ABSTRACT
Future high-speed networks are expected to carry traffic with a wide range of performance requirements. We describe two queue service disciplines, rate-based scheduling and hierarchical round robin scheduling, that allow some connections to receive guaranteed rate and jitter performance, while others receive best effort service. Rate-based scheduling is designed for fast packet networks, while hierarchical round robin is an extension of round robin scheduling suitable for use in networks based on the Asynchronous Transfer Mode (ATM) being defined in CCITT. Both schemes are feasible at rates of one Gigabit/sec. The schemes allow strict bounds on the buffer space required for rate controlled connections and can provide efficient utilization of transmission bandwidth.

Rate Control
We propose to separate the virtual circuits or connections provided by the network into two categories: rate controlled connections and best effort connections. In the literature, the use of the term "rate control" is not standardized. We use the term "rate control" to mean controlling both the rate and jitter of a connection. At call setup time, a user requesting a rate controlled connection specifies a desired average service rate and jitter bound. If the call is accepted, the network provides guaranteed bandwidth and jitter bounds, and there may be mechanisms for renegotiation if the network cannot provide a suitable connection.

In integrated networks, the design problem is more difficult because different classes of traffic have dramatically different performance requirements, expressed in terms of the desired bandwidth, latency, and jitter. Real-time video requires high bandwidth, low latency and low jitter. Voice traffic requires low bandwidth, low latency and low jitter. Computer traffic spans a wide range of requirements, from applications requiring low latency for small transfers to those which are quite insensitive to the bandwidth, latency, and jitter achieved. In addition, the operating range of future networks will be very large: we expect trunk speeds from 64 Kb/sec to over 1 Gb/sec. The burden on the network designer is that where there is contention for limited transmission bandwidth or buffer space, control mechanisms must ensure that each traffic class meets its performance objectives.

As in [ZHA], the rate of a connection is defined over an averaging interval: the rate is determined by dividing the total number of bytes transmitted during an averaging interval by the averaging interval. The value of the averaging interval will determine the allowed burstiness at the source. If the traffic source uses a large averaging interval, the network must have sufficient buffer space at the first node to accommodate a burst.

Our notion of rate control also specifies a bound on the jitter introduced by a server. We first present an intuitive definition of jitter in order to contrast it with the definition that we will use. A server is assumed to introduce delay as packets pass through it: the delay is modeled as having a constant component and a variable component. The jitter introduced by a server is computed by measuring the packet departure time from the server and subtracting the constant component of the delay from the departure time for each packet to give the normalized departure time at the server. This definition is useful in analytic modeling of the server.
We propose an alternative definition of jitter that we believe is more relevant than the above definition in systems design. Jitter is defined to be a short term average rate, where the averaging interval used in computing the jitter is different from the averaging interval used in computing the rate of a connection. We refer to the two averaging intervals as the rate averaging interval, and the jitter averaging interval. Specifically, the jitter is the maximum number of packets transmitted by the server in the jitter averaging interval. This definition captures a useful property of our rate controlled servers, namely that they smooth the output stream. In principle, either averaging interval can range from the time to send one packet to the total connection time. In practice, the rate averaging interval might be on the order of several round trip times. The jitter averaging interval would be smaller and would depend on the size of the receive buffer at the destination.

Two alternative measures of quality for real-time streams have been suggested in recent work [ER], [LL]. These two alternative measures are the source-to-destination delay bound for a packet and the variance in inter-packet gaps observed at a receiver. In transporting real-time video or voice, bounds on absolute delay for a packet are not sufficient unless the jitter is also controlled. The absolute delay time for a packet in transit seems important mainly for applications such as real-time feedback and control systems. The variance in inter-packet gaps would affect quality only when the receiver equipment does not have the intelligence and the memory required to smooth out the variation in inter-packet gaps. In an era of inexpensive microprocessors and low cost memory devices, it seems likely that receiver equipment would be designed with the capability to smooth interpacket packet gaps for a burst of packets. The real factor affecting the cost would be the length of the burst that has to be stored before decoding the data, and it is this burst length which is captured in our definition of jitter. The distinction between our definition of rate control and these alternative definitions is important since we have found that neither the bounds on the delay in a network nor the variance in interpacket gaps observed at a receiver comes cheaply, especially for networks without a synchronous time base. However, we contend that controlling the jitter and rate can be achieved at an acceptable cost.

**Traffic Categories**

In the paper, we use the following classification of sources of packet traffic. A source that injects packets into the network at fixed time intervals is referred to as a Continuous Bit Rate (CBR) source. Uncompressed video or voice sources are CBR sources of packets. Sources which use compression or adaptive techniques generate packets at variable bit rates. We refer to these types of sources as Variable Bit Rate (VBR) sources. A simple model for a VBR source has an "on-off" traffic pattern: the source sends some number of packets at its peak rate for a period of time, and is quiet for a period of time. We expect that CBR and VBR sources would use rate-controlled connections. Sources with large amounts of data to transfer, such as file transfer protocols, can simulate either a CBR source or VBR source by using rate-based admission control. It should be noted that use of rate control within a network is distinct from the admission control scheme used. A rate-controlled network could support either rate-based or window-based admission control policies.

A source that sends short messages at widely spaced intervals, is referred to as an intermittent source. Distributed computations use interprocess control messages which are typically small and hence are examples of intermittent sources. For an intermittent source, the measure of importance is low end-to-end delay and small probability of data loss. We expect that intermittent sources would use the best effort service.

**Goals of Work**

The goal of the rate control proposed in this paper is to guarantee the rate and provide an upper bound on the jitter for the rate-controlled connections, while also providing the lowest possible delay and loss probability for best-effort connections. An equally important goal for the network provider is the efficient use of network resources: memory and bandwidth. In this paper we propose two implementations of the proposed rate control scheme. The first implementation is focused on fast packet networks, and the scheduling of packets for service can be achieved by a microprocessor at very high speeds using an algorithm known as rate-based scheduling. The second implementation focuses on networks based on Asynchronous Transfer Mode (ATM), where messages are broken up and carried in small, fixed length cells that are switched and multiplexed in a uniform manner [CCITT]. This implementation, known as Hierarchical Round Robin Scheduling (HRR), yields the desired performance guarantees by an enhancement of ordinary round robin scheduling.

**Previous Work**

In a packet switched network, the main point of control is the queuing discipline used at a switch. Two simple schemes used in current wide-area networks have been widely studied: these are First-Come-First-Served (FCFS) policy and Round-Robin (RR) service among active connections. Neither of these schemes differentiates between the quality of service desired by different connections. Variations of FCFS and RR policies have been proposed to guarantee quality of service, either by use in conjunction with rate-based admission control [LL],[SLCG] or by including priorities, either static or dynamic, in each packet or implicit with each connection. One notable variation is the Earliest-Due-Date (EDD) scheme [FV], which marks each packet with a due date and schedules the packet.
whose due date is next to expire. Although EDD achieves lowest aggregate delay for a network, the per-packet processing may be prohibitive for high-speed networks. None of the schemes that we know of provides the bounds on jitter.

Various recent proposals have addressed the performance objectives for best effort connections. One approach (Jacobson [JAC] and Ramakrishnan-Jain [RJ]) assumes that sources voluntarily reduce their window size when they sense or are informed of network congestion. These schemes do not provide guarantees against data loss, and depend on the sources willingness to cooperate. In another approach [HKM], the source transmission window is adapted dynamically and the network provides guaranteed buffer space equal to the window size for each connection at each queuing point. Any of these schemes can be supported within the context of the service disciplines discussed in this paper, which multiplex rate controlled and best effort traffic at the server.

The service disciplines proposed in this paper meet our goals for rate control. Moreover, our experience in implementation suggests that they are not subsignificantly more costly in memory or CPU requirements than FCFS or RR scheduling. One of the implementations is based on a novel queue structure used to hold waiting packets. The other implementation shows how one can easily modify the basic RR server to perform rate control. These two proposed implementations are equivalent in the sense that they both meet the requirements for a rate-controlled server, and provide a packet transmission order which is similar.

Rate-Based Scheduling

This algorithm, first proposed by one of the authors [KAN], uses a tree-like structure to store packets awaiting their turn. The structure is shown in Figure 1.

We can think of the transmission on the output link as occurring in frames or rounds of fixed size. Packets are queued in each round as a singly-linked list, and there is an array which holds pointers to the packets for each round. One of the rounds is the current round from which packets are removed and transmitted. When the server reaches the end of the list, it moves on to the next round. Thus, the service procedure is quite simple. What remains to be discussed is how packets are queued in rounds and how this achieves rate control. The following algorithm is used to queue arriving packets.

For each connection, a switch keeps four parameters defining a rate and jitter for the connection. These parameters are the service quantum, the last round number, the current count and the interval. The quantum defines the number of packets belonging to this connection that can be sent per round. The last round number indicates the round to which the previous packet from this connection was added. The current count is the number of packets of this connection added so far to the last round. The interval number is an integer which is added to the last round number when the current count equals the quantum. This gives the new last round number, and is used to skip rounds when the rate of the connection is less than 1 packet/round. With these parameters, the algorithm to identify the round for a received packet takes only a few lines of code. We note that the order of packet transmission for a connection is kept the same as the order in which packets arrived.

In order to control jitter, the rounds are all kept of fixed size (bytes). The rate-based server will not insert a packet into an earlier round (a round with round number less than the last round number for the connection) even if there is an earlier round that is not full. This limitation ensures that the jitter is no more than the service quantum over an interval defined as interval number times the round transmission time. In order to handle best effort traffic with minimum delay, packets belonging to best effort sources are queued separately. Whenever a round under transmission runs short of packets, then best effort packets are served in round robin order until the fixed round size is filled. One can reserve some fraction of the bandwidth in each round for best effort packets by restricting the sum of the quanta of currently active connections to be below a predefined fraction of each round.

The algorithm described above controls rate and jitter over a wide range of desired rates. An implementation of this in software was measured to run at about 200,000 packets/sec with variable length packets, when run on a 25-MHz MC68020 Microprocessor.

Hierarchical Round Robin Scheduling

We describe below another implementation achieving equivalent order of transmission of packets over a link. This implementation was designed for use in ATM networks. We start by describing an implementation of an ordinary round robin server for fixed-size cells.

Round Robin Server

It is well known that fairness requires some sort of round robin queue service discipline as opposed to traditional first-come-first-served queuing [MOR,
The HRR server has a hierarchy of service lists, each of which has a different frame length. The topmost list has the shortest frame length, and is used to serve connections that are allocated the highest service rate. We think of each list as having some number of time slots associated with it. We partition the bandwidth on the link between service lists by allocating some fraction of the slots associated with each list to lists that are lower in the hierarchy. When all the time slots allocated to a particular list are used, we start another round through its service list. At the bottom of the hierarchy, there is a round robin server that handles the best effort traffic.

Consider two lists of connections that require service, labeled 1 and 2. The level 1 list is used for high rate connections, while the level 2 list is used for lower rate connections. (For the moment, we will ignore best effort traffic). We can achieve different service rates by having different frame lengths for the two lists. Successive rounds through the level 1 list are to be started at fixed time intervals $FT_1$ (in units of cell service time), and successive service rounds through the level 2 list are to be started at fixed time intervals $FT_2$. Moreover, we interleave the service of level 1 and 2 as follows. After serving $b_1 < FT_1$ cells in level 1, we serve $FT_1 - b_1$ cells in level 2. After serving $b_1$ cells in level 1 again, we serve the next cells in round robin order in level 2, and so on.

The next refinement is to allow for multiple service quanta each time a connection is picked for service. Given a connection $j$, we define the service quantum $a_j$, which is the number of cells served from that connection whenever it is picked for service. Thus, if VCI 29 is given a service quantum of 3, every time that VCI appears in a round robin list, three cells are served. There is an obvious problem if this VCI is on a level other than the first. Since the service of that level may be widely separated in time, we will need to remember how many cells from a VCI have been serviced at each level, and when service at that level resumes, this has to be accounted for. Thus, when connection $j$ is picked for service, we load a counter, $G_j$, with the value of the service quantum $a_j$. Every time level $i$ services a cell, $G_i$ is decremented till the service quantum is exhausted.

The final refinement of the algorithm is to add a best effort server. If in serving CIDs from any service list we run out of CIDs before exhausting the frame, we fill out the rest of the frame with best effort traffic.

We are now ready to present the complete algorithm, in a form suitable for hardware realization.

**Notation**

We number the levels $1 \cdots N$, with the highest rate connections at level 1. At level $i$, we have $n_i$ slots. As before, one slot represents one cell served on the output link. We define a boundary $b_i = n_i$, such that $b_i$ slots are allocated to lists at lower (slower) levels. The remaining $n_i - b_i$ slots at level $i$ are either allocated to

**HRR Implementation**

We will describe the implementation as a series of successive refinements on a basic idea. We then present the exact details of the algorithm in a form suitable for hardware implementation.

Again, we think of the transmission on the output link as occurring in frames of fixed size. A frame can be measured in time slots where each slot corresponds to one cell served on the output link. We modify the round robin server such that it starts service through its list of CIDs once per frame. For example, if the service quantum for each connection were one cell, the server would serve each CID on its service list at a rate of one cell each frame. Once a particular connection is served, its CID is returned to the end of the list, but we must insure that this connection is not served again until the beginning of the next frame. To do this, we implement each service list as two lists: a current list from which CIDs are being served, and a next list to which CIDs are added. At the beginning of the frame, we swap the current and next lists.
1. Initialize follows:
   - The quantum for connection \( j \) is denoted by \( a_j \). Let \( \text{queued}_j \) denote the number of cells queued at a connection at any given time.
   - We shall refer to the HRR implementation as having multiple servers, one per level, where a server is either active or inactive.
   - A server at level \( i \) is active if servers \( 1, 2, \ldots, i-1 \) are active and have served \( n_i - b_i \) cells in each of their frames.
   - Server 1 is always active. (Figure 3 gives an example of a server at level 1 with \( n_1 \) equal to 10 slots and where \( b_1 = 3 \) slots have been allocated to lower levels.)

   The frame time, \( \text{FT}_i \), for level \( i \) is defined to be the time (in units of cell service time) between two services of the first slot in the frame. During each frame time, a server makes one round through its service list.

2. Detailed Description
   - Each server \( i \) has three counters called \( \text{NB}_i \), \( B_i \), and \( G_i \). \( \text{NB}_i \) determines how many cells are served from level \( i \), \( B_i \) determines how many cells are served from all levels lower than \( i \), and \( G_i \) keeps track of service quanta larger than 1. The algorithm proceeds as follows:
     1. Initialize:
        - At the beginning of a frame at level \( i \), \( \text{NB}_i \leftarrow n_i - b_i \), \( B_i \leftarrow b_i \), and swap(next\(_i\), current\(_i\)).
     2. Server and connection selection:
        - Let \( i \) be the index of the lowest rate active server.
        - If (current service list is empty and \( \text{NB}_i \) is non-zero)
          - activate BE server for one slot
        - else
          - server \( i \) picks connection \( j \) from the head of its current service list
          - If \( G_i \) is 0
            - \( G_i \leftarrow \min(a_j, \text{queued}_j) \),
            - serve connection \( j \) for one slot; decrement \( G_i \),
            - decrement \( \text{NB}_j \) and \( B_{j-1}, \ldots, B_1 \).
          - Adjust service list:
            - If \( j \)'s data queue is empty:
              - set \( G_i \leftarrow 0 \) and clear the round robin flag
              - bit for connection \( j \)
              - else if \( G_i \) is zero
                - server \( i \) places connection \( j \) at the tail of its next service list
              - else server \( i \) places connection \( j \) at the head of its current service list
     3. Adjust service list:
        - If \( j \)'s data queue is empty:
          - set \( G_i \leftarrow 0 \) and clear the round robin flag
          - bit for connection \( j \)
          - else if \( G_i \) is zero
            - server \( i \) places connection \( j \) at the tail of its next service list
          - else server \( i \) places connection \( j \) at the head of its current service list
     4. Check for change of active server:
        - If (any of \( B_{i-1}, \ldots, B_1 \) is 0)
          - server \( i \) becomes inactive.
        - else if \( \text{NB}_1 \) is zero and \( B_1 \) is non-zero
          - server \( i \) makes server \( i + 1 \) active
          - Go to step 2.

3. Example
   - An example will illustrate. Consider two levels with \( n_1 = 10 \) slots and \( n_2 = 10 \) slots. Also, \( b_1 = 1 \) slot, which means that the level 1 server allocates 1 time slot every 10 to the level 2 server. \( b_2 = 0 \). (At 45 Mbps with ATM 53 byte cells, these frame times support rates that are multiples of 4.5 Mbps and 450 Kbps, respectively.)
   - Suppose that CID#1 has reserved 9 Mbps, CID#2 has reserved 22.5 Mbps, CID#3 has reserved 450 Kbps, and CID#4 has reserved 900 Kbps. The allocations will look like the following:

<table>
<thead>
<tr>
<th>CID#</th>
<th>allocation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Level 1:</td>
<td>1 2</td>
</tr>
<tr>
<td></td>
<td>2 5</td>
</tr>
<tr>
<td>Level 2:</td>
<td>3 1</td>
</tr>
<tr>
<td></td>
<td>4 2</td>
</tr>
</tbody>
</table>

   The HRR algorithm does not determine the service order within the frame. That is determined by the position of the CID in the service list for a particular service. For example, the above case might result in the service order shown in Figure 4.

   The HRR algorithm is able to offer guaranteed bandwidth and an upper bound on jitter to rate controlled connections. The bandwidth received by a connection \( j \) at level \( i \) is \( a_j / \text{FT}_i \) cells/sec. The jitter bound is computed as follows. In the worst case, an HRR server could serve connection \( j \) for \( a_j \) cells at the end of one frame, following by another \( a_j \) cells at the beginning of the next frame. If we consider at the interval during which the cells are sent, we see that the jitter bound is \( 2 * a_j \) cells during one frame time at that level. Thus, the HRR algorithm bounds the size of the elasticity buffer needed at intermediate nodes and at the receiver as desired. We also offer a straightforward computation of the delay bound for an HRR server at an intermediate node. Consider a cell from connection \( j \) that arrives at the server just after \( j \) was served. This cell may not get sent out until the end of the next frame, so the delay is bounded by \( 2 * \text{FT}_i \), if connection \( j \) is at level \( i \). Thus, there is a fixed delay introduced by the server as a side-effect of metering the flow.
Step 1 of the algorithm states that counters $NB$ time and double the rate. The correct solution is to set to be served twice as often, which would halve the frame $NB$ occurs when all counters $B$ the example above with a microprocessor interface to allow flexibility in setting the chip to implement the counters would be designed with to control which round robin list is served next. The chip has been built which supports sixteen round robin engines that implement per-connection queueing for 32,000 CIDs with a memory bandwidth of 1.3 Gbps. A engine that implements per-connection queueing for one of the authors and R.C. Restrick have built a queueing engine that implements per-connection queueing for 32,000 CIDs with a memory bandwidth of 1.3 Gbps. A chip has been built which supports sixteen round robin lists of CIDs. Implementation of the HRR service discipline requires support for multiple frame counters to control which round robin list is served next. The chip to implement the counters would be designed with a microprocessor interface to allow flexibility in setting the parameters of the algorithm.

**Rate Partitioning**

Consider the hierarchical round robin server in the example above with $n_1 = 10$, $n_2 = 10$, $b_1 = 1$, $b_2 = 0$. These parameter values allow up to 9 connections sending at 10% of the total bandwidth, and up to 10 connections sending at 1% of the total bandwidth. Suppose the 11th connection sends a call setup message requesting rate controlled service at 1%. As things stand, we can either allocate more bandwidth to this connection than requested, or we can deny the call. As both are unacceptable, we present a scheme which allows the menu of rates to be changed. Suppose we simply set $b_2 = 2$. This would cause the server at level 2 to be served twice as often, which would halve the frame time and double the rate. The correct solution is to set $b_1 = 2$ and $n_2 = 20$ simultaneously. Intuitively, when we double the service rate, we must also double the work.

In general, multiplying $b_{i-1}$ by $x$ should be accompanied by a corresponding change in $n_i$. Note that these changes must be atomic across different level servers. In order to describe this, we offer an alternative definition of a frame. Consider a level $\alpha$. Step 1 of the algorithm states that counters $NB_{\alpha}$ and $B_{\alpha}$ are initialized at the beginning of a frame. This occurs when all counters $NB_{\alpha}$, $NB_{\alpha - 1}$, ..., $NB_1$, and $B_{\alpha}$, $B_{\alpha - 1}$, ..., $B_1$ go to zero. Thus, a frame at level $\alpha$ can be defined to be the interval between successive occurrences of these counters going to zero. A superframe is defined to be the frame for the lowest level server, thus repartitioning must occur at the beginning of the superframe. For example, if the lowest level server had a frame time of 6 ms (600 slots), then we could repartition the menu of rates at most every 6 ms.

If the ratio $n_i / b_i$ is an integer, it is possible to write a simple formula for the superframe time. Consider a two level server. The level 1 server offers $b_1 / n_1$ of the bandwidth to the level 2 server. The level 2 server must be served $n_2$ times for a complete frame. Thus, $FT_2 = n_1 n_2 / b_1$, and in general $FT_N = n_1 n_2 \cdots n_N / b_1 b_2 \cdots b_{N-1}$. Note that if $n_i / b_i$ is initially an integer, it will retain this property through successive partitionings.

Over time, the menu may become fragmented with most slots allocated to low rate connections. When this occurs, it will be necessary to do some repacking in order to be able to quickly handle a request for a high rate connection. This fragmentation and the need for repacking are analogous to the memory fragmentation that occurs in computer systems.

**Performance Management**

In order to establish a rate-controlled connection, the current rate parameters for each of the connections passing through each switch must be considered. In this paper, we do not consider issues pertaining to establishment and dynamic adaptation of rates based on network load. These issues are faced by any packet-switched network. For example, route selection could be done at the source or by control processors in the network based on knowledge of the previously allocated rates [FV].

**Simulation Experiments**

**Network Topology and Traffic Sources**

In order to demonstrate the behavior of the rate control schemes, we present simulation results of mixed traffic in a network with HRR nodes. We also present results for FCFS nodes for comparison. The network topology is shown in Figure 5. All links are the same speed, $C = 10,000$ bits/sec. The simulation uses two rate-controlled sources (Source 1 and 2) sending to a single destination, plus a Poisson source (Source 5) to represent the effect of cross traffic.

The source parameters are set up so that the link from node 4 to node 6 is the first bottleneck. Source 1 sends at $0.4^\circ C$, and source 2 sends at $0.1^\circ C$. The mean traffic intensity of the Poisson source is increased from 0 to $1.0^\circ C$. Thus, the simulation results will show the behavior of HRR as the links move into saturation. Note that as the intensity of the cross traffic increases beyond 0.5, queue lengths would be expected to increase without bound. We model a node as having finite buffers, thus packets are dropped from all sources under.
FCFS. Under HRR, only the packets from Source 5 are dropped due to the use of per-connection queueing. The run length for the simulation was 100 seconds of simulated time, which was sufficient to reach steady state.

The results are shown for an "on-off" rate controlled source. An "on-off" source is specified by its on time the time period for which it is on, num_pkts: the number of packets sent during that time, and its off time the time for which it is off. From these numbers, it is possible to compute peak and average bandwidths. All sources used a packet size = 200 bits. The two "on-off" sources used the following parameters:

<table>
<thead>
<tr>
<th>On</th>
<th>Num_pkts</th>
<th>Off</th>
<th>Ave Rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>1:</td>
<td>450 ms</td>
<td>20</td>
<td>550 ms</td>
</tr>
<tr>
<td>2:</td>
<td>1 s</td>
<td>10</td>
<td>1 s</td>
</tr>
</tbody>
</table>

The HRR server allocates five slots to the level 1 server \( N_1 = 5 \), with one of these slots allocated to the level 2 server \( B_1 = 1 \). The level 2 server has ten slots \( N_2 = 10 \), \( B_2 = 0 \). Source 1 is allocated two slots at level 1; source 2 is allocated four slots at level 2.

**Simulation Results**

Figures 6 and 7 show the interpacket gap in seconds seen at the receiver for "on-off" Sources 1 and 2, respectively. The x-axis represents the traffic intensity from source 5 as a fraction of \( C \). The solid lines show the results for HRR scheduling; the dotted lines show the results for FCFS. We observe first that the interpacket spacing for both constant rate sources under HRR is independent of the intensity of the cross traffic. The maximum gap for Source 1 is approximately 160 ms: this is approximately the worst case jitter of the level 1 server, which has a frame time of 100 ms. The maximum interpacket gap for Source 2 is 600 ms, which is less than the worst case possible for the level 2 server. The difference between the maximum and minimum interpacket gaps at the receiver is also less than that at the source, which shows the effect of smoothing. Both these results agree with expectations.

In the case of FCFS nodes, the average gap between Source 1 and Source 2 packets can be calculated. For example, with a cross traffic intensity of 1 (total intensity of 1.5), the average gap between Source 1 packets should be 0.075 seconds, which agrees with Figure 6. For FCFS, the average interpacket gaps will increase without bound as the load increases. When the cross traffic is zero, the maximum interpacket gap for source 1 using FCFS is 550 ms: this is the maximum interpacket gap of the source output stream. As the load approaches 0.5, the maximum interpacket spacing dips for Source 1. This might be attributable the fact that packets from Source 5 delay packets from Source 1 at the end of the on time somewhat, which would reduce the maximum gap. With cross traffic intensities above 0.5, the maximum gap increases without bound. Note that the minimum interpacket gap for source 2 using FCFS is reduced below that at the source. This is due to packets being sent out back-to-back due to queueing in the FCFS nodes.

Figure 8 shows the end-to-end delay in seconds for the rate controlled traffic. The end-to-end delays using
HRR correspond to the sum of the propagation and the fixed absolute delay of the server. This delay is greater for Source 2 than Source 1 since Source 2 is placed at a lower level in the server. Note, however, that the delay is independent of the load on the network. With the FCFS server, all traffic sees essentially zero delay when the network is lightly loaded, but as the load increases, the delay increases without bound. This suggests that for traffic sources where delay (not jitter) is important, that FCFS is preferred when the network is uncongested. The HRR server can provide hard delay and jitter bounds independent of the network congestion.

Thus, with HRR nodes both the interpacket spacing and end-to-end delay are independent of load. A crucial point to be made is that these results hold even for multiple nodes connected together in tandem. One additional observation for HRR nodes with "on-off" sources is that most of the buffering for these connections occurs at the first node. The buffer size at intermediate nodes never exceeds the expected bound on the buffer size.

Conclusions

This paper proposes that the connections provided by integrated networks be separated into two categories: rate-controlled and best effort connections. Using the rate-based service disciplines, rate-controlled traffic can be carried at a wide range of rates. Traffic is metered out during each frame time, which provides rate enforcement and allows the network to provide upper bounds on the jitter in the output stream. These results hold even in the presence of congestion or when the traffic flows through multiple nodes in tandem. Strict bounds can also be placed on the amount of buffer that is required at each queueing point. Idle bandwidth is utilized by best effort traffic, insuring efficient use of network resources. Both rate controlled servers can be efficiently implemented at rates up to 1 Gigabit/sec.

Future Work

We believe that the rate-based service disciplines provide a platform for several areas of future research. These include: design of algorithms for call set up and design of traffic management policies, selection of appropriate rate partitioning and compaction algorithms (for HRR), a broader simulation study of various aspects of the design, and implementation of the schemes in experimental networks.

Acknowledgments

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