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On the throughput analysis of rate-based and window-based congestion control schemes ☆

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Abstract

The existing TCP protocols contribute to the stability of the Internet by deploying window-based additive-increase multiplicative-decrease (AIMD) congestion control. However, the window-based congestion control is not favorable in the existence of long propagation delay and for real-time multimedia traffic. Thus, the rate-based congestion control has become an attractive alternative for the links with high delay and for multimedia flows. In this paper, an analysis of the *rate-based* generic AIMD congestion control scheme is presented. The analytical model is validated via simulation experiments for a wide range of link conditions and the effects of rate-increase and decrease parameters on the performance are investigated. One important result of this analysis is that throughput performance of the rate-based schemes is inversely proportional to square-root of the propagation delay whereas that of the window-based schemes is known to be inversely proportional to the propagation delay itself. This result also justifies that the rate-based congestion control schemes have higher resilience to the high link delay than the window-based schemes.

Keywords: Rate control protocols; Rate-based congestion control; AIMD; Multimedia transport

1. Introduction

The stability of the Internet is primarily based on the deployment of congestion control by the end nodes. The existing TCP protocols deploy window-based additive-increase multiplicative-decrease (AIMD) congestion control, which requires an acknowledgment (ACK) reception to grant the injection of a new data packet to the network [3]. Despite the success of the existing TCP protocols for reliable data transport in the Internet, its window-based behavior leads to inefficiency under specific circumstances.

The window-based TCP protocols are shown to provide poor performance in the environments with high bandwidth-delay product [7]. For example, the TCP protocols experience severe throughput performance degradation in the satellite links due to the inadequacy of window-based TCP protocols in the long round-trip time

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(RTT) environments [2]. Moreover, because of the window-based ACK-controlled packet injection method, high link delays further amplify the performance degradation experienced by the TCP protocols due to misinterpretation of wireless link related packet losses as congestion. Similarly, the window-based approach is responsible for very poor utilization in extremely long propagation delay environments such as deep space communication networks [1]. For this reason, much research is currently ongoing to develop rate-based congestion control protocols for deep space networks [15].

On the other hand, the window-based TCP protocols are also not suitable for the delivery of multimedia services. The multimedia flows do not require 100% reliable transport, instead, they need timely delivery and smooth rate variation. Nevertheless, there exists a necessity of a rate control mechanism for multimedia flows to avoid unfairness to TCP sources and further congestion collapse [6]. The window-based approach, however, is not suitable for continuous multimedia streaming since its ACK-controlled packet injection method does not maintain smooth rate variation rather creates bursty traffic output. In order to address the shortcomings of the window-based approach, many rate-based schemes [4,5,8,12–14] are proposed in the literature to meet the requirements of multimedia streaming.

The throughput performance of the window-based TCP protocols are extensively studied and the steady-state throughput models for TCP-Reno [11] and window-based AIMD schemes [6,16] are developed and presented in the literature. These models provide very utile tools for the performance analysis of these protocols. The model developed for window-based TCP behavior in [11] is also adopted as a TCP-Friendliness measure to develop an equation-based congestion control in [6]. In this paper, an analysis of the *rate-based* generic AIMD congestion control scheme is presented. The analytical throughput model is developed and the effects of the rate-increase and decrease parameters on the performance are investigated. The simulation experiments validate the model for a wide range of environments, which makes it of significant value for future analysis of the rate-based AIMD congestion control schemes. One important result of this analysis is that throughput performance of the rate-based schemes is inversely proportional to square-root of the propagation delay whereas that of window-based schemes is known to be inversely proportional to propagation delay itself. This result also justifies that the rate-based schemes have higher resilience to the high link delay than window-based schemes. Furthermore, we have explored the method of AIMD parameter adaptation as an application of the analysis presented in this paper. The simulations experiments show that AIMD parameter adaptation according to the link conditions can significantly improve the throughput performance.

The remainder of the paper is organized as follows. The analysis of the *rate-based* generic AIMD congestion control scheme is presented in Section 2. The validation of the model, its preliminary results along with the effects of the AIMD parameters on the throughput performance are then discussed in Section 3. Finally, the paper is concluded in Section 4.

2. Rate-based generic AIMD congestion control

The time-dependency of the transmission rate, $S(t)$, is shown in Fig. 1. The rate-based generic AIMD scheme is assumed to increase the rate, S, additively with α at each RTT, i.e., $S = S + \alpha$. It throttles the transmission rate, S, multiplicatively by β if a packet loss is detected, i.e., $S = S \cdot \beta$. The steady-state throughput of such *rate-based* AIMD scheme is derived here.

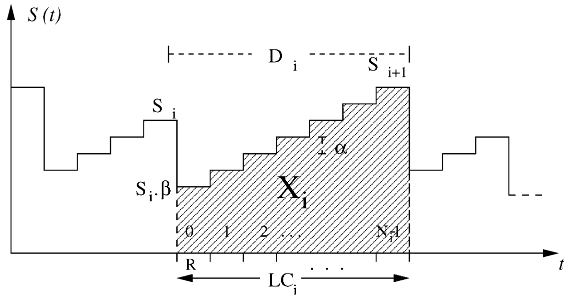

Fig. 1. Data rate change of rate-base general AIMD congestion control.

Let $X(t)$ be the total number of packets transmitted in $[0, t)$, which can be calculated by

$$X(t) = \int_0^t S(\tau)\,\mathrm{d}\tau. \tag{1}$$

For $T(t) = X(t)/t$ being the throughput achieved in $[0, t)$, the steady-state throughput achieved by the rate-based generic AIMD scheme is given by

$$T = \lim_{t\to\infty} \frac{1}{t} \int_0^t S(\tau)\,\mathrm{d}\tau. \tag{2}$$

Let X_i be the number of packets transmitted in the ith loss cycle, LC_i, which starts and ends with rate halving due to congestion decision. If the duration of LC_i is D_i, then the throughput achieved in LC_i is given by $T_i = X_i/D_i$. Assuming the evolution of the transmission rate S_i to be Markov regenerative process with rewards X_i [9], then the steady-state throughput can be calculated by

$$T = E[X]/E[D], \tag{3}$$

where $E[X]$ and $E[D]$ are the means of X_i and D_i, respectively. The transmission rate change $S_{i,k}$ during LC_i, i.e., $S_{i,k}$ being the value of the transmission rate at the end of kth RTT of the LC_i, is given by

$$S_{i,k} = S_i \cdot \beta + k \cdot \alpha. \tag{4}$$

Given that N_i is the total number of RTTs in LC_i, i.e., $k \in [0, N_i - 1]$ and the rate is decreased after the N_ith RTT is over, then the transmission rate at the end of the LC_i (before the rate decrease occurs) is expressed by $S_{i+1} = S_i \cdot \beta + N_i \cdot \alpha$. Hence, the expectation of the i.i.d random variable S denoting the transmission rate, i.e., $E[S]$, can be calculated as

$$E[S] = \frac{\alpha}{1-\beta} \cdot E[N]. \tag{5}$$

If LC_i lasts for $D_i = N_i \cdot R$, where R is the round-trip time (RTT), then the total number of packets transmitted during LC_i can be calculated by

$$X_i = \int_0^{D_i} S_i(t) \cdot \mathrm{d}t. \tag{6}$$

For a rate-based AIMD scheme, whose rate change is performed with RTT granularity, this can be calculated by replacing the integration in (6) with the discrete summation and substituting (4) as follows:

$$\begin{aligned} X_i &= \sum_{k=0}^{N_i-1} S_{i,k} \cdot R = \sum_{k=0}^{N_i-1} (S_i \cdot \beta + k \cdot \alpha) \cdot R \\ &= \frac{N_i}{2} [2 \cdot \beta \cdot S_i + \alpha \cdot (N_i - 1)] \cdot R. \end{aligned} \tag{7}$$

Here, we assume that N and S are mutually independent random variables. While higher S would seem to result in smaller N intuitively; it is, however, also not likely to reach high S (from a halved rate) with smaller N via additive rate increase with RTT period. Furthermore, N could be also very small, while S is also small, due to the congestion caused by other entering flows sharing the same bottleneck. Therefore, assuming that N and S are mutually independent random variables is reasonable and leads to an approximate solution for the calculation of the expectation of the X. Thus, the mean of X can be calculated by taking expectation of both sides of (7) and substituting (5) as follows:

$$E[X] = \frac{E[N]}{2} \cdot \left[\left(\frac{1+\beta}{1-\beta}\right) \cdot E[N] - 1\right] \cdot \alpha \cdot R. \tag{8}$$

On the other hand, the total number of packets transmitted in loss cycle i can also be calculated by $n_i + S_{i,N_i-1} \cdot R$, where n_i and $S_{i,N_i-1} \cdot R$ are the number of packets transmitted until the dropped packet and in the last RTT, respectively. Therefore, $E[X]$ can also be calculated by

$$E[X] = E[n] + E[S] \cdot R, \tag{9}$$

where the expectation of the random variable n is given by

$$\begin{aligned} E[n] &= \sum_k kP[n_i = k] = \sum_{k=0}^{\infty} k(1-p)^{k-1} p \\ &= \frac{1}{p}, \end{aligned} \tag{10}$$

where p is the packet loss probability, if loss-based congestion detection is used by the generic AIMD rate-based congestion control algorithm. By substituting (5) and (10) into (9), and equating it to (8), we solve for $E[N]$ and obtain it as follows:

$$E[N] = \frac{3-\beta}{2(1+\beta)} \left[1 + \sqrt{1 + \frac{8(1-\beta^2)}{\alpha R p (3-\beta)^2}}\right]. \tag{11}$$

Thus, it follows from (3), (8) and (11) that the steady-state throughput of the rate-based general AIMD rate control scheme as a function of rate-increase and decrease parameters, i.e., α and β, and round-trip time, R, and the packet loss probability p, can be expressed as follows:

$$\mathscr{T}^{r}_{\alpha,\beta}(p,R) = \frac{\alpha}{4(1-\beta)}\left[1+\beta + \sqrt{(3-\beta)^2 + \frac{8(1-\beta^2)}{\alpha R p}}\right]. \qquad (12)$$

3. Validation and preliminary results

In this section, the developed model is first validated via simulation experiments over a wide range of link conditions. Then, the preliminary results of the analysis are discussed and its possible applications are explored via simulation experiments in Sections 3.2 and 3.3, respectively.

3.1. Validation experiments

In order to evaluate the validity of the model, we simulate a rate-based generic AIMD congestion control scheme via *ns-2* [10] network simulator. We simulate the topology given in Fig. 3, where $N = 10$ sources are connected to the IP router, which is also connected to the backbone link with capacity of $C = 2 \cdot 10^5$ packets/s. We perform simulation experiments for a wide range of packet loss probability, p, and round-trip time, R, to assess the validity of the model in diverse environmental conditions, i.e., $10^{-4} \leqslant p \leqslant 0.5$ and $0.001\ \text{s} \leqslant R \leqslant 500$ s.

We input α and β as the configuration parameters of the protocol. The simulated rate-based protocol operates as follows: it increases the transmission rate S additively with α at each RTT, i.e., $S = S + \alpha$; and decreases the rate S multiplicatively with β in case of packet loss, i.e., $S = S \cdot \beta$.

The comparisons of the measured throughput achieved by the rate-based congestion control protocol and the throughput values estimated by the model are shown in Fig. 2(a) and (b), for $10^{-4} \leqslant p \leqslant 0.5$ and $0.001\ \text{s} \leqslant R \leqslant 500$ s, respectively. The throughput values shown in Fig. 2(a) and (b) are the average throughput achieved by the sources sharing the bottleneck link. The range of very high RTT values, i.e., $100\ \text{s} \leqslant R \leqslant 500$ s, are included in the experiments to validate (12) for the analysis of rate-based AIMD schemes in deep space communication networks. As shown in Fig. 2(a) and (b), the throughput values estimated by the model tracks the measured throughput values very accurately for all ranges of p and R, and for different set of AIMD parameters, i.e., $(\alpha,\beta) \in \{(10,0.5),(50,0.6),(100,0.7)\}$. Therefore, the model can be used to analyze rate-based AIMD congestion control schemes for very wide range of link conditions.

3.2. Preliminary results

The effects of AIMD parameters on the throughput can be directly captured from (12). For instance, it can be observed from (12) that the increase in additive-increase and multiplicative-decrease factors, i.e., α and β, lead to increase in the throughput. This is also consistent with the results of validation experiments shown in Fig. 2(a) and (b). The important difference between rate-based and window-based congestion controls is the effect of RTT on the throughput. The throughput of the window-based TCP protocols is inversely proportional to the *RTT* [11] and this leads to severe throughput degradation in the environments with long propagation delay. However, as observed in (12), the throughput of rate-based congestion control is inversely proportional to the square-root of RTT. Hence, this makes the rate-based congestion control schemes more resilient to excessive propagation delays. Therefore, the rate-based schemes are one of the most important design principles of congestion control for high propagation delay environments [15].

Another important preliminary usage of (12) can be the selection of the AIMD parameters to achieve a target throughput under certain link characteristics. Let B packets/s be the target throughput determined by the application

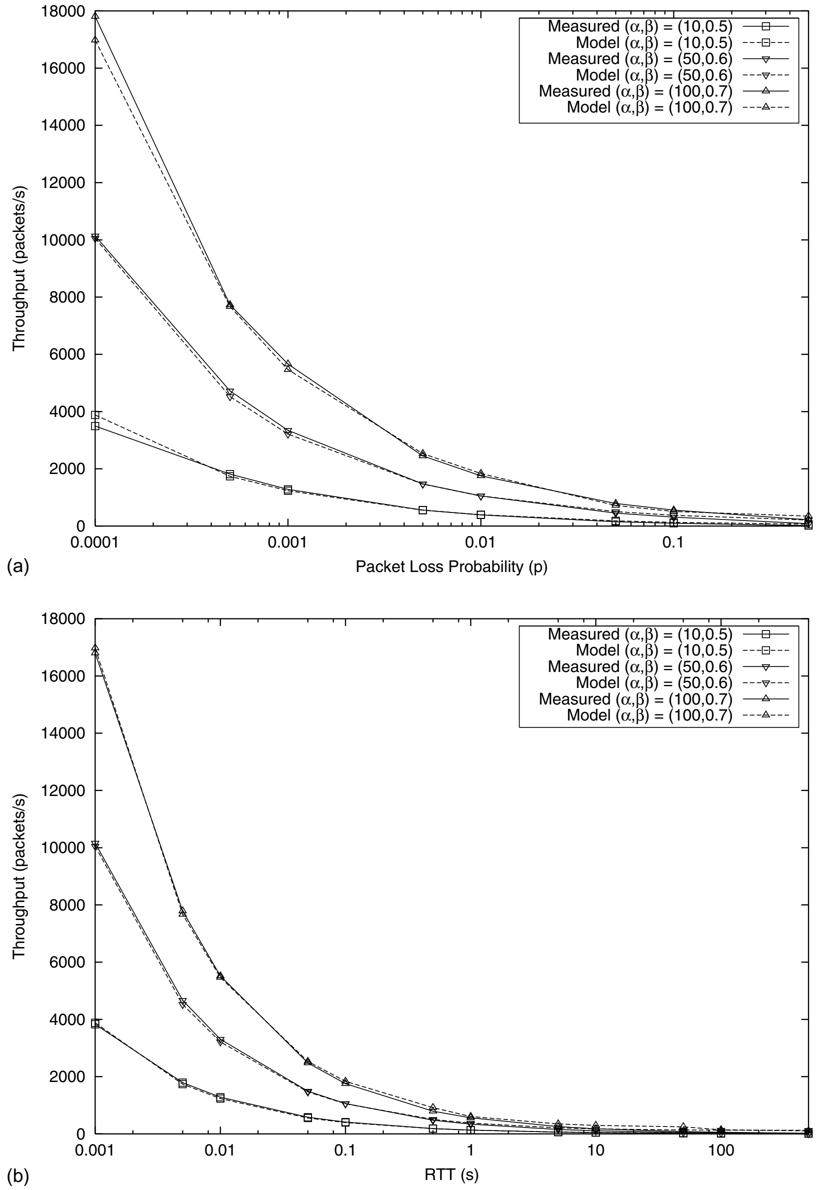

Fig. 2. The comparison of the model and the measured throughput for rate-based generic AIMD congestion scheme for (a) $R = 10$ ms and varying p and (b) $p = 10^{-3}$ and varying R.

requirements under the certain RTT of R seconds and packet loss probability of p. The target throughput can be either required data rate to achieve the highest quality of encoded multimedia streaming, the average data rate required to transmit certain amount of information within the certain delay bound, or a certain throughput upper bound to assure fairness to the TCP sources sharing the same bottleneck. Hence, the $\mathscr{T}^{r}_{\alpha,\beta}(p,R) = B$ represents a set of AIMD parameters (α,β), which achieves B packets/s for R and p. Thus, by solving (12) for the additive-increase factor α, we can obtain α as a function of target throughput, RTT and packet loss probability as follows:

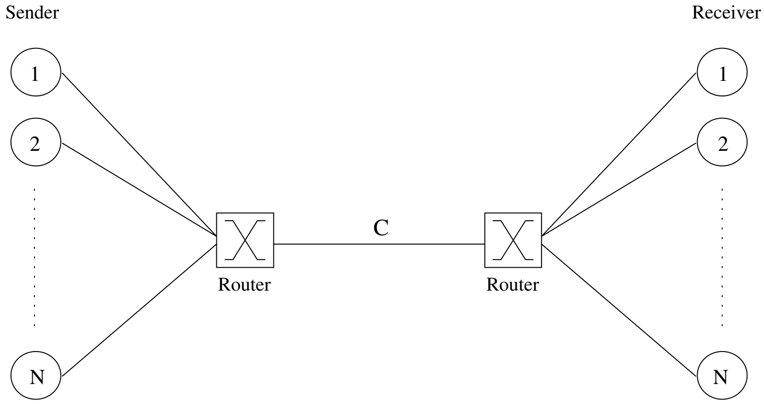

Fig. 3. Simulation scenario for rate-based throughput analysis validation.

$$\alpha = \frac{(1+\beta)}{2}\left(B+\frac{1}{Rp}\right) \times \left[\sqrt{1+\frac{8B^2(1-\beta)}{\left(B+\frac{1}{Rp}\right)^2(1+\beta)^2}}-1\right]. \quad (13)$$

Hence, by using (13), α can be chosen such that B packets/s is achieved on the link with R seconds RTT and the packet loss probability of p and $\beta \in [0.5, 1)$. One particular application could be the AIMD parameter selection for the rate-based AIMD congestion control scheme to achieve adaptation to the varying link conditions. For example, the throughput degradation due to excessive propagation delay in deep space networks can be compensated by the proper adjustment of the AIMD parameters.

3.3. Simulation experiments

To further illustrate this possible application, we run simulation experiments using the same rate-based AIMD congestion control scheme and the simulation topology used in Section 3.1. Here, we perform the simulations for varying packet loss probability p and round-trip time R. At each experiment, $N = 10$ sources are connected to the shared link with capacity of $C = 20$ Mb/s as shown in Fig. 3. The capacity of each access link connecting sources to the bottleneck router and the target throughput B are 2 Mb/s. The rate-based congestion control scheme used in the simulations has the default AIMD parameters of $(\alpha, \beta) = (2, 0.5)$. The simulations are performed with varying multiplicative rate-decrease factor, i.e, $\beta = 0.5, 0.6, 0.7, 0.8$. The additive-increase parameter α is then adjusted using (13) for each of the experiment based on R, p, B, and β.

The average throughput achieved by the rate-based congestion control sources and the corresponding AIMD parameters for varying p and R are shown in Figs. 4 and 5, respectively. As observed from Fig. 4(a), for $R = 100$ ms and wide range of p, the sources with adjusted AIMD parameters achieve significantly improved throughput performance compared to the source with default AIMD parameters. The throughput gain becomes more significant with increasing packet loss probability p. For $p = 10^{-2}$, the source with $\beta = 0.5$ improves throughput by 284% over the one with default AIMD parameters. As shown in Fig. 4(b), the corresponding additive increase parameter α is 18 packets/s. Furthermore, the performance improvement achieved by adjusting the AIMD parameters according to the link conditions decreases with increasing β. This is mainly because the additive-increase parameter α calculated using (13) also decreases with increasing β as observed from the variation of α with varying p shown in Fig. 4(b).

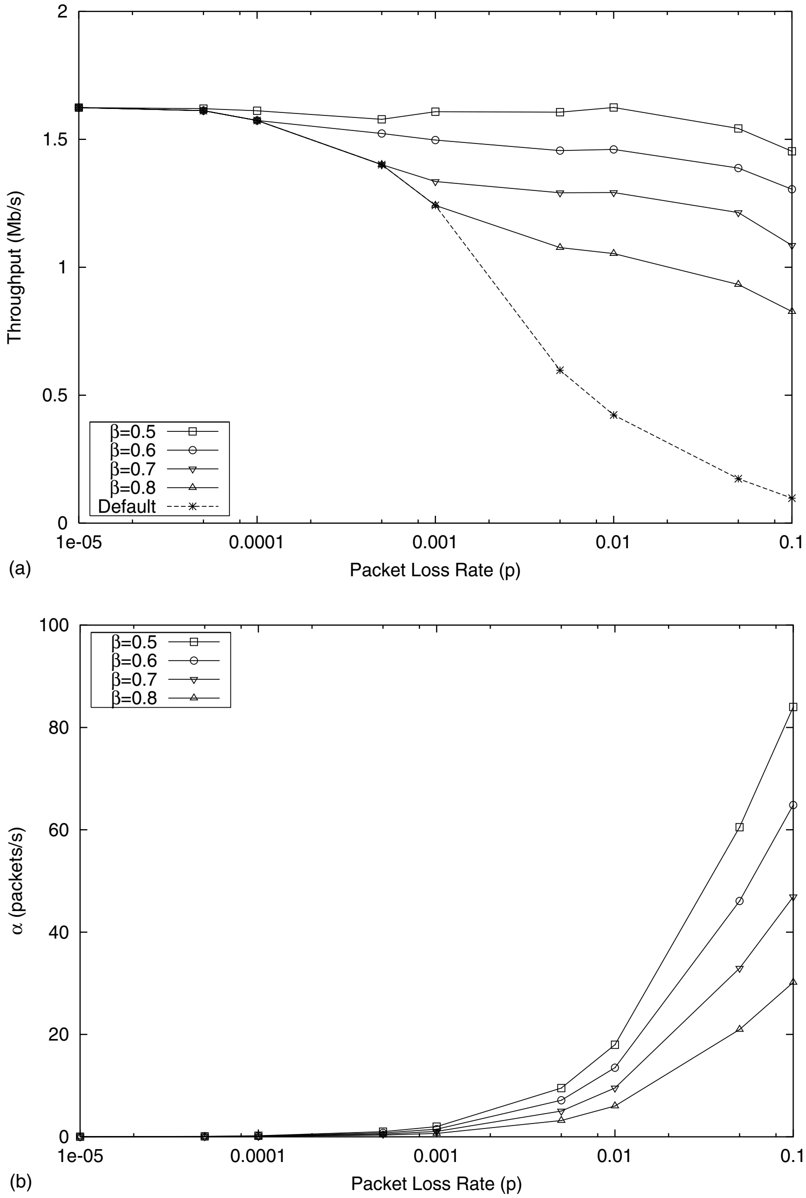

Fig. 4. (a) Throughput achieved by the rate-based congestion control sources using default and adapted AIMD parameters and (b) corresponding additive rate increase parameter α for $R = 50$ ms and varying p.

The similar patterns in the achieved throughput and the variation of α are observed for experiments with varying R shown in Fig. 5(a) and (b), respectively. For $R = 1$ second and $p = 10^{-4}$, the source with $\beta = 0.5$ uses additive-increase parameter $\alpha = 4.4803$ packets/s and achieves 78% throughput improvement over the source with default AIMD parameters. Hence, the method of AIMD parameter adaptation according to the varying link conditions can significantly improve the throughput performance.

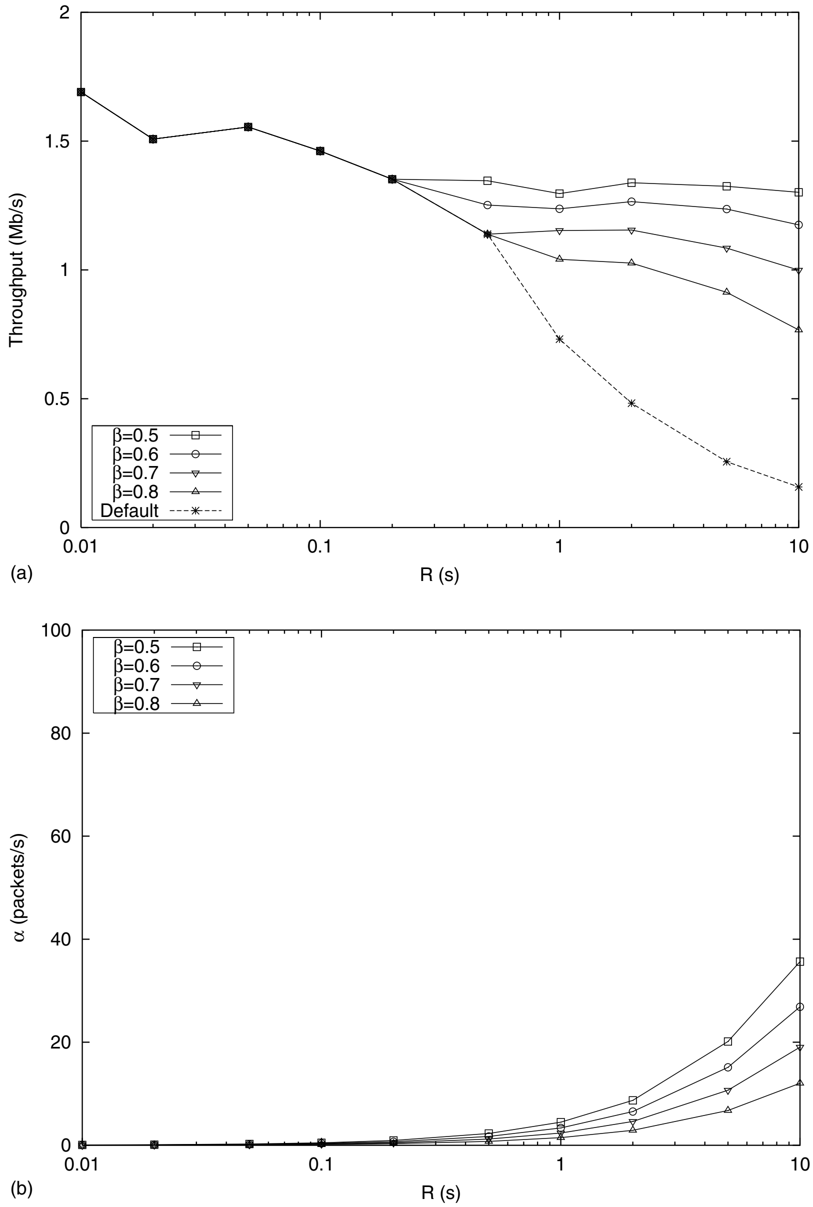

Fig. 5. (a) Throughput achieved by the rate-based congestion control sources using default and adapted AIMD parameters and (b) corresponding additive rate increase parameter α for $p = 10^{-4}$ and varying R.

On the other hand, there is a possibility that using very high α may lead to overshoot the available bandwidth during additive rate increase and hence may cause frequent rate oscillations and network instability. To study the possible effects of varying AIMD parameters on the network stability, we run simulation experiments with varying AIMD parameters and observe the variation of data transmission rate over time. The simulation parameters for each experiment scenario are summarized in Table 1.

The plots showing the data rate variation over time are given in Fig. 6(a), (b), (c), and (d) for $\beta = 0.5$, $\beta = 0.6$, $\beta = 0.7$, and $\beta = 0.8$, respectively. As shown in Table 1, the corresponding additive rate increase parameters calculated using (13) are $\alpha = 35.7360$, $\alpha = 26.9487$, $\alpha = 19.1066$, and $\alpha = 12.0731$ packets/s, respectively. As shown in Fig. 6, all of the source with the adjusted AIMD parameters respond to the congestion in the network and does not lead to significant network instability. However, it is observed from Fig. 6(a) that the source with the highest α, i.e., $(\alpha, \beta) = (35.7360, 0.5)$, frequently overshoots the available bandwidth and hence experience higher packet loss rate compared to the other scenarios. Note that for a link with high loss rate and round-trip time such as $p = 10^{-3}$ and $R = 1$ second, the required additive rate-increase parameter α is only 35.7360 packets/s. As the multiplicative-decrease factor β increases, the additive-increase factor α

Table 1
Simulation parameters

Exp. #	R (ms)	p	β	B (Mb/s)	α	T (Mb/s)
(a)	1000	10^{-3}	0.5	2	35.7360	1.5504
(b)	1000	10^{-3}	0.6	2	26.9487	1.4862
(c)	1000	10^{-3}	0.7	2	19.1066	1.3030
(d)	1000	10^{-3}	0.8	2	12.0731	1.0846

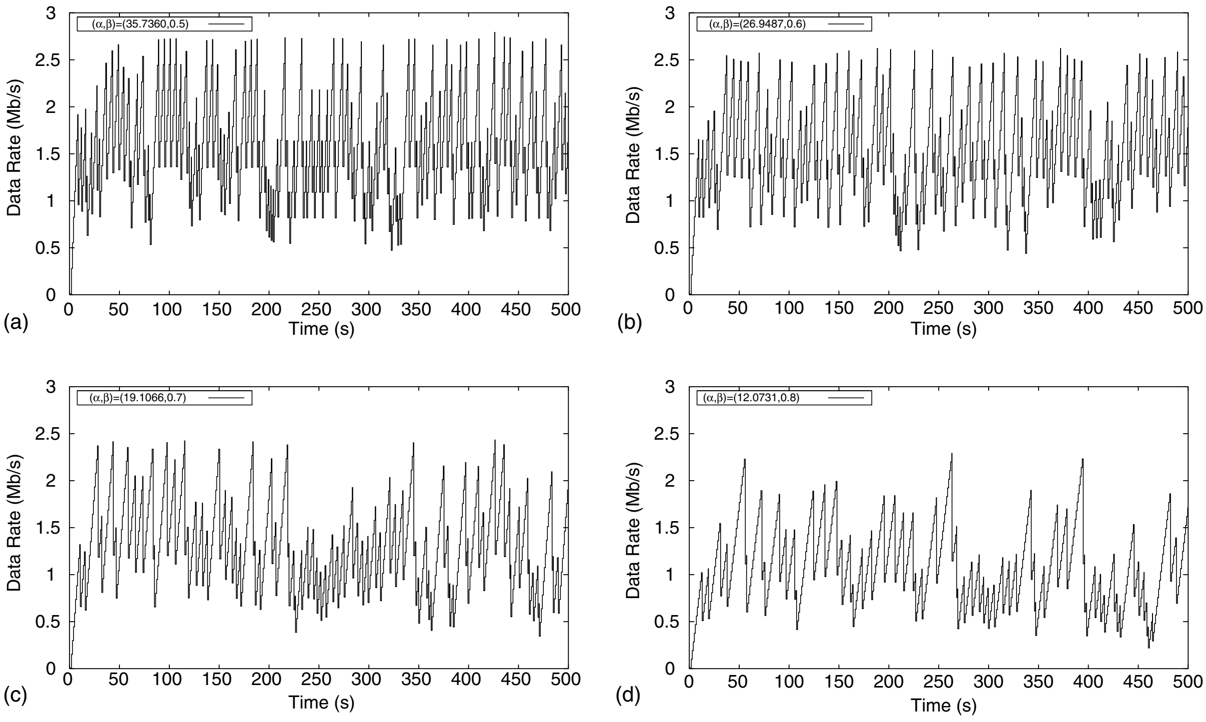

Fig. 6. Data rate change with time for the rate-based congestion control sources using different AIMD parameters over the link with $p = 10^{-3}$ and $R = 1000$ ms.

decreases and hence the less oscillatory behavior is observed in Fig. 6(b)–(d). Hence, the possible adverse effects of high α can be avoided by careful selection of both α and β parameters although the achieved throughput also decreases with decreasing α as shown in Table 1. Therefore, the selection of α can be performed such that the target throughput is achieved while minimum instability and oscillatory behavior are incurred. One conceivable approach could be to choose the target data rate B equal to the estimated TCP throughput over the same link conditions. By this way, the fairness could be also taken into account and the stability of the network would be strengthened.

4. Conclusions

In this paper, we have analyzed the throughput performance of the rate-based generic AIMD congestion control scheme and developed the steady-state throughput model. The simulation experiments show that the developed model captures the rate-based AIMD congestion control behavior for wide ranges of propagation delay and packet loss rate. One important result of this analysis is that the throughput performance of the rate-based congestion control schemes is inversely proportional to the square-root of the propagation delay whereas that of the window-based schemes is known to be inversely proportional to the propagation delay itself. This result also justifies that the rate-based congestion control schemes have higher resilience to high link delay than the window-based schemes. Therefore, the rate-based congestion control is an important alternative to the window-based schemes especially for high propagation delay environments.

Furthermore, we have explored possible applications of the analysis presented in this paper. One particular application could be the AIMD parameter selection for the rate-based AIMD congestion control scheme to achieve adaptation to the varying link conditions. The simulation experiments show that AIMD parameter adaptation according to the link conditions can significantly improve the throughput performance. However, the stability study show that the selection of the AIMD parameters should be done carefully to minimize the frequently rate oscillations and avoid possible network instability while achieving the target data rate.

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