Design and Simulation of Adaptive Measurement-based Admission Control Algorithms for Controlled-load Service

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Abstract: - The purpose of call admission control in Controlled-load Service network is to offer a Quality of Service (QoS) commitment to guarantee that QoS bounds are not violated. The traditional way of achieving this goal by declaring the worse case traffic descriptors for the incoming calls results in poor bandwidth utilization. In recent years, measurement-based admission control has become an appealing alternative to improve bandwidth utilization. It exploits the statistical gain and offers adaptivity to changing traffic condition. The measurement-based admission control first proposed by Jamin et. al in [1] was then improved by Casetti et. al in [2] with an adaptive measurement window in the algorithm. Building on the work of [2], this paper examines the algorithm design and simulation of adaptive sampling time in the algorithm. The proposed algorithms are tested through simulations under difference traffic scenarios and proven to produce higher level of utilization without violating the delay-based QoS guarantees. The simulation results are compared with the algorithm described in [2].

Key-Words: - Bandwidth Optimization, Admission Control, QoS-enabled Internet

1 Introduction

The Controlled-load Service[3] network proposed by the Internet Engineering Task Force (IETF) ensures that adequate bandwidth and packet processing resources are available to handle the requested level of traffic. Resource Reservation Protocol[4] (RSVP), for example, has been designed to provide QoS setup mechanisms for Integrated Service classes such as Controlled load service.

The Controlled-load Service does not make any guarantees about the delay but promises to approximate the end-to-end behavior of the applications to the best-effort applications under low-load. The admission control algorithm has to make sure that this approximation is met. In this paper, adaptive measurement-based admission control algorithms are used to provide this facility to controlled-load network element.

While the problem of optimizing bandwidth utilization in QoS-enabled network has been widely studied in recent years, the idea of using online measurements in the admission control algorithm has been studied in only a few selected works.

Measurement based admission control algorithm for Controlled-load service network is first studied in [5] and [1]. In [5] and [1], an admission control scheme based on delay and traffic rate measurement is used to predict the bandwidth utilization and admitting new calls. This work was further improved in [2], where the adaptive time window measurement scheme was introduced to improve the bandwidth utilization and traffic delay. As this paper builds on [2], the network model and admission control algorithm in that paper will be briefly described in Section 2 and Section 3 respectively. Interested reader is referred to [2] for further details.

2 Network Model

A single node network scenario is considered in this paper. This simplifies the network scenario to a multiple-source, single server and single traffic sink model, as shown in Figure 1.

The network model in Figure 1 consists of a link of capacity μ connecting two switches R1 and R2. As packet loss probability is not a factor in this study, each node (the switch) is assumed to have an infinite buffer. This allows us to focus on the delay and traffic rate as the performance metrics for the admission control algorithm under a specific network scenario.

Sources S connect to the first node R1 through links of infinite bandwidth. The first node concentrates (multiplexes) packets destined to the downstream node, R2. The outgoing link capacity, μ, is set as 10Mbps to preserve the simulation scenario in [2]. All traffic generated from the sources belongs to Controlled-load Service class.

The traffic sink, R2 is included to model the outgoing link capacity from server R1 in the simulation.

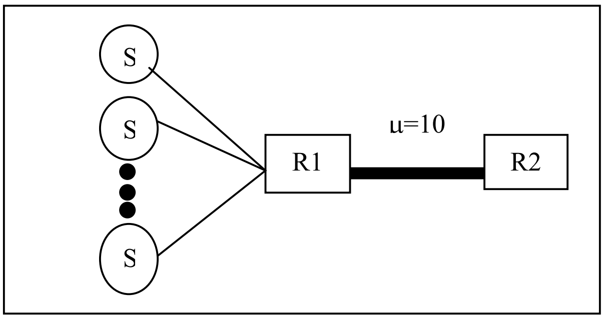

Figure 1: Network Model.

2.1 Source Model and Traffic Characterizations

Traffic can be characterized based on unicast traffic models, such as exponential ON/OFF models, or other models that are not compatible with long-range dependency. Along with many other researchers, such as [6, 7 and 8], measurement-based admissions control procedures used in this paper make as few assumptions as possible about traffic characterizations.

In our network model, traffic sources, S, are modeled as random logical processes. New calls arrive at switch R1 according to a Poisson process with mean rate r_s calls per second. The call holding times are independent and identically distributed exponential random variables with mean T_s. This will create a random number of active flows (calls) at any moment of the simulation time. The traffic generated by admitted calls is modeled by an On/Off Markov-modulated fluid process. While in the On state, a source generates traffic at peak rate r; during the Off period no traffic is generated.

In our simulations, the value is set to r_s = 10 calls/sec and T_s = 135sec. The On and Off periods are exponentially distributed with mean $1/\alpha$ = 0.3125s and $1/\beta$ = 0.3125s respectively. Depending on the scenario considered, the peak rates of admitted sources are either fixed to r=64kb/s (referred as homogeneous sources) or exponentially distributed with mean r = 64 kb/s (heterogeneous sources). Based on equivalent token bucket characterizations, the value for bucket size is chosen to be 1 to produce a token bucket rate which is equivalent to the peak rate. As has been noted in [2], with the choice of the parameters above, a network without any admission control mechanism would achieve a near 100 % utilization.

The motivation for choosing fluid sources, rather than a discrete, packetized source model is that it contributes to the simplicity and accuracy in source modeling and traffic measurement. By virtue of this choice, the simulation is not committed to any packet size. However, tokens are representative of whole data units such as cells and packets. In this simulation model, a token is equivalent to a 1000bit data unit. The source is modeled as a Markov-modulated fluid process whose smallest unit is bit, as proposed in [2], for the same reason.

Fluctuating traffic arrival pattern has also been included in the model to test the robustness of admission control algorithms operating in extreme network environment. A Different source creation pattern is used to create the fluctuating traffic environment. The network is exposed to extremely low call arrival rate for a significantly long period of time (causing the network to be under-loaded) before the call arrival rate is switched abruptly to its normal high value which will result in network overload. This method is particularly useful to test the effectiveness of adaptive measurement-based admission control algorithm in preserving QoS commitment when there is an abrupt change in call arrival rate. A responsive measurement-based admission control will react (adapt) quickly to the sudden change in call arrival rate, and make accurate admission control decisions by not over-admitting flows into the network immediately after the change occurs.

To model the fluctuating traffic arrival pattern, the call arrival rate is switched every 100 seconds on average, between 10 calls per second with a probability of 2/3 and 0.01 call per second with a probability of 1/3. The value of rate switching interval is an independent identically distributed random variables with a mean of 100 seconds. This parameter setting results in a fluctuating call arrival pattern, thus, creates fluctuating network traffic.

2.2 Server Model

The switches (R1 and R2) in Figure 1 are modeled as a First-in-First-out (FIFO) server with infinite buffer. With this simplified model, packet loss probability is not a factor in the simulation. In essence, this server emulates a M/M/1/∞/FIFO system in all aspects, except that its service time is fixed rather than exponentially distributed. The rationale of modeling the server as a FIFO server rather than a priority server is that all arriving calls are seeking admission to Controlled-load Service (and thus, all the admitted traffic belongs to the

Controlled-load Service class). The FIFO server has to facilitate the measurement of traffic arrival rate and the delay experienced by incoming traffic waiting to be served. This can be done by calculating the traffic arrival rate, and queuing all arriving traffic into a switch buffer while waiting to be served by the FIFO server. The queue's serving rate of the FIFO server is set to be equivalent to the outgoing link capacity, which is μ = 10Mbps.

The FIFO server also provides packet-forwarding service. As all outgoing traffic from the server is directed to switch R2, the server model requires no information about routing.

3 Algorithm Design and Simulation

The goal of admission control process is to admit a call only if there are sufficient resources to satisfy the QoS requirements of the new call, while at the same time not violating QoS commitment made to the admitted calls.

The proposed admission control algorithm can be decomposed into three logically separable modules, i.e. the 'measurement mechanism', the 'admission algorithm' and 'tuning algorithm' used to tune the parameter used for making the traffic measurement, which is referred to as the 'tuning algorithm'. In essence, the algorithms proposed in this paper extend the 'tuning algorithm' and 'admission algorithm' in [2].

3.1 Admission Algorithm

There exist a few variants of measurement-based admission control algorithms as well as measurement mechanisms. The 'measured sum' algorithm is adopted in this paper as the admission algorithm, with a 'time-window' measurement mechanism.

The 'measured sum' algorithm uses measurement to estimate the load of existing traffic. This algorithm admits the new flow if the following test succeeds:

$$\upsilon + r(\alpha) < \nu\mu,, \qquad (1)$$

where

υ is the estimated load of existing traffic

$r(\alpha)$ is the peak rate requested by the flow α

μ is the outgoing link bandwidth

ν is the user-defined utilization target

Upon admission of a new flow, the load estimate is

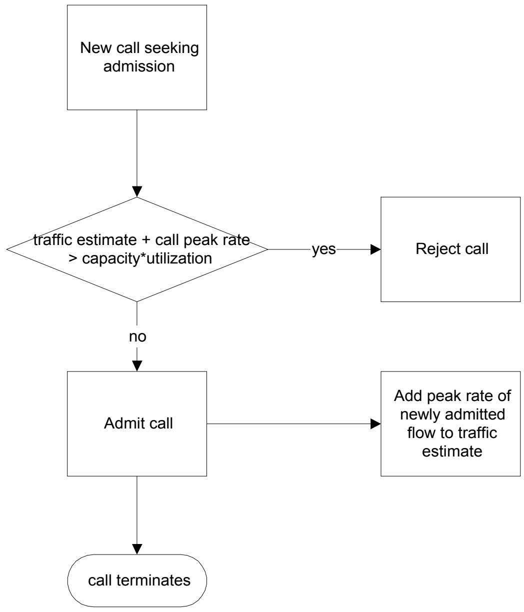

Figure 2: 'Measured sum' algorithm as the admission algorithm.

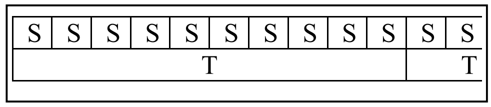

Figure 3: The Relationship between Measurement Window, T and Sampling Period, S.

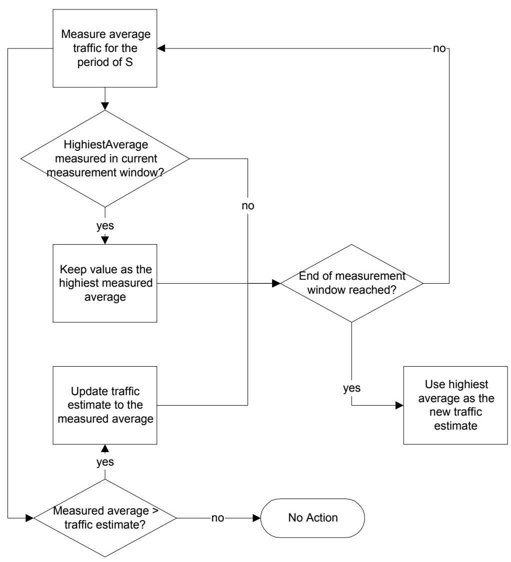

Figure 4: Traffic estimate using 'time window' measurement mechanism.

increased to $\upsilon' = \upsilon + r(\alpha)$. The 'Measured sum' algorithm as the admission algorithm is shown in Figure 2.

3.2 Measurement Mechanism

The traffic measurement is made by computing an average load every sampling period, S. At the end of a measurement window T (T>10S), the highest average load from the just ended T is used as the load estimate for the next measurement window. The relationship between T and S is shown in Figure 3. The traffic estimate is made using a measurement mechanism shown in the flow chart in Figure 4. When a new flow is admitted to the network, the estimate is increased by the peak rate of the new request. If a new computed average (from measurement), during any S, is greater than the estimate, the estimate is immediately raised to the new average. At the end of every T, the estimate is adjusted to the highest measured average in the previous T.

3.3 Tuning Algorithm

Previous work done by Casetti et. al in [2] has shown that a longer measurement window T yields a larger reading as a longer history of sampling periods is kept. This results in a more conservative admission control (less calls admitted). On the other hand, if the measurement window T is shorten, then more calls can be admitted as the estimated traffic tends to be smaller in value. This results in higher bandwidth utilization. However, higher utilization comes at the risk of increasing delay violation for traffic in admitted flows. It has also been shown that a smaller sampling period S gives higher maximal averages, resulting in a more conservative admission control algorithm, and vice versa.

These results have led to the development of an adaptive measurement-based admission control in which the measurement window T is adaptively adjusted according to the instantaneous network traffic condition. A two-level algorithm is used as a tuning algorithm in [2] to achieve this goal. The tuning algorithm introduced in [2] uses an adaptive measurement window T and a fixed sampling period S. The algorithm is illustrated in Figure 5 and Figure 6.

3.3.1 Level One Algorithm

The idea behind the level one algorithm is to continually shrink the length of the measurement window, T(by using a window reshaping factor, f_{down}). This results in admission control decisions that work towards increasing link utilization. The trigger value in Figure 5 is a value set to be smaller that the bandwidth allocated for the controlled-load traffic class, to provide an early warning that the system is about to reach a full load level, where it may be requested to readjust the admission control criteria. The level one algorithm will then reacts by enlarging the measurement window (by using window reshaping factor f_{up}) until the measured rate drops below the trigger, at which the window may be shrunk again. Their values are set to $f_{up} = 1.2$ and $f_{down} = 0.9$.

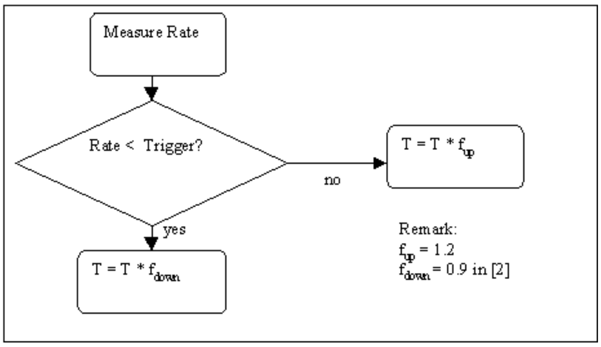

Figure 5: Level One of The Adaptive Measurement Window Algorithm

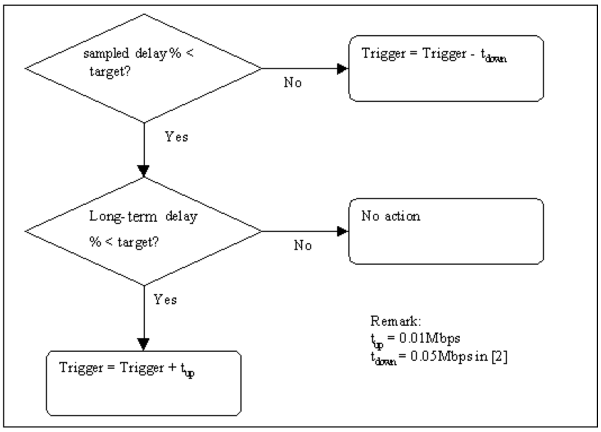

Figure 6: Level Two of The Adaptive Measurement Window Algorithm

3.3.2 Level Two Algorithm

The level two algorithm (Figure 6) takes charge of the trigger value, by lowering or raising the trigger value in search of a good operating point in response to traffic fluctuation. The targeted value for the sampled delay violation percentage and long-term delay violation percentage is preset at time 0.

The first delay measurement is the "sampled delay violation percentage" over a sampling period. It provides an indication of the percentage of bits that violats the delay bound in the pass sampling period. The second delay measure is the "long term delay violation percentage", which keeps track of the percentage of late bits observed since time 0.

Whenever the level one algorithm attempts to modify the window length, it checks the flag indicating whether the sampled delay percentage has been detected to be higher than the targeted delay percentage during any sampling period during the pass window. If this is the case, the admission control algorithm has been behaving too aggressively. Therefore the trigger rate value is lowered, resulting in more conservative admission control. If the sampled delay percentage never exceeded the target, the algorithm checks whether the long term delay violation percentage is currently under the target value. If it is, the trigger value is raised. Otherwise, no action is performed. The last option is motivated by the fact that if the sampled target delay is not violated, the trigger is sufficiently low and the overall delay only needs more time to settle under the target. Trigger increment and decrement is indicated by t_{up} and t_{down} in Figure 6. The value used for t_{up} and t_{down} under the 10Mbps link capacity is t_{up} = 0.01Mbps and t_{down} = 0.05Mbps in [2]. The same value is preserved in our model.

An interesting parameter in the level two algorithms is the delay bound which is used to detect the delay violation of bits being served by the FIFO server. The delay bound is the maximum time that any bit can be queued on the server's buffer before being declared as a late bit. Late bits indicate violation of service commitment and implies a network overload. The targeted delay percentage is guaranteed over the period of the simulation time. The value of delay bound can be chosen by network administrators as it is not quantitatively specified for provision of Controlled-load Service. However, the value of the delay bound must be no greater than the flow's "burst time" and there should be minimal loss rate averaged over time-scales larger than "burst time" [3], where the "burst time" is defined as the time required to serve a flow's maximum burst at the flow's reserved rate.

In the simulation, the value of delay bound is chosen to be 16ms all experiments to facilitate meaningful performance comparisons between different admission control algorithms. Based previous works, especially [2] and [9], the following constraints are applied to the parameter setting in our adaptive algorithm model:

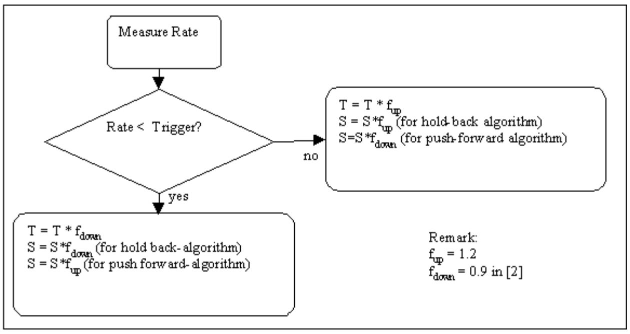

Figure 7: The extended level one algorithm.

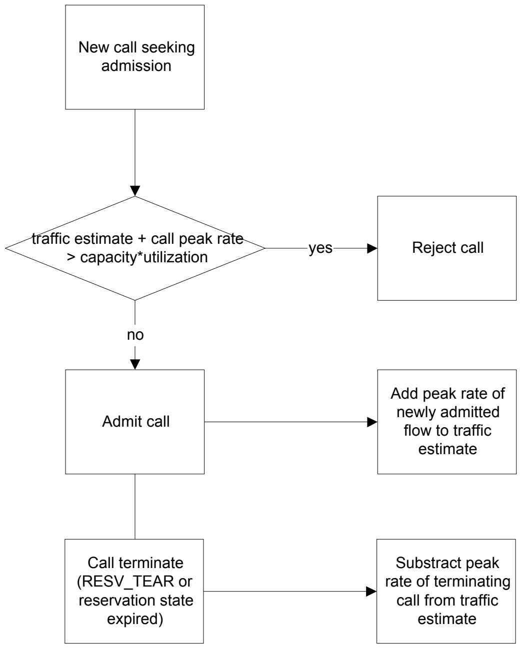

Figure 8: The Extended 'measured sum' algorithm.

1. The measurement window, T, is not allowed to shrinking below 1 second to avoid overly aggressive admission scheme.
2. The sampling period, S, of value less than T/10 is disallowed to ensure statistically meaningful number of sampled rate value to make traffic estimate. A default value of S = T/20 is recommended.
3. The initial value of target utilization, ν in Equation (1), is set to 0.95 or 95%. The decision to set high target utilization is made based on the fact that the admission control algorithm is well protected by the 'trigger' value in the level one algorithm.

3.4 Proposed Extensions

In this paper, the algorithm proposed in [2] is extended in two important directions. The extensions are:

1. Adaptive tuning of sampling period, S, concurrently when the measurement window, T, is reshaped.
2. Dynamic update of traffic estimate as admitted sources terminate.

The algorithm in the first extension is referred as 'adaptive measurement window and adaptive sampling period measurement-based admission control algorithm'. The intent here is to examine the effects of adaptively tuning the sampling period S together with the adaptive tuning of the measurement window, T. This requires only a minor modifications to the level one algorithm of the 'tuning algorithm' shown previously in Figure 5.

There are two approaches in which the sampling period can be tuned adaptively.

Firstly, the sampling period, S, can be shrunk by the same reshaping factor as the measurement window shrinks, and vice-versa. Shrinking the sampling period will result in a higher measured average, whereas shrinking the measurement window will result in a lower traffic estimate. Thus, it is expected that by shrinking the sampling period, the aggressiveness resulting from the decrease in the measurement window of the admission scheme can be reduced. It is worth noting that, on the other hand, the sampling period will be increased as the measurement window is increased. This, on the contrary, will reduce the conservativeness of the admission scheme when the measurement window is increased. In essence, this scheme is expected to have the effect of holding back the admission control algorithm from behaving too aggressively or too conservatively. For this reason, it is referred as 'holding-back algorithm' hereafter. The actual performance of the algorithm is examined in through simulations.

In the second approach, the sampling period is shrunk as the measurement window is increased, and vice-versa. It is expected that this approach will have the effect of increasing the conservativeness of the admission scheme whenever the measurement window is expanded to increase the traffic estimate, while increasing the aggressiveness of the scheme whenever the measurement window is shrunk to decrease the traffic estimate. For this reason, this algorithm is referred as 'pushing-forward algorithm'. It is expected that this algorithm can increase the speed by which the algorithms react to changing network environment, such as fluctuating traffic pattern. The algorithm is also examined in the simulation.

The level one algorithm for both the 'holing-back algorithm' and 'pushing forward algorithm' is shown in Figure 7.

The second extension to the algorithm is to dynamically update the traffic estimate as admitted sources terminate. Source termination signal can be provided through QoS setup mechanisms such as RSVP or SNMP. This is a useful information for the admission control algorithm in making traffic estimate. In all previous studies [6, 7, 8, 9 and 10], when a new flow is admitted to the network, the estimate is increased by the parameter (peak rate) of the new request, but the traffic estimate is not decreased by the same parameter (peak rate) when a source terminates. This is because the resulting admission scheme is overly aggressive. However, it also results in inaccurate estimate when sources terminate, which in turn makes the admission scheme more conservative in admitting new calls.

The addition of this feature does not change the level one and level two algorithm described earlier. However, the dynamic update of traffic estimate requires a slight change in the 'measured sum' algorithm. The extended 'measured sum' algorithm is shown in Figure 8. The new algorithms ware simulated to test the effects of dynamic update on traffic estimate as sources terminate.

From the combination of these two extensions, six of algorithms have been developed. They are:

Algorithm 1:
Fixed-sampling rate algorithm (as proposed in [2])

Algorithm 2:
Pushing Forward Algorithm.

Algorithm 3:
Holding Back Algorithm.

Algorithm 4:
Fixed-sampling rate algorithm with Source Termination Update.

Algorithm 5:
Pushing Forward Algorithm with Source Termination Update.

Algorithm 6:
Holding Back Algorithm with Source Termination Update.

The performance of all the proposed algorithms is examined through simulations. The results are presented in the next section.

3.5 Simulations

The simulations have been implemented in PARSEC, a C-based parallel discrete event simulation language. The design of the simulation is mainly based on PARSEC approach of adopting message passing among simulation entities to simulate the occurrence of logical processes in the real network.

The simulations have shown encouraging results in terms of performance for the proposed algorithms. It has been proven through simulations that adaptive tuning of sampling period concurrently with the tuning of measurement windows has resulted in either improved utilization percentage or late bit percentage, or both at the same time.

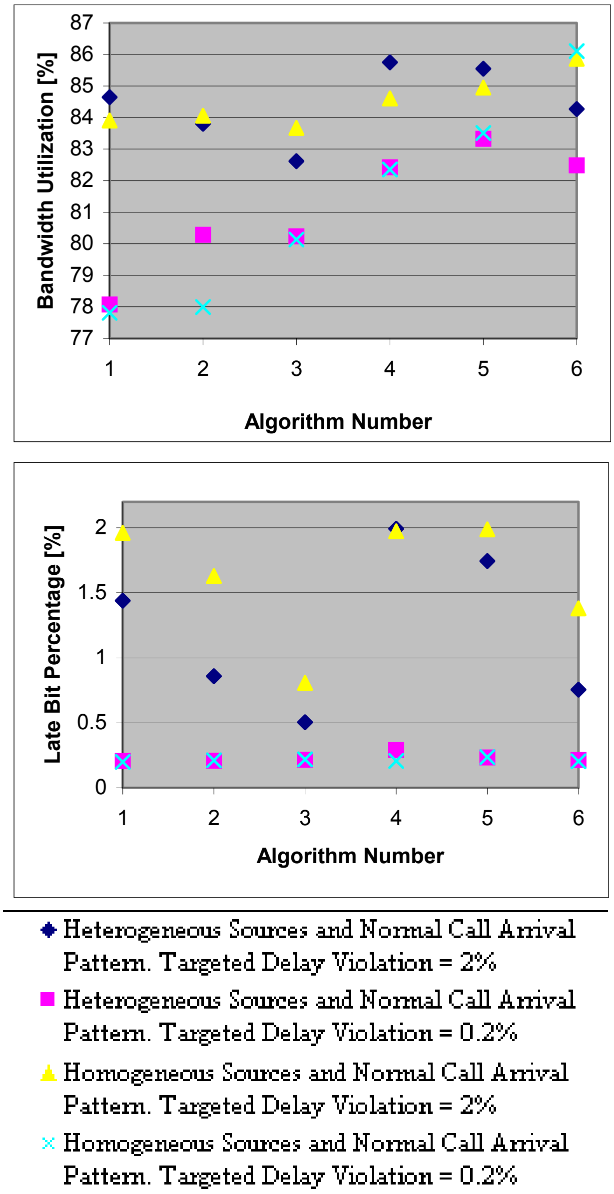

Figure 9: The Bandwidth Utilization and Late Bit Percentage Performance under Normal Call Arrival Pattern.

Table 1: The Performance of Proposed Algorithms under Fluctuating Call Arrival Pattern.

Algorithm No.	Targeted Delay Violation [%]	Late Bits [%]
1	2.0	0.6075
2	2.0	0.6048
3	2.0	0.4363
4	2.0	1.3893
5	2.0	0.8647
6	2.0	0.4167
1	0.2	0.2052
2	0.2	0.2023
3	0.2	0.2121
4	0.2	0.2366
5	0.2	0.2028
6	0.2	0.2524

Figure 9 and Table 1 show the performance of all the proposed algorithm compared to Algorithm 1 proposed in [2].

Based on the observations and analysis made on the simulation results, there are a few important findings:

1. If a high utilization percentage is the major goal of an admission control algorithm, the algorithm should include source termination update the traffic rate estimate. Even though Algorithm 4, 5 and 6 produce a high percentage of late bits served by the network node, the resulting late bit percentage is kept below the targeted value. The holding back algorithms apparently out-perform the pushing forward algorithms in terms of late bit percentage.
2. For algorithms that include source termination update in traffic rate estimate, (i.e. Algorithm 4, 5 and 6), there is an significant improvement in late bit percentage especially when the targeted delay violation percentage is set to a high value (2%). In cases where the targeted delay violation percentage is tight (0.2%), adaptive tuning of sampling period not only results in improved late bits percentage, but also a higher utilization percentage.
3. Algorithm 6 shows a good tradeoff between the performance goals in an admission control algorithm with a relatively high utilization and being able to keep the late bit percentage at the lowest level.
4. Adaptive tuning of sampling period generally exhibits a behavior known as "complementary behavior". When the targeted delay violation percentage is high, the use of adaptive sampling period results in improved late bit percentage in

long run at the expense of lower utilization percentage. However, when the targeted delay violation percentage is low, the use of adaptive sampling period will results in improved utilization percentage. The pushing forward algorithms (Algorithm 2 and Algorithm 5) achieve a higher utilization percentage at the expense of a higher late bits percentage, when compared to the holding back algorithms (Algorithm 3 and Algorithm 6).

5. Algorithms that adaptively tune their sampling period (i.e. Algorithm 2, 3, 5 and 6) suffer from massive fluctuation in late bit percentage over the simulation runs. This is particularly obvious for the pushing forward algorithms (Algorithm 2 and Algorithm 5), which experience fluctuation of late bit percentage over a wider range of values compared to the holding back algorithms (Algorithm 3 and Algorithm 6). Algorithms that use a fixed sampling period (Algorithm 1 and Algorithm 4) do not experience such problem.
6. In a fluctuating traffic environment (where fluctuating call arrival pattern is used), 'pushing forward' algorithms (Algorithm 2 and 5) outperform the other algorithms in terms of late bit percentage performance. It is worth noticing that 'holding back' algorithms (Algorithm 3 and 6) demonstrate poorer performance compared to fixed sampling period algorithms (Algorithm 1 and 3) under fluctuating traffic environment.

4. Conclusion

The design and simulation of measurement-based admission control with an adaptive sampling period has been examined in this paper. In addition, the effects of source termination update in traffic rate estimate have also been addressed. The simulation results have shown that the proposed algorithms achieve significant performance improvements compared to the existing measurement-based admission control algorithm in [2] in both bandwidth utilization and QoS commitment. Despite simplicity in approach, the proposed extensions to the algorithms have also improved the performance of the admission control for controlled-load service in fluctuating traffic environment, which has proved difficult to be handle, particularly under tight QoS commitment. However, the algorithm sensitivity to the parameters in its tuning algorithm remains a topic to be further examined.

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