An Efficient Flow Control Algorithm for Multi-rate Multicast Networks

Naixue Xiong, Yanxiang He
The State Key Lab of Software Engineering,
Computer School of Wuhan University
Wuhan, PR. China
N.Xiong@mail.ccnu.edu.cn, yxhe@whu.edu.cn

Yan Yang
Department of Computer Science,
Central China Normal University,
Wuhan, PR. China
Y.Yang@mail.ccnu.edu.cn

Jiannong Cao
Department of Computing,
Hong Kong Polytechnic University,
Hung Hom, KLW, Hong Kong, PR. China
csjcao@comp.polyu.edu.hk

Chuan Lin
The State Key Lab of Software Engineering,
Computer School of Wuhan University
Wuhan, PR. China
chlin@wuhee.edu.cn

***Abstract*—This paper describes a novel multi-rate multicast congestion control scheme based on the well-known proportional plus integrative control technique, where the control parameters can be designed to ensure the stability of the control loop in terms of source rate. The congestion controller is located at the next upstream nodes of multicast receivers and has explicit rate (ER) algorithm to regulate the rate of the receivers. We further analyze the theoretical aspects of the proposed algorithm, show how the control mechanism can be used to design a controller to support many-to-many multi-rate multicast transmission based on ER feedback, and verify its agreement with simulations in the case of bottleneck link appearing in a multicast tree. Simulation results show the efficiency of our scheme in terms of the system stability, high link utilizations, fast response, scalability, high throughput and fairness.**

Keywords- explicit rate, multicast congestion control, multi-rate multicast, QoS (quality of service), rate-based congestion control

I. INTRODUCTION

With the ever-increasing multicast data applications, considerable research efforts have been focused on the design of congestion control scheme for multicast service. Multicast improves the efficiency of multipoint data distribution by building a distribution tree from a sender, or multiple senders, to a set of receivers [1]. However, the widely used multicast transport protocols, which are layered on top of the IP multicast layer, could cause congestion or even congestion collapse if they do not provide adequate congestion control. Congestion control thus plays an important role in traffic management of multicast communications, such as teleconference and information dissemination services.

There are two approaches of multicast congestion control schemes proposed so far, namely, Single-rate Multicast Congestion Control (SR-MCC) and Multi-rate Multicast Congestion Control (MR-MCC). One obvious limitation of the SR-MCC is that all the receivers must receive data at a single rate, which is the slowest path rate of all relevant paths. This presents a serious practical limitation if we consider, for example, users who are willing to pay more to access at a higher speed. Furthermore, due to the diverse characteristics and requirements of the different receivers within a multicast group, and for greater flexibility in resource allocation, it is desirable to have multicast sessions in which different receivers receive data at different rates. This inflexibility is overcome by MR-MCC that can allocate different rates to different users.

Several multicast congestion control approaches [1, 2-19, 22-24] have been proposed recently. One class of approaches [7-9] adopts a simple hop-by-hop feedback mechanism. Although the simple hop-by-hop feature seems to be an advantage, these approaches often lead to the so-called consolidation noise problem [10, 11] due to incomplete feedback information. To overcome this drawback, [11] and [12] proposed the concept of feedback synchronization, at each branch point, by accumulating feedback from all downstream branches. These schemes of [11] and [12] then introduce another problem of slow transient response since the feedback from the congested branch may have to needlessly wait for the feedback from "longer" paths. Such delayed congestion feedback can cause excessive queue build-up and packet loss at the bottleneck link. The authors of [13] and [14] suggested that only a carefully chosen set of receivers, instead of all receivers, send their feedbacks to the sender. Zhang et al. [15] proposed an optimal second-order rate control algorithm to deal with control packet round-trip time (RTT) variation in multicast communications, which defined that the data transfer rate is adjusted at the source depending on the available bandwidth at the bottleneck.

More recently, several studies (such as [16-19]) have focused on the design of MR-MCC protocols. However, all of them have drawbacks. Some designs cause over-subscription

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and high packet losses. Some are slow to converge and unresponsive. Some designs are too complex and infeasible [24].

To address the above problems, this paper describes a novel MR-MCC congestion control scheme based on the proportional plus integrative (PI) controller. The incoming flow rate of a session, at every branching point in its tree, is enforced to be the maximum of the rates that can be accommodated by its participating branches. By doing so, the sending rate at the source will eventually be the maximum of the rates that can be accommodated by the entire paths to individual receivers. Since the source sends data at the maximum path rate, it is necessary to reduce the rate of an incoming flow at every branching point to the value that can be accommodated by its participating branches [24]. The PI controllers are located at the next upstream branch node of the receivers.

The relevant gain parameters of the PI controller are determined by the system stability. Each branch point in our scheme only receives feedbacks from the direct downstream nodes instead of all downstream nodes, thus it greatly decreases the number of feedbacks to be aggregated at one node. As a result, our scheme can avoid the so-called feedback explosion problem [24] to a great extent. Simulation results show the efficiency of the proposed scheme in terms of system stability, high link utilizations, quickly response, scalability, high system transport rate, intra-session fairness and inter-session fairness. Simulation results verify the efficiency of the proposed MR-MCC scheme.

Our scheme is very versatile. It can support sessions where receivers are added and depart. It can manage the traffic to guarantee stability, in real time, even if considerable changes occur in the source-receivers tree.

The rest of this paper is organized as follows: In section 2, we present the congestion control model. In section 3, we propose a novel congestion control algorithm, and give the specific pseudocode description of the proposed algorithm. In section 4, we present the optimal choice of the PI control gains and describe the implementations of the PI controller in detail and in section 5, we use simulations to validate and evaluate the performance of our scheme. Finally, in section 6, we end the paper in the conclusions and the future works.

II. THE CONGESTION CONTROL MODEL

To analyze the performance and characteristic of the multicast, we focus on the following system model as shown in figure 1, where we have two classes of sources, i.e., one multicast source and one end-to-end CBR source. The PI controllers are located at the next upstream nodes of the receivers, i.e., the routers from RT 1 to RT m, and compute the expected rates used to adjust the multicast receiving rates of the downstream receivers. The receiver ji represents the i^{th} receiver corresponding to the j^{th} router (RT j). We provide rate adaptation functionality at every branch point of each session. This rate adaptation scheme is determined on the basis of the fact that the multicast tree will eventually receive data at an independently trimmed rate allowed by its entire path. So we acquire the above computed maximum value as the effective sending rate of the multicast source. The sending rate is necessary to convert down the rate of an incoming flow at every branching point to the values that can be accommodated by its participating branches to individual receiver. The considered multicast elastic service in the network-assisted environments and the relevant parameters are described as follows:

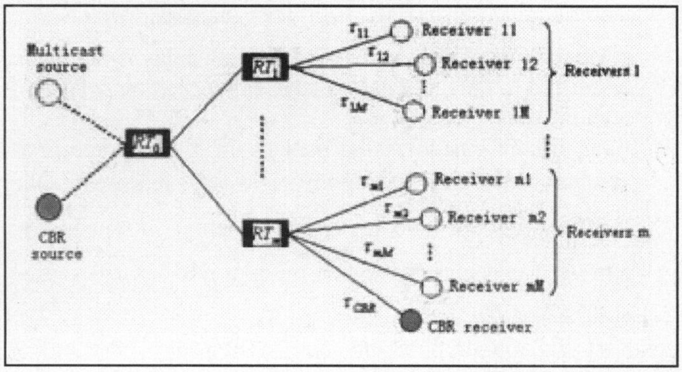

Figure 1. A multi-rate multicast transmission model with single point to multiple points

(1) The network is a connection-oriented one, and time is slotted with the duration $[n, n+1)$ by the sampling period T. The associated data is transferred by a fixed size packet.
(2) In every sampling period, the multicast source issues and transmit forward control packet (FCP) that is turned around by the branch node and destinations, and the backward control packet (BCP) is sent back to the source.
(3) After the multicast source receives the BCPs aggregated by the branch nodes, they will take appropriate action to adjust their transmitting rates of multicast traffic based on the maximum sending rate in BCP.
(4) The branch node of the multicast session replicates each data packet including FCP and transfers these packets to all its downstream nodes. Moreover, the branch nodes consolidate the BCPs that carry all the available rates and the relevant link bandwidth from different branches into one BCP and feedback the new BCP to their upstream node.
(5) After receiving the data packets coming from the network, the receivers construct the BCPs and send them back to the network.
(6) The component M is the total number of receivers corresponding the node RT $i, (i = 1, 2, \cdots, m)$. The maximum number of packets which is allowed by the multicast source to be transmitted into the network in one interval T is denoted by μ.
(7) For each receiver ji, the buffer occupancy is denoted by $x_{ji}(n)$ at time slot n and the desired buffer level is denoted by $\bar{x}_{ji}$.

(8) The packet number sending out by the receiver ji in one interval T is denoted by O_{ji}, the receiver ji has the forward delay τ_{ji} from the next upstream branch node to itself, the round-trip delay (RTD) between the next upstream branch node to itself $\tau_{Rji} = 2\tau_{ji}$, and $\tau_{Rj} = \max(\tau_{Rj1}, \tau_{Rj2}, \cdots, \tau_{RjM})$. We further assume that τ_{ji} and τ_{Rji} are integers, which are reasonable by adjusting T. And the link delay is dominant compared to the other delays such as proceeding delay and queuing delay, etc.

(9) Each router schedules the packets in a first-come-first-service way. The component $R_j(n)$ represents the receiving rate of the computed receivers j at time slot n, and $R_c(n)$ denotes the transmission rate at CBR source at time slot n.

Under the above notations and assumptions, the buffer occupancy of the receiver ji is determined by

$$x_{ji}(n+1) = Sat_{K_{ji}}\{x_{ji}(n) + R_j(n-\tau_{ji}) - O_{ji}\} \quad (1)$$

Where K_{ji} is the buffer size, $x_{ji}(n)$ is the buffer occupancy of the receiver ji at time slot n, and $R_j(n-\tau_{ji})$ is the sending rate of the next upstream branch node to the receiver ji.

$$Sat_{K_{ji}}\{x_{ji}\} = \begin{cases} K_{ji}, & x_{ji} > K_{ji}; \\ x_{ji}, & 0 \le x_{ji} \le K_{ji}; \\ 0, & x_{ji} < 0; \end{cases}$$

After lifting the saturation restriction, equation (1) can be written into

$$x_{ji}(n+1) = x_{ji}(n) + R_j(n-\tau_{ji}) - O_{ji} \quad (2)$$

III. THE ALGORITHM

The key component of the proposed congestion control algorithm is the way to compute the required source rate matching with the destinations' buffer. If $x_{ji}(n)$ is too high, it often leads to buffer overflowing and packet loss. If $x_{ji}(n)$ is too low, it increases the likelihood of link underutilization during occasionally idle period. Thus the router buffer occupancy plays an important role in the congestion control. In this paper, we firstly propose the following PI controller, which is located at the next upstream node of each receiver and is updated upon every T epoch for MR-MCC.

$$R_j(n) = \mu + \sum_{k=1}^{M} a_k\left(x_{jk}(n-\tau_{jk}) - \bar{x}_{jk}\right) + \sum_{k=1}^{\tau_{Rj}} b_k R_j(n-k) \quad (3)$$

where a_k and b_k are proportional and integrative parameters, which are to be determined by the stability criteria. The coefficient is used to locate all the poles of the closed-loop system (2) and (3) within the unit circle to ensure the stability. The component $\bar{x}_{ji}$ is the target queue length. In (3) it is seen that, if the buffer occupancy of the receiver ji is measured at the instances $n-\tau_{ji}$, after the feedback delay τ_{ji} the BCP reaches the controller located at the next upstream node RT j $(j = 1, 2, \cdots, m)$ and the router then takes out the buffer occupancy of the destination nodes at time $t = n$. By doing so, the designed controller can be expected to have flexibility to cope with the sharp oscillation in buffer occupancy which will lead the network to lose packets. In addition, the calculation in (3) is completely independent of virtual connections travelling through the multicast tree, which means the scheme has scalability.

Figure 1 depicts the proposed congestion control model of single point to multiple points in a multicast network. The sending rate at the source will eventually be the maximum of the rates that can be accommodated by the entire paths to individual receiver. So it is necessary to convert down the rate of an incoming flow at every branching point to the values that can be accommodated by its participating branches. The incoming flow rate at every branching point is enforced to be the maximum of the rates that is also expected by corresponding receivers [24]. i.e., the receivers j send the BCPs (Back Control Packets) to the next upstream branch node RT j as soon as they receive the FCPs (Forward Control Packets), then PI controller located at RT j computes the expected rate $\hat{R}_j$ for the receivers at the next period T according to equation (3). Moreover, every branch node merges BCP including the sending rate $\hat{R}_j$ and selects the maximum of the rate $\hat{R} = \max(\hat{R}_1, \hat{R}_2, \cdots, \hat{R}_m)$, then feedback the value to the next upstream node until the value $\hat{R}$ is feedbacked to the multicast source. Then, the multicast source adjusts the incoming flow rate in terms of the feedback maximum sending rate. Since the source sends data at the maximum rate including FCP, the independently trimmed rate $\hat{R}$, which is the rate allowed by its entire path, satisfies the best-effort service for all the receivers j.

Before we present the algorithm in details, we specify the following variables. The *multicasttree*[*i*] = 1(0) means the i^{th} branch point receive (don't receive) FCP or BCP control packet; while *receivertree*[*i*] = 1(0) means the i^{th} branch point receive (don't receive) confirmations of all receivers. Based on the above specifications, the pseudocode of the proposed router and source algorithms are given in Figure 2.

Source Algorithm

Upon every T epoch
Transmit data including FCP;
Upon multicast source receives a consolidation BCP from its downstream
Adjust the transmitting rates in terms of min(the maximum receiving rate of corresponding receivers in the consolidated BCP, the bandwidth of the connective link);

Router Algorithm

```
If multicasttree[i]= =1
{
 if the packet is an FCP
  {
   Put the data packet in the buffer;
   Multicast the data packet including FCP to the downstream
   nodes;
  }
else
{
  If the node is the next upstream node of the receiver j
  {
   Compute the expected sending rate R_j for the receivers i
      using PI controller;}
  else
  {
   Select the maximum expected incoming rate of the next
     downstream node; }
   Construct the BCP based on the received BCPs
     and the relevant case;
   Feedback it to the upstream node;
   if receivedtree[i]==1
     { Delete the data packets from the buffer; }
   else {
     Maintain the data packets in the buffer
     until Receive all confirmations of the receivers;}
  }
}
```

Receiver Node Algorithm

Upon receipt of an FCP
Put the data packets into the buffer;
Construct the BCP based on the current case of the receiver nodes;
Feedback the BCP to the upstream branch point;

Figure 2. Pseudocode of source/ router/ receiver algorithms

IV. IMPLEMENTATION OF PI CONTROLLER

A. Control Gain Selections

We use a rate-based rather than a window-based adaptation algorithm to achieve congestion control in MR-MCC tree. The window-based scheme has extra complexity in maintaining and synchronizing the congestion window across all receivers, it usually generates data bursts periodically. In our proposed PI control scheme, the rate adaptation takes into account the receivers' buffer occupancies as well as the variation of RTTs. The controller parameters are designed to guarantee the stability of rate, which ensures a smooth dynamic of rate adaptation to minimize the packet loss rate. This in turn brings an obvious advantage of the proposed scheme over the widely adopted AIMD (addictive increase and multiplicative decrease) (see, for example [20]). For example, in AIMD, it is difficult to choose the appropriate increase and decrease factors to guarantee the system's stability and then to obtain smooth and healthy rate adaptation and good link utilization.

In this section, the stability of the proposed PI congestion control scheme is analyzed as follows.

Considering the proposed controller of equation (2), if z-transformation is applied, one can easily arrive at

$$(z-1)X_{ji}(z) = z^{-\tau_{ji}} R_j(z) - O_{ji}D(z), \qquad (4)$$

where the z-transformation of $x_{ji}(n)$, $R_j(n)$ are described by

$$X_{ji}(z) = \sum_{n=0}^{+\infty} x_{ji}(n)\cdot z^{-n}, \quad R_j(z) = \sum_{n=0}^{+\infty} R_j(n)\cdot z^{-n},$$

and $D(z) = \sum_{n=0}^{+\infty} z^{-n} = \dfrac{z}{z-1}$.

Taking the z-transform of equation (3), one yields

$$R_j(z) = \mu D(z) + \sum_{k=1}^{M} a_k (z^{-\tau_{jk}} X_{jk}(z) - \overline{X}_{jk}D(z)) + \sum_{k=1}^{\tau_{Rj}} b_k z^{-k} R_j(z) \quad (5)$$

From equation (4) and (5), one has

$$\Delta z \cdot R_j(z) = -\sum_{k=1}^{M} a_k (O_{jk}D(z) z^{-\tau_{jk}} + \overline{X}_{jk}D(z)(z-1))$$

and

$$\Delta z = (1 - \sum_{k=1}^{\tau_{Rj}} b_k z^{-k})(z-1) - \sum_{k=1}^{M} a_k z^{-\tau_{Rjk}} \qquad (6)$$

The coefficients a_k and b_k $(k = 1, 2, \cdots, \tau)$ are determined by the stability criteria of the control theory. From a control-theoretic view when all the zeros of (6) lie within the unit disc, the original network system (2) with the controller (3) is stable in terms of in terms of the source's emitting rate. Stability in terms of sending rate is a prerequisite in congestion control to ensure that the network has no oscillation of sending rate and thus minimizes the packet loss rate.

Without loss of generality, we group those nodes into one class, which have a small variation of time delays and sending rates. Thus we divide receivers j $(j = 1, 2, \cdots, m)$ into q groups based on the RTTs, and in each group, the RTT is assumed to be equal, i.e.,

$$\{\tau_{Rj1}, \tau_{Rj2}, \tau_{Rj3}, \cdots, \tau_{RjM}\} = \{t_1, \cdots, t_1, t_2, \cdots, t_2, \cdots, t_q, \cdots, t_q\}$$

and we set n_l is the number of the RTT t_l $(l = 1, 2, \cdots, q)$ corresponding to the l^{th} group receivers, then $M = \sum_{l=1}^{q} n_l$).

We set $a_i = \dfrac{\tau\varepsilon - 1}{M}$, $b_j = j\varepsilon - 1 - na$, $j \in [1, \tau_{Rj}], i \in [1, M], n \in [0, M]$. if $j \le \tau_{R1}$, $n = 0$;

else $\tau_{R1} \le \tau_{Ri} < j \le \tau_{R(i+1)}$ and $j \le \tau_{RM}$, we can get

$$\Delta(z) = z^{-\tau_{Rj}}[z^{(\tau_{Rj}+1)} - \varepsilon(z^{\tau_{Rj}} + z^{(\tau_{Rj}-1)} + z^{(\tau_{Rj}-2)} + \cdots + z + 1)] \quad (7)$$

From [21], when $\varepsilon < 1/(\tau + 2)$, all the zeros of (7) lie within the unit disc, the original network system (2) with the controller (3) is stable.

V. SIMULATION RESULTS

To evaluate the performance of the studied multicast congestion control method, we focus upon the following simulation model and are mostly interested in analyzing the transient behaviors of the network. In the performance analysis, the duration of response time, link utilization, receiving rate of receivers and steady state of buffer occupancy are the main concerns. Simulation is carried out over a wide range patterns and propagation between two different nodes can lie in LAN (Local Area Network) case or the WAN (Wide Area Network) case.

In simulations, we process the nodes together, which have a small extent change of the time delay and sending rate. Then we make the time delay and sending rate to be unified respectively.

Since the situation of every node in each group is similar, we only choose one node from each group as a representative. We assume that the link delay is dominant compared to the other delays such as processing delays and queuing delay. The relevant notations and assumptions are listed in the following table 1, and pertain to the simulation. We further set the bandwidth of the links l_1, l_2, l_3, l_4 to be 12 Mbps, 6 Mbps, 12 Mbps and 8 Mbps respectively. According to the introduced stability test to select the control gains, one computes the relevant parameters *a*, *b*, *c*.

For group 1, $\varepsilon = 1/6$, $a_1 = a_2 = \cdots a_{200} = -4/1200$, $b = [b_1, b_2] = [-5/6, -4/6]$;

For group 2, $\varepsilon = 1/10$, $a_1 = a_2 = \cdots a_{200} = -4/2000$ and

$$b = [b_1, b_2, b_3, b_4, b_5, b_6] = [-9/10, -8/10, -7/10, -6/10, -5/10, -4/10];$$

For group 3, $\varepsilon = 1/14$, $a_1 = a_2 = \cdots a_{200} = -4/2800$ and

$$b = [b_1, b_2, b_3, b_4, b_5, b_6, b_7, b_8, b_9, b_{10}] = [-13/14, -12/14, -11/14, -10/14, -9/14, -8/14, -7/14, -6/14, -5/14, -4/14];$$

$= [-16/17, -15/17, -14/17, -13/17, -12/17, -11/17, -10/17, -9/17, -8/17, -7/17, -6/17, -5/17, -4/17, -3/17]$;

For group 4, $\varepsilon = 1/17$, $a_1 = a_2 = \cdots a_{200} = -3/3400$ and

$$b = [b_1, b_2, b_3, b_4, b_5, b_6, b_7, b_8, b_9, b_{10}, b_{11}, b_{12}, b_{13}, b_{14}]$$

TABLE I. THE PARAMETERS IN THE SIMULATION MODELS

Receiver Node / Variables	Group 1		Group 2		Group 3		Group 4	
	r_{11}	r_{21}	r_{12}	r_{22}	r_{13}	r_{23}	r_{14}	r_{24}
O_{ji} (Mbps)	2	1	3	4	2	4	5	5
$\overline{x}_i$ (Kb)	70		80		120		220	
τ_{ji} (msec)	1		3		5		7	
ε	1/6		1/10		1/14		1/17	
τ_{Rji} (msec)	2		4		10		14	

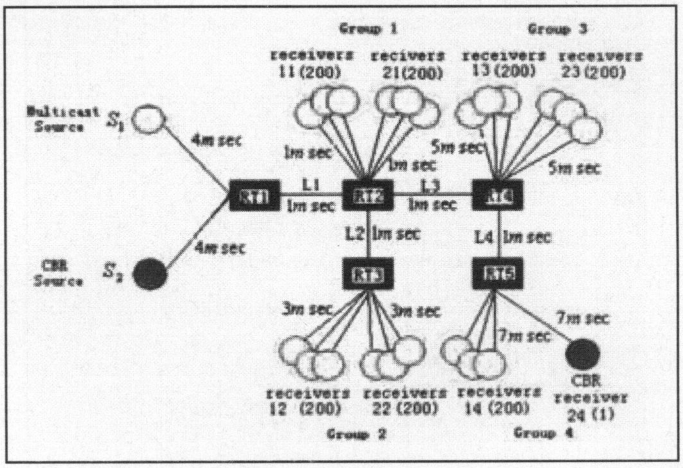

Figure 3. The simulation model

The simulation of the paper is shown in the figure 4 -12. Figure 4 shows the sending rate of sources. CBR source sends data at the constant rate of 3 *Mbps*. The initial sending rate of multicast sources is 6 *Mbps*; As time goes on, the sending rates are gradually adjusted and have some fluctuation. Then sending rate of the multicast source is quickly stable at the value of 5 *Mbps* during 88 *m* sec.

Figure 5 and figure 6 respectively show the buffer occupancy of the corresponding receivers of multicast source 1. These receivers all have some fluctuation in the beginning. Then the buffer occupancy gradually is stabilized.

The specific description of simulation results is given in Table 2.

Figure 7-10 show the receiving rates of the corresponding receivers of one multicast source and CBR source. These three figures demonstrate the dynamic change of the network's transmitting rate. Though the rate has fluctuation at first, it can be rapidly stable and be the same as the maximum of output rate. Thus the high utilization of the network link can be achieved. Figure 11 shows the utilization of the link L_1, L_2 and figure 12 shows the utilization of the link L_3, L_4. In the stable period, they achieve the maximum utilization at the maximum of 100%.

TABLE II. THE SPECIFIC DESCRIPTION OF SIMULATION RESULTS

Parameters / Objects	Buffer occupancy in stable period (*M bps*)	The response time of the buffer occupancy (*m sec*)	Receiving rate in stable period (*M bps*)	The response time of receiving rate (*m sec*)
Receiver 11	71.5	43	2	118
Receiver 12	58.25	63 ~ 113	3	89
Receiver 13	91.5	113	2	114
Receiver 14	4.2	71 ~ 102	5	91
Receiver 21	75.25	58	1	102
Receiver 22	46	65	4	76
Receiver 23	42	74	4	78
Receiver 31	0	14	3	14

From Figure 4 –12, one observes that in the time interval [0, $5+\tau_1$], the sources transmit data packets at the initial rates, and at this time the receive nodes have not acquired data packets because the packets of multicast traffic have not arrived at the receiver during this interval. But in the time interval $[5+\tau_1, 5+2\tau_1)$, data packets begin to be accumulated in the buffer of receiver group 1 because the CP of receiver group 1 have not still reached the router RT2, the sending node still transmits packets at the initial rate. After the delay $(5+2\tau_1)$, the feedback information of receiver group 1 returns to the RT2, the PI controller located at RT2 starts to compute the expected receiving rate of the receiver group 1, and the BCP including the computed receiving rate is fed back to its upstream node. As time goes, the consolidated BCPs with the computed receiving rate of every receiver node return to the sources of the corresponding session. Then the source of every multicast session acquires the maximum receiving rate of all corresponding receivers from the consolidated BCP as the sending rate of the next time, and adjusts the sending rate gradually to make the buffer occupancy and the sending rate become steady.

VI. CONCLUSIONS AND FUTURE WORK

This paper presents a theoretic analysis and design method of MR-MCC using explicit rate feedback mechanism to satisfy the different needs of the multiple users. The PI controller, whose control parameters can be designed to ensure the stability of the control loop in terms of buffer occupancy on the basis of control theory, is used in the next upstream node of the receivers to regulate the receiving rate. Relevant pseudocodes for implementation have subsequently been developed. It is clearly that the proposed MR-MCC scheme solves intra-protocol unfairness and low link utilization of SR-MCC. Simulations have been carried out with a multicast source and a CBR source. Simulation results demonstrate the efficiency of our scheme in terms of the system stability, high link utilizations, fast response, scalability, high unitary throughput, intra-session fairness and inter-session fairness. Future research would investigate into the TCP-friendly related issues in multicast congestion control along this line of study.

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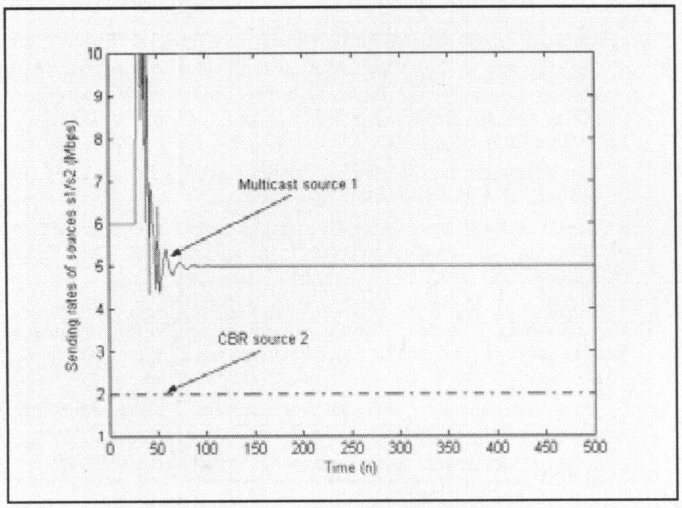


Figure 4. The sending rate of the sources

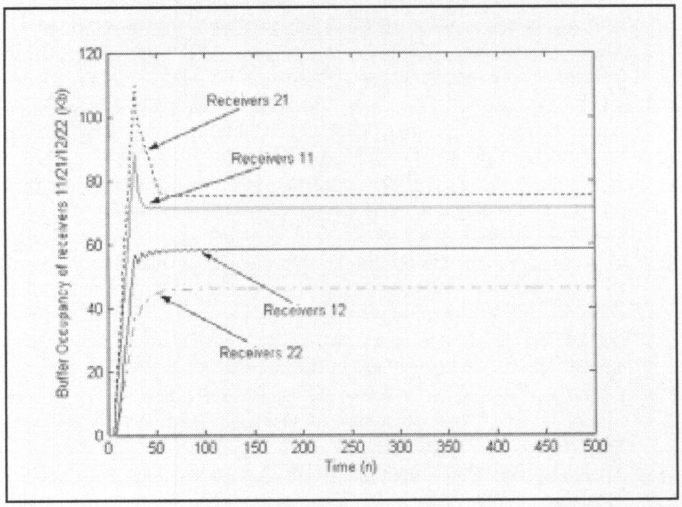


Figure 5. The buffer occupancy of receivers 11, receivers 21, receivers 12 and receivers 22

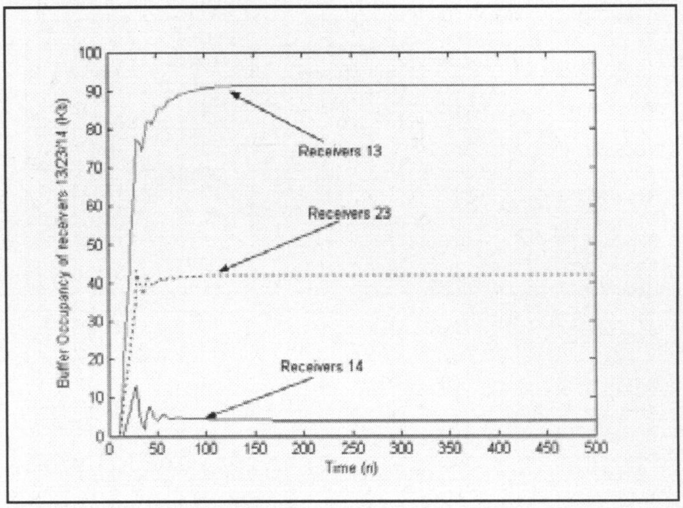


Figure 6. The buffer occupancy of receivers 13, receivers 23 and receivers 14

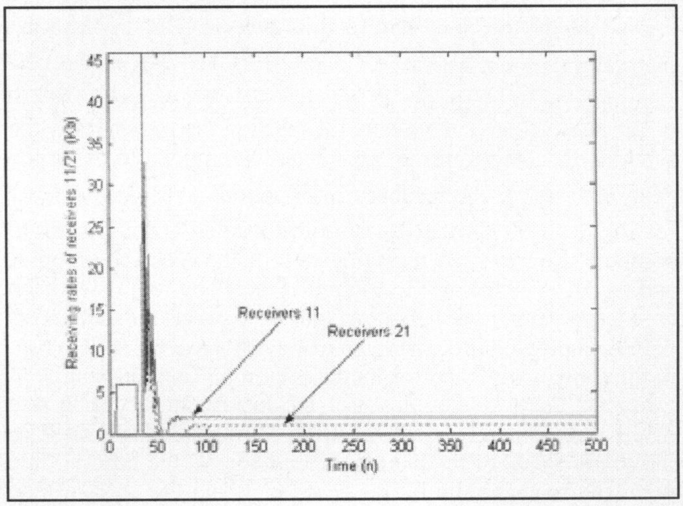


Figure 7. The receiving rates of receivers 11and receivers 21

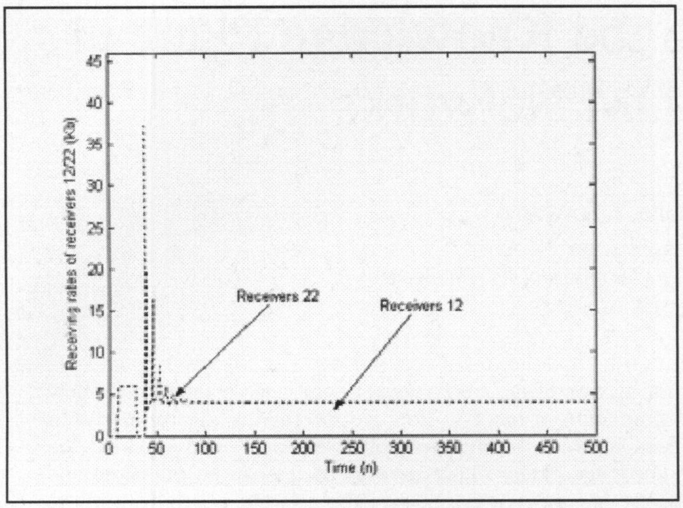


Figure 8. The receiving rates of receivers 12 and receivers 22

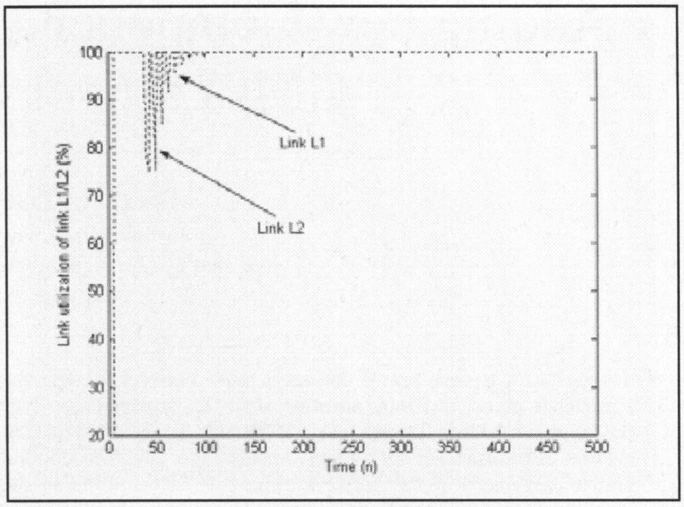


Figure 11. The link utilization of link L1 and link L2

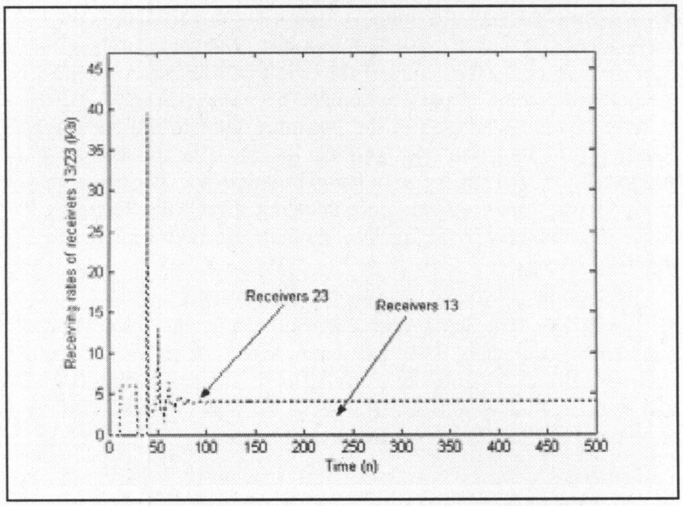


Figure 9. The receiving rates of receivers 13 and receivers 23.

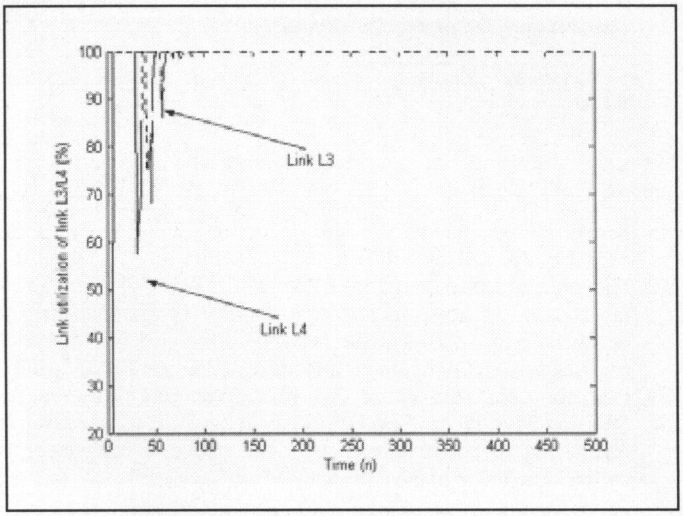


Figure 12. The link utilization of link L3 and link L4

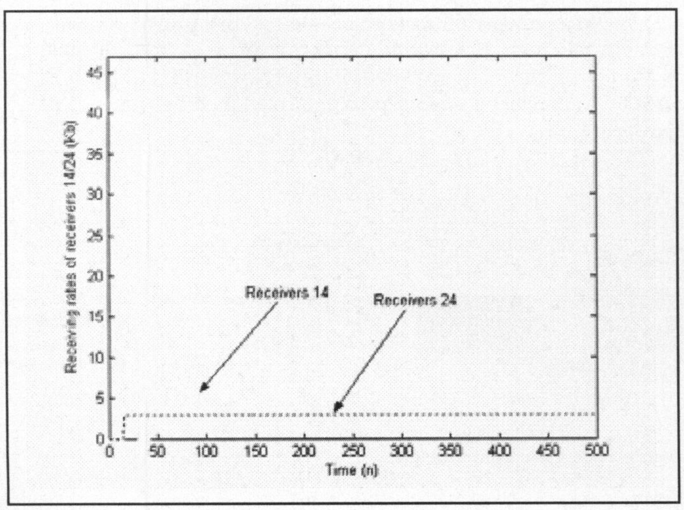
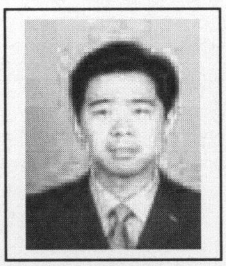
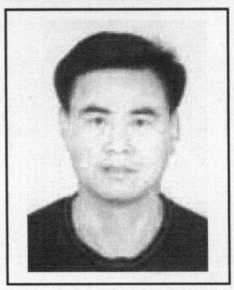


Figure 10. The receiving rates of receivers 14 and receivers 24



Nai-xue Xiong (1978 -) is PHD. Student in Computer School of Wuhan University, the State Key Lab of Software Engineering, PR. China. His main research interests lie in neural network, flow control and protocol in computer networks.
Email: N.Xiong@mail.ccnu.edu.cn, xiongnaixue@hotmail.com.



Professor **He Yanxiang** (1952-) is now a Full Professor, Dean of Computer School and the associate head of State Key Lab of Software Engineering, Wuhan University, PR China. Professor He received his M.S. and Ph.D. degrees in computer science from the Oregon University, U.S. and Wuhan University respectively. He has published over 100 papers in related journals and conference proceedings and 15 academic works and teaching books. His main research interests lie in neural network, communication networks