Journal of Zhejiang University SCIENCE A
ISSN 1009-3095 (Print); ISSN 1862-1775 (Online)
www.zju.edu.cn/jzus; www.springerlink.com
E-mail: jzus@zju.edu.cn

# TCP-friendly source adaptation for multimedia applications over the Internet

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Received Dec. 12, 2005; revision accepted Feb. 20, 2006

**Abstract:** In this paper, we propose a simply TCP-friendly source adaptation framework to provide a proportional bandwidth sharing service. Our scheme is based on the framework of Monotonic Response Function (MRF), can be used to bound the sending rate of a source within a predefined interval and provide very smooth traffic, and is suitable for multimedia applications over the Internet. As our scheme is also very simple and TCP-friendly, it is easy to be deployed over the current Internet. We verify our scheme with experimental studies.

**Key words:** Source adaptation, TCP-friendly, TCP-compatible, Multimedia-friendly

**doi:**10.1631/jzus.2006.AS0001 **Document code:** A **CLC number:** TN919.8

## INTRODUCTION

Many new applications being widely deployed in the Internet include voice over IP, video conference, video streaming, audio streaming, mission-critical financial data, and so on. Essentially, these applications have different requirements on the bandwidth from those applications using transmission control protocol (TCP) which dominates the current Internet. For example, multimedia applications have the following two distinguished requirements: (1) The sending rate should be adjusted smoothly; (2) The sending rate should be in a predefined interval, i.e., there usually exists an upper bound and a lower bound for the sending rate.

There are two possible approaches to address these problems. One is to enhance the Internet with resource reservation, admission control and so on to provide QoS over Internet. The other is to adopt suitable source adaptation schemes such that the sending rate of an application is adjusted according to the current network condition (Crowcroft and Oechslin, 1998). Compared to the first approach, the second approach cannot provide guaranteed services but is still an attractive approach due to its simple structure and better utilization of the available time varying network resources.

The current Internet is dominated by TCP traffic which employs the Additive-Increase Multiplicative-Decrease (AIMD) scheme for congestion control. To reduce the impact on the existing services, the new rate control protocol is recommended to be TCP-friendly, i.e. with the same Round Trip Time (RTT), the bandwidth share should not be greater than what a typical TCP will get (Handley *et al*., 2003) (This is also called TCP-compatible). Due to the two requirements, especially the second one of multimedia applications, TCP-friendliness and multimedia-friendliness "contradict" with each other when the load of the Internet is heavy (Wang and Schulxrinne, 2005). Therefore, these new applications present new problems for the Internet.

Recently, an interesting framework named Monotonic Response Function (MRF) was proposed

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in (Chen *et al*., 2005) and sufficient conditions are derived for the MRF to be TCP-friendly and TCP-compatible with respect to a weighting factor. The framework can be used to design both TCP-friendly source adaptation schemes and multimedia friendly source adaptation schemes. It was shown in (Chen *et al*., 2005) that TCP-friendliness and multimedia-friendliness are consistent in the sense that multimedia-friendliness is actually TCP-friendliness with respect to a weighting factor, which controls the bandwidth share that a source will be received in steady state. Furthermore, a simple way is provided by our MRF to extend the existing congestion control algorithms, such as AIMD, general AIMD (GAIMD) and Binomial Congestion Control (BCC), to provide proportional bandwidth sharing service. This is very helpful for different applications over the Internet.

Let $c$ be the link capacity of the bottleneck link. It was shown in (Chen *et al*., 2005) that the sending rate of MRF source $s$ with weighting factor $w_s$ converges to the throughput which is given as

$$r_s = \frac{cw_s}{\sum_{\text{all sources } i} w_i}. \tag{1}$$

Clearly, the actual throughput a source received depends on the weighting factors of other sources. Since it is impossible for a source to know in advance the weighting factors of other sources and the Internet traffic is time varying, it is very difficult for a multimedia application to properly choose its weighting factor such that the actual throughput is within a predefined interval. It is thus necessary to adjust the weighting factor according to the bounds and the network status.

To address this issue, we propose a simple source adaptation (SA) scheme in this paper. Our SA model keeps on monitoring the throughput and adjusts the weighting factor when the (average) sending rate is not in the predefined interval. Our scheme can be used to provide a smooth traffic with sending rate in the predefined interval. This is very attractive for multimedia applications over the Internet.

The rest of this paper is organized as follows. Section 2 includes some preliminary knowledge. Our SA scheme is presented in Section 3. Extensive experimental results are provided in Section 4 to verify the efficiency of our scheme. Finally, several concluding remarks are listed in Section 5.

## PRELIMINARY KNOWLEDGE

### Existing source adaptation schemes

AIMD is the most popular source adaptation scheme (Jacobson and Karels, 1990). In this scheme, the window size is increased by 1 when a window size of packets is acknowledged, and decreased by half when a loss is detected, i.e.,

$$\begin{aligned} &I:\ x_s(k+1) \leftarrow x_s(k)+1, \\ &D:\ x_s(k+1) \leftarrow x_s(k) - x_s(k)/2, \end{aligned} \tag{2}$$

where $I$ and $D$ stand for the corresponding increase policy and decrease policies, respectively. It was shown in (Chiu and Jain, 1989) that the AIMD scheme converges to fairness and efficiency when all flows in the network get the same feedback simultaneously.

The GAIMD scheme extends AIMD with the following increase and decrease policies (Yang and Lam, 2000):

$$\begin{aligned} &I:\ x_s(k+1) \leftarrow x_s(k)+\alpha,\ 0<\alpha, \\ &D:\ x_s(k+1) \leftarrow x_s(k) - \beta x_s(k),\ 0<\beta<1. \end{aligned} \tag{3}$$

To be TCP-compatible, $\alpha$ and $\beta$ must satisfy (Floyd *et al*., 2000)

$$\alpha=3\beta/(2-\beta). \tag{4}$$

The BCC (Bansal and Balakrishnan, 2001) scheme further extends the GAIMD scheme with the following policies of $I$ and $D$:

$$\begin{aligned} &I:\ x_s(k+1) \leftarrow x_s(k)+\alpha / x_x^p(k),\ 0<\alpha, \\ &D:\ x_s(k+1) \leftarrow x_s(k) - \beta x_s^l(k),\ 0<\beta<1, \end{aligned} \tag{5}$$

with a further condition of $p+l=1$ to ensure TCP-friendliness, where $x^p$ denotes the $p$th-power of $x$. The SQRT ($p=l=0.5$) and IIAD ($p=1$, $l=0$) schemes are two special cases of BCC.

**1-responsiveness**

Let $\Delta_I$ and $\Delta_D$ be the increase and the decrease in window size resulting from a single application of $I$ and $D$, respectively. A flow is said to be 1-responsive if $\Delta_I \le \Delta_D$.

In other words, the decrease in window size from a single application of $D$ must at least wipe out the previous increase resulting from the last application of $I$.

It can be easily shown that AIMD, GAIMD, BCC are 1-responsive. Thus, in the rest of this paper, unless otherwise stated, the protocols are supposed to be 1-responsive.

**Weighted fairness**

A weighted fairness index is given by (Chen *et al.*, 2005)

$$F=\sum_{s=1}^{n}\frac{x_s^2}{w_s^2}-\frac{1}{n}\left(\sum_{s=1}^{n}\frac{x_s}{w_s}\right)^2. \tag{6}$$

Fig.1 illustrates the phrase plot of an example of weighted fairness in the case of 2 sources. When $w_1=w_2$, it yields the normal fairness. Otherwise, it is a line with slope equal to $w_2/w_1$. It is easy to derive the following proposition:

**Proposition 1** Let $r_s$ be the average sending rate of source $s$. If two sources are fair with respect to weighting factors $w_1$ and $w_2$, the average sending rates then satisfy $r_1/r_2 \approx w_1/w_2$.

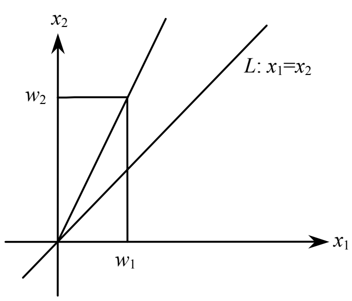



**Fig.1 The phase plot of weighted fairness index**

**MRF**

The same as the existing schemes, the MRF is also composed of $I$ and $D$ policies that are defined as

$$\begin{aligned}&I:\ x_s(k+1)\leftarrow x_s(k)+w_s f[x_s(k)/w_s],\\&D:\ x_s(k+1)\leftarrow x_s(k)-w_s g[x_s(k)/w_s],\end{aligned} \tag{7}$$

where $f(x)$ is a monotonically non-increasing function and $g(x)$ is a monotonically nondecreasing function (at least one of them is strictly monotonic), with $f(x)>0$, $x \ge g(x)>0$.

In (Chen *et al.*, 2005), the following theorems are derived:

**Theorem 1** MRF converges to the weighted fairness for $n \ge 2$ flows.

**Theorem 2** A smooth MRF flow is TCP-compatible w.r.t. $w_s$ in the steady state if and only if

$$f(x/w)=\frac{3(1+f'(x/w))g(x/w)}{2x/w-(1+f'(x/w))^m g(x/w)},$$

where $m$ is given as

$$m=\frac{g(x/w)(1+f'(x/w))}{f(x/w)}.$$

Using the above result, we can easily derive Eq.(4) for GAIMD.

It is easy to show that the existing algorithms such as AIMD, GAIMD and BCC are special cases of the MRF.

We shall now illustrate the features of our MRF through an example. Consider the corresponding weighted algorithms of AIMD and GAIMD as follows:

$$\begin{aligned}&I:\ x_s(k+1)\leftarrow x_s(k)+w_s,\\&D:\ x_s(k+1)\leftarrow x_s(k)-x_s(k)/2,\end{aligned} \tag{8}$$

and

$$\begin{aligned}&I:\ x_s(k+1)\leftarrow x_s(k)+w_s/5,\\&D:\ x_s(k+1)\leftarrow x_s(k)-x_s(k)/8.\end{aligned} \tag{9}$$

By letting $w_s$=5, we get the following DECbit (Jain *et al.*, 1987) scheme from Eq.(9):

$$\begin{aligned}&I:\ x_s(k+1)\leftarrow x_s(k)+w_s,\\&D:\ x_s(k+1)\leftarrow x_s(k)-x_s(k)/8.\end{aligned} \tag{10}$$

Obviously, the DECbit is not TCP-compatible but is TCP-compatible with respect to 5. It is shown in Fig.2 the dynamics of the windows size for three different MRFs, the normal AIMD, a WAIMD with weighting factor 5 and the DECbit. Obviously, both the WAIMD and DECbit get the same throughput that is 5 times that of the normal AIMD while the sending rate of DECbit is much smoother than that of the WAIMD.

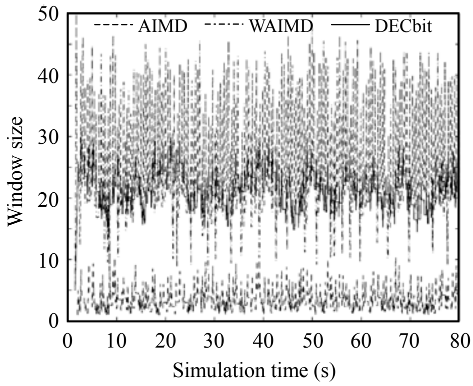



**Fig.2 An example of AIMD, weighted AIMD and weighted GAIMD**

## SOURCE ADAPTATION FOR MULTIMEDIA APPLICATIONS

In this section, we introduce our source adaptation scheme for multimedia applications over the Internet. There are an upper bound $R_s^+$ and a lower bound $R_s^-$ for the sending rate of multimedia application (also source) $s$. Let $c$ be the link capacity of the bottleneck link. The same as in (Low and Lapsley, 1999), we require that

$$\sum_{\substack{\text{all SA sources}\\ \text{at bottle-neck link}}} R_s^- < c.$$

As shown in Eq.(1), for MRF source $s$ with a fixed weighting factor $w_s$, the throughput is related with all the weights on the link. Since it is impossible for a source to know in advance the weighting factors of other sources and the network is time varying, it is very difficult for a source to properly choose its weighting factor such that the actual throughput is within a predefined interval. We require a simple source adaptation (SA) scheme to address this issue.

Fig.3 shows that the overall structure of our scheme requires an SA model and an MRF at the source side. The MRF works as follows: it sends out packets, receives feedback information from the corresponding destination and adjusts its window size accordingly.

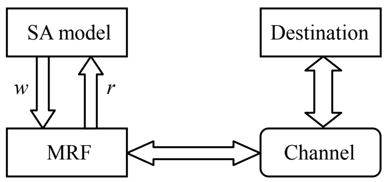



**Fig.3 Structure of the source adaptation scheme**

The SA model keeps on monitoring the sending rate of the MRF algorithm during each time period $T_s$ and compares it with the lower bound $R_s^-$ and the upper bound $R_s^+$. If the sending rate is within the interval, the weighting factor is not adjusted. Otherwise, the SA model adjusts the weighting factor of the MRF accordingly.

Specifically, SA model records the bytes of data sent out during the period and then computes an average value by the following sliding window method:

$$\overline{r}_s(k+1) = r_s(k)a_s + \overline{r}_s(k)(1-a_s), \tag{11}$$

where $a_s \le 1$ is a positive factor.

$\overline{r}_s$ is then compared with the bounds. If $\overline{r}_s > R_s^+$ or $\overline{r}_s < R_s^-$, then $w_s$ is adjusted by:

$$w_s(k+1) = w_s(k) + \gamma_s\left(1 - \frac{2\overline{r}_s(k+1)}{R_s^+ + R_s^-}\right). \tag{12}$$

Otherwise $w_s$ does not change. In the next section, we will verify our scheme through experimental results.

It should be mentioned here that the service we provide is not a strict one. Although the user defines an upper bound and a lower bound, there is still a chance that the service can go beyond the requirement. However, the possibility is very small and there is usually a buffer associated with a multimedia application to smoother the effect of this event with small possibility. Moreover, compared to other structures providing different bandwidth sharing services, our scheme is much simpler and does not require any change in the routers. Also, our scheme can work harmoniously with networks dominated by TCP traffic, such as the current Internet. Thus our scheme is very helpful for the improvement of the current Internet.

## EXPERIMENTAL RESULTS

In this section, we will verify our scheme by extensive experimental results. We have implemented our scheme in ns2. The topology used in our simulation is a typical dumbbell as illustrated in Fig.4. The certain bottleneck link has a capacity of 1 Mbps and employs a RED scheme to provide random dropping. There are 10 applications sending data from $s_i$ to $d_i$ where 9 of them are normal FTP which generate TCP-Reno traffic and the other is our source adaptation scheme. The Round Trip Time (RTT) is 0.01 s. Our source adaptation scheme employed the following weighted GAIMD (WGAIMD) scheme:

$$\begin{aligned} I&: x_s(k+1) \leftarrow x_s(k) + 0.007519 w_s(k), \\ D&: x_s(k+1) \leftarrow x_s(k) - 0.005 x_s(k). \end{aligned} \quad (13)$$

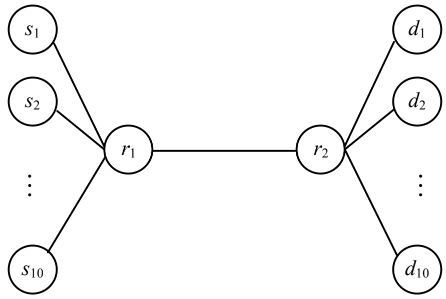


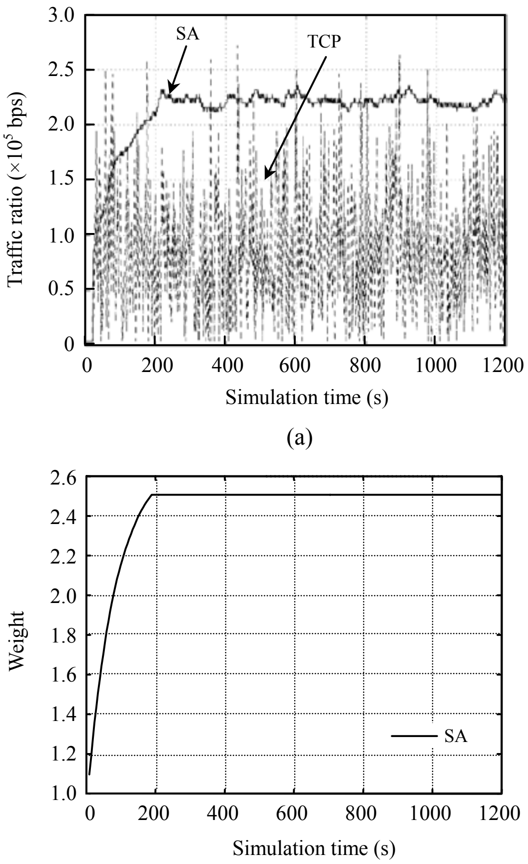

**Fig.4 Topology of the simulation**

The adjust time period, i.e. $T_s$, is set at 5 s. The upper bound, i.e. $R_s^+$, is 250 kbps and the lower bound, $R_s^-$ is 200 kbps. The adjusting step, $\gamma_s$, is selected as 0.1 and the value of $a_s$ is set at 0.2. The simulation is run for 1200 s.

**Service**

We first test the scenario that all the sources start at random time in the first 20 s. Fig.5 shows the traffic rate and the weighting factor of the source adaptation scheme under the typical setting. It can be shown that our scheme can be used to adjust the sending rate to the desired interval. Furthermore, compared to TCP traffic, our scheme provides much smoother traffic.

We then test the scenario that there is an abrupt change to the network load: only 5 FTP applications are opened in the beginning and the other 4 FTP applications are started at around 400 second. It can be seen from Fig.6a that the sending rate of our source adaptation application decreases at about 400 second. The weighting factor is thus increased by our scheme to get the desired throughput (Fig.6b).




(a)

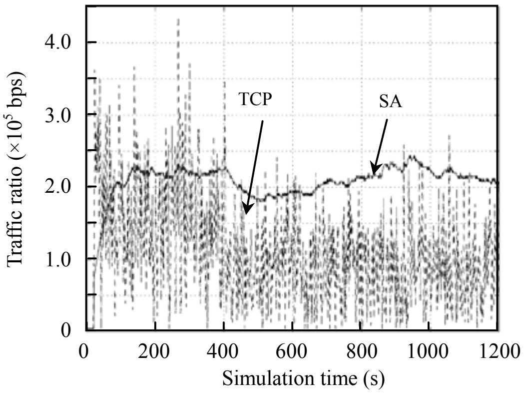
(b)

**Fig.5 The sending rate (a) and the weighting factor (b) of source with WGAIMD under the typical setting**

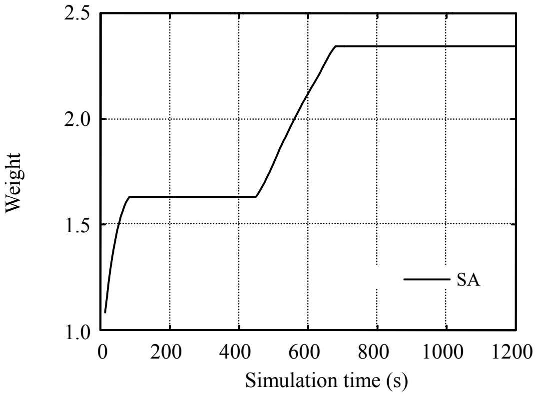
We also test the scenario of two SA sources. The settings are the same except that one of them now has the upper and lower bounds as 350 kbps and 300 kbps, respectively. Fig.7 shows the traffic rates and the weighting factors. Clearly, our scheme can adjust the traffic rates to the desired intervals.

## CONCLUSION

In this paper, we have proposed a simple source adaptation framework to provide a relative bandwidth sharing service. Our scheme is based on the MRF. We verify our scheme with experimental results showing that our scheme can bound the traffic rate to within the predefined interval and provide a very smooth traffic.

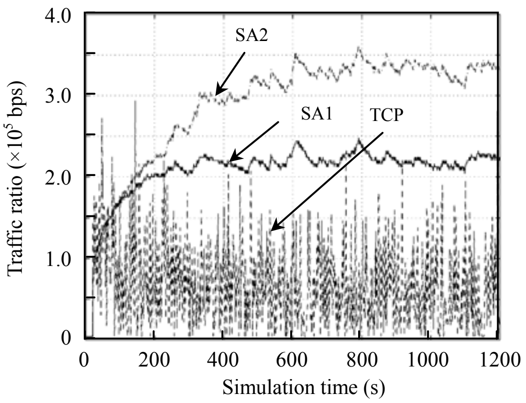



(a)

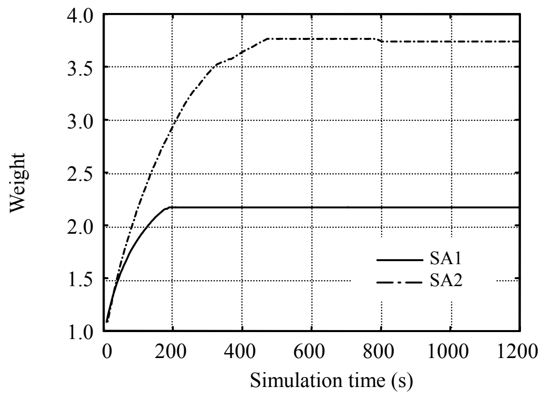



(b)

**Fig.6 The sending rate (a) and the weighting factor (b) of source with WGAIMD in case of abrupt change**




(a)




(b)

**Fig.7 The sending rates (a) and the weighting factors (b) of 2 SA sources with the same WGAIMD**

## References

Bansal, D., Balakrishnan, H., 2001. Binomial Congestion Control Algorithms. IEEE INFOCOMM, p.631-640.

Chen, C., Li, Z.G., Soh, Y.C., 2005. MRF: A framework for source and destination based bandwidth differentiation service. *IEEE/ACM Trans. on Networking*, in Press.

Chiu, D.M., Jain, R., 1989. Analysis of the increase and decrease algorithms for congestion avoidance in computer networks. *Computer Networks and ISDN Systems*, **17**:1-14. [doi:10.1016/0169-7552(89)90019-6]

Crowcroft, J., Oechslin, P., 1998. Differentiated end-to-end Internet services using a weighted proportional fair sharing TCP. *ACM SIGCOMM Computer Communication Review*, **28**(3):53-67. [doi:10.1145/293927.293930]

Floyd, S., Handley, M., Padhye, J., 2000. A Comparison of Equation-Based and Aimd Congestion Control. http://www.aciri.org/trfc/.

Handley, M., Floyd, S., Padhye, J., Widmer, J., 2003. TCP Friendly Rate Control (TFRC): Protocol Specification. RFC 3448.

Jacobson, V., Karels, M., 1990. Congestion avoidance and control. *ACM Comput. Commun. Rev.*, **18**(4):314-329. [doi:10.1145/52325.52356]

Jain, R., Ramakrishnan, K., Chiu, D., 1987. Congestion Avoidance in Computer Networks with a Connectionless Netowrk Layer. Tech. Rep. DEC-TR-506~34, Digital Equipment Corporation.

Low, S.H., Lapsley, D.E., 1999. Optimization flow control—I: Basic algorithm and convergence. *IEEE/ACM Trans. on Networking*, **7**(6):861-874. [doi:10.1109/90.811451]

Wang, X., Schulxrinne, H., 2005. Incentive-compatible adaptation of Internet real-time multimedia. *IEEE Journal on Selected Areas in Communications*, **23**(2):417-436. [doi:10.1109/JSAC.2004.839399]

Yang, Y.R., Lam, S.S., 2000. General AIMD Congestion Control. IEEE Int. Conf. Network Protocols. Osaka, Japan, p.187-198.