overhead as in the feedback case. The feedforward information from all the active senders gives the receiver global knowledge about the original sending load. It is worth noting that, this global knowledge equips the receiver with great flexibility that also allows it to execute any kind of more advanced user-centric or service provider-centric fairness criteria. Special fairness criteria may be required, for example, when the server is experiencing denial of service attack instead of regular overload.

6.2 Dynamic Load Performance

Figure 12 depicts the arrival pattern for our dynamic load test. We used the step function load pattern because if the algorithm works in this extreme case, it should work in less harsh situations. The three UAs each starts and ends at

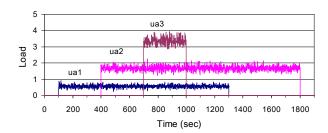


Fig. 12. Dynamic load arrival

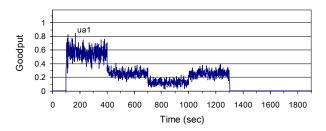


Fig. 13. win-cont UA1 goodput with dynamic load

different time, creating an environment of dynamic source arrival and departure. Each source also has a different peak load value, thus allowing us to observe proportional fairness mechanisms when necessary.

For dynamic behavior, our simulation shows that all algorithms except winauto adapts well to the offered dynamic load, showing little transition difference during new source arrival and existing source departure as well as at load change boundaries. As far as fairness is concerned, the rate-occ by default can provide user-centric fairness; the basic rate-abs, win-disc and win-cont algorithms are capable of basic service provider centric fairness by allocating equal amount of capacity to each SE. After implementing our double-feed architecture with sources reporting the original load to the RE, we are able to achieve user-centric fairness in all rate-abs, win-disc and win-cont algorithms through a proportional allocation of total RE capacity according to SEs' original incoming load. In addition, having knowledge of the incoming load proportion from each SE could also help us refine the algorithms when necessary. For example, in the win-cont case, we can improve the window allocation accuracy by using "weighted fair processing", i.e., available processing resources are probabilistically assigned to the SEs based on their proportional share of total incoming load. The improved algorithm is illustrated below:

$$w_i^0 := W_0$$
 where $W_0 > 0$
 $w_{left'}^0 := 0$

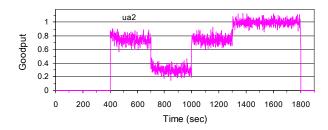


Fig. 14. win-cont UA2 goodput with dynamic load

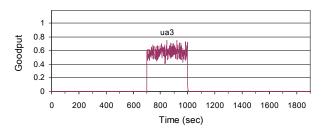


Fig. 15. win-cont UA3 goodput with dynamic load

```
\begin{split} w_i^t &:= w_i^t - 1 \text{ for INVITE received from } SE_i \\ w_{left}^t &:= N_{sess}^{max} - N_{sess} \text{ upon msg processing} \\ w_{left}^t &:= w_{left}^t + w_{left'}^t \\ if(w_{left}^t \geq 1) \\ w_{left'}^t &= (frac)(w_{left}^t) \\ w_{share}^t &= (int)(w_{left}^t) \\ \text{assuming the proportion of original load from } SE_i \text{ is } P\% \\ w_i^t &:= w_{share}^t \text{ with probability } P/100 \end{split}
```

Results of the win-cont algorithm with user-centric fairness are shown in Figure 13 through Figure 15. As can be seen, UA1 starts at the 100th second with load 0.57 and gets a goodput of the same value. At the 400th second, UA2 is started with load 1.68, three times of UA1's load. UA1's goodput quickly declines and reaches a state where it shares the capacity with UA2 at a one to three proportion. At the 700th second, UA3 is added with a load of 3.36. The combination of the three active sources therefore has a load of 5.6. We see that the goodputs of both UA1 and UA2 immediately decrease. The three sources settle at a stable situation with roughly 0.1, 0.3, and 0.6 goodput, matching the original individual load. At the 1000th second, the bulk arrival of UA3 ends and UA3 left the system. The allocation split between UA1 and UA2 restores to the similar situation before UA3's arrival at the 700th second. Finally, at the 1300th second, UA1 departs the system, leaving UA2 with load 1.68 alone. Since the

load is still over the server capacity, UA2 gets exactly the full capacity of the system with a goodput of 1.

The graph for service-provider centric fairness is similar, with the total allocation equally shared by the current number of active sources during each load interval.

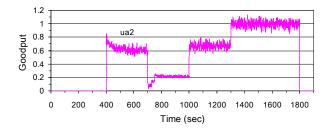


Fig. 16. win-auto UA2 goodput with dynamic load

We also evaluated the dynamic performance of the simplest win-auto algorithm. We found that with source arrival and departure, the system still always reaches the maximum goodput as long as the current load is larger than the server capacity. A difference from the other algorithms is that it could take a noticeably longer adaptation time to reach the steady state under certain load surge. For example, we show in Figure 16 the goodput for UA2. At the 700th second when the load increases suddenly from 2.25 to 5.6, it took over 60 s to completely stabilize. However, the good thing is once steady state is reached, the total goodput of all three UAs adds up to one. Moreover, performance under source departure is good. At the 1300th second, when UA2 becomes the only UA in the system, its goodput quickly adapts to 1. There is, however, one specific drawback of the win-auto mechanism. Since there is basically no processing intervention in this algorithm, we found it hard to enforce an explicit share of the capacity. The outcome of the capacity split seem to be determined by the point when the system reaches the steady state which is not easy to predict. Therefore, win-auto may not be a good candidate when explicit fairness is required. But because of its extreme simplicity, as well as near perfect steady state aggregate performance, win-auto may still be a good choice in some situations.

7 Conclusions and Future Work

The SIP server overload problem is interesting for a number of reasons: first, the cost of rejecting a request is not negligible compared to the cost of serving a request; Second, the various SIP timers lead to many retransmissions in overload and amplify the situation; Third, SIP has a server to server application level routing architecture. The server to server architecture helps the deployment of a

pushback SIP overload control solution. The solution can be based on feedback of absolute rate, relative rate, or window size.

We proposed three window adjustment algorithms win-disc, win-cont and win-auto for window-based feedback and resorted to two existing rate adjustment algorithms for absolute rate-based feedback rate-abs and relative rate-based feedback rate-occ. Among these five algorithms, win-auto is the most SIP specific, and rate-occ is the least SIP specific. The remaining three win-disc, win-cont, and rate-abs are generic mechanisms, and need to be linked to SIP when being applied to the SIP environment. The common piece that linked them to SIP is the dynamic session estimation algorithm we introduced. It is not difficult to imagine that with the dynamic session estimation algorithm, other generic algorithms can also be applied to SIP.

Now we summarize various aspects of the five algorithms.

The design of most of the feedback algorithms contains a binding parameter. Algorithms binding on queue length or queuing delay such as win-disc, win-cont and rate-abs outperform algorithms binding on processor occupancy such as rate-occ. Indeed, all of win-disc, win-cont and rate-abs are able to achieve theoretical maximum performance, meaning the CPU is fully utilized and every message processed contributes to a successful session, with no wasted message in the system at all. On the other hand, occupancy based heuristic is a much coarser control approach. The sensitivity of control also depends on the extra multiplicative increase and decrease parameter tuning. Therefore, from steady load performance and parameter tuning perspective, we favor algorithms other than rate-occ.

The adjustment performed by each algorithm can be discrete time driven such as in win-disc and rate-abs, rate-occ or continuous event driven such as in win-cont and win-auto. Normally the event-driven algorithm could have smaller number of tuning parameters and also be more accurate. But with a sufficiently short discrete time control interval the difference between discrete and continuous adjustments would become small.

We found that all the algorithms except win-auto adapt well to traffic source variations as well as bulk arrival overload. When we further look at the fairness property, especially the user-centric fairness which may be preferable in many practical situations, we found the rate-occ algorithm realizes it by default. All other algorithms except win-auto can also achieve it with our introduction of the double-feed SIP overload control architecture.

Finally, win-auto frequently needs to be singled out because it is indeed special. With an extremely simple implementation and virtually zero parameters, it archives a remarkable steady load aggregate output in most cases. The tradeoff to this simplicity is a noticeable load adaptation period upon certain load surge, and the difficulty of enforcing explicit fairness models.

Our possible work items for the next step may include adding delay and loss property to the link, and applying other arrival patterns as well as node failure models to make the scenario more realistic. It would be interesting to see whether and how the currently closely matched results of each algorithm may differ in those situations. Another work item is that although we currently assumed percentage-throttle for rate-based and window-throttle for window-based control only, it may be helpful to look at more types of feedback enforcement methods at the SE and see how different the feedback algorithms will behave.

8 Acknowledgments

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