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Computer Networks 38 (2002) 553–575

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# MTCP: scalable TCP-like congestion control for reliable multicast<sup>☆</sup>

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Received 28 September 2001; accepted 15 October 2001

Responsible Editor: I.F. Akyildiz

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## Abstract

We present MTCP, a congestion control scheme for large-scale reliable multicast. Congestion control for reliable multicast is important, because of its wide applications in multimedia and collaborative computing, yet non-trivial, because of the potentially large number of receivers involved. Many schemes have been proposed to handle the recovery of lost packets in a scalable manner, but there is little work on the design and implementation of congestion control schemes for reliable multicast. We propose new techniques that can effectively handle instances of congestion occurring simultaneously at various parts of a multicast tree.

Our protocol incorporates several novel features: (1) hierarchical congestion status reports that distribute the load of processing feedback from all receivers across the multicast group, (2) the relative time delay concept which overcomes the difficulty of estimating round-trip times in tree-based multicast environments, (3) window-based control that prevents the sender from transmitting faster than packets leave the bottleneck link on the multicast path through which the sender's traffic flows, (4) a retransmission window that regulates the flow of repair packets to prevent local recovery from causing congestion, and (5) a selective acknowledgment scheme that prevents independent (i.e., non-congestion-related) packet loss from reducing the sender's transmission rate. We have implemented MTCP both on UDP in SunOS 5.6 and on the simulator ns, and we have conducted extensive Internet experiments and simulation to test the scalability and inter-fairness properties of the protocol. The encouraging results we have obtained support our confidence that TCP-like congestion control for large-scale reliable multicast is within our grasp. © 2001 Elsevier Science B.V. All rights reserved.

*Keywords:* Reliable multicast; Congestion control

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<sup>☆</sup>This work was supported by a grant from the NCSU Center for Advanced Computing and Communications (CACC). An earlier version of this paper appeared in the Proceedings of Infocom'99.

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## 1. Introduction

As the Internet becomes more diversified in its capabilities, it becomes feasible to develop and offer services and applications that were not possible under earlier generations of Internet technologies. The Multicast Backbone (MBONE) and

IP-multicast are two Internet technologies that have enabled a wide range of new applications. Using multicast, large-scale conferencing involving hundreds to thousands of participants is possible over the Internet. As multicast technologies become more widely deployed, we expect to see new multicast-based applications that demand more bandwidth and higher speed. Many of these applications will require reliable data transfer.

Multicast traffic generated by these applications can be of two types: quality-of-service guaranteed and best effort. QoS guaranteed traffic requires the underlying network to provide per-flow resource reservation and admission control services. Unless these services become widely deployed over the Internet and made sufficiently inexpensive for general use, they will likely be available only to a small fraction of future Internet traffic, and multicast traffic will be primarily best effort. This work is concerned with the flow and congestion control of best-effort multicast traffic.

Congestion control is an integral part of any best-effort Internet data transport protocol. It is widely accepted that the end-to-end congestion control mechanisms employed in TCP [1] have been one of the key contributors to the success of the Internet. A conforming TCP flow is expected to respond to congestion indication (e.g., packet loss) by drastically reducing its transmission rate and by slowly increasing its rate during steady state. This congestion control mechanism encourages the fair sharing of a congested link among multiple competing TCP flows. A flow is said to be TCP-compatible or TCP-like if it behaves similar to a flow produced by TCP under congestion [2]. At steady state, a TCP-compatible flow uses no more bandwidth than a conforming TCP connection running under comparable conditions.

Unfortunately, most of the multicast schemes proposed so far do not employ end-to-end congestion control. Since TCP strongly relies on other network flows to use congestion control schemes similar to its own, TCP-incompatible multicast traffic can completely lock out competing TCP flows and monopolize the available bandwidth. Furthermore, multicast flows insensitive to existing congestion (especially congestion caused by their own traffic) are likely to cause simultaneous

congestion collapses in many parts of the Internet [3]. Because of the potential far-reaching damage of TCP-incompatible multicast traffic, it is highly unlikely that transport protocols for large-scale reliable multicast will become widely accepted without TCP-like congestion control mechanisms.

The main challenge of congestion control for reliable multicast is scalability. To respond to congestion occurring at many parts of a multicast tree within a TCP time-scale, the sender needs to receive immediate feedback regarding the receiving status of all receivers. However, because of the potentially large number of receivers involved, the transmission of frequent updates from the receivers directly to the sender becomes prohibitively expensive and non-scalable.

Another challenge is the isolation of the effects of persistent congestion. As a single multicast tree may span many different parts of the Internet, TCP-like congestion control will reduce the sender's transmission rate upon indication of congestion in any part of the tree. While such a feature fosters fairness among different flows (*inter-fairness*), it does not address the issue of fairness among the receivers in the same multicast group (*intra-fairness*) [4]. Specifically, it would be unfair for non-congested receivers to be subject to a low transmission rate just because of some isolated instances of congestion.

In this paper, we introduce multicast TCP (MTCP), a new congestion control protocol for reliable multicast that addresses the inter-fairness and scalability issues. The issue of intra-fairness is outside the scope of this paper, and it will be addressed in future work. Our protocol is based on a multilevel logical tree where the root is the sender, and the other nodes in the tree are receivers. The sender multicasts data to all receivers, and the latter send acknowledgments to their parents in the tree. Internal tree nodes, hereafter referred to as sender's agents (SAs), are responsible for handling feedback generated by their children and for retransmitting lost packets. MTCP incorporates several novel features, including:

1. hierarchical congestion status reports that distribute the load of processing feedback from all receivers across the multicast group,

2. the relative time delay (RTD) concept which overcomes the difficulty of estimating round-trip times (RTTs) in tree-based multicast environments,
3. window-based control that prevents the sender from transmitting faster than packets leave the bottleneck link on the multicast path through which the sender's traffic flows,
4. a retransmission window that regulates the flow of repair packets to prevent local recovery from causing congestion, and
5. a selective acknowledgement scheme employed at SAs to prevent independent (i.e., non-congestion-related) packet loss from reducing the sender's transmission rate.

We have implemented MTCP both on UDP in SunOS 5.6 and on the simulator ns, and we have conducted extensive Internet experiments and simulation to test the scalability and inter-fairness properties of the protocol. The encouraging results from these experiments indicate that MTCP is an effective flow and congestion control protocol for reliable multicast.

Tree-based protocols are not new and have been studied by many researchers [5–9]. However, little work has been done on congestion control for these protocols. Instead, most previous work has focused on the issues of error recovery and feedback implosion. In Refs. [5,10] it has been analytically shown that tree-based protocols can achieve higher throughput than any other class of protocols, and that their hierarchical structure is the key to reducing the processing load at each member of the multicast group. Tree-based protocols such as RMTP [6] and TMTP [8] do not incorporate end-to-end congestion control schemes and do not guarantee inter-fairness. In Refs. [9,11] it was proposed to use a tree structure for feedback control, and a detailed description of how to construct such a tree was provided, but no details on congestion control were given. A more detailed discussion on related work can be found in Section 5.

This paper is organized as follows. In Section 2 we provide an overview of MTCP, and in Section 3 we present a detailed description of the protocol and its operation. In Section 4 we present results

from Internet experiments and simulation. In Section 5 we discuss related work, and we conclude in Section 6.

## 2. Overview of MTCP

MTCP was designed with two goals in mind: TCP-compatibility and scalability. Compatibility with TCP traffic is needed because TCP is the most commonly used protocol in the Internet, and also because the utility of TCP depends on all other network flows being no more aggressive than TCP congestion control (i.e., multiplicative decrease on congestion occurrence, and linear increase at steady state). Non-TCP-compatible flows may lock out TCP traffic and monopolize the available bandwidth. Scalability is necessary because the target applications of reliable multicast may involve a very large number of receivers. Below we give an overview of MTCP, and in the next section, we provide a detailed description of the protocol.

*Packet loss detection and recovery via selective acknowledgments:* A sender multicasts data packets using IP-Multicast [12] to all receivers. SAs in the logical tree store packets received from the sender in their buffers, and set a timer, called *retransmission timer*, for each packet they buffer. The sender also sets a retransmission timer for each of the packets it transmits. Each receiver may send a positive acknowledgment (ACK) or a negative acknowledgment (NACK) to its parent in the tree. Received packets are reported in ACKs and missing packets are reported in NACKs. An SA (or the sender) discards a buffered packet when it receives an ACK from all of its children. On the other hand, an SA (or the sender) retransmits a packet via unicast (a) upon receiving a NACK reporting that the packet is missing, or (b) if it does not receive a ACK for the packet from all its children in the logical tree before the timer associated with the packet expires.

*Hierarchical congestion reports:* Each SA independently monitors the congestion level of its children using feedback received from them. When an SA sends an ACK to its parent, it includes in the ACK a summary of the congestion level of its

children (called *congestion summary*). The parent then summarizes the congestion level of its own children, sends the summary to its parent, and so on. The sender regulates its rate based on its own summary. The congestion summary carries an estimate of the minimum bandwidth available along the multicast paths to the receivers contained in the subtree rooted at the SA that sends the summary. An SA computes its summary using the summaries it has received from its children and a TCP-like congestion window (cwnd) maintained using feedback from its children. As a result, the summary computed at the sender represents the current available bandwidth in the bottleneck link on the paths to all receivers in the multicast group. By sending only as much data as the bottleneck link can accommodate, the sender will not aggravate congestion anywhere in the network.

*TCP-like congestion window:* Each SA (including the sender) estimates the minimum bandwidth available in the multicast routes from the sender to its children by maintaining a TCP-like cwnd. An SA maintains its cwnd using TCP-Vegas [13] congestion control mechanisms such as slow start and congestion avoidance. The only differences with TCP-Vegas are that (1) the cwnd is incremented when an SA receives ACKs for a packet from all of its children, and (2) receivers send NACKs for missing packets, and an SA immediately retransmits the packets reported missing. For instance, slow start is implemented as follows. When an SA receives a new packet from the sender, it buffers the packet and sets the retransmission timer (RTO) of the packet to an estimated time before which the packet should be acknowledged. At the beginning of transmission or after a RTO expiration, an SA sets its cwnd to one. The SA increments its cwnd by one when a packet is acknowledged by all of its children. Once the cwnd reaches a threshold called *ssthresh*, congestion avoidance is invoked and the cwnd is incremented by  $1/\text{cwnd}$  each time a new packet is acknowledged by all of its children.

*Congestion summary:* The congestion summary sent by an SA whose children are leaf nodes, consists of two pieces of information: the size of its cwnd, and the estimated number of bytes in transit from the sender to the SA's children (transit window, *twnd*). *twnd* is initially set to zero, it is in-

cremented when a new packet is received from the sender, and it is decremented when a packet is acknowledged by all of the SA's children. The congestion summary of the other SAs consists of (1) the minimum of their cwnds and the cwnds reported by their children (*minCwnd*), and (2) the maximum of their *twnds* and the *twnds* reported by their children (*maxTwnd*). *maxTwnd* estimates the number of unacknowledged bytes in transit to the receivers in the tree and *minCwnd* estimates the congestion level of the bottleneck link on the multicast routes to the receivers in the tree. The sender always transmits data in an amount less than the difference between the values of *maxTwnd* and *minCwnd* that it computes. This window mechanism prevents the sender from transmitting faster than packets leave the bottleneck link.

*Relative time delay:* Unlike TCP where the sender maintains the cwnd based on feedback received from one receiver, MTCP does not provide closed-loop feedback. This is because SAs have to adjust their windows based on the ACKs for packets that another node (the sender) transmitted. The main problem with this open-loop system is that an SA cannot accurately estimate the RTT of a packet. This problem arises due to the unpredictable delay variance in the network and the fact that the sender's and SA's clocks are not synchronized. In MTCP, we measure the difference between the clock value taken at the sender when a packet is sent, and the clock value taken at the SA when the corresponding ACK is received from a child node. We call this time difference the RTD. The RTD to a child receiver can be easily measured by having each ACK carry the transmission time of the packet being acknowledged. Thus, RTD measurements can be taken every time the SA receives an acknowledgment. A weighted average of RTDs is used to estimate the retransmission timeout value (RTO) of packets. An SA sets the RTO of a packet to expire only after the sum of the send time of the packet and the RTO of the SA becomes less than the current clock value of the SA. The use of RTD for this purpose is appropriate because the protocol uses only the relative differences in RTDs.

*Retransmission window for fast retransmission:* Retransmission may cause congestion if many packets are lost in a loss burst and an SA re-

transmits them without knowing the available bandwidth between itself and its children. Recall that the *cwnd* at the SA only estimates the amount of data that can be sent from the sender to the SA's children. Because new and repair packets may travel over different routes, the *cwnd* cannot be used to regulate repair traffic. In MTCP, each SA maintains another window, called the retransmission window, used only for repair packets. The retransmission window is updated in the same way as *cwnd* (i.e., slow start, congestion avoidance, etc.). Since SAs receive ACKs for the packets they retransmit, maintaining the retransmission window does not incur significant overhead.

*Handling of independent loss:* MTCP decreases the sender's transmission rate if any link of the multicast routing tree is congested. This may raise a concern that the protocol is too sensitive to independent packet loss: since for large multicast groups, almost every transmitted packet may experience independent loss, it might be argued that the overall throughput will be reduced to zero. However, in MTCP, most occurrences of independent loss trigger NACKs to SAs which immediately retransmit the lost packets. Only packet loss accompanied by indication of congestion, such as retransmission timeouts or several consecutive duplicate NACKs, reduces the *cwnd*. Simulation results to be presented later confirm that independent packet loss is immediately recovered by SAs and does not have a negative effect on the overall throughput.

### 3. Detailed description of MTCP

#### 3.1. Selective acknowledgment scheme

In MTCP we use a selective acknowledgment (SACK) scheme in which each feedback contains information about all received packets. We also adopt a delayed acknowledgment scheme in which each ACK is delayed for a few tens of milliseconds before its transmission. Since an SA can quickly detect the packets lost by a receiver and retransmit them, these schemes reduce the number of acknowledgments and retransmissions. Also, our

SACK scheme provides a good means to recover from independent, uncorrelated losses.

When an SA (or the sender) receives an indication that a packet is lost, it immediately unicasts the missing packet to the receiver that sent the NACK, unless the same packet was (re)transmitted to this receiver within a time period equal to the current estimated RTT between the SA and the receiver. Since the SACK scheme indicates exactly which packets have been received, the SA can determine the packets that were lost by its children and retransmit only them, avoiding unnecessary retransmissions. Each ACK also contains the send time of the last packet received, the congestion summaries, and the number of buffers available at the receiver. This information is used for flow and congestion control as described below.

#### 3.2. Relative time delay measurement

In MTCP, an SA sets a retransmission timer for each packet it receives. The timer for a packet must be set to the mean time period between the time, the packet was transmitted by the sender and the time, that the SA expects to receive an ACK from all its children. This time period is hard to measure because of the clock differences between the SA and the sender. To overcome this difficulty, we introduce the RTD, defined as the difference between the clock value, taken at the SA, when the ACK for a packet is received from a child, and the clock value, taken at the sender, when the same packet was transmitted. MTCP requires that each data packet carry its transmission time at the sender, and that each ACK carry the transmission time of the last packet received, making it easy for SAs to compute the corresponding RTDs.

The RTD is used in the same way that the RTT is used in TCP-Vegas. For instance, the difference between the minimum measured RTD and the currently measured RTD to a child is used to estimate the number of packets in transit from the sender to the child. Also, a weighted average of RTDs and their deviations is used to set the retransmission timeout value  $RTO_{RTD}$ . Given a value for  $RTO_{RTD}$ , MTCP sets the retransmission timer of a packet to expire only after the sum of the send time of the packet (according to the

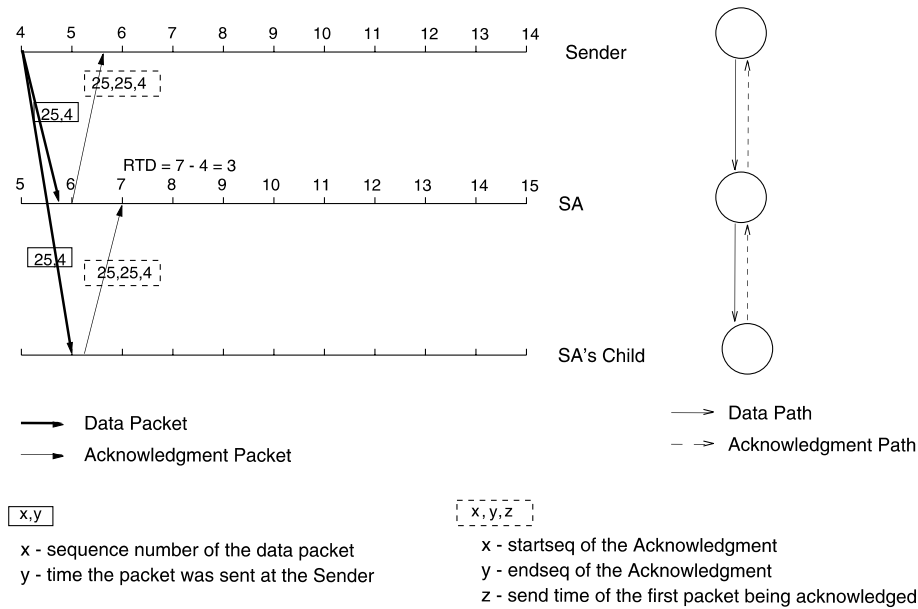


Fig. 1. Example of RTD measurement.

sender’s clock) plus the  $RTO_{RTD}$  of the SA becomes less than the current clock value of the SA.

Fig. 1 illustrates how the RTD is measured. In this example, the clocks of the sender and the SA are not synchronized, but the SA’s clock is ahead of the sender’s by one time unit. The sender transmits a packet at time 4 which is received by both the SA and its child, which then send an acknowledgment to their respective parents in the logical tree. Recall that the data packet includes its send time, and that the receivers copy this send time in their acknowledgments. Then, the RTD measured at the SA is 3 in this example, since the packet was transmitted by the sender at time 4, and it was received by the SA at time 7.

### 3.3. Round-trip time measurement

An SA will retransmit a packet whose timer has expired, at which time it needs to reset the retransmission timer. Recall, however, that retransmissions are unicast from an SA to its children and may follow a different path than the original multicast transmission from the sender. As a result, while the retransmission timeout  $RTO_{RTD}$  for

the *first* retransmission of a packet should be based on the RTD, for subsequent retransmissions an SA must adjust its timer to the RTT between itself and its children.

In order to estimate the RTT, an SA periodically polls its children by sending a probe packet. In the current implementation of MTCP, the polling period is set to 2 s. Upon receiving this probe, a child immediately sends an acknowledgment to its parent. The SA can then measure the difference between the time when it sent a probe and the time it received the corresponding acknowledgment. These RTT measurements are used to set the retransmission timeout  $RTO_{RTT}$  for packets retransmitted at least once.

### 3.4. Estimation of RTD- and RTT-based retransmission timeouts and timer backoff

The timeout and retransmission mechanisms employed in MTCP require the estimation of retransmission timeout values  $RTO_{RTD}$  and  $RTO_{RTT}$ . As these values can change over time because of routing and network traffic changes, MTCP needs to track these changes and modify its timeout val-

ues accordingly. We use the techniques employed by TCP [1] to set the  $RTO_{RTD}$  and  $RTO_{RTT}$  based on the average measured length of RTDs and RTTs, respectively, and their deviations in time. In MTCP, each SA (including the sender) sets its  $RTO_{RTD}$  (respectively,  $RTO_{RTT}$ ) to the maximum of the estimated  $RTO_{RTDs}$  (respectively,  $RTO_{RTTs}$ ) for all of its children.

When a packet is initially transmitted by the sender, or when it is received for the first time by an SA, it is buffered by the sender or SA and the retransmission timeout of the packet is set to  $RTO_{RTD}$ , which represents the estimated amount of time for the packet to travel through the multicast tree to the children of the SA and for the corresponding ACKs to arrive at the SA. As we explained in Section 3.2, the retransmission timer for the packet is set to expire when the sum of the send time of the packet plus the  $RTO_{RTD}$  becomes less than the current time. When a packet is retransmitted because of a NACK or the expiration of its retransmission timer, the latter is set to  $RTO_{RTT}$ , since retransmissions are performed by

an SA via unicast. Following this retransmission, whenever the timer expires, the timeout value is multiplied by two (exponential backoff) until an acknowledgment from all children is received.

Figs. 2 and 3 illustrate how the timeouts are used in a part of a multicast tree involving three nodes: the sender, an SA who is a child of the sender in the logical tree, and a child of the SA. Fig. 2 shows how the RTD-based timeouts are computed at the SA. In this scenario, the sender’s clock is assumed to be ahead of the SA’s clock by three time units. The sender transmits packet 25 at time 5; the send time is included in the packet header. The SA and its child transmit their ACKs, which also include the send time of the packet, to their respective parents. The SA receives the ACK at time 4, and calculates the RTD as  $-1 (= 4 - 5)$ . Assume that the estimated  $RTO_{RTD}$  is also  $-1$ . The sender then transmits packet 26 at time 7.2. The SA receives the packet and buffers it for possible retransmission. It also sets its retransmission timer to the send time of the packet plus the  $RTO_{RTD}$ , i.e., to  $7.2 + (-1) = 6.2$ . The packet

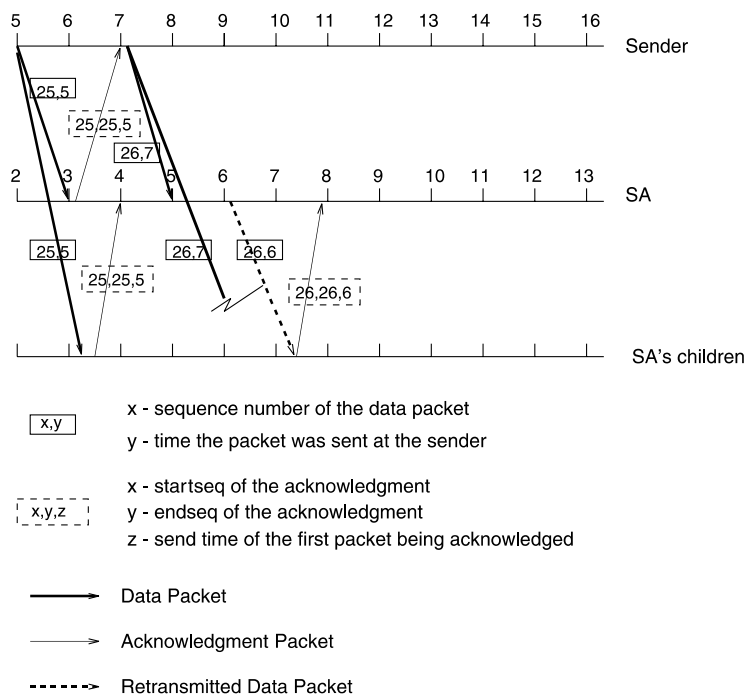


Fig. 2. Example of timeouts based on RTD.

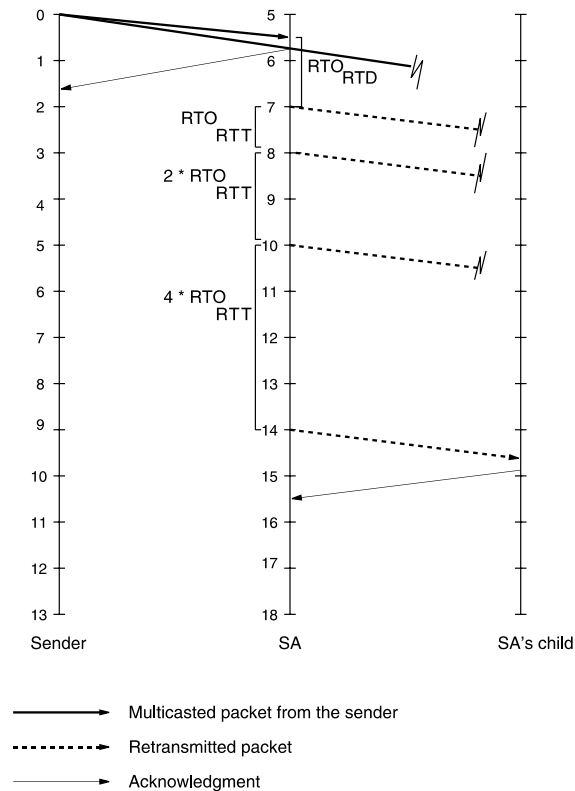


Fig. 3. Example of timeouts and exponential retransmission timer backoff.

is lost on the route to the SA's child. Since the SA has not received an ACK for the packet when the timer expires at time 6.2, it retransmits packet 26 to its child. At that time, the SA also sets its retransmission timer to  $RTO_{RTT}$  (not shown in the figure).

In Fig. 3, the  $RTO_{RTD}$  and  $RTO_{RTT}$  values are assumed to be 7 and 1, respectively, and the SA's clock is five units of time ahead of the sender's clock. The sender sends a packet at time 0, which reaches the SA at time 5.5. When the SA buffers the packet, it sets the retransmission timer of the packet to expire at time 7, based on the sum of the send time and the  $RTO_{RTD}$ . Since the SA does not get an acknowledgment within that time period, it retransmits the packet at time 7. At that time, the SA also sets the retransmission timer of the packet to expire at time 8, based on the  $RTO_{RTT}$  value of 1. Since it does not receive an ACK within the

timeout period, it retransmits the packet again at time 8 and doubles the timeout value to 2. This doubling of the timeout value continues until an ACK for the packet is received.

### 3.5. Slow start and congestion avoidance

Slow start is used by TCP [1] to find the available bandwidth on the path of a connection. Slow start is invoked at the beginning of transmission or after the retransmission timer of a packet expires. As in TCP, MTCP also employs a slightly modified version of slow start to estimate the available bandwidth in the multicast paths. Since during slow start cwnd increases exponentially, this process itself can cause congestion. MTCP uses the slow start mechanisms of TCP-Vegas [13] to detect congestion during slow start. MTCP also adopts the congestion avoidance mechanisms of TCP-Vegas [13]. During congestion avoidance, cwnd increases or decreases linearly to avoid congestion.

### 3.6. Retransmission window

The packets retransmitted by an SA to one or more of its children may take a different route than the multicast path these packets followed when originally transmitted by the sender. Consider the situation arising when an SA receives an ACK reporting a list of lost packets. If the SA is allowed to retransmit a large number of packets regardless of the available bandwidth between itself and its children (recall that cwnd estimates only the bandwidth between the sender and the SA's children), it may cause another instance of congestion. To overcome this problem, each SA maintains another window, called the retransmission window for each child, which is used only for retransmitted packets. Maintaining the retransmission window is possible because SAs receive ACKs for the packets they send (i.e., a closed-loop system). In the current implementation, the initial window size of the retransmission window is set to one. The size of the window changes in the same way that TCP-Vegas modifies cwnd. However, the size of the retransmission window is not reset to 1 even if retransmission is quiescent. Intermittent retrans-



mission can be accommodated immediately with the window set for the previous retransmission session (or burst). Unless the previous retransmission session undergoes slow start, the window will be larger than 1.

### 3.7. Window-based flow control

Flow control ensures that the sender does not send more data than the current capacity of a receiver, which is indicated by the number of buffers available. In MTCP, each receiver advertises the number of buffers available to its parent. We define the advertised window of a node to be the minimum of the buffers reported by the children of the node. The sender always sends no more than its own advertised window.

Another issue which affects the congestion control mechanism is how fast the sender transmits packets. In MTCP, the sender uses ACKs as a “clock” to strobe new packets in the network. Each SA maintains a variable called *transit window*, *twnd*. *twnd* is initially set to zero, it is incremented when a new packet is received from the sender, and it is decremented when a packet is acknowledged by all of the SA’s children. Whenever a retransmission timeout occurs, *twnd* is set to zero. The information about *twnd* is propagated up the tree to the sender and it is used to regulate the transmission rate of the sender, as explained below.

The congestion summary an SA sends to its parent consists of two parameters: (1) parameter *minCwnd*, which is the minimum of the SA’s *cwnd* and the *cwnd*s reported by its children, and which estimates the congestion level of the bottleneck link on the multicast routes to the receivers in the tree, and (2) parameter *maxTwnd*, which is the maximum of the SA’s *twnd* and the *twnd*s reported by its children, and which estimates the number of unacknowledged bytes in transit to the receivers in the tree. The difference between *maxTwnd* and *minCwnd* is called the *current window*. The sender always transmits data in an amount no more than the current window. This window mechanism prevents the sender from transmitting faster than packets leave the bottleneck link.

### 3.8. Hierarchical status reports

If the sender is allowed to determine the amount of data to be transmitted based only on its own *cwnd* and *twnd*, which it maintains using feedback from its immediate children alone, it is highly likely that the sender will cause congestion somewhere in the multicast routes. This possibility arises from the fact that the sender’s *cwnd* and *twnd* provide information only about the multicast paths to the sender’s immediate children in the logical tree; the sender receives no first-hand information about the congestion status of the multicast paths to other nodes. To ensure that an MTCP session will not cause congestion anywhere in the multicast routes, we require that the sender regulate its rate based on the congestion status of all receivers in the tree. This is accomplished by using a hierarchical reporting scheme, in which information about the status of each receiver propagates along the paths of the logical tree from the leaves to the root (the sender) in the form of the congestion summaries discussed in the previous subsection.

Fig. 4 illustrates how congestion summaries propagate from leaf receivers to the sender along the edges of the logical tree. In the figure, *cwnd* and *twnd* are expressed in units of number of segments (they are normally defined in bytes). SAs whose children are leaf nodes, send their *cwnd*s and *twnd*s to their parents. Referring to Fig. 4, node 7 sends its *cwnd*(30) and *twnd*(8) to its parent (node 3), node 9 sends its *cwnd*(15) and *twnd*(10) to its parent (node 3), and node 1 sends its *cwnd*(25) and *twnd*(5) to its parent (node 0). Upon receiving this information, the parent SAs send to their own parents the minimum of the received *cwnd*s and their own *cwnd*, and the maximum of the received *twnd*s and their own *twnd*. In Fig. 4, for instance, node 3 sends to node 0 the minimum of the *cwnd*s(15) and the maximum of the *twnd*s(10). The sender at node 0 computes the minimum of all *cwnd*s in the tree to be 15 and the maximum of all *twnd*s to be 10. Therefore, the current window is  $15 - 10 = 5$  segments. The sender sends no more than the minimum of the current window and the advertised window.

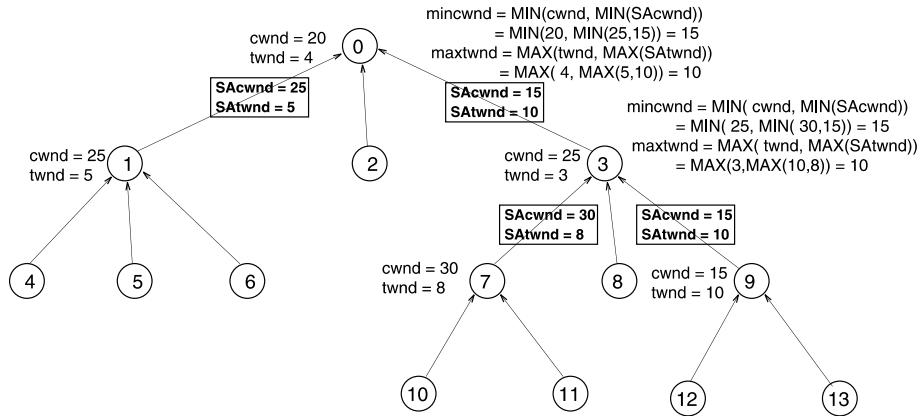


Fig. 4. Example of hierarchical status reports.

Note that the minimum bandwidth within the subtree rooted at a given SA is computed based only on information reported in the congestion summaries sent to this SA by its children. Since the maximum number of the children of a node is limited to a small constant, this scheme achieves good load balancing.

We also note that the delay between the time when an SA detects congestion and the time when the sender reduces its transmission rate in response to this congestion, may be longer than the TCP time-scale. Since the congestion status report (in the form of a summary) has to travel all the way to the root of the tree, this delay can be larger than a round-trip delay. However, unless congestion is reported directly to the sender (an approach that inevitably leads to ACK implosion), this extra delay is unavoidable. Furthermore, as it has been pointed out [14], extra delays up to a few seconds can be tolerated because network links where a single flow can create severe transient congestion are likely to employ an appropriate queue management mechanism such as random early detection (RED) [2,15]. We have observed through Internet experiments and simulation that the delay in MTCP is well within this range; more details are given in Section 4.

### 3.9. Window update acknowledgments

In MTCP, congestion summaries are normally piggybacked on every ACK and NACK. Thus,

congestion summaries are reported by an SA whenever a new packet is received (recall also that ACKs/NACKs are delayed for a few tens of milliseconds for an SA to receive ACKs from its children). However, if congestion summaries are reported only upon reception of data packets, deadlocks are possible since the window size at an SA may change even if the sender does not transmit any packets. Consider the following scenario. The sender transmits a number of packets and receives ACKs for the packets from all of its children. One of the SAs, say, SA A, on the other hand, does not receive ACKs for these packets from its children. Thus, SA A will report a high value for the twnd in the congestion summary it sends to its parent, and which will propagate to the sender. It is possible that this high twnd value will reduce the size of the current window of the sender to zero, in which case the sender will not transmit any more packets. Since the SA will not receive any packets from the sender, it will not send any ACKs either. When the SA finally receives ACKs from its children, its twnd decreases from the previously high value. Since no ACKs on which to piggyback this window update are generated, the sender will never learn of this updated window, and in turn, it will not send any packets at all, resulting in a deadlock.

To overcome this problem, we require each receiver to periodically send a congestion summary to its parent. This information is called a window update acknowledgment, and it is sent *only if* a

congestion summary has not been sent within the last period. In the current implementation, the period within which a window update acknowledgment is sent is initially set to 500 ms, and it is incremented by 1 s each time the receiver sends the acknowledgment. The period is reset to 500 ms when a new packet is received from the sender. This periodic window update information effectively resolves the deadlock problem.

#### 4. Internet experiments and simulation

We have implemented MTCP on top of UDP in Posix Threads and C, in SunOS 5.6. For the routing of MTCP packets, we have implemented a special process called *mcaster*, whose function is similar to that of *mroutd* in the MBONE. An *mcaster* simply “tunnels” incoming packets by first multicasting them to its own subnet via IP-multicast, and then forwarding them to the *mcasters* of its child sites in the tree via UDP. The members of the multicast group in the Internet experiments were distributed in five different sites: North Carolina State University, Raleigh, NC (NCSU), University of North Carolina, Chapel Hill, NC (UNC), Georgia Institute of Technology, Atlanta, GA (GaTech), Emory University, Atlanta, GA (Emory), and University of Texas, Austin, TX (UTexas). We have used an assortment of SPARC Ultras, SPARC 20s and SPARC 5s at each site, organized into the logical tree shown in Fig. 5.

We have also implemented MTCP on the network simulator *ns* to test the protocol on a larger scale. The experiments presented in this section were designed to evaluate the potential of MTCP for large-scale reliable multicast by addressing three key features of the protocol: scalability, inter-fairness, and sensitivity to independent loss.

##### 4.1. Scalability

Scalability is an important issue in reliable multicast, since the target applications may involve hundreds or even thousands of receivers. In the case of MTCP, the maximum fanout and the height of the logical tree are two important parameters in determining its ability to scale to large

numbers of receivers. In this section, we study the effect of these parameters on the performance of MTCP.

Let us first discuss the significance of the height of the tree. It is well known that the scalability of reliable multicast protocols is directly related to the degree of throughput degradation as the number of receivers increases. Since MTCP emulates TCP on a logical tree, the throughput behavior of MTCP is similar to that of TCP and can be approximated as [3]:  $T = cs/RTT\sqrt{p}$ , where  $s$  is the packet size,  $RTT$  is the round-trip time,  $p$  is the packet loss rate, and  $c$  is some constant. When the maximum fanout is limited to a small constant, the only factor in the expression for  $T$  affected by the number of receivers is the  $RTT$ . In MTCP, the  $RTT$  grows linearly with the height of the tree, since the sender recognizes congestion through feedback that propagates from a congested node to the sender via the ancestors of the node. In the best case, the throughput of MTCP will degrade in proportion to  $\log_f n$ , where  $f$  is the maximum fanout of the tree and  $n$  is the number of receivers. The worst case occurs when the height of the tree grows linearly with  $n$ . Consequently, we expect MTCP to achieve a high throughput, even for large numbers of receivers, when a well-balanced tree with a relatively small number of levels is employed.

The second parameter of interest is the number of children that an SA can accommodate, which determines the maximum fanout of the logical tree. In light of the limitations on the tree height, it is desirable to construct trees with a large fanout in order to support a large number of receivers. On the other hand, the larger the number of children of an SA, the higher the load imposed on the SA who has to receive and process feedback (ACKs, NACKs and congestion summaries) from its children. Thus, unless the maximum fanout of the tree is bounded, SAs may become overloaded and the throughput of MTCP will suffer.

##### 4.1.1. Internet experiments

Our first experiment investigates the maximum fanout of a logical tree that can be supported by MTCP without inducing an excessive load on each SA. The experiment involved a sender transmitting

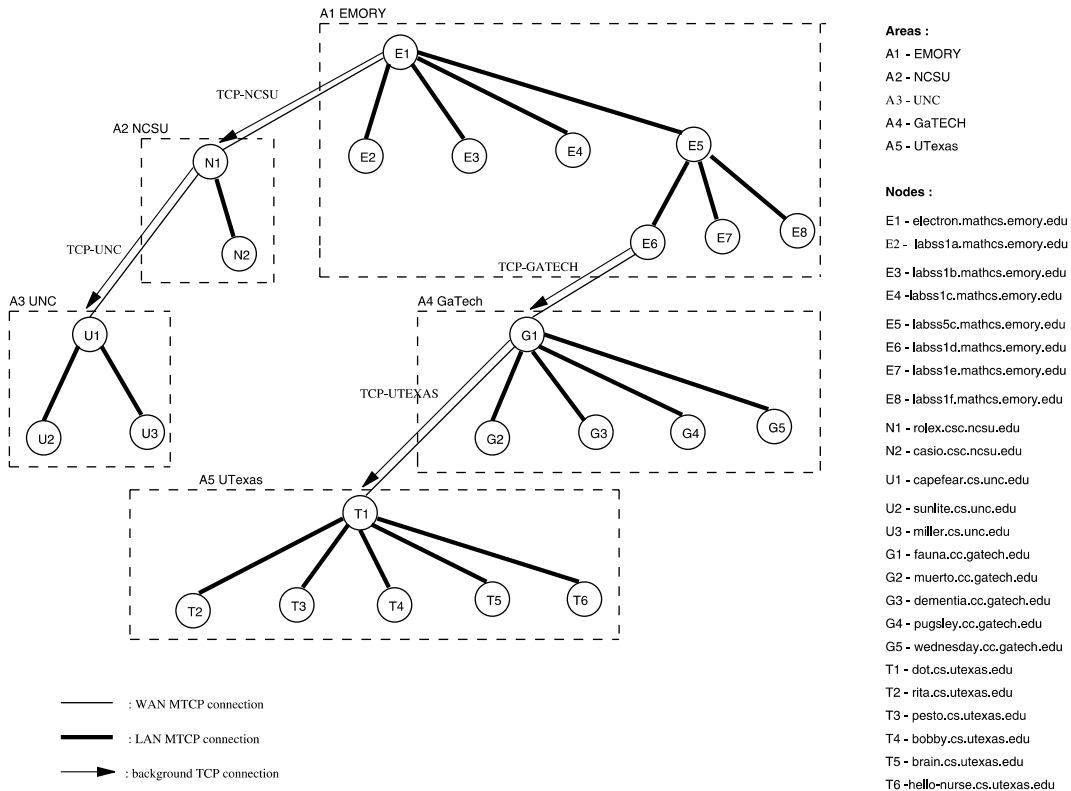


Fig. 5. Tree used for Internet experiments.

a 70 MB file to multiple destinations on the same LAN. The nodes were organized in a one-level tree rooted at the sender, with all receivers on the same level. We measured the throughput and CPU load at the sender as we increased the number of receivers. We limited the number of receivers to 16, since if each SA can accommodate 16 children, MTCP can support 69,904 receivers organized in a four-level tree. All the machines used in the experiment were Ultra-Sparc Model 250 attached to a 100 Mbps LAN.

Fig. 6 plots the throughput, the total transfer time, and the CPU time of the sender, against the number of receivers. The CPU time represents the amount of time that the CPU is used during the transfer of the file, while the total transfer time is the time it takes to transfer the file and includes the time spent by the sender waiting for ACKs. We observe that as the number of receivers increases, the throughput does decrease, but not signifi-

cantly. We also see that the CPU load (i.e., the CPU time as a fraction of total time) also decreases with the number of receivers. This behavior can be explained by observing that, as the number of receivers increases, the sender spends a greater amount of time waiting for ACKs, and thus total transfer time also increases. Our results indicate that even if the sender and the SAs have as many as 16 children, the processing of ACKs does not pose a problem. In view of the fact that the experiment was performed in a high-speed LAN (where the sender can transmit at a fast rate, and also receives ACKs at a fast rate), the number 16 appears to be a reasonable upper bound on the number of children each SA can have in the logical tree, suggesting that MTCP is suitable for large-scale implementation.

The purpose of our second experiment was to test whether the protocol can respond to congestion within a TCP time-scale, as well as to measure

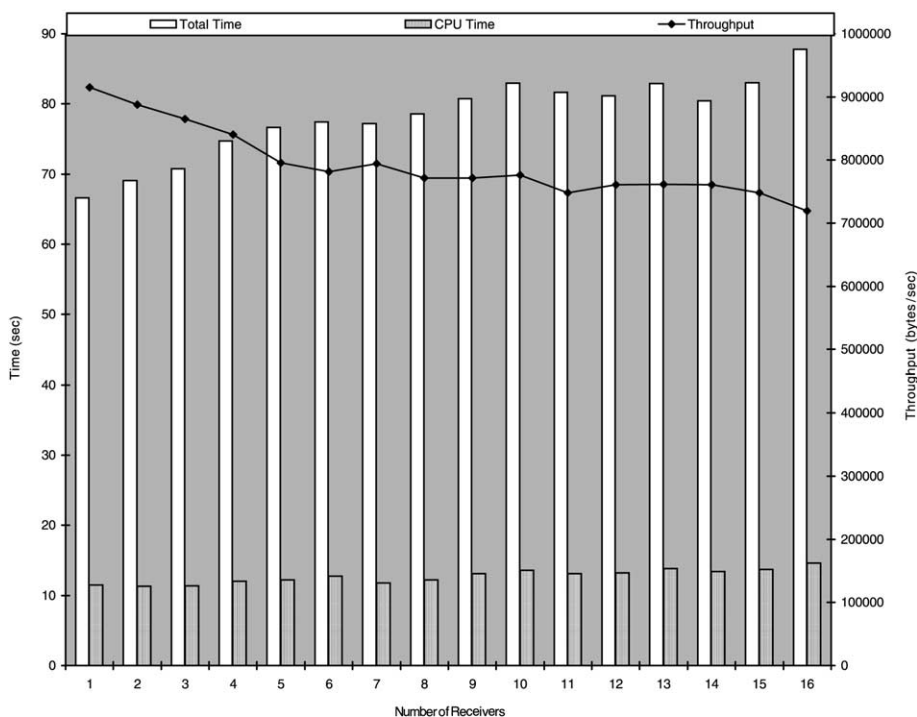


Fig. 6. One-level scalability test—LAN experiment.

the time delay involved in responding to congestion. To this end, we set up a four-level tree and examined how long it takes for the cwnd of the sender to be adjusted in response to changes in the cwnd of SAs in the path to the congested receiver. The tree involves one machine from each of the following sites: NCSU (the sender), Emory (the first SA, SA1), GaTech (the second SA, SA2), and UTexas (the leaf receiver). The experiment involved a source transmitting a 70 MB file to the three destinations. During the experiment we recorded the cwnd sizes at the sender and the SAs.

Fig. 7 shows a five second segment of the experiment. In this experiment we found UTexas to be the bottleneck, which caused SA2 (at GaTech) to have the smallest window size. The sender's window size is the largest. Recall that in MTCP, the sender regulates its transmission rate based on the minimum of all the reported congestion summaries and its own window. Let us call this minimum the transmission window. As we can see, the trans-

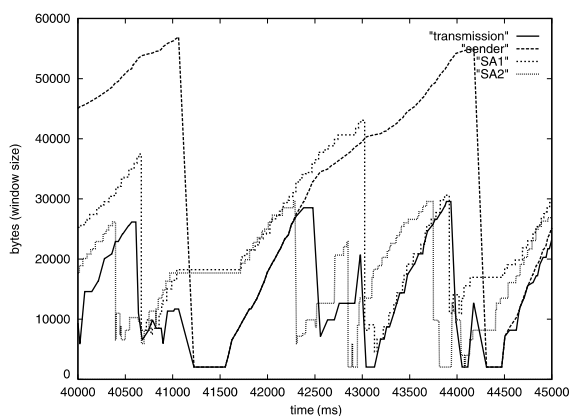


Fig. 7. Multilevel response time test—Internet experiment.

mission window closely follows the window of SA2. Furthermore, we observe that whenever any site runs into a slow start, the sender reduces the size of its transmission window drastically within about 200–250 ms. For example, in Fig. 7 we see that SA2 initiated a slow start at around 43 s, and

that about 250 ms later the transmission window also dropped to match the window of SA2.

4.1.2. Simulation

We used simulation to investigate the throughput behavior of MTCP on a larger scale. We simulated the network topology shown in Fig. 8, consisting of 101 nodes (one sender and 100 receivers) with each node having at most five children. The links connecting the nodes have a bandwidth of 1.5 Mbps and a delay of 10 ms. The queue size at the nodes was set to 8, and the DropTail queuing discipline was used.

We conducted two experiments to study the effect of the logical tree structure on the throughput of MTCP. In the first experiment, the nodes were organized in a one-level tree with the sender at the root and all receivers as children of the sender. With this arrangement, the sender has to process feedback from all receivers. In the second experiment, the nodes were organized in a perfect multilevel logical tree in which all non-leaf nodes have exactly five children (except, perhaps, the rightmost non-leaf node at the lowest level which may have fewer than five children). As a result, the load of processing feedback was evenly distributed among the SAs in the tree. During each experiment we recorded the throughput of MTCP as we increased the number of receivers from 1 to 100. The total simulation time for each run was 250 s.

Fig. 9 plots the data throughput (not including header and other control overhead) against the number of receivers. As we can see, when the nodes are arranged as a one-level tree, the throughput

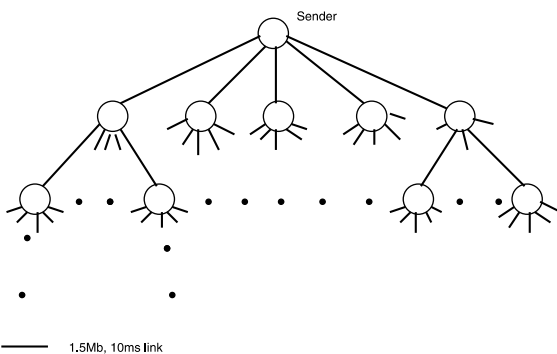


Fig. 8. Network topology for scalability tests—simulation.

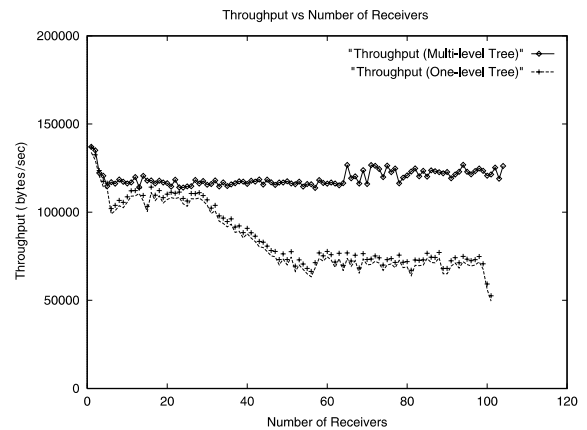


Fig. 9. One-level and multilevel scalability tests—simulation.

decreases drastically when the number of receivers increases beyond 25. This result is due to the significant increase in the load of the sender who has to process feedback from all receivers. On the other hand, when a multilevel tree is employed, the throughput does decrease when the number of receivers becomes greater than two, but not significantly. More importantly, the throughput remains at around 125 KBps (1 Mbps) as the number of receivers increases from 3 to 100, despite the corresponding increase in the height of the tree from one to three levels. These results indicate that the tree structure is the key to reducing the processing load at the sender, and that MTCP provides a high degree of scalability when a well-balanced tree is employed. We also note that MTCP achieves a data throughput of approximately 1 Mbps on 1.5 Mbps links, which demonstrates that MTCP is successful in capturing the available network bandwidth.

4.2. Inter-fairness

A protocol is said to be inter-fair if it uses no more bandwidth than a conforming TCP traffic would use on the same link. We have run both Internet experiments and simulation to investigate the inter-fairness properties of MTCP. The Internet experiments test whether MTCP can successfully co-exist with competing TCP traffic, while the simulations test the behavior of competing MTCP flows.

4.2.1. Internet experiments

We have conducted a large number of experiments over the part of the Internet shown in Fig. 5 in order to study the interaction between MTCP and TCP traffic under real-world scenarios. In Figs. 10–13 we show results from three different experiments. Each experiment involves independent TCP connections running over the WAN routes in the tree of Fig. 5. Recall that MTCP packets are routed over the WAN via UDP, thus, the TCP and MTCP traffic between the same sites in our experiments take the same WAN routes. Since WAN links are the ones most likely to be a bottleneck, this setup is appropriate for studying

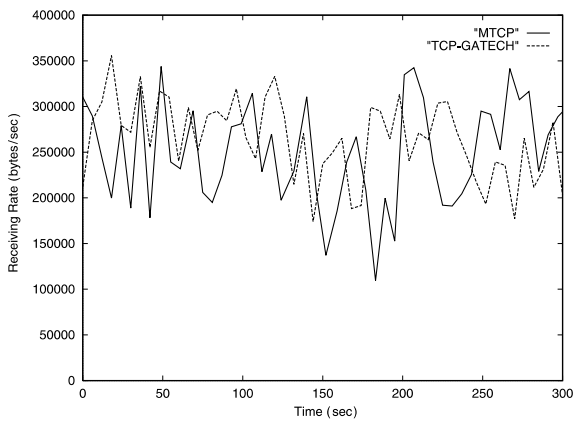


Fig. 10. Receiving rates averaged over 5-s intervals (first Internet experiment, areas A1 and A4).

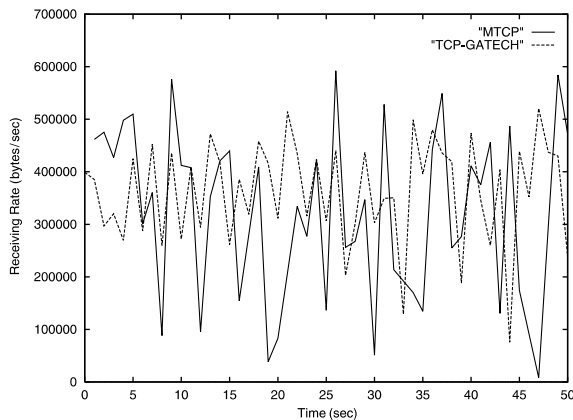


Fig. 11. Receiving rates recorded every second (first Internet experiment, areas A1 and A4).

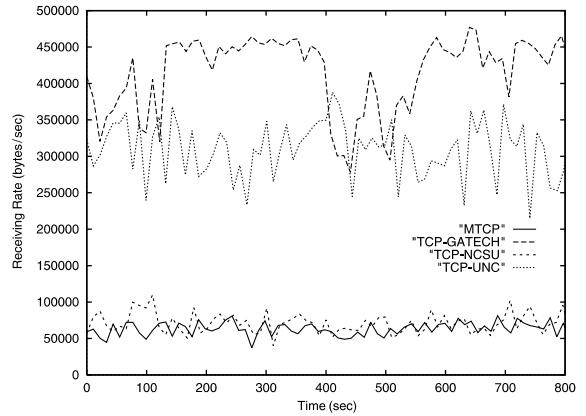


Fig. 12. Receiving rates averaged over 5-s intervals (second Internet experiment, areas A1, A2, A3 and A4).

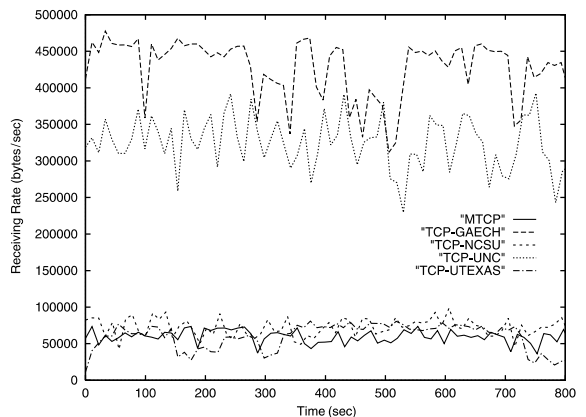


Fig. 13. Receiving rates averaged over 5-s intervals (third Internet experiment, areas A1, A2, A3, A4 and A5).

how the bandwidth of a link is shared between MTCP and TCP connections.

The first experiment involves areas A1 and A4 (refer to Fig. 5), the second experiment involves areas A1, A2, A3, and A4, and the third involves the entire tree. In these experiments, the MTCP sender and TCP senders transmit data as fast as it is allowed by their congestion control protocols. Each TCP sender starts transmitting at approximately the same time as the MTCP sender. We expect MTCP to match its sending rate to the minimum bandwidth available in the tree, therefore every MTCP receiver should receive at

approximately the same rate as the TCP receiver on the bottleneck connection in the tree.

The results of the first experiment are shown in Figs. 10 and 11. We run MTCP and TCP connections for 300 s and recorded the receiving rates of MTCP and TCP receivers. Fig. 10 shows the receiving rates of the MTCP and TCP and receivers averaged over 5-s intervals, while Fig. 11 shows the receiving rates recorded every second for the first 50 s of the experiment. It is evident from both graphs that MTCP and TCP share approximately the same bandwidth of about 280 KBps.

The receiving rates recorded during the second experiment are shown in Fig. 12; only the average rates over 5-s intervals are plotted in this case. From the figure, it is clear that the route from Emory to NCSU is the bottleneck because the TCP connection between these two sites gives the minimum receiving rates. MTCP matches the TCP receiving rate over this bottleneck route at around 70 KBps.

The results of the third experiment are shown in Fig. 13, where again we plot the receiving rates averaged over 5-s intervals. As we can see, the TCP connection between GaTech and Utxas has the minimum receiving rate of about 60 KBps, indicating that the route between these two sites is the bottleneck link in the whole tree. Again, we observe that MTCP is successful in matching its rate to the receiving rate of the TCP connection on the bottleneck link.

These three experiments indicate that MTCP uses no more bandwidth than a TCP connection uses on the bottleneck route of a given tree configuration. Although a bottleneck link may be located several levels away from the root, MTCP is capable of adjusting its rate according to the available bandwidth on that link. In all experiments, the fluctuation of MTCP's receiving rate is not perfectly synchronized with that of TCP's. This is because MTCP and TCP are not the same protocol, and the way that they detect congestion is different. In addition, MTCP reacts to every instance of congestion within a tree while TCP reacts to congestion only between two end points.

To study the performance of MTCP when sharing a link with multiple TCP connections, we

run a fourth experiment involving areas A1 and A4. In this experiment, while MTCP was transmitting, we run three TCP connections, all along the WAN route between Emory and GaTech, each of which is started at a different time. Specifically, the three connections TCP1, TCP2, and TCP3 were started at around 150, 300, and 410 s, respectively, after MTCP was started. All TCP connections were made between different host machines to eliminate the effect of computational overhead. We expect to see MTCP adjust its rate to match the current level of bandwidth available over the link between Emory and GaTech.

Fig. 14 shows the results of this experiment. When MTCP runs alone, its receiving rate reaches around 400 KBps. When TCP1 is added, MTCP reduces its rate from 400 to 300 KBps while TCP1 traffic slowly increases its rate to around 300 KBps. As soon as TCP2 is added, both TCP1 and MTCP reduce their rates. TCP1 goes down to 180 KBps while MTCP matches its rate with TCP2 around 240 KBps. When TCP3 is added, both MTCP and TCP2 reduce their rates slightly. MTCP still does not use more bandwidth than TCP2. As soon as TCP1 finishes its transmission, MTCP's rate bounces up to match that of TCP2. TCP3 also increases its rate. It appears that TCP3 always uses less bandwidth than TCP2. The difference is about 50 KBps. There could be a couple of reasons for this difference. First, although the two TCP connections use the same route, their end

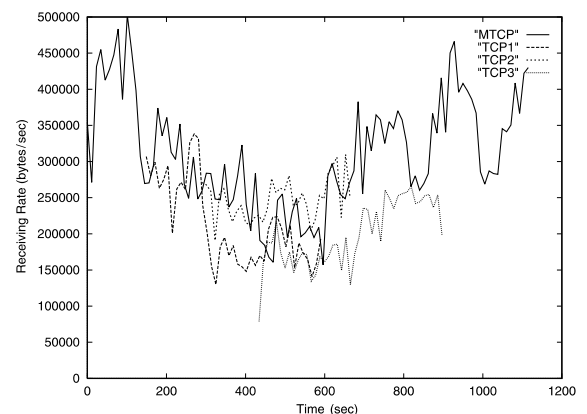


Fig. 14. Receiving rates averaged over 5-s intervals (fourth Internet experiment, areas A1 and A4).



points are different. So there could be other background job activities at the end points of TCP3 affecting its overall receiving rate. Second, TCP itself sometimes can be too conservative in its estimate of the available bandwidth. When TCP2 ends, both TCP3 and MTCP increase their rates quite a bit. MTCP settles at around 330 KBps while TCP3 goes up to 260 KBps. The difference is close to that between the receiving rates of TCP2 and TCP3. As soon as TCP3 ends, MTCP restores its rate quickly to 400 KBps. From this experiment, we observe that MTCP seems to adjust its rate as quickly as TCP, according to the current available bandwidth on the bottleneck link in a given tree.

#### 4.2.2. Simulation

To gain insight into the inter-fairness properties of multiple competing instances of MTCP, we have used ns to simulate a large number of network topologies. In this section we present results for the topology shown in Fig. 15, which is very similar to the one used in Ref. [16, Fig. 2]; results obtained from other topologies are very similar to the ones presented here. There are four senders and nine receivers in the network, and each MTCP instance involves one of the senders and all nine receivers. We run three different experiments. In experiment  $i$ ,  $i = 1, 2, 3$ , each sender node was involved in exactly  $i$  MTCP sessions, so that a total of  $4i$  distinct MTCP connections were simulta-

neously active in the network. The same logical multilevel tree was used for all instances of MTCP, with the sender at the root and each node in the tree having at most three children. The total simulation time for each experiment was 250 s.

In each experiment, we calculated the fairness index defined by Jain [17]. The fairness index is a quantitative measure of fairness for a resource allocation system, which is independent of the amount of resource being shared. The fairness index, based on throughput, for the bottleneck link is defined as

$$FI = \frac{\left[ \sum_{x=0}^{N-1} T(x) \right]^2}{N \sum_{x=0}^{N-1} T(x)^2} \tag{1}$$

where  $T(x)$  is the throughput of the  $x$ th protocol instance, and  $N$  is number of protocol instances sharing the resource. The fairness index always lies between 1 (indicating that all instances get an equal share of the link bandwidth) and  $1/N$  (when one of them gets all the bandwidth and all others starve).

Our results are presented in Table 1, where we show, for each experiment, the throughput of each MTCP session and the corresponding fairness index. The bottleneck link is the one between nodes 5 and 8 in Fig. 15, since it has a capacity of only 1

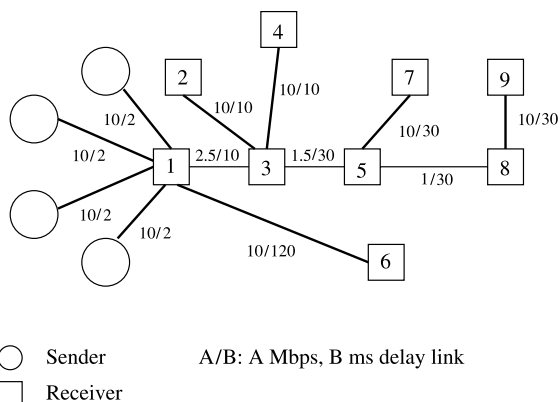


Fig. 15. Network topology for inter-fairness experiment—simulation.

Table 1  
Inter-fairness results—simulation

	Experiment 1	Experiment 2	Experiment 3
	4 MTCP sessions	8 MTCP sessions	12 MTCP sessions
MTCP throughput (bps)	122976.32 120163.84 121892.08 118946.12	71762.67 111270.22 119783.11 137866.67 130666.67 130311.11 145478.22 126960.00	65616.889 64457.778 96119.111 85045.333 92062.222 97333.333 68444.444 71111.111 83555.556 65777.778 97777.778 41452.764
FI	0.999836	0.970422	0.955753

Mbps. As we can see, the fairness index was always very close to 1 indicating that MTCP sessions fairly share the available bandwidth.

*4.3. Sensitivity to independent loss*

To study the impact of independent packet loss and random background TCP traffic on the throughput of MTCP, we have conducted simulation involving an MTCP session with 100 receivers. For the results presented in this section, we have used the network topology shown in Fig. 16, which is an extension of the topology used in Ref. [18]. This network was generated using Georgia Institute of Technology’s Internetwork Topology Models (GT-ITM) tool [19], which has been shown to

generate topologies that closely resemble the structure of the Internet. The links connecting the nodes were one of three types, as shown in the figure, the queuing discipline at the routers was DropTail, and the queue size was 20. Several TCP connections were running in the background between the various end-points indicated in Fig. 16. TCP connections were started at random times, and each lasted for only a short time period. The MTCP receivers were arranged in a logical tree as shown in Fig. 17, with each SA having at most five children.

We run two experiments, each involving a single MTCP connection and random TCP-Reno traffic in the background. In the first experiment, there were no losses at MTCP receivers. In the second experiment, with probability  $p = 0.01$ , a packet was

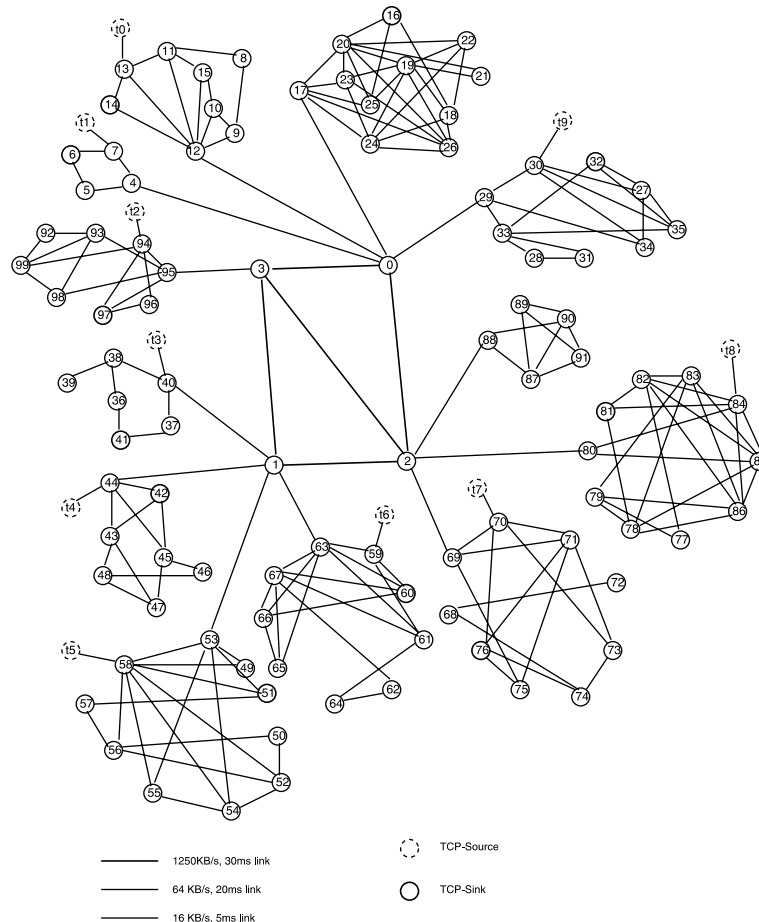


Fig. 16. Network topology for experiments with random TCP background traffic—simulation.

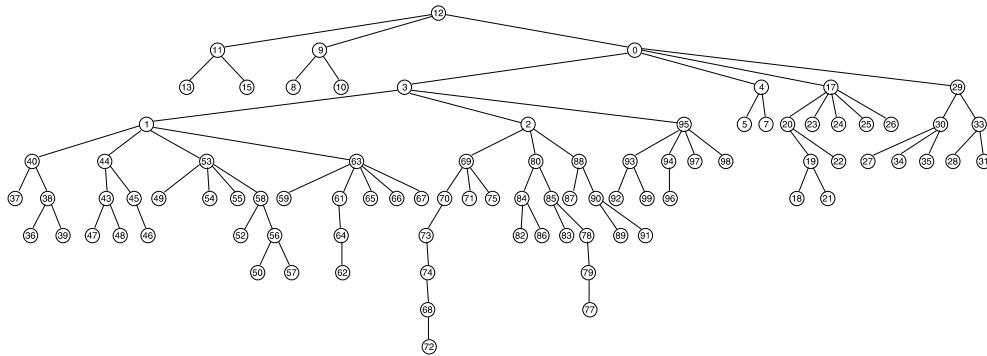


Fig. 17. Logical multicast tree for the topology of Fig. 16—simulation.

lost at a MTCP receiver, independently of other receivers and packets. During each experiment, which lasted 1500 s, we recorded the throughput at MTCP receivers and TCP sinks. These simulations help us evaluate the throughput of MTCP on a realistic topology under loss patterns that have been observed in real multicast environments.

Figs. 18 and 19 plot the throughput of MTCP and of the TCP background connections over time for the no-loss and 1%-loss experiments, respectively. In both cases, MTCP shows about 12 KBps throughput which was a little less than the bandwidth of the bottleneck links in the simulated network topology. When MTCP ran with 1% independent loss, the throughput of MTCP was slightly reduced, but not significantly. The impact of the background TCP traffic also seems very

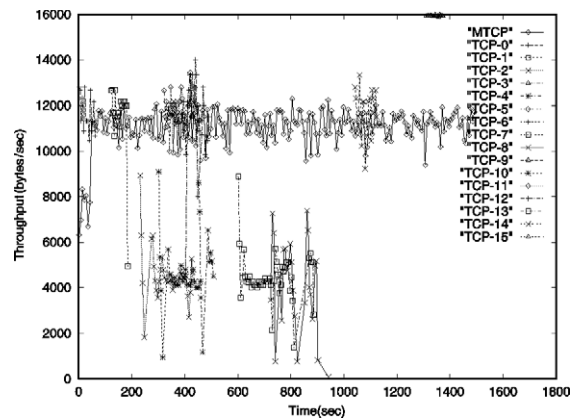


Fig. 19. Throughput of MTCP and of background TCP traffic—1% loss rate at MTCP receivers.

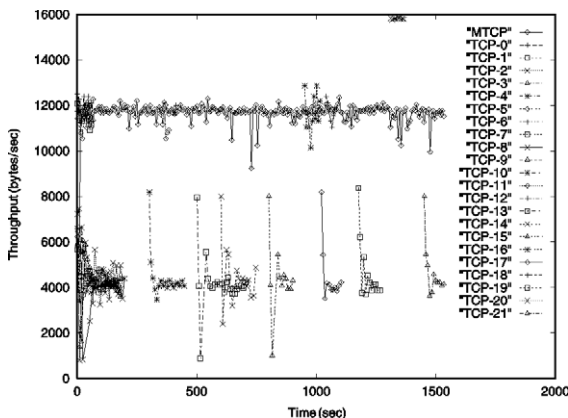


Fig. 18. Throughput of MTCP and of background TCP traffic—no loss.

marginal. The graphs show that MTCP does not reduce its rate because of independent uncorrelated losses or due to random TCP traffic. Comparing Figs. 18 and 19 we find that the receivers subject to 1% loss have throughput comparable to that of receivers not subject to loss.

### 5. Related work

Many reliable multicast protocols have been proposed in the literature [6–8,20–29]. For the purposes of our discussion, we classify these protocols into three broad categories: *unstructured*, *structured* and *hybrid*. We examine the protocols in each category with an emphasis on their congestion control techniques.

Unstructured protocols do not impose any structure among receivers, and Pingali et al. [30] further classify them into *sender-based* [20,22,26–29] and *receiver-based* protocols [24,25]. In sender-based protocols, every receiver sends ACKs or NACKs directly to the sender, and the sender retransmits lost packets reported in NACKs. The main problem with sender-based protocols is the *feedback implosion* problem: if many receivers send ACKs or NACKs to the sender at the same time, the sender may quickly become overloaded. This problem is especially severe when losses occur near the sender, in which case a large number of receivers will experience packet loss. In a system involving more than a few receivers, the load imposed by the storm of acknowledgments limits the function of the sender.

In a receiver-based protocol, each receiver multicasts NACKs to all members of the group, and any receiver that has received the requested packets multicasts them to the group. Typically, the protocols incorporate randomized NACK and retransmission suppression timers to reduce the number of duplicate NACKs and retransmissions. We find three main shortcomings with receiver-based protocols.

First, Yajnik et al. [31] report that most packet loss in the Internet MBONE occurs not at the backbone, but near end receivers, and that even excluding packet loss occurring near the sender, a small, but still significant, amount of loss (about 1–30%) involves more than two receivers. This study suggests that it is highly likely for two receivers not sharing common multicast routes to lose the same packets. A randomized NACK suppression technique may cause some uncorrelated NACKs to suppress correlated NACKs which actually report about a congested link. Since NACKs are multicast, the sender would get NACKs, but possibly from a different receiver each time: it may appear to the sender that NACKs are completely uncorrelated. Thus, the sender may not distinguish correlated packet losses from uncorrelated ones. It is unclear whether the sender should respond to all the NACKs by controlling its rate or ignore “seemingly” uncorrelated NACKs. Either approach seems unreasonable.

Second, in most receiver-based protocols that primarily use NACKs to detect congestion, the absence of NACKs is considered as no congestion or congestion clearance. Some Internet studies [31,32], however, reveal that almost every experiment trace includes one or more extremely long bursts of packet loss lasting from a few seconds up to a few minutes. During these bursts, no packets are received. As a result, receivers do not detect any packet loss, and do not send any NACKs. A similar scenario arises when the return path from receivers to the sender is congested, so that all feedback is lost. In either case, the sender would incorrectly translate the lack of feedback as no congestion.

Third, the randomized NACK suppression techniques employed by the receiver-based protocols require each receiver to estimate the RTT to every receiver in the group. This approach requires  $O(n^2)$  RTT estimations by every receiver, thus imposing limits on scalability. Grossglauser [33] proposed a distributed deterministic timeout estimation protocol that does not require global information. However the protocol assumes that the end-to-end delay variation is bounded and a priori known to all receivers.

Structured protocols impose a logical structure among group members. Two commonly studied structures are *rings* and *trees*. In ring protocols [23], a logical ring of group members is formed. Typically, a token is passed around the ring and only the process with the token may send feedback to the sender. RMP [23] supports TCP-like congestion control based on both ACKs and NACKs. However, since only the token holder can send an ACK, it is unclear how the ACKs are used for purposes of congestion control when there is a large number of nodes in the ring. In RMP, since NACKs are also multicast to suppress other NACKs, the protocol suffers from problems similar to those arising in receiver-based protocols.

In a tree protocol [5–9], a logical tree structure is imposed on the multicast group, with internal nodes acting as representative receivers for the group. While the sender multicasts data to the entire group, a receiver sends feedback only to its parent. The representatives buffer packets received from the sender, and retransmit any packets re-

ported lost by their children. Since the maximum degree of each node is fixed to a small constant, each node, including the sender, receives only a small amount of feedback within a RTT. In the following, we discuss the congestion control schemes of RMTP [6] and TMTP [8], since the LBRM [7], LGC [9] and LORAX [5] tree protocols do not incorporate (or do not give much detail about) a congestion control scheme.

The main problem with RMTP is that it does not provide end-to-end feedback. The sender only gets feedback from its own children (called designated receivers (DR)) about their receiving status. Hence, the sender has little information about the congestion status of leaf receivers. When congestion occurs at leaf receivers, it may not be possible for the sender to detect the congestion, especially if the DRs and the leaf receivers do not share the same network path. In this case, the sender will continue to transmit at the same rate, aggravating the existing congestion. As a result, RMTP traffic can be completely unresponsive to congestion and may cause congestion collapse.

TMTP also does not provide end-to-end feedback. This protocol relies on a back pressure effect caused by lack of buffers at representative nodes (called domain managers (DM) in TMTP terminology). In TMTP, DMs store the packets received from the sender until they receive ACKs for the packets from their children. When the buffers at a DM fill up because of congestion, the DM drops the next incoming packet. Its parent will continue to retransmit the packets not acknowledged by the DM until the parent's buffers also fill up. The sender detects congestion only when the buffers of all DMs between the sender and the congested nodes are completely full. So the congestion is completely neglected until the sender feels the pressure. Since each DM typically maintains a large number of buffers to reduce the number of ACKs returned to it, it could take a long time before the sender feels the pressure and reduces its rate. The fact that TMTP continues to transmit at a fixed rate despite the congestion is unfair to TCP-compatible flows which reduce their rates at the first indication of congestion.

Hybrid protocols [18,34] combine the packet recovery techniques used in structured and un-

structured protocols. As in receiver-based protocols, a receiver can multicast NACKs suppressing other NACKs, while other receivers may respond to the NACKs by retransmitting lost packets. In addition, a small number of representative receivers multicast their feedback immediately without any delay or suppression. The sender uses this feedback to control its transmission rate.

Delucia and Obraczka [18] proposed a hybrid congestion control technique in which the size of the representative set is fixed, but the actual nodes in the set change over time based on the congestion status of receivers. Assuming that a small set of bottleneck links always causes the majority of the congestion problem, the protocol solves the feedback implosion problem, as well as other problems associated with SRM [24] (such as the RTT estimation problem). The scalability and utility of the protocol highly depend on this basic assumption, namely, that the representative set is always small. This assumption may not be realistic, however, since several group members can be independently and simultaneously congested although they do not share the same congested links. No safeguard against this situation is provided.

Handley [34] also recently proposed a hybrid congestion control architecture. His technique works as follows. A small set of representative receivers is selected based on their loss characteristics, and each representative forms a subgroup along with receivers that share similar loss characteristics. For each subgroup, one relay receiver is chosen to receive data from the sender and play them out at a slower rate suitable for the receivers in the subgroup. The idea of representatives is similar to that in Ref. [18], but the subgroup idea is new and promising. However, the overhead, complexity, and efficacy of dynamic subgroup formations are not yet explored, justified or proven. In addition, since the group structure is essentially two level, it is not clear whether the protocol is scalable to very large numbers of receivers.

Other types of protocols that do not fall within the above categories include receiver-driven layered multicast protocols [16,35,36]. These protocols implement congestion control by encoding the transmitted data into multiple layers and

transmitting each layer to a different multicast group. By joining and leaving different multicast groups, each receiver can control its own receiving rate. Initially, the layering technique was proposed for continuous multimedia data streams which can tolerate some loss. Recently the technique was applied to a reliable bulk data multicast by Vicisano et al. [16]. However, the technique is applicable only when a large portion of the data is available for encoding prior to transmission, but not when data is generated in real-time such as during synchronous collaborative conferences.

## 6. Concluding remarks

We have presented MTCP, a set of congestion control mechanisms for tree-based reliable multicast protocols. MTCP was designed to effectively handle multiple instances of congestion occurring simultaneously at various parts of a multicast tree. We have implemented MTCP, and we have obtained encouraging results through Internet experiments and simulation. In particular, our results indicate that (1) MTCP can quickly respond to congestion anywhere in the tree, (2) MTCP is TCP-compatible, in the sense that MTCP flows fairly share the bandwidth among themselves and various TCP flows, (3) MTCP is not affected by independent loss, and (4) MTCP flow control scales well when an appropriate logical tree is employed. Thus, we believe that MTCP provides a viable solution to TCP-like congestion control for large-scale reliable multicast. We are currently working on designing and implementing a set of mechanisms for addressing the intra-fairness problem of reliable multicast protocols, i.e., to prevent a slow receiver from slowing down the whole group.

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