Multimedia Traffic Management and Congestion Control in Satellite ATM Networks

by

S. Annukka Piironen

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Author..
Department of Electrical Engineering and Computer Science
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Certified by..
Dr. Steven G. Finn
Thesis Supervisor (Academic)

Certified by..
Anil K. Agarwal
Company Supervisor (COMSAT Laboratories)

Accepted by..
Prof. Frederic R. Morgenthaler, Chairman
Departmental Committee on Graduate Students

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Abstract

ATM networks provide service for a variety of traffic types (voice, video, and data) with different quality of service (QOS) requirements. When both constant and variable bit rate traffic sources are multiplexed together on common satellite and terrestrial links, effective proactive congestion control mechanisms are needed in order to satisfy QOS requirements and to maximize network throughput and utilization.

This work presents a study of a proactive, rate based congestion detection and notification algorithm. Four different traffic classes were considered together with two alternative rate control policies at traffic sources. Furthermore, an access policing and control mechanism, and two-priority queue management policies were incorporated into the network. The access policing consisted of leaky bucket queues and peak rate controllers.

The performance of the above mechanisms was studied via simulation. The proposed mechanisms satisfied a near zero cell loss QOS requirement for both real-time and nonreal-time traffic as well as a delay jitter requirement for voice and video. Also, the data throughput was maximized while achieving a link utilization of 80-90%.

Thesis Supervisors: Dr. Steven G. Finn
Department of Electrical Engineering

Anil K. Agarwal
Manager, Advanced Networking Department
COMSAT Laboratories

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Table of Contents

1. Introduction

1.1 Asynchronous Transfer Mode (ATM) Networks

Asynchronous transfer mode (ATM) is the intended transfer mode for Broadband ISDN, a digital network supporting various applications. ATM is a high-bandwidth, low-delay, packet-like switching and multiplexing technique. An ATM network is intended to provide service for multiple users via ATM switches (Figure 1).

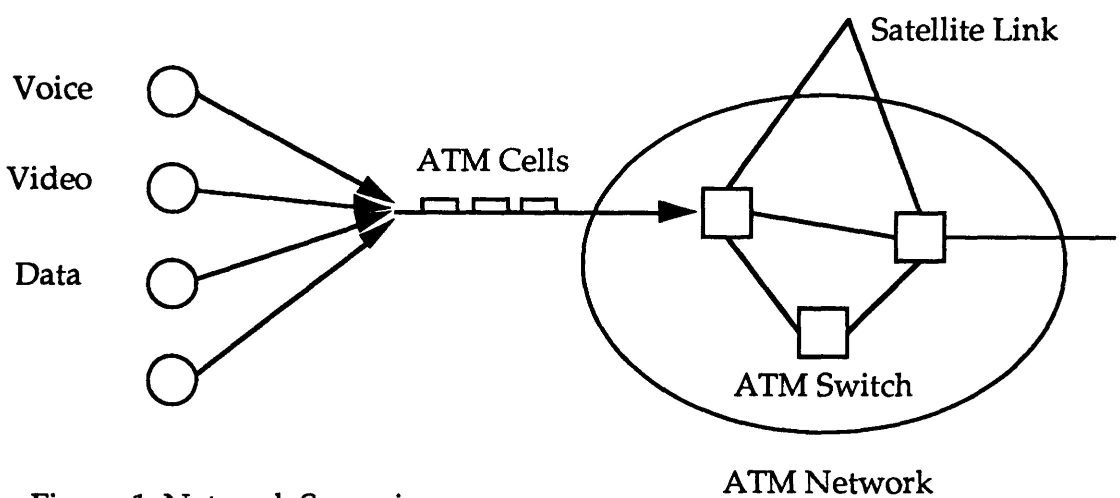

Figure 1. Network Scenario

In traditional circuit switching networks, once a connection is established, the required bandwidth is normally reserved for the entire duration of the session, regardless of the traffic pattern: a voice call typically consists of active periods and silence periods during which no traffic is sent, and the reserved bandwidth is wasted. In ATM networks, all traffic is carried in cells consisting of a 5-byte header and a 48-byte payload: no fixed amount of bandwidth is reserved for a session but rather ATM cells are sent only when there is traffic to be sent, for example during the active periods of a voice call, in other words, ATM cells are allocated to services on demand. This leads to more efficient use of the network resources, and also makes ATM particularly suitable for bursty traffic. The links in an ATM network include fiber optic links, as well as satellite links.

Commercial networks support many different types of traffic (Figure 2), which can be classified into two general categories, namely real-time traffic, such as voice and video, and nonreal-time data traffic. Voice and video traffic is very sensitive to delay as well as variations in delay, i.e. jitter. Data traffic is generally delay-insensitive and hence rate-flexible, meaning that the traffic can be shaped by reducing the transmission rate at the cost of an increased delay. The limit on jitter is considerably less stringent as compared to voice and video traffic. Voice traffic consists of bursts with a peak rate, resulting in an average rate that is considerably lower than the peak rate. Video traffic can be either variable or constant rate: variable rate video has a highly irregular traffic pattern that depends on the specific video application to be transmitted with rates ranging between a minimum and a maximum. The transmission rate for constant rate video is between 1.5 Mbps and 10 Mbps. Data traffic includes various kinds of traffic patterns resulting from different data applications such as e-mail, file transfers, and database accesses.

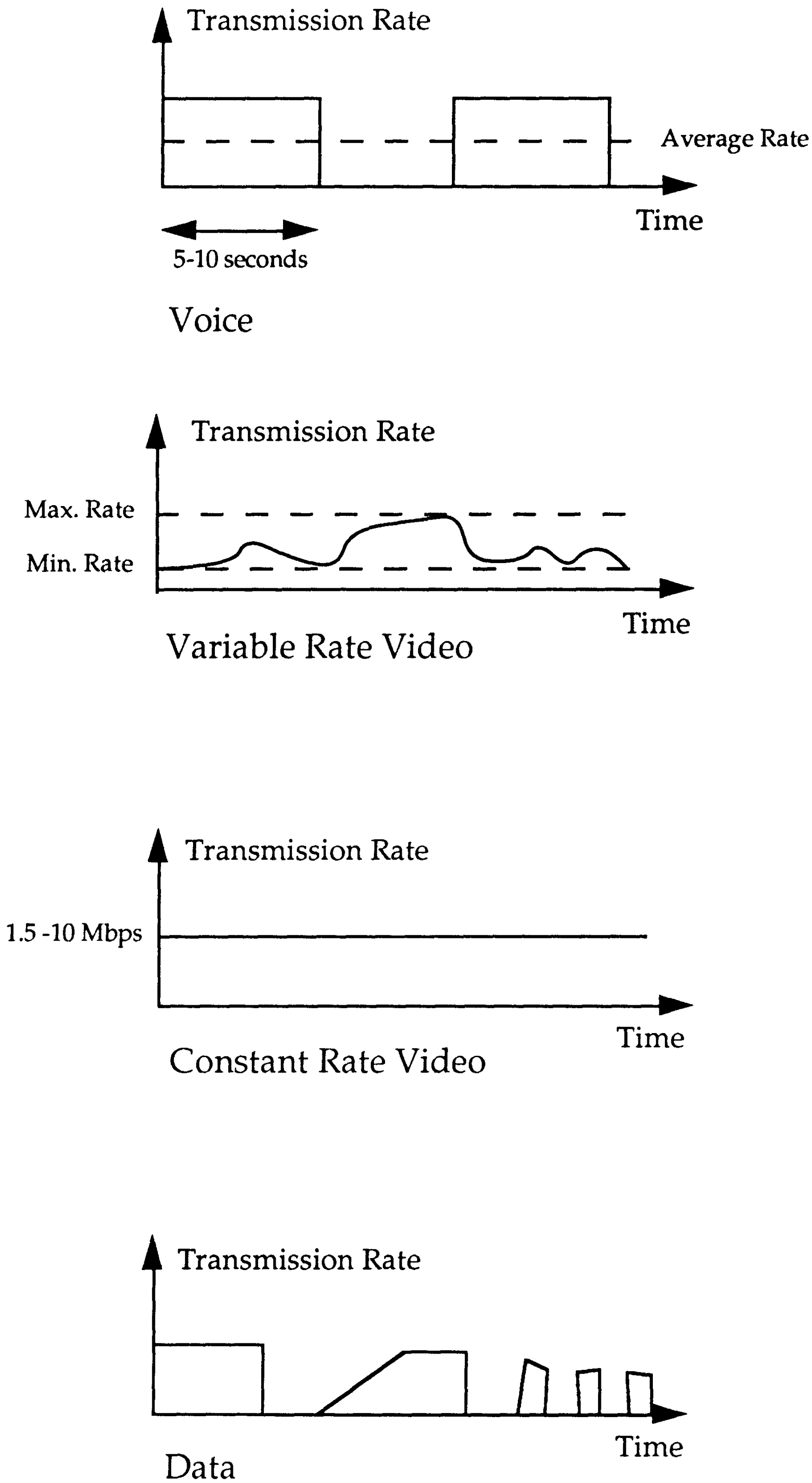

Figure 2. Traffic Types

ATM has several advantages. As discussed above, instead of allocating a fixed amount of bandwidth for the entire duration of a telephone call, in ATM, capacity is allocated on demand, as cells are only sent when there is traffic to be sent, and hence the bandwidth utilization is maximized. For new types of traffic, such as variable rate compressed video, and compressed voice, which also has a variable rate due to compression, definition and allocation of fixed bandwidth is not applicable due to the varying transmission rates. ATM provides a way of incorporating these types of traffic in a network.

1.2 Problem Definition

Given the above network scenario and the various types of traffic described, the question arises as to how to multiplex the different types of traffic on the common links without overloading the network, while at the same time providing the users with an acceptable quality of service (QOS). A simple solution would be to use deterministical multiplexing, namely to allocate bandwidth equal to the peak rate for all connections, in which case congestion would never occur in the network. However, this solution clearly does not make efficient use of the network resources as it underutilizes bandwidth. The bursty nature of traffic expected in the ATM network suggests the use of statistical multiplexing, according to which users are allocated bandwidth that is lower than the peak transmission rate of that particular connection. As different types of traffic are statistically multiplexed together, there is a finite probability that at some link, the sum of the peak rates of all connections on that link exceeds the total bandwidth of the link. If the situation persists, cells are discarded due to overflowing buffers inside the

ATM network. We define congestion on a link or network to be the state when cells are being discarded.

As discussed in 1.1, voice and video traffic is very sensitive to the delay jitter. When the buffers inside the ATM network overflow, the delay jitter experienced by the voice and video cells also increases, as cells experience relatively larger delays as compared to when the buffers are virtually empty.

There are two conflicting interests in determining how to multiplex different types of traffic in the same network. The network provider would like to send as much traffic as possible, as that translates directly to the profit of the network. On the other hand, the users' quality of service requirements must be met.

In traditional packet-switching networks, window-based congestion control mechanisms, relying heavily on end-to-end exchange of control messages, have been used. As the transmission speeds have increased to the Gigabit/sec ranges with the introduction of fiber optic links, the propagation delays have started to dominate over the other delay components, such as the switching and buffering delays: due to the large propagation delays, the congestion control mechanism in ATM networks cannot depend heavily on feedback information as the feedback cannot be updated fast enough to react to the congestion situation [1, 2]. The introduction of multiplexing different types of traffic with varying delay and cell loss requirements, as discussed above, has also created the need for new congestion control mechanisms to replace the traditional window-based schemes.

In this thesis, done at COMSAT Laboratories, the networks studied include satellite links, as the problem of introducing ATM on satellite links is important to COMSAT. Satellite links are characterized by large propagation delays, close to 270 milliseconds. This creates an additional constraint on possible feedback-based congestion control mechanisms, as discussed above. Therefore, proactive mechanisms are needed in a satellite environment to prevent congestion.

The problem of multiplexing different types of traffic in the network can be divided into two categories, namely call admission and congestion management. Call admission addresses the issue of accepting traffic into the network in such a way as to prevent the network from reaching a state of congestion. Congestion management, on the other hand, includes pro-active traffic enforcement and reactive congestion control. The function of traffic enforcement is to police the bit rates of the users such that they do not violate the contract made at the call setup time. Reactive congestion control deals with the issues of how to react when the network is already congested.

1.3 Motivation and Goal

As discussed earlier, the ATM network is intended to provide service for two main types of traffic, namely real-time voice and video traffic, and nonreal-time data traffic. The difficulty in multiplexing these different types of traffic on the same links arises from the fact that voice and video traffic, and data traffic have conflicting goals, which must be simultaneously satisfied.

Goals for Voice and Video:

- Low jitter
- Near zero cell loss
- Fairness
- Protection against badly behaving sources
- Variable, controllable rate video

For voice and video traffic, the delay jitter requirement is stringent, as discussed in 1.1. Similarly, a cell loss rate of as close to zero as possible is desirable. Fairness refers to the issue that cell loss should be biased towards the users that violate their contract in terms of the average bit rate. In other words, the cells from users that violate their contract should be preferentially discarded before the cells from other users. Similarly, the network needs mechanisms to protect it from users that violate their contract in terms of the average bit rate. Finally, the rates of voice and video should be controllable, i.e. they should be reduced if the network is approaching a state of congestion.

Goals for Data:

- Data should not affect voice and video
- High throughput
- Near zero cell loss
- Fairness
- Adaptive algorithm allowing overuse

Data traffic should not affect the quality of voice and video traffic, especially the delay jitter requirement. For data users, the throughput should be maximized. As for voice and video, the issues of near zero cell loss and fairness are also applicable to data. The data sources should be allowed to exceed their nominal bit rates to make effective use of the excess bandwidth available, in order to maximize link utilization. The allocation of the excess capacity should be in the ratio of the nominal capacities, and furthermore, cell discarding should be biased towards users that exceed their nominal bit rate.

In this thesis work, hybrid terrestrial and satellite ATM networks are considered: discarding cells in the case of congestion, namely buffer overflows inside the ATM switches, is not desirable, as the users' quality of service requirements are not met. Therefore, a cell loss ratio close to zero is desired. Furthermore, the performance of the network should be optimized from both the users' as well as the network's point of view: for users, this refers to maximizing the throughput, and satisfying the delay and jitter requirements of a specific traffic type, as discussed above, whereas for the network, the link utilization should be maximized. In order to achieve these performance goals, effective congestion management mechanisms are needed.

The goal of this thesis is to propose new congestion management mechanisms and evaluate via simulation the effect of these congestion management mechanisms on network throughput and utilization, and users' quality of service. An access policing and control mechanism together with transmit link queue management is proposed. The goal of the access policing is to protect the network from badly behaving sources, whereas the goal of the queue management is to promote fairness among the users by biasing cell discarding towards the users that violate their average transmission rate contract with the network.

As part of the congestion management, a proactive, rate-based source control scheme is proposed to allow nonreal-time data sources to increase their bit rates to effectively utilize most of the bandwidth available. It is crucial that the behaviour of the data sources does not adversely affect the quality of service requirements for voice and video.

Overall Goals for the Solution:

- High network utilization
- Low implementation complexity
- Standards compatible
- Work well over satellite links
- Algorithms and parameter values independent of network size and link types

In summary, the network utilization should be maximized as that relates to the profit of the network. A low implementation complexity is desired, as that reduces both the hardware and software costs. In addition, the solution should be standards compatible and take advantage of the network structure and the ATM cell structure. As discussed before, the solution should work well in a satellite environment, in which the propagation delays are significant. Finally, the algorithms and parameter values should be designed so that they are not dependent on the network size and the link types.

2. Call Admission

The function of call admission is to decide whether to accept new calls into the network depending on the availability of resources, namely capacity, at any given time.

The call admission is usually done at the access node at the call setup time. The user requests service from the network by stating certain traffic descriptors, which generally include parameters such as the peak and average bit rates, and the maximum burst length. The network determines, based on these parameters and the available network resources, whether the connection should be admitted into the network. The connection is accepted only if sufficient resources are available. The call admission algorithm is not considered in this thesis, but rather it is assumed to be already in place in the network.

3. Usage Parameter Control (UPC)

The function of the access policing and control mechanism or the usage parameter control (UPC) is to enforce the contract made between the user and the network by controlling the admission rate of traffic of a virtual connection (VC) into the network. The policing function studied in this project is included for each VC, and it consists of a peak rate controller with a queue, and a leaky bucket with an optional queue in front of it [2, 3, 4, 5] (Figure 3). The function of the peak rate controller is to ensure that the connection does not exceed its agreed peak bit rate, whereas the leaky bucket

[6, 7, 8, 9] polices the average bit rate as well as the burstiness of the connection.

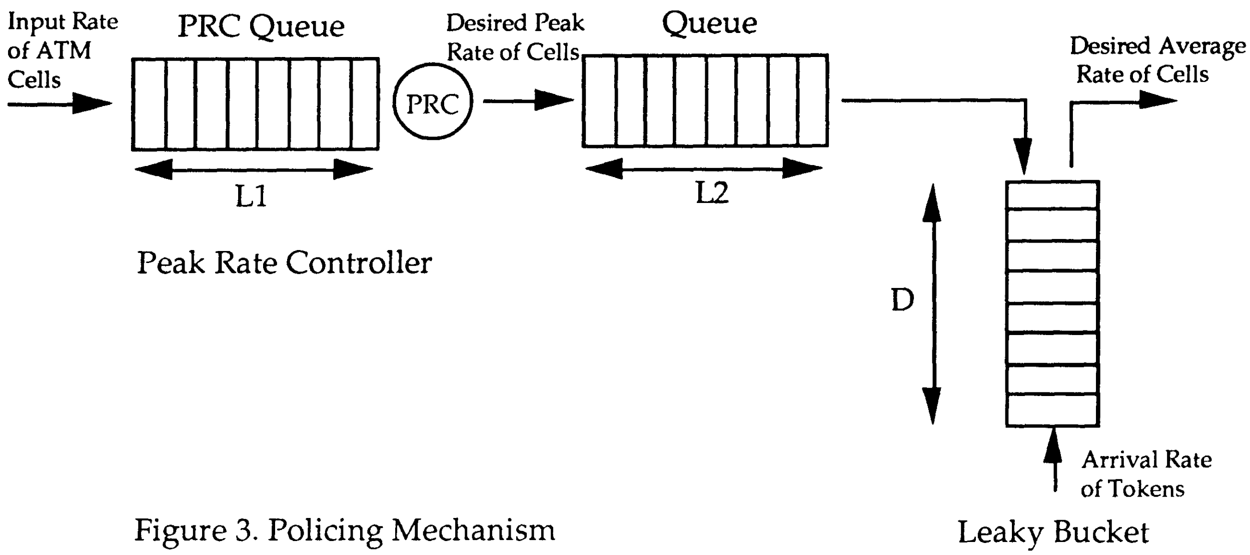

Figure 3. Policing Mechanism

3.1 Leaky Bucket

The purpose of the leaky bucket is to ensure that the sources do not exceed their agreed average bit rates. In the leaky bucket, tokens are generated at the "drain rate" which is equal to the connection's average bit rate or customarily at 110% of the average rate [10]. They are stored in a token pool, which has a maximum depth, D, and equal to the number of cells arriving in a burst in excess of the drain rate:

$D = (R_{arr} - R_{drain})$*average burst length/ATM cell size,

where

R_{arr} = arrival rate of traffic into the leaky bucket in bps

R_{drain} = drain rate in bps

The average burst length is in terms of seconds, and the cell size in bits. The token count is incremented every time a token is accepted to the pool, and tokens arriving to a full pool are discarded. When an ATM cell arrives to the leaky bucket, it is sent out if the token pool is not empty (i.e. the token count is not equal to zero), and the token count is decremented by one. If the token pool is empty, the ATM cell is queued in a first-in-first-out (FIFO) buffer to wait for a token. The queue has a maximum length, L_2, and a cell arriving to a full queue can be either

a) discarded

b) inserted into the queue, while a cell at the head of the queue is marked by setting the cell loss priority (CLP) bit in the ATM cell header, and sent out of the leaky bucket into the transmit link queue via the router in the ATM switch [3].

The implication of CLP=1 is to indicate that the respective cell should be dropped first at a later stage in the network if buffer overflows necessitate dropping of cells. This ensures that the virtual connections violating the contract are penalized more heavily in case the network becomes congested. By setting the maximum queue length $L_2=0$, the scenario described above reduces to a leaky bucket in which a cell arriving to an empty token pool is either discarded, or marked (CLP=1) and sent out.

The relevant parameters of the leaky bucket are the depth of the token pool, D, the generation or drain rate of tokens, and the maximum queue length, L_2.

3.2 Peak Rate Controller

The function of the peak rate controller is to enforce the peak bit rate of a connection, and at the same time smooth the traffic coming from the users.

As discussed above, the leaky bucket policing mechanism ensures that the average rate of a VC obeys the contract made at the call setup time. However, the mechanism allows users to transmit a burst of traffic at a much higher rate for a shorter duration as long as the average rate is still within the value agreed in the contract [8]. In other words, the peak rate of a VC cannot be policed by the leaky bucket mechanism discussed above. Multiple users transmitting at high peak rates could potentially cause the network to become congested, if several users are simultaneously transmitting on the same link such that at one time instant the sum of their peak rates exceeds the maximum bandwidth of the link. Additionally, by causing the queues in the ATM switches to fill up, users that exceed their peak rates for some period of time, increase the delay jitter of other users in the network. Therefore, the average rate as well as the peak rate of the connections must be enforced at the access node.

The peak rate enforcement can be accomplished by adding a peak rate controller (PRC) in front of the leaky bucket mechanism described above. The PRC includes a queue which operates on a first-in-first-out (FIFO) basis. The queue has a maximum length, L_1, and a voice or video cell arriving to a full queue is discarded, whereas for data, the cell at the head of the queue is marked by setting CLP=1, and the arriving cell is inserted at the tail of the queue: voice and video sources are not allowed to exceed their nominal peak bit rates, whereas data traffic sources are allowed to exceed their nominal transmission rates in order to effectively utilize all the available excess bandwidth, namely the bandwidth not utilized by voice and video at one time instant. The PRC schedules the departures of the cells from the queue such that the interdeparture times are above a minimum dictated by the inverse of the agreed peak rate in terms of cells/second.

The relevant parameters of the peak rate controller are the maximum size of the buffer in terms of the number of cells (L1), and the peak rate of the connection in terms of bps, which is used by the peak rate controller to determine the minimum interdeparture times between cells in seconds.

In previous work [8, 11, 12], a second leaky bucket to control the peak rate of the connection has been considered, but in this thesis project, a peak rate controller as described above is proposed in order to smooth the traffic coming from the users, thereby reducing the delay jitter of the ATM cells.

4. Queue Management

The function of the queue management is to provide fairness in the network, in terms of biasing cell discarding towards users that have violated their contract about the average and peak bit rates, and whose cells have been therefore marked as CLP=1 by the peak rate controllers and the leaky buckets.

After being admitted to the network, the ATM cells are queued at the output of ATM switches before being sent out on the trunks. One queue is included for each outgoing trunk. The queues operate on a first-in-first-out (FIFO) basis, and when the queue overflows, subsequently arriving cells will cause the queue to either discard all arriving cells until the queue length decreases below the maximum length, L, or to preferentially discard a CLP=1 cell, that is to say a cell belonging to a VC that violated its contract in terms of the average transmission rate. In the second approach, if the cell arriving to a full queue has CLP=0, a cell with CLP=1 must be searched from the queue in order to discard that cell to allow room for the arriving CLP=0 cell. Because of the necessity to search for a CLP=1 cell among all the cells already present in the queue, the second approach is computationally expensive, and therefore a faster implementation is needed to approximate the desired selective cell discarding behaviour.

Selective cell discarding [1] can be achieved by having a threshold L_o in addition to the maximum transmit link queue length, L, where $L>L_o$, and a count of the number of cells inserted into the queue, N. Upon a cell arrival, if the current cell count is less than the threshold ($N<L_o$), the arriving cell is inserted into the queue, and the cell count is incremented. Otherwise, the CLP bit in the cell header is checked: if CLP=1, the cell is discarded. If CLP=0 and the cell count is less than the maximum queue length ($N<L$), the cell is inserted into the queue, and the cell count is incremented by one. The queue operates on a first-in-first-out (FIFO) basis with a deterministic service time corresponding to the transmission time of the cell and the propagation delay of the outgoing link. Once a cell is sent out, the cell count, N, is decremented by one. Even though this approach approximates the second one discussed

above, the two are not identical: a CLP=0 cell arriving to a full queue is discarded in the approximate approach instead of searching and discarding a CLP=1 cell already present in the queue among the first L_0 cells.

4.1 Priority Scheme

According to the goals for data traffic, data should not affect the quality of service, namely the delay jitter of voice and video. The function of the priority scheme is to ensure that this goal is met by assigning voice and video traffic a higher priority as compared to data.

In order to achieve efficient use of network resources, namely bandwidth, users sending data traffic are allowed to increase their bit rates from the nominal value in order to utilize as much bandwidth as possible. Data traffic is delay-insensitive, and therefore, if one user sends data at a higher rate than the nominal rate, it is desirable and fair that only that particular data user's cells are delayed without affecting the quality of service of the real-time users. A scheme with two priorities can be used to achieve the desired behaviour: priority 1 is the low-priority traffic from the data users, which occasionally exceed their nominal transmission rates, whereas priority 0 traffic consists of the cells from the real-time voice and video sources that have stringent delay and delay jitter requirements. The priority 0 (real-time) traffic is always sent before the lower priority (data) traffic. The simplest way to implement this protocol is by having a separate transmit queue for each priority (Figure 4). The priority 0 queue is served until it becomes empty, and the priority 1 queue is served only when the priority 0 queue is empty. If a priority 0 cell arrives while the priority 1 queue is being served, the server starts serving the priority 0 queue after sending out the priority 1 cell currently being served. With the exception of the high-priority cell arriving to an empty priority 0 queue, while a priority 1 cell is being transmitted, the low priority traffic does not influence the delay performance of the priority 0 traffic, and therefore the delay performance of the high-priority traffic should be better in the case when priorities are supported, as compared to the situation of no priority scheme. Since the main idea behind the priority scheme is to delay only the non real-time data cells, the maximum length of the low priority queue can be made fairly large .

The important parameters for each of the two outgoing ATM transmit link queues include the maximum queue lengths, L^0 and L^1, and the queue thresholds, $L_o{}^0$ and $L_o{}^1$, for priorities 0 and 1 respectively. As the maximum queue length is increased, the delays experienced by the ATM cells are expected to increase. On the other hand, a smaller L is likely to cause more cells to be dropped due to buffer overflows.

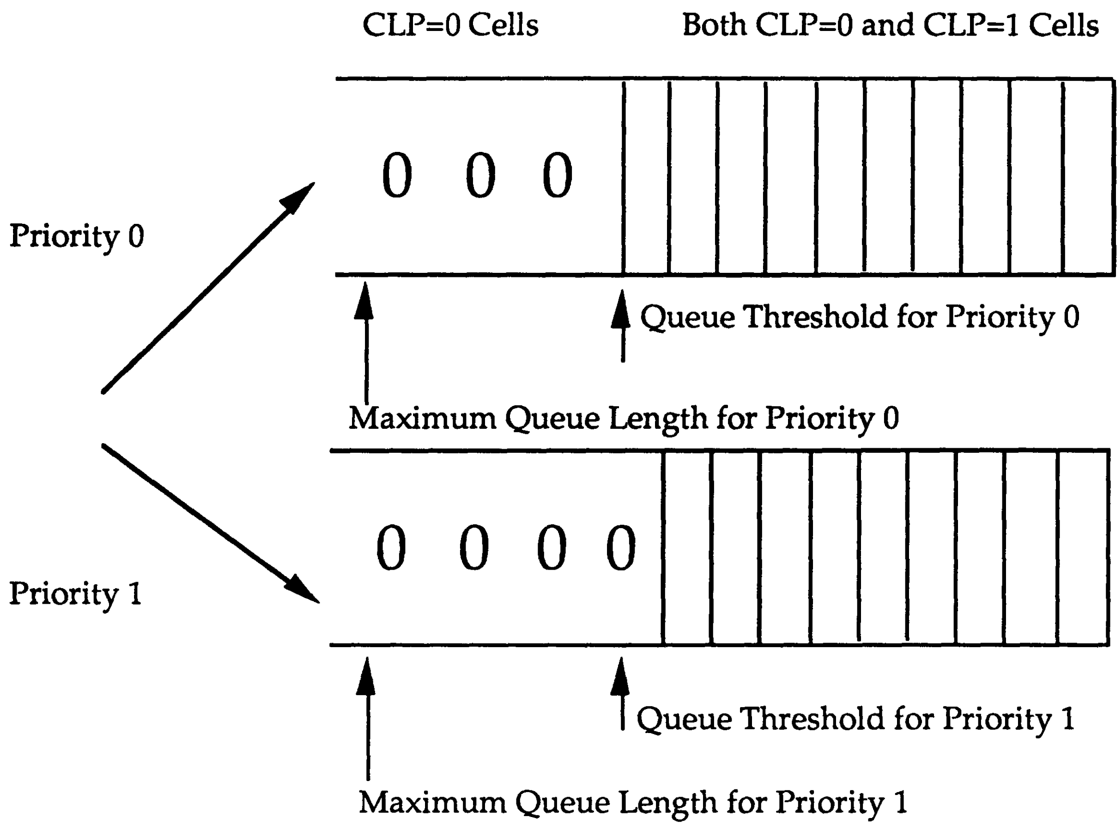

Figure 4. Outgoing Queue

5. Congestion Control

Due to the large propagation delays in a satellite environment, congestion control must act as to prevent congestion in a network, rather than to react to a congested network state. When the network is approaching a state of congestion, there is first of all a need for a detection mechanism to indicate the status of the network. Once the onset of congestion state is detected, the traffic sources, namely the users, must be informed that the network is approaching a congested state. As a last step of the preventive congestion control scheme, the traffic sources must react to the network status information delivered, by, for example, decreasing their transmission rates or completely stopping their transmission.

5.1 Rate-Based Congestion Detection Mechanism

As the links in the ATM network are approaching the state of congestion, the queue lengths on the outgoing links at the ATM switches increase and eventually the queues overflow and the link becomes congested. Therefore, approaching a congestion state can be detected by observing the queue lengths. A proactive congestion management mechanism can be triggered when the queues are, say, 90% occupied, and when no cells have yet been discarded due to queue overflows. This may start the management mechanism in time to react to the overload in the network. A simple congestion detection mechanism, implemented in most current ATM switches, considers the link to be congested when a cell is discarded at the outgoing transmit link queues.

In this thesis, we suggest a proactive congestion detection mechanism which is based on measuring the input rate of traffic to the transmit link queues. When the total input rate is 90% of the link rate, congestion notifications are sent to the users according to the following algorithm.

The traffic is divided into four different classes based on the priority and the CLP (cell loss priority) fields in the ATM cell header:

Class 1. Priority 0 (Real-time traffic), CLP=0

Class 2. Priority 1 (Data), CLP=0

Class 3. Priority 0 (Real-time traffic), CLP=1

Class 4. Priority 1 (Data), CLP=1

The highest class consists of priority 0 (higher priority) and CLP=0 cells, whereas the priority 1 and CLP=1 cells belong to the lowest class. The current input rate to the link is measured for each class by counting the number of bits (number of cells times the cell size of the particular class) arriving to the link queue over a defined measurement interval, T_m, divided by the link transmission rate. The link is regarded as congested when the total "filtered" input rate (defined below) of the link is greater than 0.9, and a congestion message is sent to the source of each arriving cell, if the following conditions are met:

> 1) the arriving cell belongs to one of the lowest classes, counting upwards from class 4, whose total input rate is at least 20 % of the link capacity, and
>
> 2) no congestion message has already been sent to that particular source within a time period T_m.

This ensures that the source does not receive congestion messages spaced out by less than the measurement interval, T_m, thus giving the source time to react to the first message.

The measurement interval T_m is taken to be 50 milliseconds, which represents a reasonable end-to-end delay in fiber optic terrestrial links of approximately 10,000 km. It is important to note that this congestion control algorithm is designed independently of the link type; even though the measurement interval is derived from terrestrial links, the algorithm is general and applicable for satellite links. If T_m was chosen based on a reasonable end-to-end delay on a satellite link, the resulting measurement frequency would be too low for terrestrial fiber optic links, but when T_m is based on the fiber optic end-to-end delay, the congestion detection algorithm not only applies to fiber optic links, but with proper rate control at the sources, the algorithm is also applicable for satellite and hybrid networks. This is the general goal of the congestion control algorithm presented in this thesis work.

A filter which takes a weighted average of the current calculated input rate value and the previous value is used to obtain a filtered input rate value. The filter values have been set at 0.8 for the current input rate values, and 0.2 for the previous ones. These values were chosen so that the current link input rate has more weight in the calculation, as it represents the actual condition of the link. The previous values are taken into account to filter out the noise and sudden temporal fluctuations in the input rate of traffic that may not represent a true state of congestion; however, the previous values should not have a significant weight in the calculation, as the congestion detection mechanism should respond fast to sudden true changes in the input rate of traffic, even after a period of low input rates. A filter value of 0.2 for the previous input rate values was chosen to meet these criteria. The input rates for each of the classes are calculated, and the total input rate is the sum of the input rates for the individual classes.

The main idea of the congestion detection scheme is to provide fairness among the four traffic classes: users that violated their contract in terms of the average and peak bit rates, and have their cells marked CLP=1, are always sent congestion messages first, as classes 3 and 4 are among the lowest classes whose input rate to the link queue is 20% of the link transmission rate. In other words, the traffic belonging to classes 3 and 4 should be penalized first, and if they are not utilizing 20% of the link capacity, the next class to be sent a congestion notification is class 2, low priority, nonreal-time traffic. Only in a case when the classes below class 1, namely classes 2, 3, and 4, are not utilizing 20% of the link capacity, is class 1, the highest class, sent a congestion notification. The order in which sources are penalized is from class 4 to class 1, meaning that the sources whose cells belong to class 4 are penalized first.

It is worthwhile to note that for the congestion detection mechanism described above, input rate measurements are only done for the four classes rather than for each virtual circuit. This is a significant reduction in computation as well as complexity of the algorithm.

In addition to the input rate -based congestion detection scheme, a different type of congestion message is sent to a source whose cell is dropped, provided that no congestion message was sent to that particular source within a time period equal to the measurement interval T_m. A simulation in which only the cell discard detection scheme is used serves as a benchmark in evaluating the performance of the congestion detection based on the filtered input rates of traffic to the link.

5.2 Congestion Information Delivery

Once the onset of congestion has been detected at the ATM switches, the information must be conveyed to the sources so that they can then react to bring the network traffic load to an acceptable level. Two mechanisms to accomplish the information delivery are the Explicit Backward Congestion Notification (EBCN) and the Explicit Forward Congestion Notification (EFCN) [13]. In both schemes, the ATM switches accomplish the congestion detection, but once the link is approaching a congested state, the congestion information delivery is done differently. In EFCN, the ATM switches mark the arriving cells by setting the forward congestion indicator (FCI) [14] bit in the ATM cell header. The destination node is then responsible for informing the sources of the congestion status based on the number of arriving cells with FCI=1, as compared to the total number of arriving cells. In EBCN, the ATM switches are responsible for congestion information delivery. EBCN results in a better congestion control mechanism, especially if congestion is detected at a node close to the source, and the propagation delays in the network are large, as in a satellite network. The sources are able to receive the congestion information in less time and therefore react to the situation faster than in the case of EFCN. In general, EBCN provides a more independent congestion information delivery mechanism, as the end-users are not required to know anything of the delivery mechanism and FCI setting, whereas in EFCN, all the end-users must have knowledge of the FCI setting policy, as well as when and how to send a congestion notification to the source. Moreover, if not all the end-users have the same information on setting the FCI bit and when to send a congestion notification, the entire mechanism is prone to failure.

As discussed in conjunction with the congestion detection mechanism, the congestion information delivery in this thesis is similar to EBCN. Based on the input rate measurement algorithm discussed above, the transmit link queues send a congestion message to the node's router, which then forwards it to the appropriate transmit link to be sent to the proper source. In the congestion notification scheme, all routed paths are assumed to be full duplex, so that the intermediate nodes between the source and the destination are able to send congestion notifications back to the source. The congestion message is an ATM cell with the same header format as all the other cells in the network: the cell is assigned priority 0 (same priority as real-time traffic), and the Type field is set to 0 for congestion messages based on the filtered input rates of traffic to the link, and 2 for messages when traffic cells are dropped.

It is worthwhile to note that this congestion notification algorithm takes advantage of the ATM cell standard by using the same ATM cell structure for congestion notifications. Furthermore, no special transmit link queues or queue management is included for the congestion notification ATM cells, but rather they are treated as regular high-priority (priority 0) cells. It is conceivable that the transmit link queues containing congestion notifications overflow, in which case the arriving congestion notifications are discarded. In that case, according to the congestion detection and notification algorithm, the source will have to wait for the next congestion notification, separated from the previous one by at least 50 milliseconds.

5.3 Rate Control Policy

When a congestion message has been delivered to a source, the source needs to react to bring the network traffic load down to an acceptable level. The rate control policy has two distinct phases, namely to first decrease the sending rate to alleviate the congestion situation, and second to increase the rate to achieve efficient use of network resources, after no congestion is detected. The simplest rate control mechanism is an on/off policy, according to which a source is simply turned off when a congestion control message is received. Even though this policy clearly brings the network traffic to acceptable levels, it is likely to underutilize network resources. According to another policy, a source decreases its transmission rate to some percentage of the current value [15]. Due to the propagation delays in the network, this policy gradually alleviates the network congestion. When no more congestion control messages are received at the source for some period of time, the source can gradually increase its sending rate: this is usually done by adding a constant amount, generally a network-wide constant, to the current value of the bit rate. The relevant parameters in the rate control policy include the percentage decrease, the constant increase, and the time intervals after which the increase and decrease will take place when congestion control messages have been received by the source.

In this thesis, sources reduce their bit rates by 20% of their current rate when a congestion control message based on the input rate of traffic to the link (Type field in the ATM cell header is 0) is received. When a congestion notification based on cell discarding (Type 2) is received, the sources reduce their bit rates by 50% of their current rate. Every time interval T_{rt}, the sources increase their bit rates by 2% of their nominal rate.

When a congestion notification is received, the sources that belong to the lowest classes whose total input rate is at least 20% of the link capacity, are slowed down by 20% of their current rate, thus reducing the overall link utilization by at least 4%: the total link utilization is therefore reduced from approximately 90% (necessary condition for congestion control, as discussed earlier) to 86%. The goal of the congestion control scheme is to keep the link utilization between 80 and 90 percent, a link utilization that is as high as possible without cells being discarded. To provide a certain quality of voice or video service once a connection has been established, the voice and video bit rates can only be reduced to some minimum rate due to congestion. The minimum acceptable bit rate is taken to be 50% of the nominal value. Due to their nonreal-time nature, the bit rates of the data sources can be reduced to zero, if necessary.

The sources increase their bit rates by 2% of the nominal value every time interval T_{rt} after a congestion notification has been received, and until further congestion messages requesting the rate to be reduced are received. T_{rt} is taken to be the roundtrip propagation delay from the source to the destination, and since the roundtrip propagation delay to nodes that are closer to the source is less than T_{rt}, the sources can receive further congestion notification from the ATM switches, after reducing their rates due to the previous message, in a time period less than T_{rt}.

The idea of the rate control algorithm is to decrease the bit rate by a relatively large amount (20%), when congestion occurs, and the rate should be slowly increased after no subsequent congestion notification messages are received. It is important to note that the constant by which the sources increase their bit rates after congestion is not a network-wide constant as in TCP for instance [16], but rather it is constant unique to each VC, as virtual circuits can have different nominal rates. Additional congestion control messages based on link utilization are ignored for the duration of T_{rt} following the congestion message.

When a congestion message based on a discarded cell (Type field is 2) is received, the sources are slowed down to 50% of their current value, or for voice and video, to their minimum value, as the rates cannot be reduced below the minimum value. A Type 2 congestion message arriving during the roundtrip time T_{rt} following a Type 0 congestion message is not ignored, as it is considered a more severe congestion indication. When a Type 2 notification is received, the rates are reduced by 50%, as opposed to 20%, as more aggressive congestion control is needed. The congestion detection and rate control algorithm based on the input rate of traffic to the link try to act to prevent congestion (cell discarding) from happening by requesting the sources to slow down as the path approaches a congested state, whereas the discard detection algorithm merely reacts to a situation in which the link is already congested.

It is important to notice that the rate control takes place at the traffic sources, where the traffic bit rates are decreased and increased according to the algorithm described above. The peak rate parameter of the peak rate controller (section 3.2), and the drain rate, derived from the average rate of traffic of a connection, of the leaky bucket (section 3.1), do not change according to the rate control algorithm, but rather they are set to the nominal values of each connection at the call setup time. The nominal peak bit rates and the resulting average values for each traffic type are discussed further in section 6.2.1.

6. Simulation Model

All simulations in this thesis are done by using the OPNET (Optimized Network Engineering Tools) simulation package by MIL 3, Inc. The network topology consists of a linear network of 2 ATM nodes and 2 users, one of which consists of multiple traffic sources, the other one having an equal number of sinks (Figure 5). The link in the middle of the network is a satellite link of 6 Mbps with a propagation delay of 270 milliseconds. The links connecting the users and the ATM switches are fiber optic links of 21 Mbps with a propagation delay of 5 milliseconds. All links are bidirectional to accommodate the congestion notification algorithm.

The link sizes are reduced approximately by a factor of eight for simulation purposes. The simulated time is significantly reduced, as less traffic is needed to saturate the link: reduction of overall traffic and traffic sources are translated into less simulation events, which in turn decreases the simulation time. It is important to simulate the network in conditions when the link is close to saturation, and it is expected that scaling the link pipe size down should not significantly affect the qualitative behaviour and effect of the proposed congestion management algorithms on the network performance. The non-scaled down satellite link has a capacity of 45 Mbps, whereas the terrestrial links have a capacity of 155 Mbps. As part of Suggestions for Future Work, the effect of scaling the link size on the results should be assessed further by running simulations with the true link sizes.

The network topology chosen is similar to previous work [17], as it represents one link in a more complex network. The reason for choosing this topology is that it provides a simple way to study the effects of the proposed congestion management mechanisms on the network performance, and the users' quality of service requirements. It is important to note that in designing the access policing, and the congestion detection, and rate control, nothing was assumed about the network topology, but rather the algorithms are general, independent of the network topology and link types. In this thesis work, the algorithms are tested for the linear network described above; as discussed further in conjunction with Suggestions for Future Work, the algorithms need to be tested for larger networks that include crosstraffic.

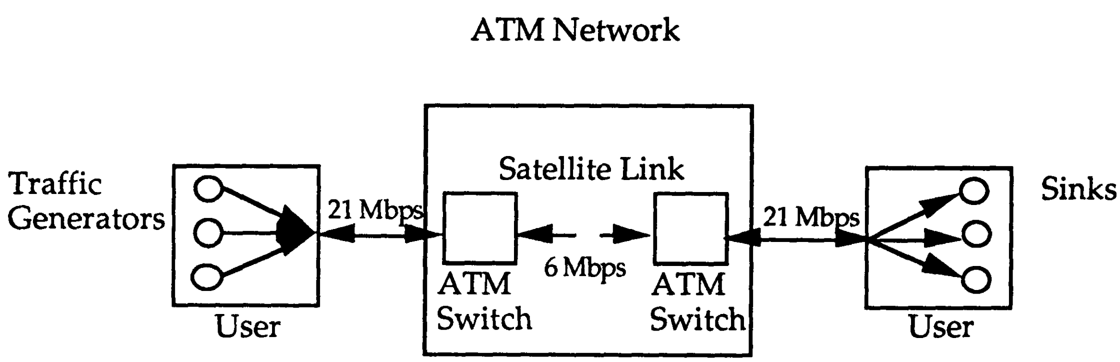

Figure 5. Network Topology

6.1 Modules in the ATM Node

The users and nodes have unique ID numbers, and each of the sources within a user are uniquely numbered (Figure 6). Link numbers are included for node-to-node and user-to-node links. A leaky bucket and a peak rate controller is included for each individual source, and the first number in the name refers to the user number, the second one being the source number within that user.

6.1.1 Leaky Bucket

The function of the leaky bucket is to enforce the average bit rate of a connection, while at the same time allowing some burstiness.

The leaky bucket, included for each source, consists of a first-in-first-out (FIFO) queue, into which the arriving cells are inserted. If the buffer is full, the arriving cell can either be marked (CLP=1), if cell loss priorities are supported, or it can be discarded. Cells are sent from the head of the queue and the bucket_count is decremented, as long as there are tokens in the bucket and the queue is not empty. The maximum queue length can be set to zero, in which case an arriving cell is either marked or dropped, if the bucket is empty; otherwise the cell is sent and the bucket_count is decremented. The bucket_count is initialized to the bucket_depth, which is the maximum allowed number of tokens in the bucket at one time, and the bucket_count is incremented at periodic intervals dictated by the drain rate of tokens, which is 110% of the average bit rate of the connection.

The parameters to be specified at the simulation time include the bucket_depth, D, the maximum queue length, L_2 >= 0, and the drain rate of tokens, R. Cell loss priorities are supported by setting clp_support=1. The leaky bucket module can be included in the ATM node by setting leaky_bucket_on to 1. Similarly, by setting leaky_bucket_on to 0, the cells arriving into the ATM node arrive directly at the router, as the leaky buckets are not included in the node.

6.1.2 Peak Rate Controller

The function of the peak rate controller is to enforce the peak bit rate of a virtual connection, while smoothing the traffic to improve the delay jitter.

The peak rate controller, included for each source, is a FIFO queue into which the arriving cells are inserted: for voice and video, cells arriving to a full queue are discarded, whereas for data, the cell at the head of the queue is sent out with CLP=1, and the arriving cell is inserted at the tail of the queue. A timer is started for the duration of the effective transmission time of the cell at the agreed peak rate of the connection whenever a cell is sent. The effective transmission time dictates the minimum interdeparture interval between cells: while the timer is on, the server is considered to be busy, and the next cell at the head of the queue must wait until the timer expires before it can be sent. This ensures that the peak rate of the connection is not exceeded.

The parameters to be specified at the simulation time include the peak bit rate of the connection, R_{peak}, and the maximum queue length, L_1. The peak rate controller can be turned on or off by setting prc_on to 1 or 0, respectively.

6.1.3 Router

The router performs the function of an ATM switch: arriving packets are routed to the respective transmission links based on the destination node number and the Type field included in the ATM cell header: congestion notifications are routed to the sources, whereas traffic cells are sent to their respective destinations. The routing table is a two-dimensional array which includes a row of transmission link numbers corresponding to each node. The router includes no processing, queueing, or delay: packets are routed instantaneously in the order of arrival. Priorities are taken into account at the transmit link.

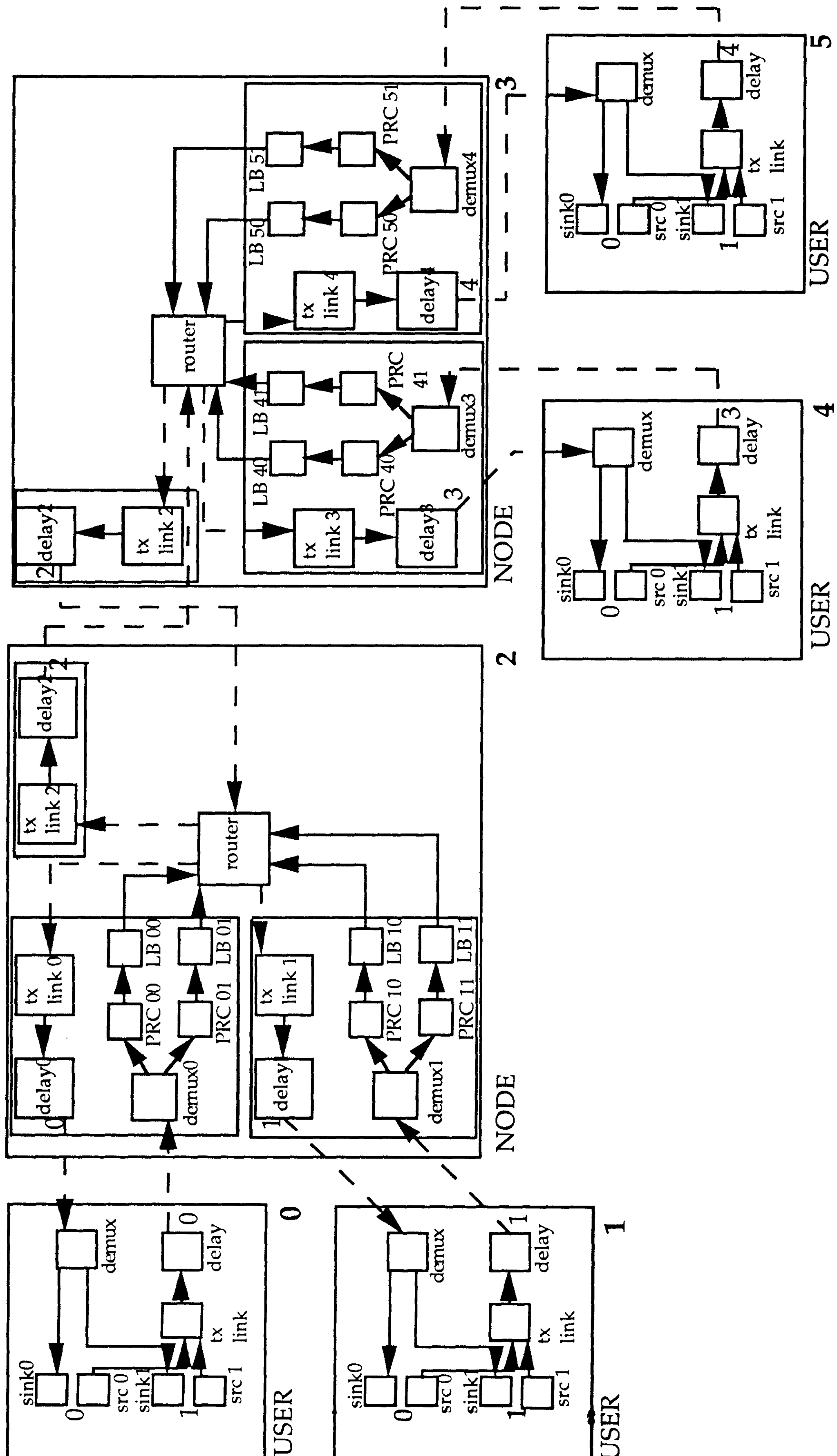

Figure 6. Simulation Network Modules

6.1.4 Transmit Link

The function of the transmit link is to model the link by sending out cells so that they are spaced out by an amount dictated by the transmission rate of the link. The transmit link ensures that the data traffic does not affect the delay jitter of voice and video traffic by maintaining queues for two different traffic priorities. Furthermore, cells marked CLP=1 are preferentially discarded when the queues overflow.

The transmit link module includes subqueues for two different priority classes (priorities 0 and 1 are provided to support real-time traffic (voice and video) and nonreal-time traffic (data)). Arriving cells are inserted into the respective subqueues based on the priority bit included in the ATM cell header while the current subqueue length is less than the subqueue threshold. If the subqueue threshold is reached or exceeded, the cell loss priority (CLP) bit in the ATM cell header is checked: if CLP=1, indicating that the particular connection exceeded its agreed average bit rate, the cell is discarded. Otherwise, the cell is inserted into the appropriate subqueue, provided that the subqueue is not full. Cells arriving to a full subqueue are discarded.

At the beginning of a cell transmission, if the subqueue 0 is not empty, a timer for priority 0 is started for the duration of the cell transmission at the link transmission rate. During this time, a server busy flag is set and no other cells are transmitted. When the timer expires, the cell from the head of the priority 0 subqueue is sent, and the server becomes idle. If at this point the subqueue 0 is not empty, the transmission of the cell at the head of the subqueue is started. Subqueue 1 is serviced only when the subqueue 0 is empty.

According to the reactive congestion control mechanism, the current input rate of traffic to the transmit link queue is measured for all the four traffic classes every 50 milliseconds. Congestion messages are sent to the sources of the arriving cells when the criteria (based either on input rates of traffic to the transmit link queue, or cell discarding, as stated earlier) for sending congestion notifications is met. When a congestion message is to be sent, the transmit link module creates an ATM cell and sets the Type field to 0 for input rate -based congestion messages, and 2 for cell discarding messages. The destination node and VC numbers are obtained from the traffic cell which caused a congestion notification. The ATM congestion management cell is then forwarded without any delay to the router, which sends the cell on the correct transmit link. The transmit link that initially creates and sends out the congestion notification keeps a table in a form of an array, which holds the time when the last congestion message was sent to a particular source. Only if the difference between the current time and the time in the table is greater than 50 milliseconds, will a congestion message be sent and its time recorded.

The parameters to be specified at the simulation time include the link transmission rate, the subqueue thresholds for the two priorities, and the maximum subqueue lengths. The input rate-based congestion control can be turned on and off by setting congestion_control_on to 1 or 0, respectively. Similarly, the simple congestion control scheme can be turned on and off by setting simple_congestion_control_on to 1 or 0.

6.1.5 Delay Link

This OPNET module models the propagation delay on the link. The arriving cells are inserted into a FIFO queue, and a timer is started for the duration of the propagation delay. During the timer, the server busy flag is set so that no other cells can be serviced simultaneously. This ensures that cells leave this queue module consecutively as they arrive from the transmit link. The timer for each cell is set to start from the time that particular cell is inserted into the queue, and since in OPNET a cell leaving from the transmit link arrives instantaneously at the delay link queue, the delay link module introduces no additional queueing delays to the overall cell delay. As the timer expires, the cell is sent out, and the server becomes idle and hence available to start a timer for the next cell in queue.

The parameters to be specified at simulation time include the link propagation delay. The maximum queue length is set to infinity because no cells should be dropped at the delay link, as it is merely a simulation representation of the propagation delay rather than a physical buffer in an ATM switch.

6.1.6 Demultiplexer at the Network Access

The cells arriving via the link from the user node are demultiplexed to the respective policing mechanisms according to the source VC number (essentially the source number) included in the ATM cell header.

6.2 Modules in the User Node

6.2.1 Modeling of Traffic Sources

In this thesis, three types of traffic sources are considered, namely voice, variable rate video, and data, each of which generate ATM cells. The traffic generators attach a predefined cell header to the cells, including the source and destination node and VC numbers (Figure 7). The header is not the standard ATM cell header but rather a simulation cell header. The CLP (cell loss priority) is originally set to 0, and the priorities are set to 0 for voice and video traffic, and 1 for data. The Type field in the ATM cell header is set to 1 for traffic cells and 0 or 2 for congestion notification cells.

Type (Data or Congestion Notification)
Source Node Number
Source VC Number
Destination Node Number
Destination VC Number
CLP (Cell loss priority)
Priority (0, 1)

Figure 7. ATM Cell Header

6.2.1.1 Voice

A voice session consists of ON and OFF periods. An ON period corresponds to an active period during which ATM cells are sent at a constant rate, normally 64 kbps, whereas an OFF period is the silence period (Figure 8). The lengths of the ON and OFF periods are taken to be exponentially distributed with a mean of 0.5 seconds. Therefore, the average rate for a voice session is 32 kbps. The duration of a voice session, as well as the time between separate voice sessions, are also modeled as exponentially distributed with a mean of 15 seconds. The durations for the average lengths of the ON and OFF periods, as well as the average durations of the sessions, are scaled down for simulation purposes. Realistic voice session durations are likely to be in the order of a few minutes. In order to run simulations with voice sessions of three minutes, the simulation times for this work would have to be increased by a factor of twelve.

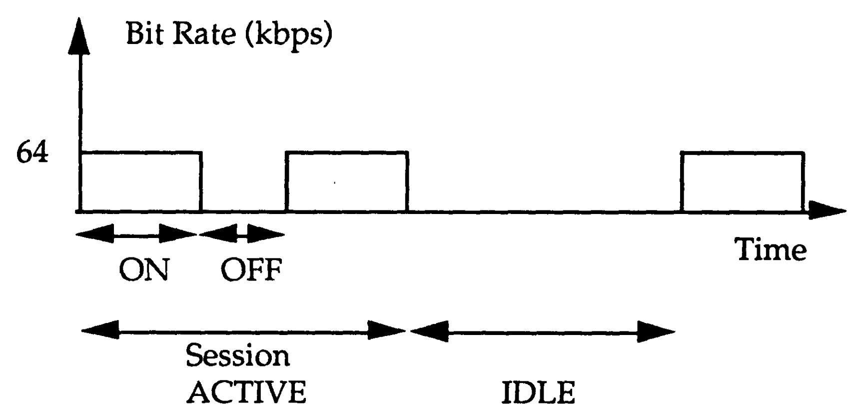

Figure 8. Voice Traffic Model

6.2.1.2. Variable Bit Rate Video

The bit rates of a variable rate video session generally range between a minimum and a maximum, and the traffic pattern is highly irregular depending on the specific video application. A video session is modeled to consist of periods of transmission at the higher bit rate (512 kbps) and periods of transmission at the lower rate (192 kbps), where these respective time lengths are exponentially distributed with a mean of 0.5 seconds (Figure 9). The average video rate for a single session is 352 kbps. The duration of a video session and the length of the time period between sessions, are also exponentially distributed with a mean of 15 seconds. As discussed in conjunction with the voice modeling, the video session average durations, and the lengths of the high and low periods are scaled down for simulation purposes.

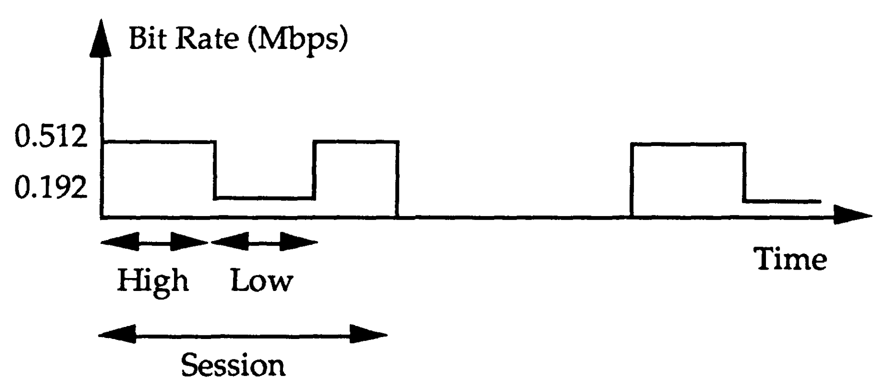

Figure 9. Variable Rate Video Traffic Model

6.2.1.3 Data (File Transfer)

When no form of congestion control is present in the network, files are transferred at a constant bit rate of 0.5 Mbps, and the file size is a constant of 1 Mbyte. The time between the starting times for two consecutive file transfers from the same source is exponentially distributed with a mean of 16 seconds; however, if the timer scheduled to mark the starting time of a new file transfer expires before the previous file has been sent, the timer is started again, and the next file transfer is started when the timer expires, after the current file has been sent. 16 seconds is the time it would take a source to send a file of 1 Mbyte if the bit rate was constant at 0.5 Mbps, the nominal value.

With congestion control present in the network, the file transfer bit rate for each individual file transfer session is allowed to increase by 2% of the nominal value every T_{rt} (Figure 10), while no congestion is detected. There is no maximum bit rate for file transfers, as this allows the data sources to make efficient use of the available excess bandwidth by utilizing as much of it as possible. The bit rates are decreased according to the congestion control algorithm.

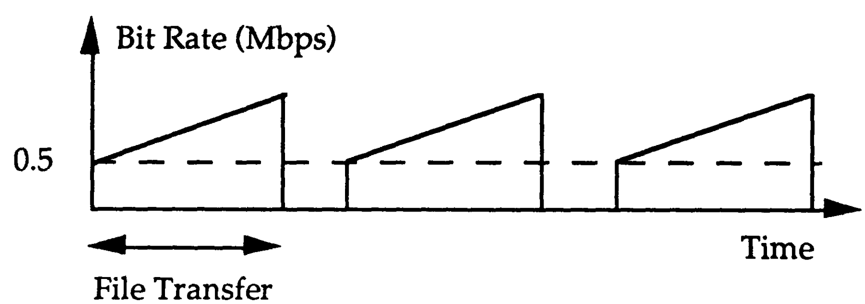

Figure 10. Data Traffic Model with Congestion Control

6.2.1.4 Slow Start Algorithm

A slow start algorithm [18] is present for voice, video, and data sources. When a session is initiated, the bit rate starts at 50% of its nominal value. It is increased by 10% of the nominal value every time interval equal to the roundtrip propagation delay from the source to the destination, until the nominal value is reached or until a congestion message is received. After the first congestion message is received or the source reaches its nominal value, the increase percentage is reduced to 2% of the nominal value. The purpose of the slow start mechanism is to smooth the traffic and eliminate sudden step increases in the bit rate from zero to some nominal value (slow start not shown in the figures).

As data traffic is delay-insensitive, the traffic can be smoothed by reducing the rate at the cost of an increased delay. This is done by reducing the input load to the network and hence the source's throughput. For voice and video traffic, the slow start algorithm implies a lower quality of service for a few seconds at the beginning of a session.

6.2.1.5 Startup of Sources

When a simulation is started, a Bernoulli probability distribution is used to determine which voice, video, and data sources start in an ACTIVE or IDLE state. The probability of starting in an ACTIVE state is determined by dividing the average length of the ACTIVE state duration by the sum of the average lengths of the ACTIVE and IDLE states. Because for our work the average durations of the ACTIVE and IDLE states are equal for the voice and video sources, the probability of starting in an ACTIVE state is 0.5. The same probability is applied for the data sources. The same method is also used for the video sources to determine whether the bit rate starts at the high or the low value when a session is started. Again, because the average lengths of the high and low periods are equal, the probability of starting at a high bit rate is 0.5.

The purpose of this startup algorithm is to try to eliminate possible startup effects in the simulation due to a large number of sources starting at the same time.

6.2.2 Sink

The final destination of a cell is represented by a sink, in which the total end-to-end delay of a cell from the time of its generation is recorded.

6.2.3 Transmit Link

The transmit link module includes subqueues for two different priority classes (priorities 0 and 1). The arriving cells are inserted into the respective subqueues based on the priority bit included in the ATM cell header. At the beginning of a cell transmission, if the subqueue 0 is not empty, a timer for priority 0 is started for the duration of the cell transmission at the link transmission rate. During this time, a server busy flag is set and no other cells are transmitted. When the timer expires, the cell from the head of the priority 0 subqueue is sent, and the server becomes idle. If at this point the subqueue

0 is not empty, the transmission of the cell at the head of the subqueue is started. The subqueue 1 is serviced only when the subqueue 0 is empty.

The parameters to be specified at the simulation time include the link transmission rate, and the maximum queue length. The maximum queue length should be large enough so that no cells arrive to a full buffer, and are therefore dropped at the transmit link buffer in the user node.

The delay link in the user is identical to that described above for the ATM node.

6.2.4 Demultiplexer

The Type 1 traffic cells arriving via the link to the user node are demultiplexed to the respective destination sinks according to the destination VC number (essentially the sink number) included in the ATM cell header. Similarly, the Type 0 or 2 congestion notification cells are sent to the respective traffic sources.

7. Simulation Scenarios

Three different traffic scenarios are simulated:

Scenario 1: only real-time traffic (video and voice sources)

Scenario 2: only non real-time traffic (data sources)

Scenario 3: mixture of real-time and non real-time traffic (video, voice, and data sources)

As the traffic mix for future ATM networks is not currently known, the above traffic mixes are chosen to represent reasonable traffic scenarios. For Scenario 1 video traffic is 80-85 % and voice 15-20 % of the total unconstrained traffic. For Scenario 3 the total of voice and video traffic is 60-70 %, and the data is around 30 % of the total unconstrained traffic. All the traffic scenarios are chosen such that at higher loads the input rate to the transmit link queue is greater than 90 % of the link rate, leading to an actual link utilization of 80-90 % of the link rate.

For each of the above traffic scenarios, three different congestion control cases are simulated, where the first case is the so called base case, as it includes no access policing or congestion control (Table 1). The traffic bit rates for each traffic types follow their nominal rates, as described in 6.2.1. The second case includes access policing (leaky buckets and peak rate controllers), CLP setting at the leaky buckets and the peak rate controllers with the proper queue management at the transmit link queues, and the complete congestion control with congestion detection based on the input rate of traffic to the transmit link queue. The more severe congestion messages, when a cell is discarded, are also included. The data rates are increased and decreased according to the algorithm described earlier. The third case is identical to the second one except that the congestion control is solely based on discarded cells.

	Case 1.	Case 2.	Case 3.
Leaky bucket	off	on	on
Peak rate controller	off	on	on
CLP setting	off	on	on
Priorities	on	on	on
Data rate	nominal	adaptive	adaptive
Congestion notification	off	link loading and cell discard	cell discard only
Slow Start	on	on	on

Table 1. Simulation Cases

7.1 Traffic Scenario 1 (Voice and Video) Output Measures

The load in the network is varied by adding five voice sources for each video source added. For voice and video, the cell loss ratio, defined as the number of cells dropped at the transmit link queue of the satellite link, divided by the total number of cells sent by the sources, is plotted against the total number of traffic sources and compared for the three simulation cases. Similarly, the percentage of the total number of cells sent, for which the difference between the end-to-end delay and the minimum end-to-end delay exceeds 20 milliseconds, is graphed against the number of sources. The minimum end-to-end delay for each virtual connection is defined as the shortest end-to-end delay measured for that particular VC during a simulation, and recorded by the sinks.

7.2 Traffic Scenario 2 (Data) Output Measures

The data load is varied by adding data sources. The cell loss ratio for data cells is graphed against the total number of traffic sources and compared for the three simulation cases. The congestion control algorithm studied in this thesis relieves congestion by decreasing the bit rates of the sources, and thereby decreasing the carried load and the throughput of data. Therefore, the throughput for data traffic is graphed against the number of sources. The throughput is defined as the number of bits received by the sinks over the entire simulation time, and normalized by the satellite link rate, 6 Mbps. Similarly, the average file transfer time, defined as the time needed by the source to generate a file transfer at the data bit rate, is graphed against the number of sources: as the data bit rate varies according to the congestion control mechanism, the average file transfer time is an indication of the average data bit rate achieved by a source, as the file size is held constant.

7.3 Traffic Scenario 3 (Voice, Video, and Data) Output Measures

The load in the network is varied by adding five voice sources for each video and data source added. The results collected are the same as for the voice and video, and data simulations described above.

7.4 Simulation Runs

Each simulation run, defined by three parameters, namely traffic scenario, simulation case, and number of traffic sources, is completed for three different random seeds. The simulated time is taken to be three minutes for each run, as this corresponds to an average of twelve voice and video ON and OFF session combinations, where the average duration of a session is 15 seconds, as described in 6.2.1.1 and 6.2.1.2. The average time for a source to send out a file of 1 Mbyte at the nominal rate of 0.5 Mbps is also around 16 seconds, thus leading to an average of ten file transfers during the simulated time. Three minutes of simulated time also represents approximately 320 roundtrip propagation delays in the network described in Chapter 6.

7.5 Simulation Parameters

In selecting the queue sizes of the various queues in the simulation network, an important tradeoff needs to be considered. In order to ensure near zero cell loss throughout the network, the queue sizes should be made as large as possible. On the other hand, large queue sizes increase the end-to-end delay and delay jitter, which are important quality of service (QOS) requirements for the real-time traffic. The queue sizes must be such that they satisfy both the requirement for the near zero cell loss, and the delay and jitter requirements for the users.

- Transmit link queue inside the ATM nodes
  - Queue size for priority 0: 512
  - Queue threshold for priority 0: 256
  - Queue size for priority 1: 1024
  - Queue threshold for priority 1: 512
- Transmit link queue in the user node
  - Queue size for priority 0: 1000
  - Queue size for priority 1: 5000
- Queue size for peak rate controllers: 512
- Peak rate for peak rate controllers
  - voice: 64 kbps

video: 512 kbps

data: 500 kbps

- Queue size for leaky buckets: 0
- Drain rate for leaky buckets

voice: 36 kbps

video: 391 kbps

data: 556 kbps

Table 2. Simulation Parameters

The transmit link queue size for all links, except for the access link from the user, is 512 cells for priority 0 with a threshold of 256 for discarding CLP=1 cells, and 1024 cells for priority 1 cells with a threshold of 512. The queue size for the lower priority, namely priority 1 nonreal-time data traffic, is made twice as large as the queue size for voice and video traffic (priority 0), as data traffic is considered delay insensitive. The larger queue size accommodates sudden bursts in the traffic pattern, thus minimizing the cell loss ratio at the expense of a slight increase in end-to-end delay. The queue size for the transmit link in the user node is 1000 cells for priority 0, and 5000 cells for priority 1, so that no cells are dropped in the user node.

The queue size for all peak rate controllers is 512 cells, and the leaky bucket queue size is set to zero. Initial test simulations with a finite leaky bucket queue were conducted. Adding a queue in front of the leaky bucket increases the overall end-to-end delay in the network, and the users' delay jitter QOS cannot be met. To reduce the number of cells marked CLP=1 at the leaky buckets, the bucket depth was increased instead.

The traffic source parameters for voice, video, and data are described in sections 6.2.1.1, 6.2.1.2, and 6.2.1.3, respectively.

8. Results and Discussion

8.1 Voice and Video

As discussed earlier, the load in the network was changed by adding five voice sources for each video source added, and the total number of traffic sources varied from 72 to 120. For Case 1, the amount of traffic sent by the sources was calculated by using the average parameters for the sources presented in Chapter 6, as no congestion or rate control is present. For Cases 2 and 3, the amount of traffic sent by the sources was measured during the simulations. Figure 11 shows the total traffic load, as a percentage of the satellite link capacity, sent by the traffic sources in the case where no congestion control or rate control policy is present, namely in Congestion Control Case 1. For a number of sources equal to or greater than 96, the amount of traffic sent by the sources (in Mbps) exceeds the link capacity. Figure 12 shows the amount of traffic sent by the sources in Congestion Control Case 2, in which the full input rate -based congestion control and rate policy scheme is present. For all the numbers of sources shown, the load is fairly constant between 70 and 80% of the link capacity. Comparing Figure 12 to Figure 11, both voice and video traffic are requested to slow down their bit rates in Congestion Control Case 2. Figure 13 shows the traffic load for Congestion Control Case 3, in which sources are requested to slow down their rates only when a cell is discarded. Therefore, the load is higher in this case as compared to Case 2, but still significantly lower as compared to the unconstrained case, namely Case 1. In both Case 2 and 3 the traffic load sent by the sources is less than the link capacity of the satellite link.

The goal for voice and video traffic, as stated in section 1.3, is to satisfy the quality of service requirements (QOS) in terms of delay and especially delay jitter. The delay jitter requirement is in the order of tens of milliseconds. As also discussed, cell discarding is not acceptable. The results for Scenario 1 consisting of voice and video traffic are presented in Figures 14 through 19.

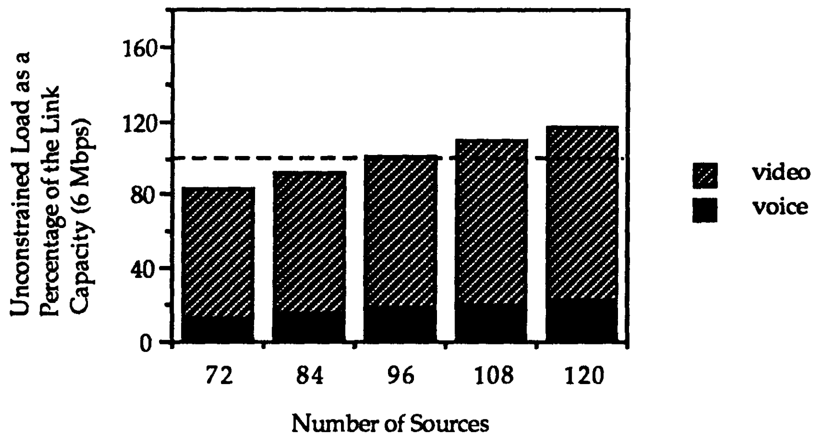

Figure 11. Traffic Scenario 1 Congestion Control Case 1

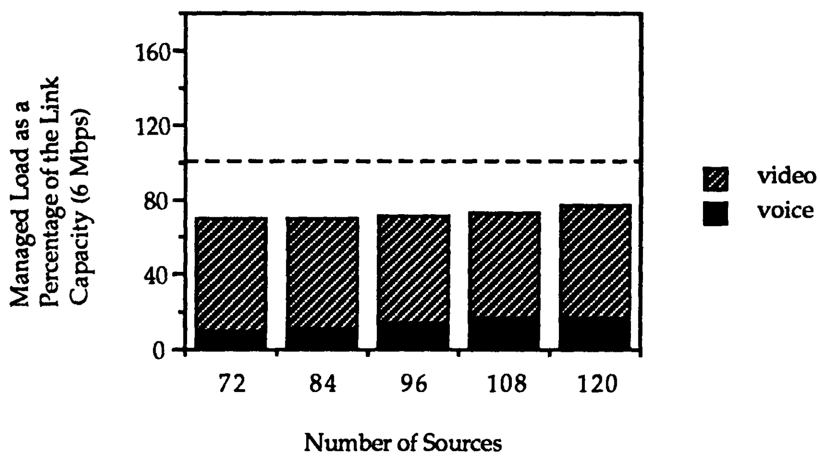

Figure 12. Traffic Scenario 1 Congestion Control Case 2

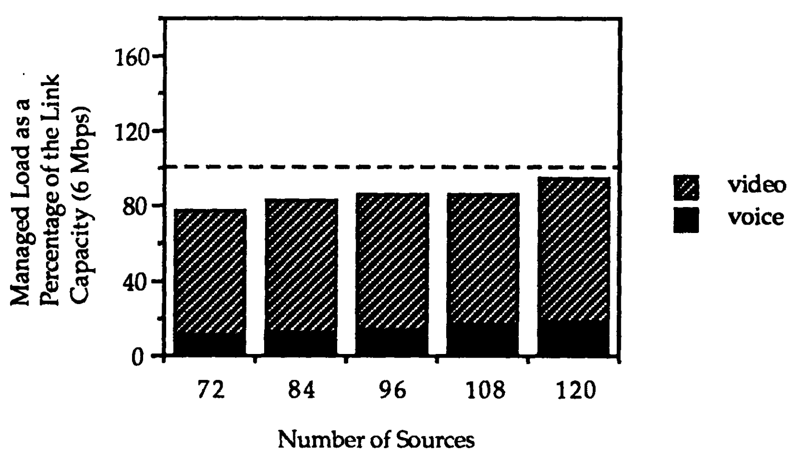

Figure 13. Traffic Scenario 1 Congestion Control Case 3

As seen in Figure 14, the cell loss ratio in Case 1, the base case, increases as a function of the load, namely the number of sources, and at a loading of around 1.2 of the link capacity, the cell loss ration is as high as 13%. Furthermore, the cell loss graphs for different random seeds do not closely follow each other, which is likely to be a result of the randomness of the traffic: as discussed in conjunction with the traffic models, the session and active period durations for each source are exponentially distributed based on the random seed used. Figure 15 shows the percentage of cells for which the end-to-end delay exceeds the minimum measured end-to-end delay by more than 20 milliseconds on a virtual circuit (VC) basis. The minimum end-to-end delay is the minimum delay measured for a VC during a simulation. The delay statistic shown in the graphs is a measure of the delay jitter experienced by the ATM cells, measured over the entire duration of the simulation, rather than pertaining to a single voice session. As the number of traffic sources is increased, the percentage of cells exceeding the jitter requirement increases: at a load of 1.1 of the link capacity, the percentage of cells is above 60%. It is clear from Figures 14 and 15 that the goals set for voice and video traffic cannot be satisfied in the case where no access policing or congestion control is present.

Figures 16 and 17 present the same cell loss ratio and delay statistics for Case 2, which includes access policing with full rate-based congestion control. Zero cells are discarded for all the numbers of sources simulated, and similarly no cells exceed the jitter requirement; therefore, the goals set for voice and video are fully satisfied. The simulation results verify that the congestion control algorithm works as predicted: congestion detection based on the input rate of traffic to the transmit link queue acts to prevent cell discarding by requesting the traffic sources to slow down their traffic bit rates already when the total input rate of traffic to the link is 90%. It should be noted that voice and video is allowed to slow down their transmission rates to 50% of the nominal value. Further work is needed to determine whether voice and video traffic transmitted occasionally at 50% of the nominal rate actually meets the audio and visual quality requirements, respectively.

Figures 18 and 19 show the measured statistics for Case 3. The only difference between Case 2 and 3 is that for Case 3 congestion detection at the transmit links is solely based on cell discarding. Congestion cannot be prevented before cells are discarded, but rather the situation can only be alleviated after cells have been discarded. This is demonstrated by the worse cell loss ratio and delay statistic of Case 3 as compared to the results for Case 2. As some cells are dropped, the queues must be overflowing at times, causing increases in the end-to-end delays of cells. The performance of Case 3, as measured by the statistics collected, is clearly superior to that of Case 1, but Case 3 does not satisfy the goals for voice and video traffic in terms of the jitter requirement and the cell loss ratio.

It is important to note that looking at Figures 12 and 13, the traffic load in Case 3 is approximately 10% higher as compared to Case 2, and therefore it could be expected that the voice and video traffic is slowed down less and the throughput is higher in Case 3, as the link utilization is higher. The cell loss ratios in Case 3 are on the order of 10^{-4}. It is conceivable that Case 3 would lead to a more efficient link utilization if a coding algorithm that supports cells loss ratios of 10^{-4} is available. However, Figure 19 shows that the percentage of cells for which the delay jitter exceeds 20 milliseconds can be as high as 6% in Case 3, which clearly violates the users' quality of service requirements. As stated earlier, only Congestion Control Case 2 fully satisfies the users' QOS requirement of both near zero cell loss and minimal delay jitter.

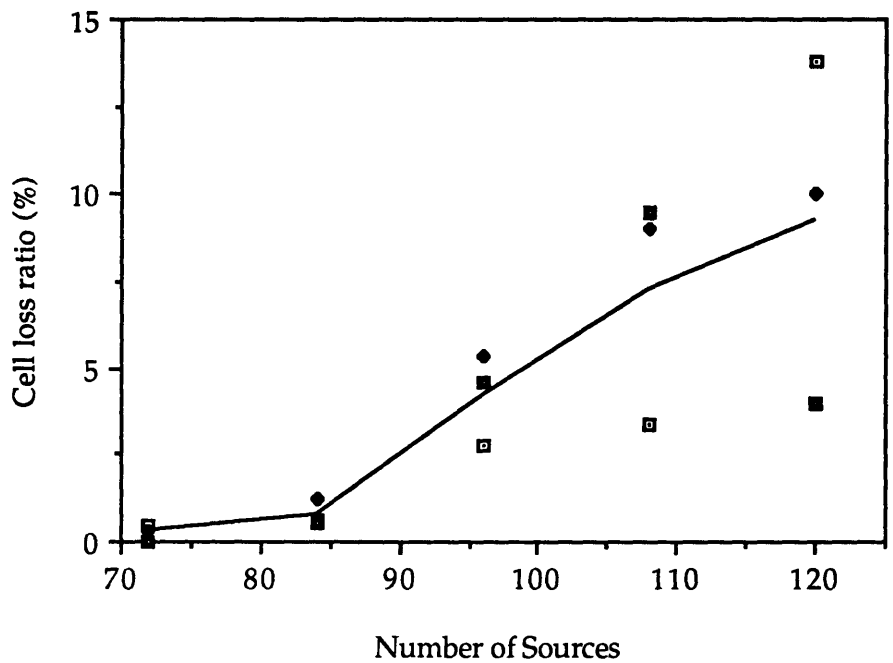

Figure 14. Cell Loss Ratio as a Function of the Number of Sources for Voice and Video Traffic Scenario 1 Congestion Control Case 1

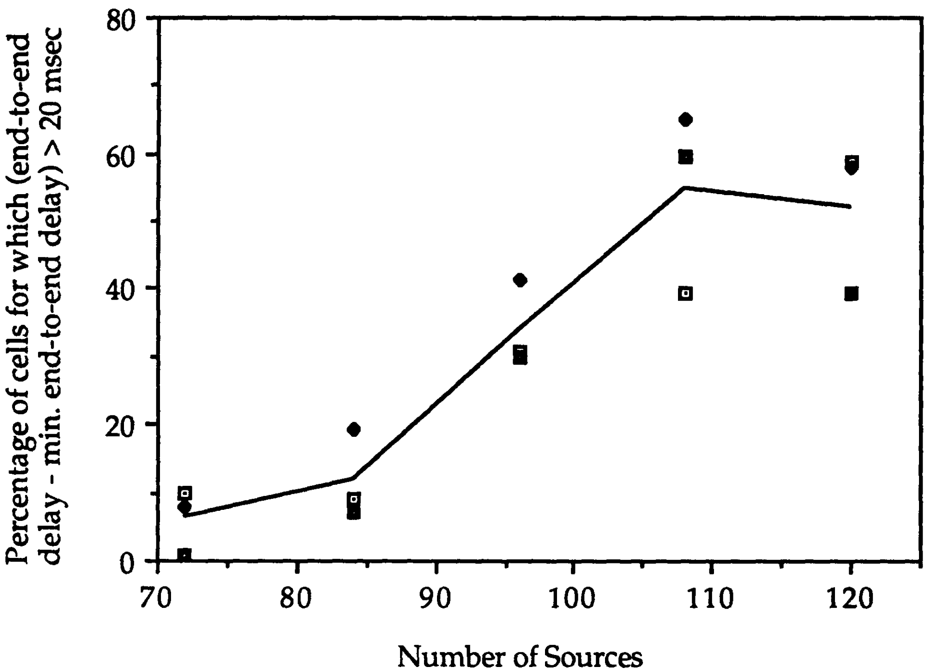

Figure 15. Percentage of Cells for Which the Difference Between End-to-end Delay and Minimum End-to-end Delay Exceeds 20 Milliseconds as a Function of the Number of Sources for Voice and Video Traffic Scenario 1 Congestion Control Case 1

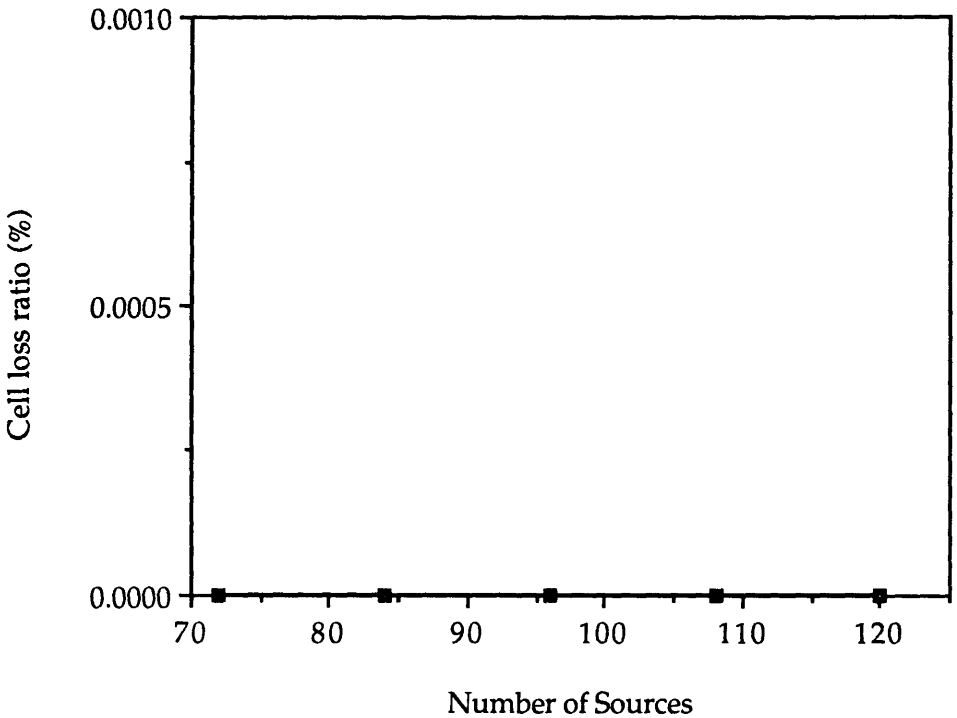

Figure 16. Cell Loss Ratio as a Function of the Number of Sources for Voice and Video Traffic Scenario 1 Congestion Control Case 2

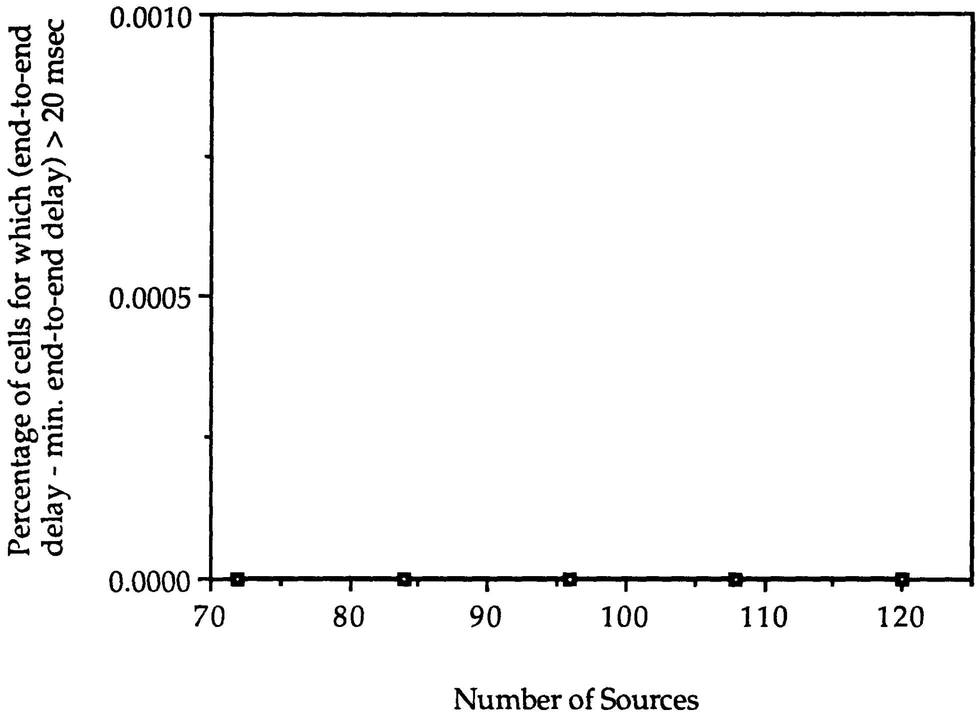

Figure 17. Percentage of Cells for Which the Difference Between End-to-end Delay and Minimum End-to-end Delay Exceeds 20 Milliseconds as a Function of the Number of Sources for Voice and Video Traffic Scenario 1 Congestion Control Case 2

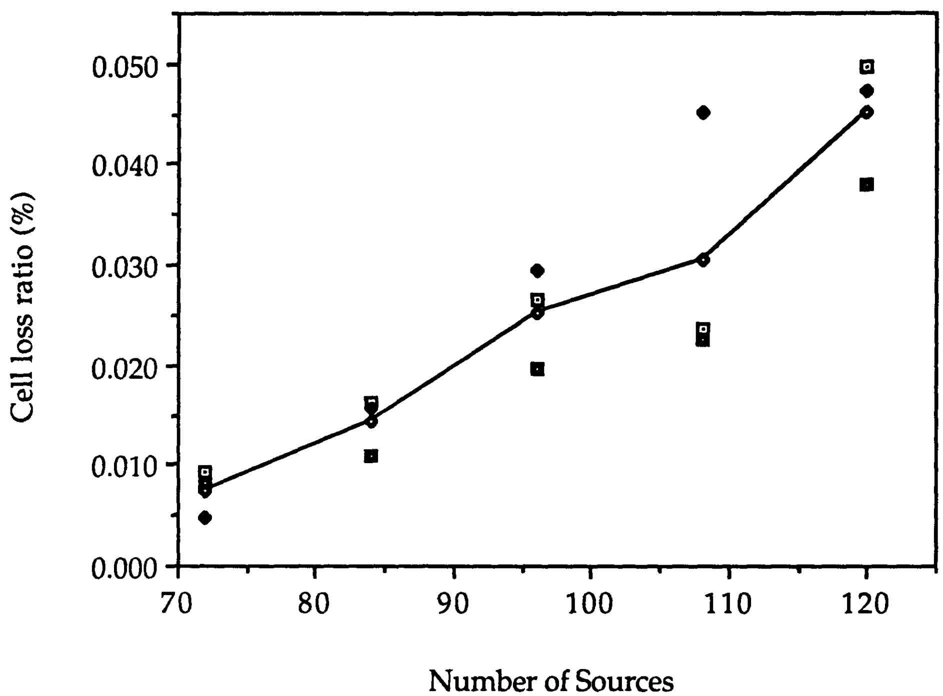

Figure 18. Cell Loss Ratio as a Function of the Number of Sources for Voice and Video Traffic Scenario 1 Congestion Control Case 3

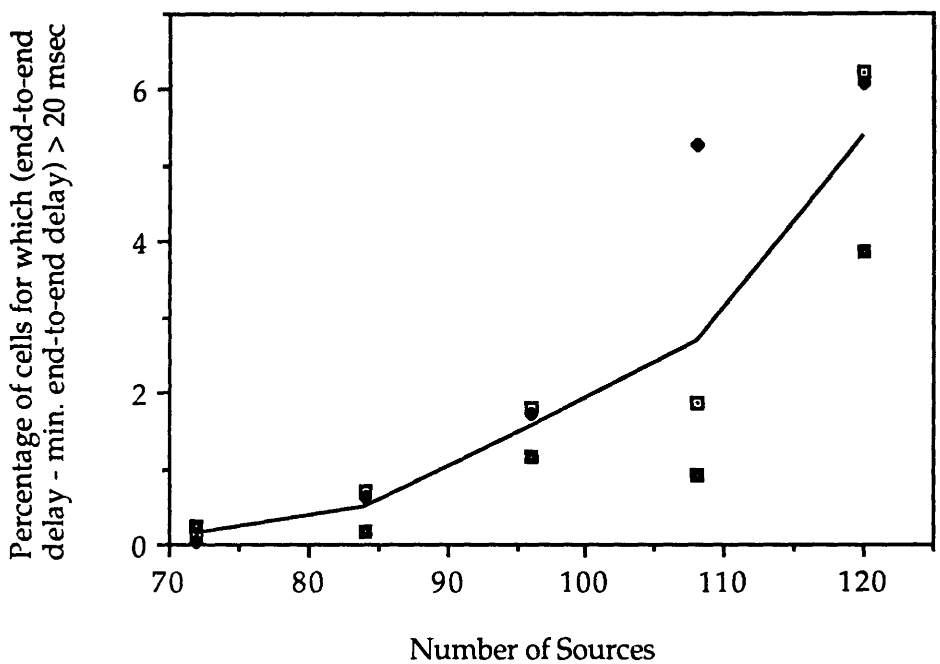

Figure 19. Percentage of Cells for Which the Difference Between End-to-end Delay and Minimum End-to-end Delay Exceeds 20 Milliseconds as a Function of the Number of Sources for Voice and Video Traffic Scenario 1 Congestion Control Case 3

8.2 Data

As discussed earlier, the loading in the network is changed by adding data sources, and the total number of sources varied from 15 to 29. The amount of traffic sent by the sources in each congestion control case was measured during the simulations. Figure 20 shows the traffic load sent by the sources in Case 1 as a percentage of the satellite link capacity for the different number of sources. In Case 1 there is no congestion or rate control policy, and for a number of sources equal to and greater than 25 the amount of traffic sent by the sources exceeds the link capacity. Figure 21 shows the loading for Case 2, in which the sources are requested to slow down their bit rates according to the input rate-based congestion control algorithm. The traffic load is fairly constant around 75% of the link capacity. Figure 22 shows the loading in Case 3, in which the sources are slowed down only when a cell is discarded. The load in Case 3 is again over 10% higher than in Case 2.

As discussed earlier, near zero cell loss is the QOS requirement of the users. Furthermore, for data traffic (file transfers), the goal is to maximize throughput. In this work, throughput is defined as the number of information bits received by the destination over the entire duration of the simulation, and normalized by the satellite link rate, namely 6 Mbps. It is crucial that the throughput should be maximized such that the QOS cell loss requirement is not violated. High throughput is only valuable if it is consistent with QOS requirements. The results for Scenario 2 consisting of data file transfers are presented in Figures 23 through 31.

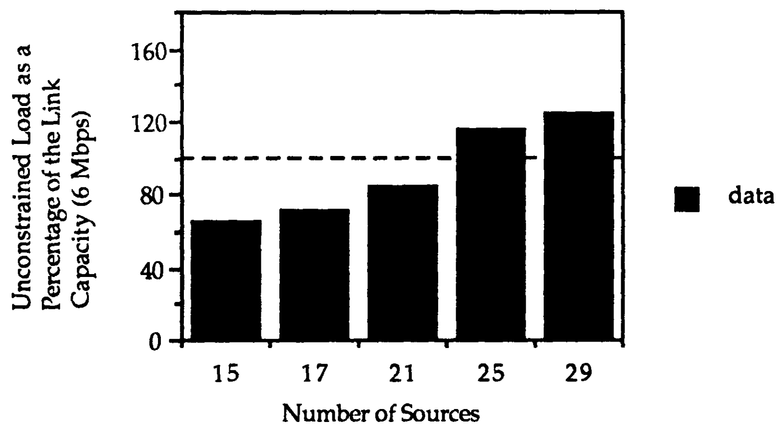

Figure 20. Traffic Scenario 2 Congestion Control Case 1

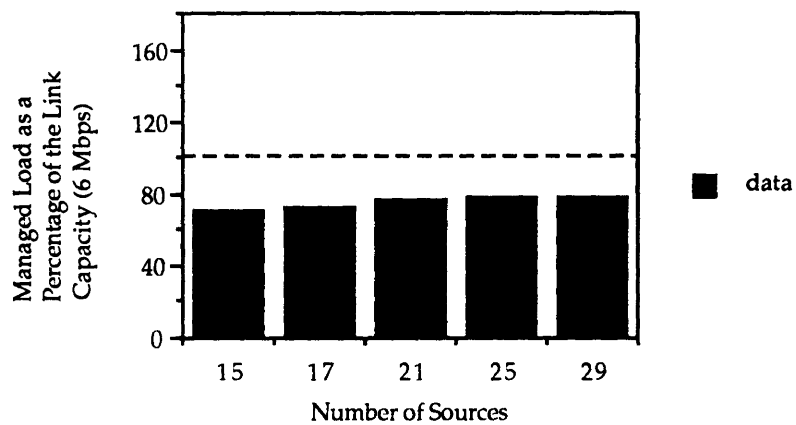

Figure 21. Traffic Scenario 2 Congestion Control Case 2

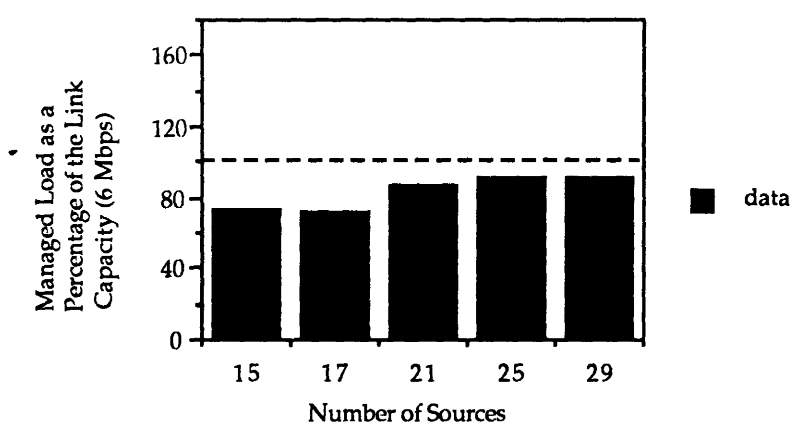

Figure 22. Traffic Scenario 2 Congestion Control Case 3

The cell loss ratio in the base case (Case 1, which does not include any access policing or congestion control) increases as a function of the number of sources; the cell loss ratio is close to 20% at loads equal to 1.25 of the link capacity. Even though the cell loss performance in Case 3 is clearly superior to that in Case 1 (see Figures 23. and 29.), the near zero cell loss QOS requirement is only met in Case 2. Case 2 succeeds in preventing congestion and therefore cell discarding, by requesting the sources to slow down their data bit rates when the total input rate of traffic to the transmit link is 90% of the link transmission rate.

The average file transfer times for each of the three cases are shown in Figures 24, 27, and 30, respectively. The average file transfer time refers to the time it takes a sources to send out a file of 1 Mbyte at its data bit rate, averaged over all the file transfers in the network. For Case 1, the files are transferred at the nominal rate of 0.5 Mbps, which is reflected in the constant file transfer time of approximately 16 seconds. The file transfer time is slightly above 16 seconds as each file transfer includes a slow-start algorithm according to which the data bit rate gradually increases from 50% of the nominal value to the nominal value. For Case 2, the average file transfer time increases as the load to the network, namely the number of sources, is increased: as the traffic load in the network is higher, the sources are requested to slow down their traffic bit rates according to the congestion management mechanism. At loads close to 80% of the link capacity (29 sources), the average file transfer time is approximately 35 seconds, leading to an average traffic bit rate of 0.23 Mbps, which is roughly 46% of the nominal file transfer bit rate of 0.5 Mbps. For Case 3, the average file transfer time is shorter as compared to Case 2. At loads of 90% of the link capacity (29 sources), the average file transfer for Case 3 is around 25 seconds leading to an average data bit rate of 0.32 Mbps, which is 64% of the nominal bit rate. Even though the average file transfer times are shorter in Case 3 versus Case 2, the cell loss QOS requirement is not satisfied in Case 3. Case 2 not only satisfies the near zero cell loss requirement, but it also provides the highest throughput, given the cell loss requirement. High throughput is only valuable if the QOS cell loss requirement is satisfied. The idea behind the congestion control algorithm is to regulate the input load of traffic into the network, by requesting sources to slow down, such that all the cells sent by the sources can be transmitted to the destinations without cell discarding.

Figure 29 shows that the cell loss ratio in Case 3 is on the order of 10^{-4}. If a coding algorithm supporting this cell loss ratio is available, it is conceivable that Case 3 may lead to over 10% higher throughput as compared to Case 2. This can been seen by comparing the load graphs (Figures 21 and 22) and the throughput (Figures 28 and 31) for Cases 2 and 3.

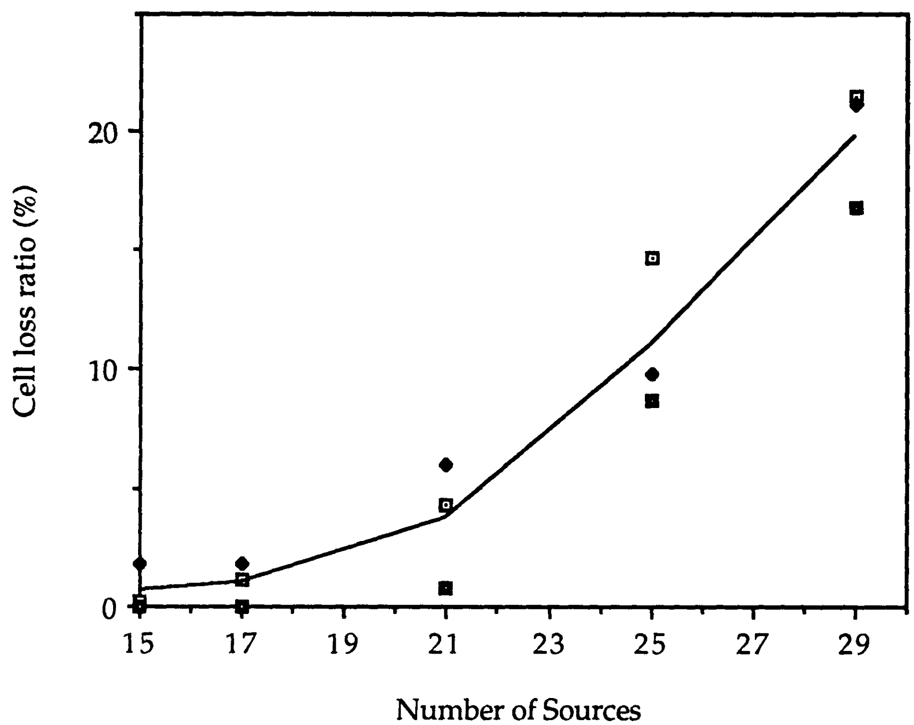

Figure 23. Cell Loss Ratio as a Function of the Number of Sources for Data (File Transfer) Traffic Scenario 2 Congestion Control Case 1

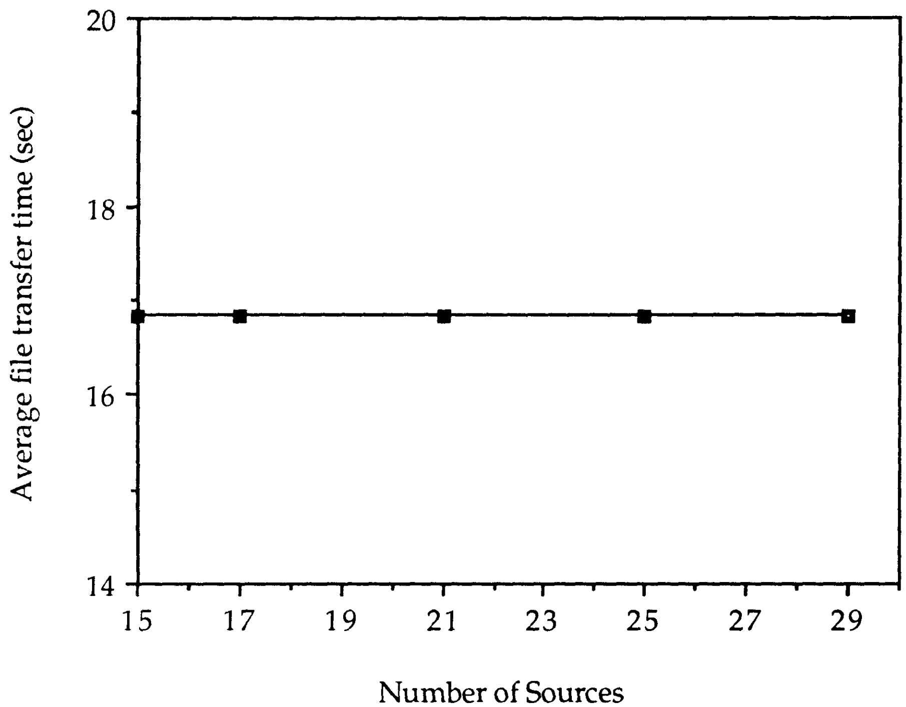

Figure 24. Average File Transfer Time as a Function of the Number of Sources for Traffic Scenario 2 Congestion Control Case 1

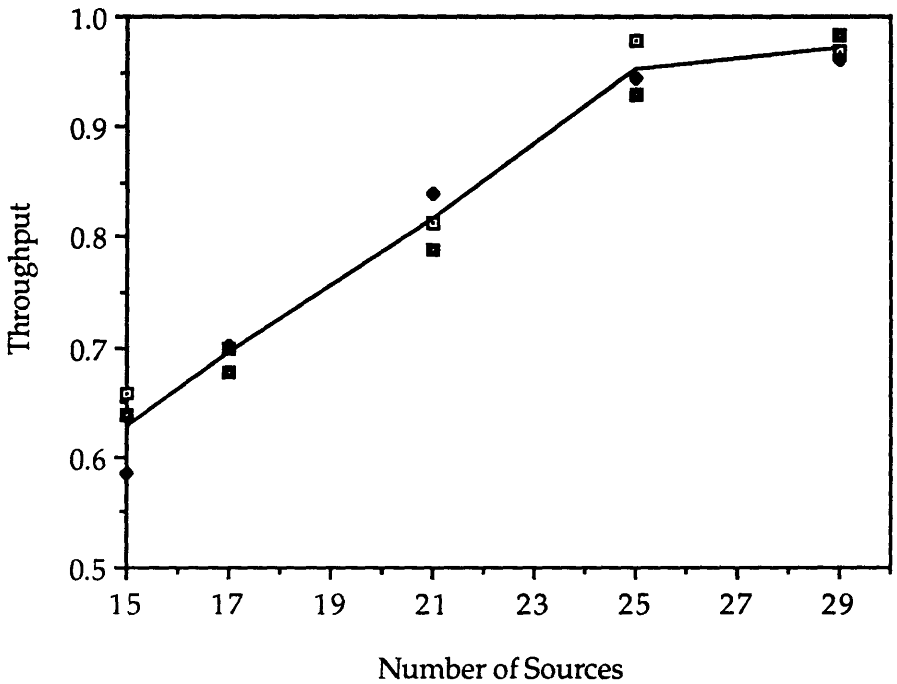

Figure 25. Throughput of Data Traffic (File Transfers) as a Function of the Number of Sources for Traffic Scenario 2 Congestion Control Case 1

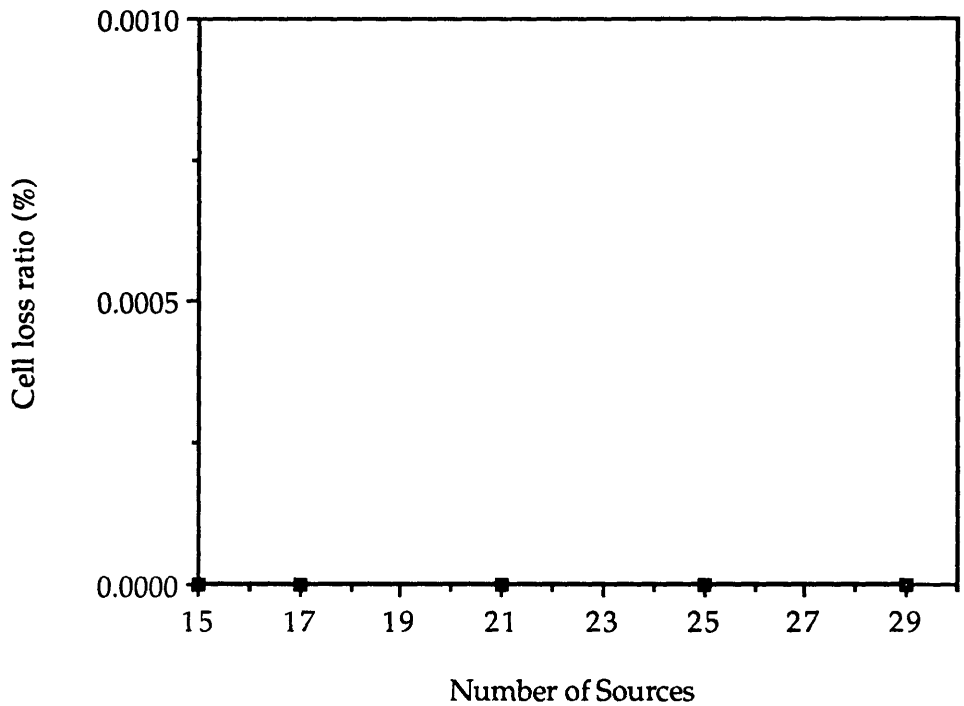

Figure 26. Cell Loss Ratio as a Function of the Number of Sources for Data (File Transfer) Traffic Scenario 2 Congestion Control Case 2

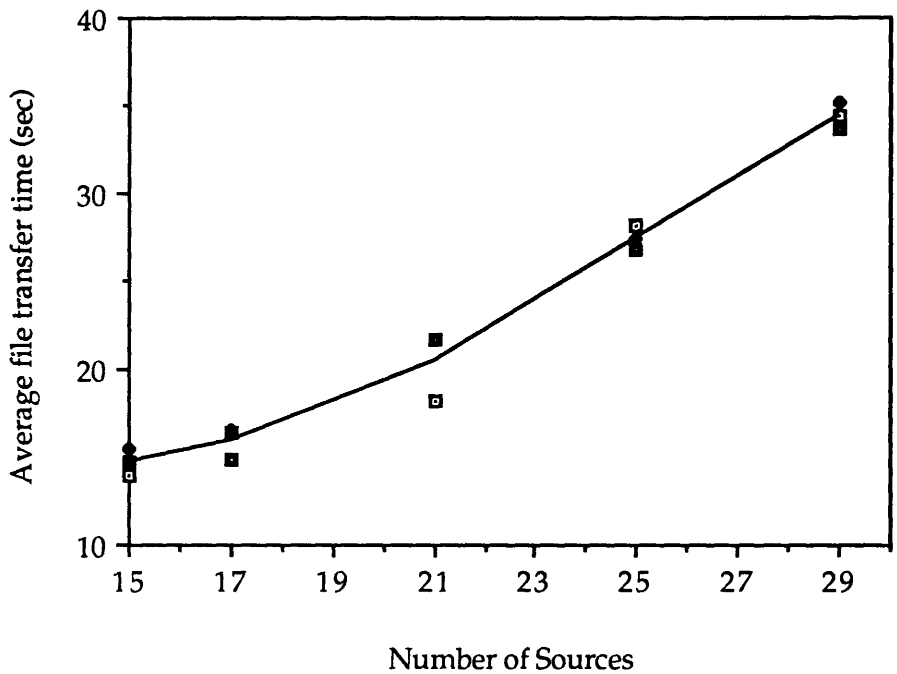

Figure 27. Average File Transfer Time as a Function of the Number of Sources for Traffic Scenario 2 Congestion Control Case 2

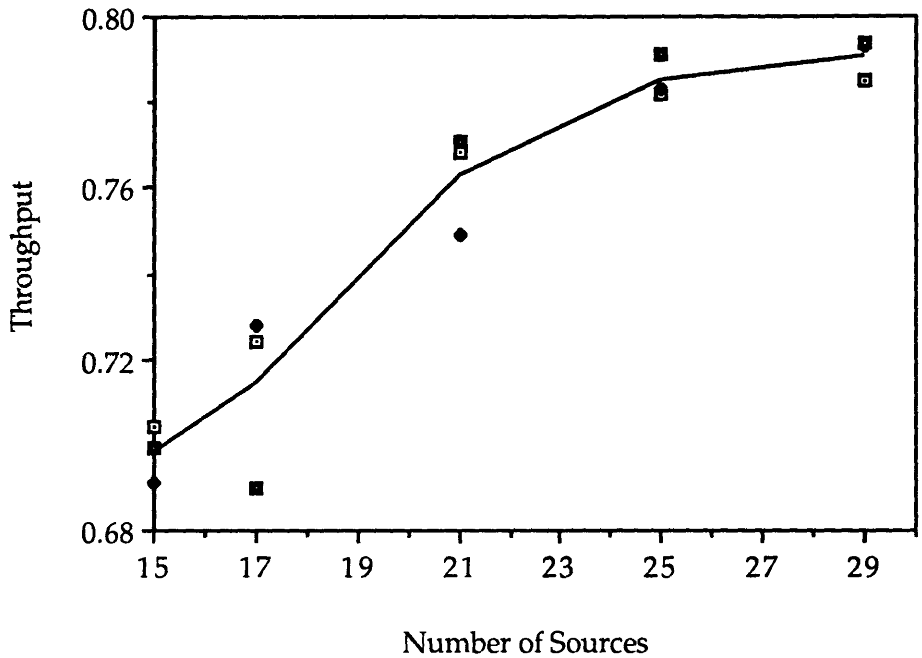

Figure 28. Throughput of Data Traffic (File Transfers) as a Function of the Number of Sources for Traffic Scenario 2 Congestion Control Case 2

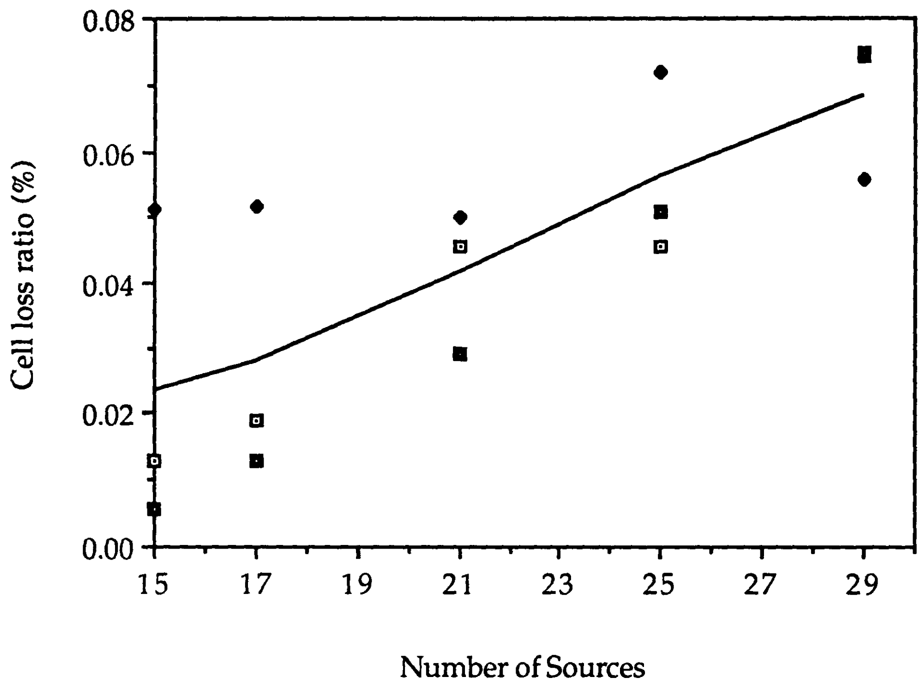

Figure 29. Cell Loss Ratio as a Function of the Number of Sources for Data (File Transfer) Traffic Scenario 2 Congestion Control Case 3

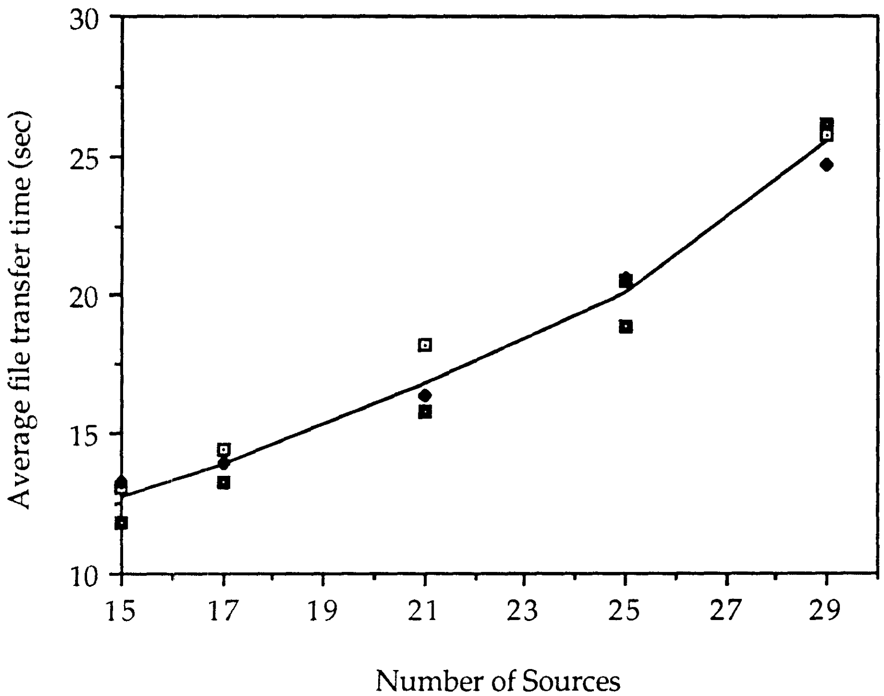

Figure 30. Average File Transfer Time as a Function of the Number of Sources for Traffic Scenario 2 Congestion Control Case 3

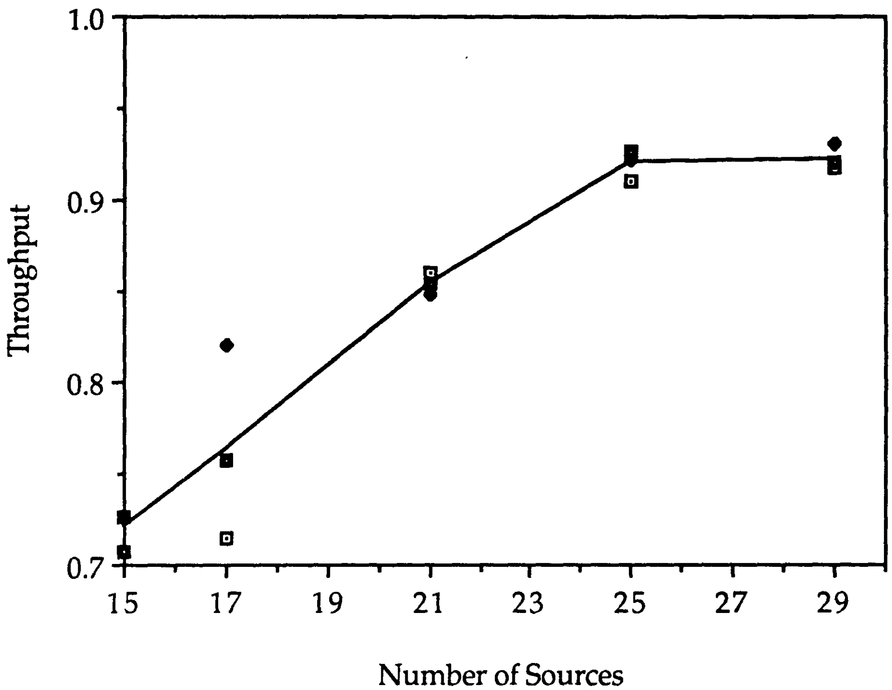

Figure 31. Throughput of Data Traffic (File Transfers) as a Function of the Number of Sources for Traffic Scenario 2 Congestion Control Case 3

8.3 Mixture of Voice, Video, and Data

The load in the network was changed by adding five voice sources for each video and data source added. The total number of traffic sources varied from 29 to 113. For Case 1, in which no congestion or rate control is present, the total amount of traffic sent by the voice and video sources was calculated using the average parameters of the sources, as discussed earlier. Data traffic for Case 1, and all traffic in Cases 2 and 3, sent by the sources was measured over the duration of the simulation. Figure 32 shows the traffic load as a percentage of the satellite link capacity for the different number of sources. For a number of sources greater than or equal to 57, the amount of traffic sent by the sources exceeds the satellite link capacity. Figure 33 shows that for Case 2 the loading levels off to around 80% of the link capacity at the larger number of sources. In Figure 34, the load for Case 3 is over 10% higher as compared to Case 2. This is due to the fact that in Case 3, the sources are requested to slow down only when a cell is discarded, i.e. when the satellite link is already congested. Case 2 acts to prevent congestion by slowing down the sources when the input rate of traffic is 90% of the link capacity. Comparing Figures 33 and 34 to Figure 32, it is clear that all three types of traffic are slowed down in Cases 2 and 3, but data is slowed down considerably more than voice and video, which is consistent with the congestion control algorithm.

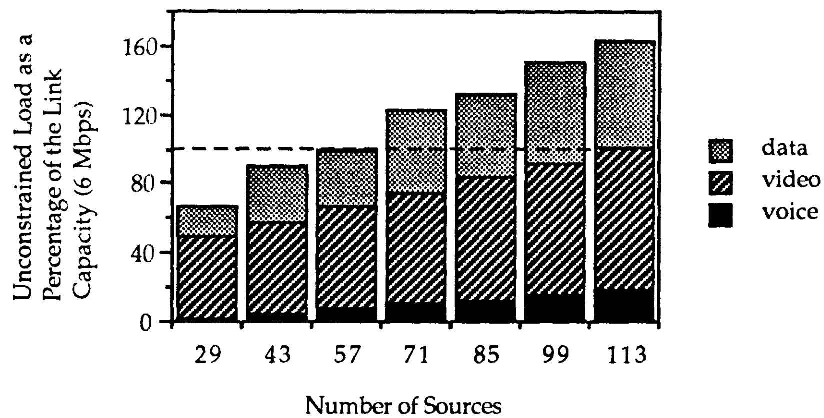

Figure 32. Traffic Scenario 3 Congestion Control Case 1

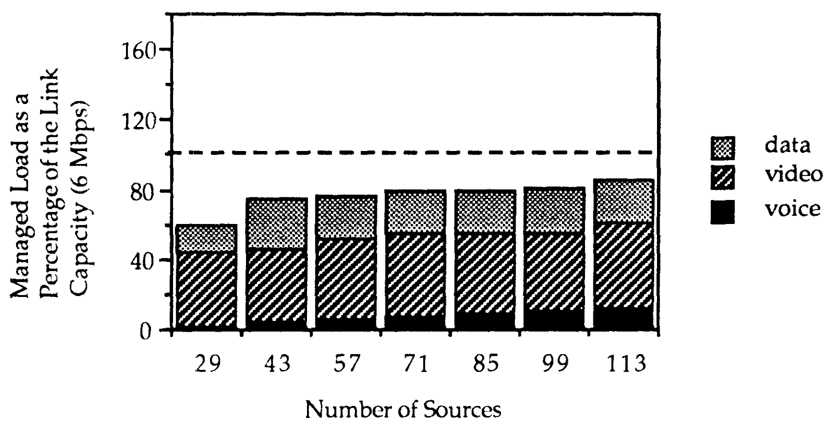

Figure 33. Traffic Scenario 3 Congestion Control Case 2

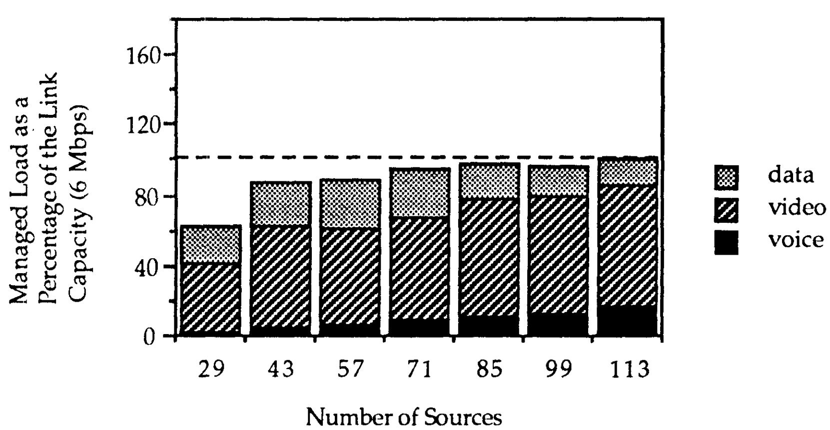

Figure 34. Traffic Scenario 3 Congestion Control Case 3

The results for Scenario 3, namely a mixture of voice, video, and data (file transfer) traffic are shown in Figures 35 through 49. The goal for a zero cell loss ratio for both voice and video, and data, is achieved in Case 2 with access policing and the proposed congestion control scheme. The cell loss ratios in Cases 1 and 3 are not acceptable as the users' QOS requirement is not satisfied. The delay jitter requirement is also only achieved in Case 2. It is worthwhile to note that in Case 2, the cell loss ratio and delay jitter for voice and video traffic are not affected by the data file transfer traffic, as was predicted earlier in conjunction with the priority scheme. This is an important goal of the congestion control scheme and queue management, as it is crucial that allowing the data sources to exceed their nominal bit rates does not occur at the expense of the high-priority, real-time voice and video traffic.

Figures 38, 43, and 48 show the average file transfer time for all three cases. In Case 1, the 1 Mbyte files are transferred at the nominal bit rate, 0.5 Mbps. The average file transfer time, averaged over all the data sources, is around 16 seconds. In Case 2, the average file transfer time increases as the number of sources is increased. The data sources are requested to slow down more when the load in the network is higher, in order to prevent congestion. At a load of close to 0.8 of the link capacity (113 sources), the average file transfer time in Case 2 is around 85 seconds, and so the average data bit rate for a source is 0.1 Mbps, which is 20% of the nominal value of 0.5 Mbps. The same trend of increasing average file transfer time with increasing load is observed for Case 3. However, the average file transfer times are longer as compared to Case 2. In Case 3, the sources are requested to reduce their transmission rate by 50% instead of 20% of the current bit rate, when a congestion notification is received. At a load of 1.0 of the link capacity (113 sources), the average file transfer time in Case 3 is almost 150 seconds, leading to an average data bit rate of 0.05 Mbps, which is only 10% of the nominal value.

It is desirable that the average file transfer time be as short as possible as this leads to the highest data throughput. However, throughput can only be maximized to the extent that the more stringent requirement of near zero cell loss is not violated. Case 2 satisfies the cell loss requirement and also provides a shorter average file transfer time than Case 3. More importantly, in Case 2 sources are requested to slow down when the total input rate to the transmit link queue is 90% of the link rate, and when the link is not yet congested. Case 2 prevents congestion and cell discarding, whereas Case 3 reacts to congestion when some cells have already been discarded.

The throughput for the three cases is shown in Figures 39, 44, and 49. As discussed above, it is important to note that the data, namely file transfer throughput should only be maximized so that the requirement about the zero cell loss ratio is not violated. At higher loads, the throughput for Case 2 is clearly superior compared to the other two cases, and it settles around 24% of the link rate. Deterioration of the throughput performance at higher loads for Cases 1 and 3 is likely to be due to the large number of data cells discarded at that loading. At some points in the graph, the throughput is actually higher in Cases 1 and 3, but the file transfer cell loss ratio is not zero in Cases 1 and 3, and therefore they do not satisfy the QOS cell loss requirement for satellite connections.

In addition to satisfying the zero cell loss requirement for file transfer data traffic, Case 2 also provides the best throughput performance. Data sources are requested to slow down their bit rates to prevent congestion and cell discarding. The main idea of Case 2 is to control the input load of the data sources into the network, by controlling the bit rates of the sources, so that all data cells transferred by the sources are received at the destination sinks without any cell discarding. According to the algorithm, the input rate of traffic is controlled so that it is around 90% of the link transmission rate.

Comparing Figures 33 and 34, voice and video are slowed down more in Case 2 as compared to Case 3. Figure 45 shows that the cell loss ratio for voice and video in Case 3 is on the order of 10^{-4}. It is therefore conceivable that Case 3 would lead to a higher link utilization as compared to Case 2, if a coding algorithm supporting a cell loss ratio of 10^{-4} is available. However, Figure 46 shows that for Case 3, the percentage of cells for which the jitter exceeds 20 milliseconds is as high as 1.5%, which violates the users' jitter requirement. Therefore, Case 2 with the preventive congestion and rate control policy is the only policy that satisfies the users' QOS requirements, both in terms of cell loss ratio and delay jitter.

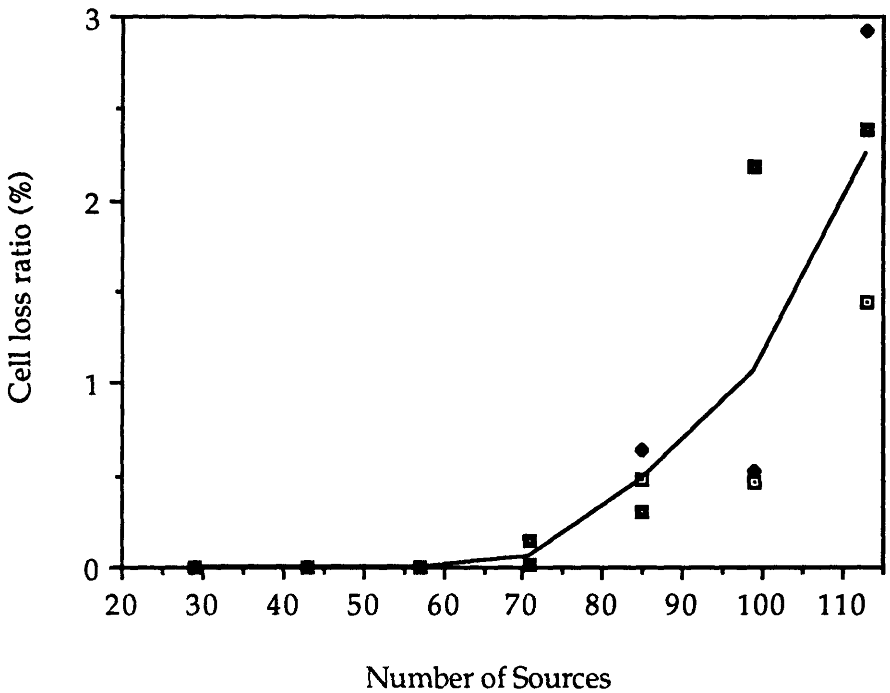

Figure 35. Cell Loss Ratio as a Function of the Number of Sources for Voice and Video in Traffic Scenario 3 Congestion Control Case 1

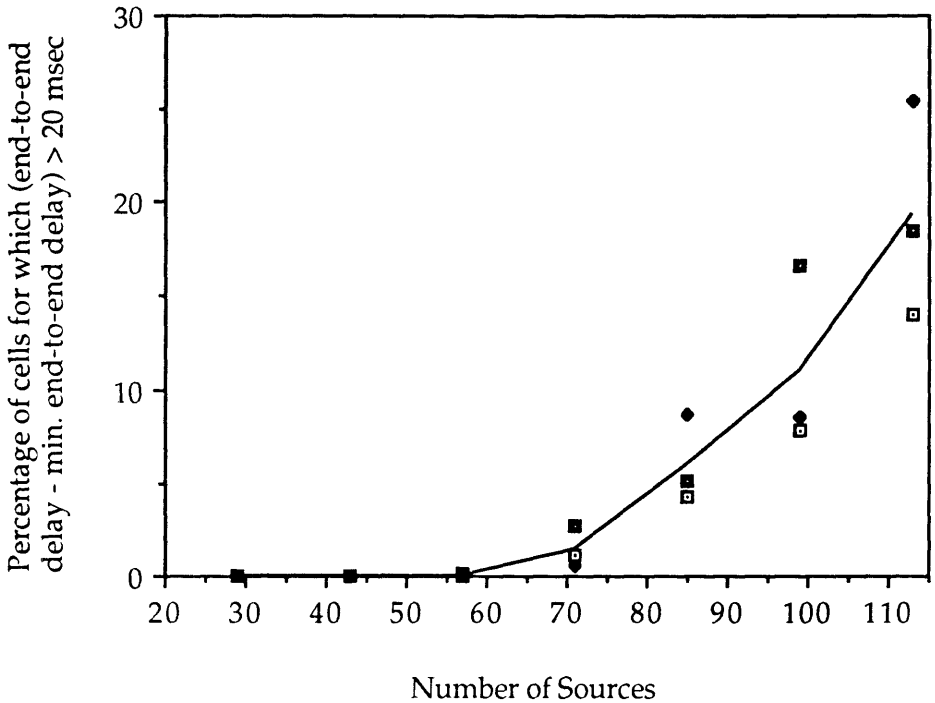

Figure 36. Percentage of Cells for Which the Difference Between End-to-end Delay and Minimum End-to-end Delay Exceeds 20 Milliseconds as a Function of the Number of Sources for Voice and Video in Traffic Scenario 3 Congestion Control Case 1

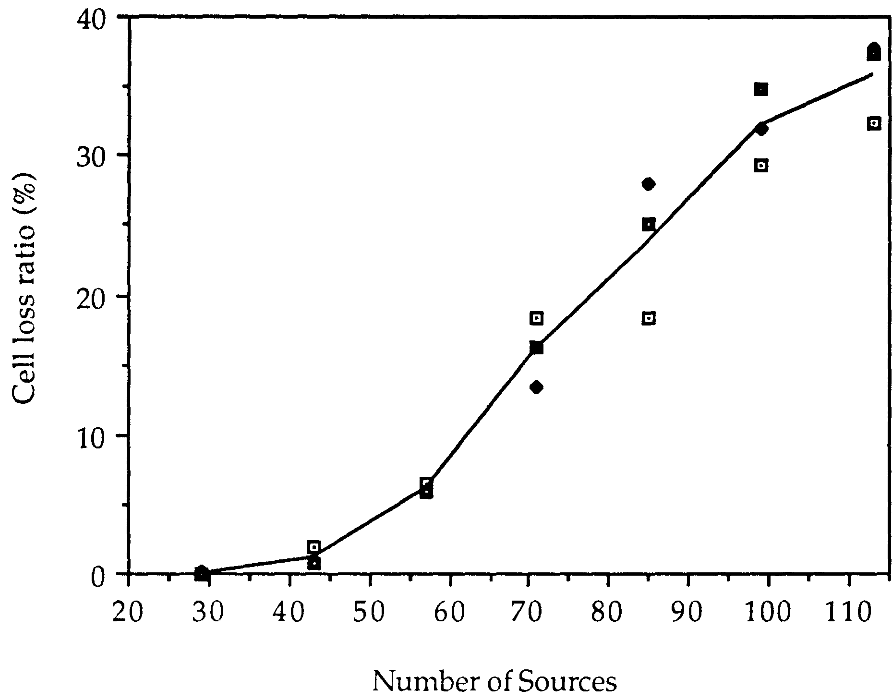

Figure 37. Cell Loss Ratio as a Function of the Number of Sources for Data (File Transfers) in Traffic Scenario 3 Congestion Control Case 1

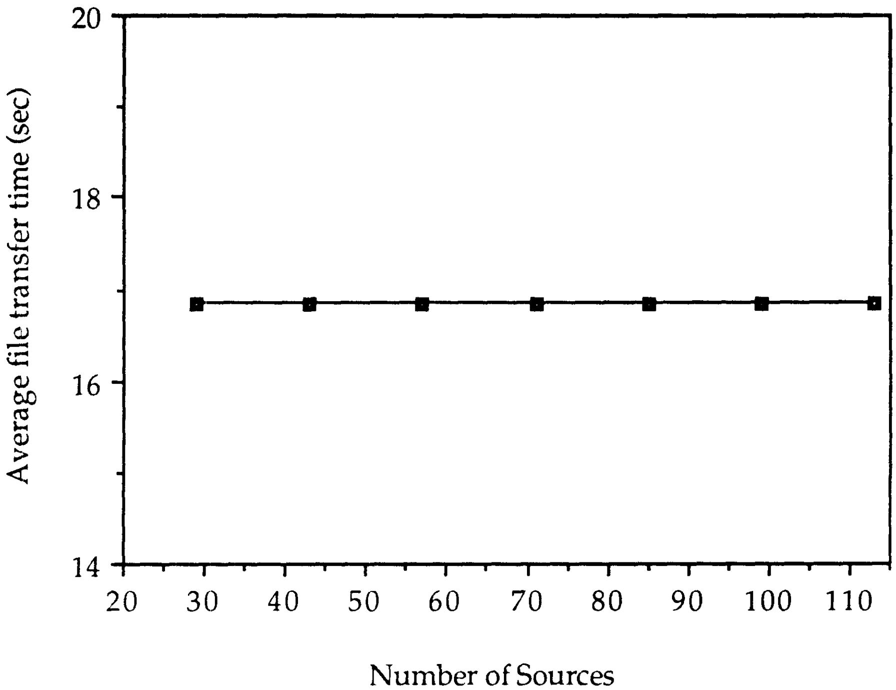

Figure 38. Average File Transfer Time as a Function of the Number of Sources for Traffic Scenario 3 Congestion Control Case 1

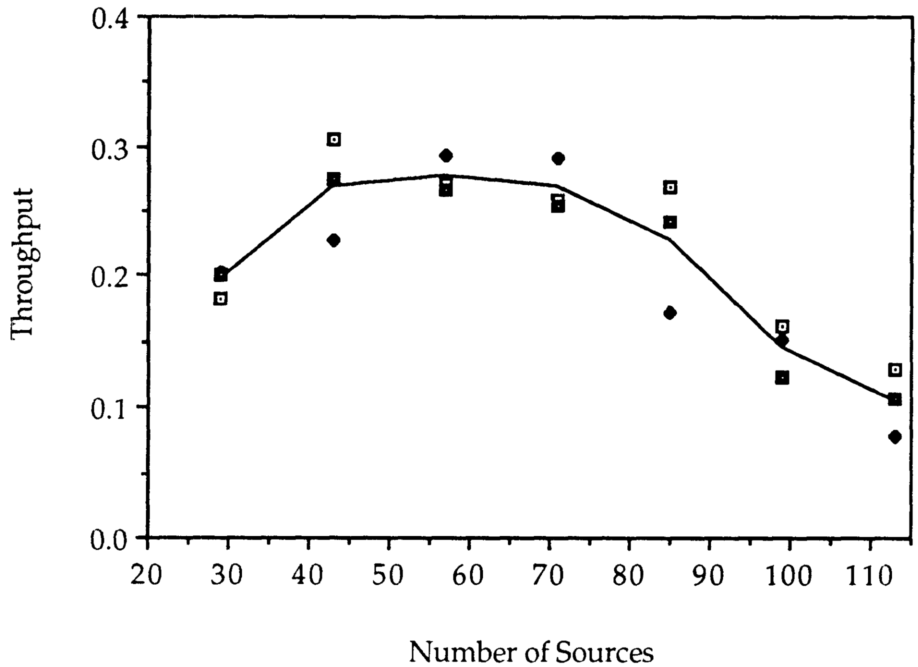

Figure 39. Throughput of Data (File Transfers) as a Function of the Number of Sources for Traffic Scenario 3 Congestion Control Case 1

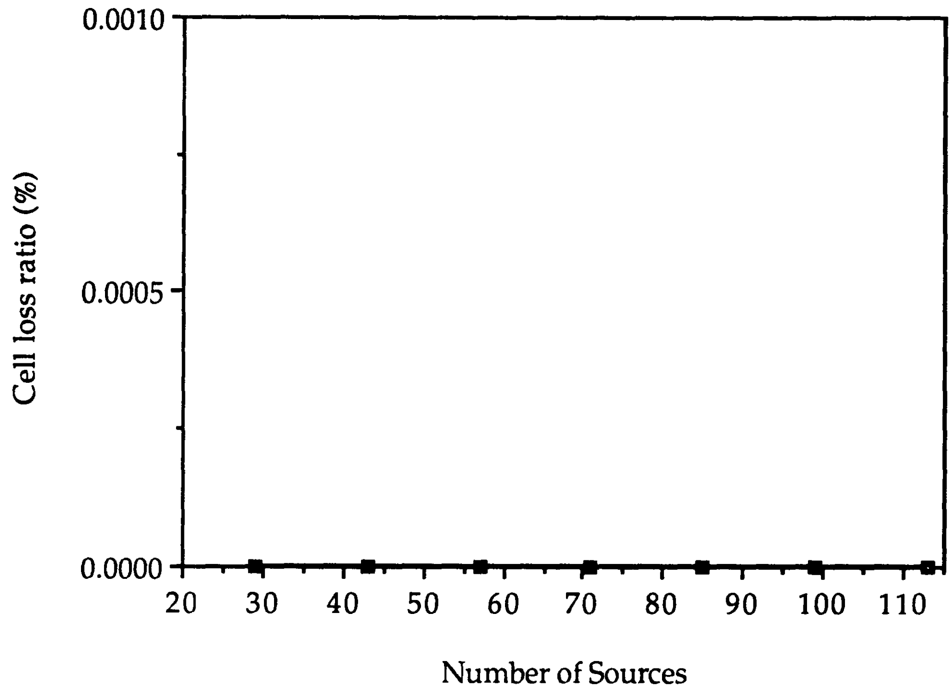

Figure 40. Cell Loss Ratio as a Function of the Number of Sources for Voice and Video in Traffic Scenario 3 Congestion Control Case 2

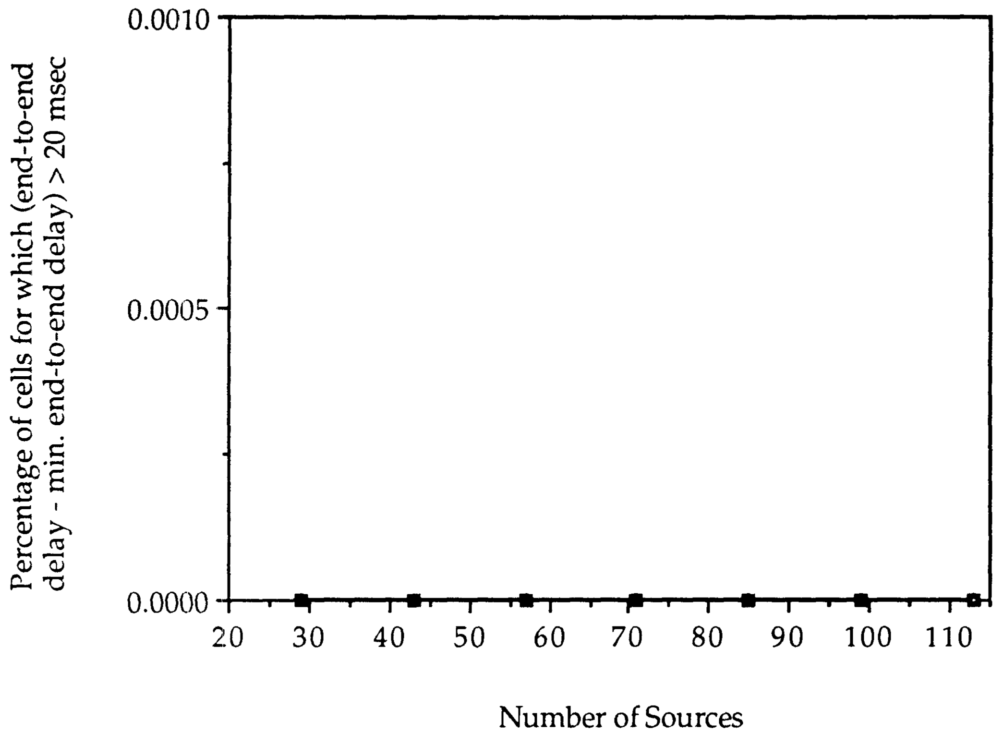

Figure 41. Percentage of Cells for Which the Difference Between End-to-end Delay and Minimum End-to-end Delay Exceeds 20 Milliseconds as a Function of the Number of Sources for Voice and Video in Traffic Scenario 3 Congestion Control Case 2

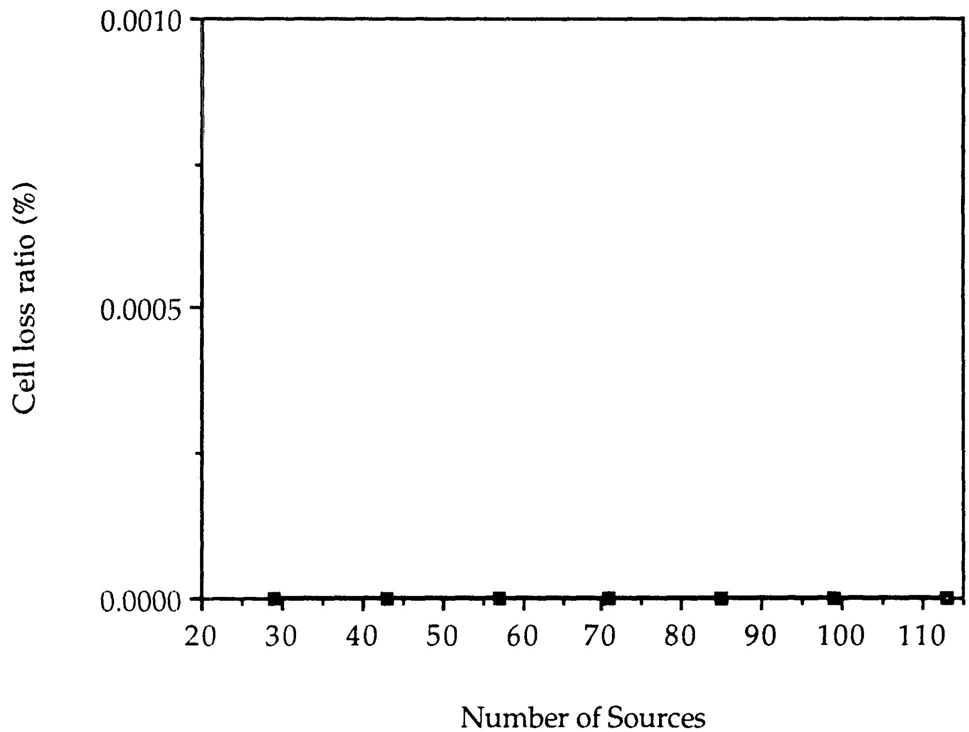

Figure 42. Cell Loss Ratio as a Function of the Number of Sources for Data (File Transfers) in Traffic Scenario 3 Congestion Control Case 2

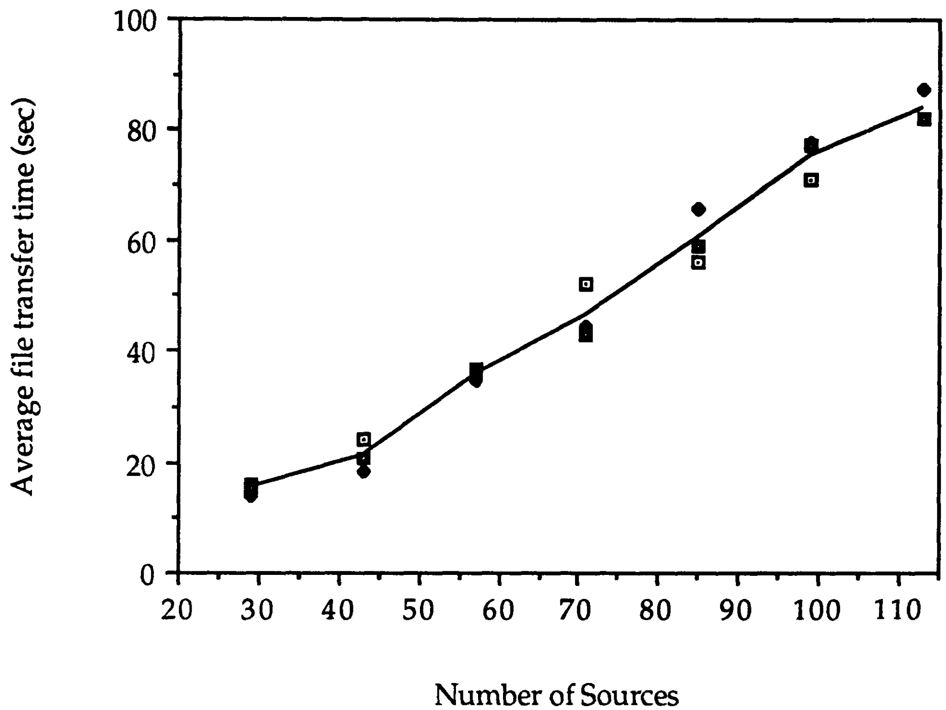

Figure 43. Average File Transfer Time as a Function of the Number of Sources for Traffic Scenario 3 Congestion Control Case 2

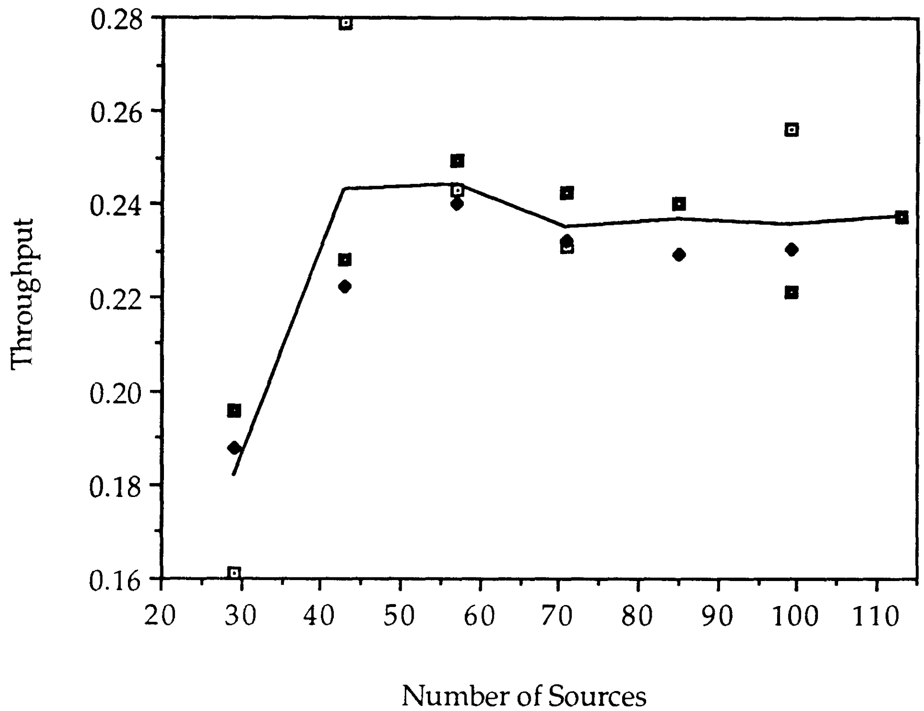

Figure 44. Throughput of Data (File Transfers) as a Function of the Number of Sources for Traffic Scenario 3 Congestion Control Case 2

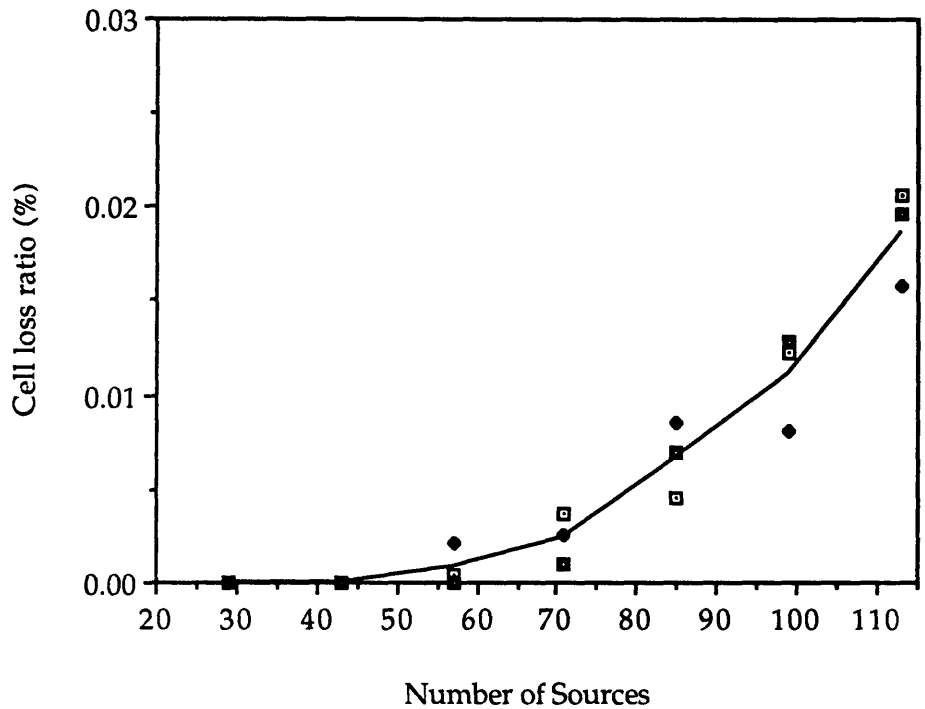

Figure 45. Cell Loss Ratio as a Function of the Number of Sources for Voice and Video in Traffic Scenario 3 Congestion Control Case 3

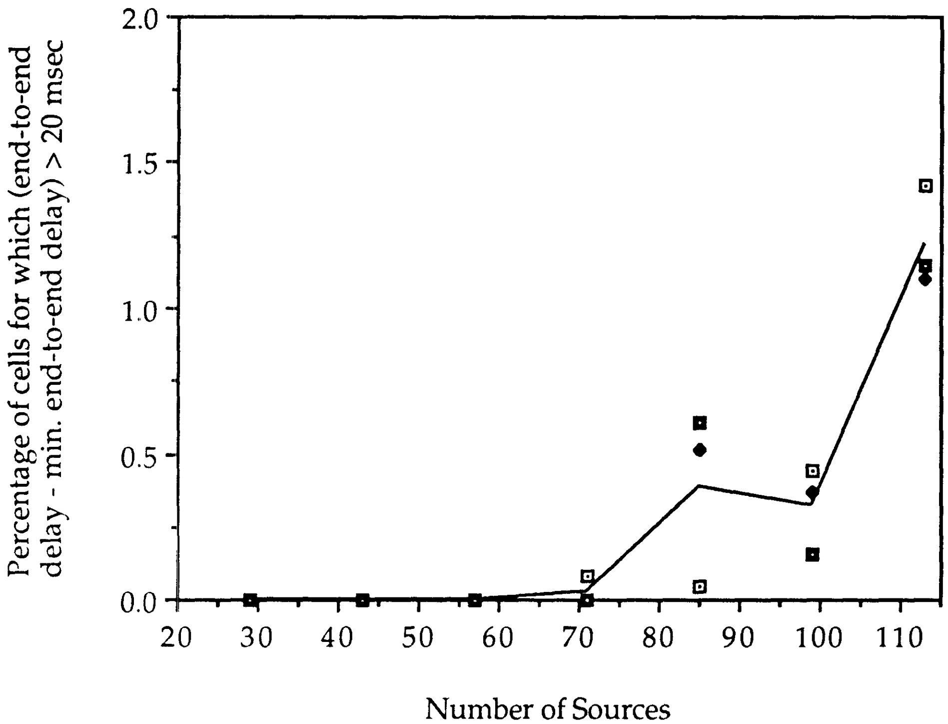

Figure 46. Percentage of Cells for Which the Difference Between End-to-end Delay and Minimum End-to-end Delay Exceeds 20 Milliseconds as a Function of the Number of Sources for Voice and Video in Traffic Scenario 3 Congestion Control Case 3

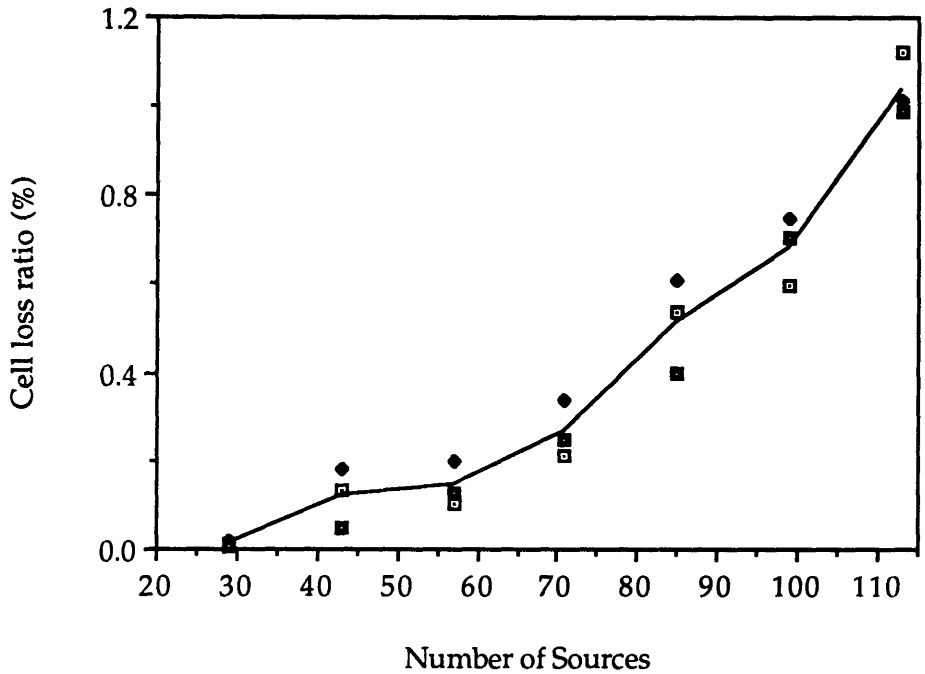

Figure 47. Cell Loss Ratio as a Function of the Number of Sources for Data (File Transfers) in Traffic Scenario 3 Congestion Control Case 3

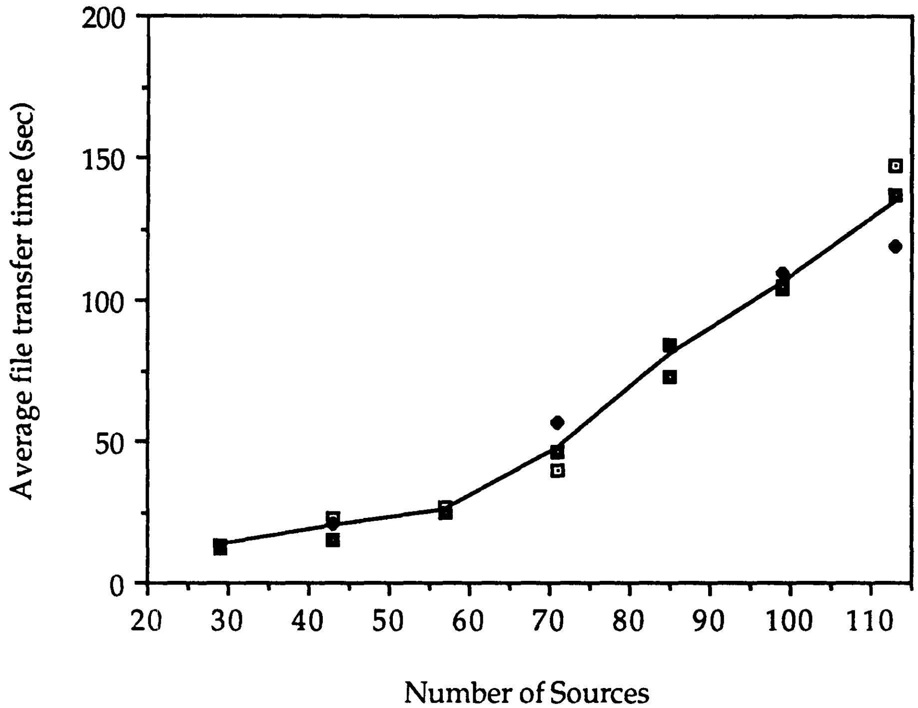

Figure 48. Average File Transfer Time as a Function of the Number of Sources for Traffic Scenario 3 Congestion Control Case 3

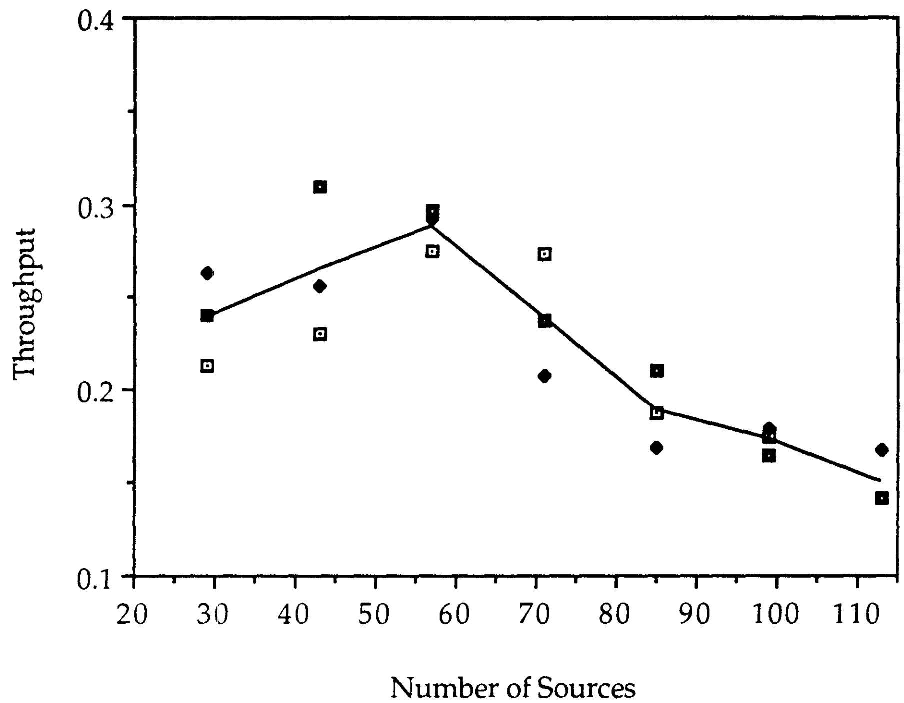

Figure 49. Throughput of Data (File Transfers) as a Function of the Number of Sources for Traffic Scenario 3 Congestion Control Case 3

9. Conclusion and Suggestions for Future Work

9.1 Conclusion

In this thesis, a peak rate controller, which polices the peak rate of a connection by buffering and spacing the cells was presented. Furthermore, a proactive congestion detection and notification algorithm based on four internal traffic classes was presented. The algorithm allowed the network to request data sources which exceed their nominal bit rates to slow down before affecting real-time voice and video traffic. A congestion detection mechanism based on the input rate of traffic into the transmit link queue was utilized, together with a rate control policy at the sources. Leaky buckets were incorporated in the access policing

As shown in Chapter 8, the proposed congestion control algorithm satisfied the users' near zero cell loss QOS goals for both real-time and nonreal-time traffic. Furthermore, the delay and delay jitter requirements for voice and video traffic were also satisfied and the data throughput for file transfers was maximized. Also, the goal of link utilization between 80 and 90 percent was achieved. In conclusion, the current work showed superior network performance of rate-based congestion control over a simple congestion control algorithm based on cell discarding.

It is important to note from the discussion in Chapter 8, that while the congestion control scheme based on discarded cells (Case 3) seemed to provide higher throughput for voice and video, the users' cell loss and delay jitter

QOS requirement was not satisfied. In order for ATM to provide both nearly zero cell loss and small jitter for real-time traffic, and nearly zero cell loss and maximum throughput for data traffic while at the same time maximizing link utilization, policing mechanisms (such as leaky buckets and peak rate controllers) and preventive congestion and rate control algorithms appear to be necessary.

9.2 Suggestions for Future Work

The congestion control algorithm presented in this thesis is designed to be general and independent of the network topology, size and link type. This thesis work presents the results of the algorithm tested for a linear network depicting one satellite connection and near zero control information delay. The algorithm needs to be further tested for larger networks with cross traffic and more realistic control message delays to verify its potential as a general congestion detection and rate control algorithm.

The simulations do not take into consideration the fact that file transfer cells that are discarded must be retransmitted by the sources. It is worthwhile to note that possible retransmissions would not affect the results for Case 2, as the data cell loss ratio is zero. Therefore, the conclusion that the proposed input rate-based congestion detection algorithm and the rate control at the sources, namely simulation Case 2, provides superior network performance, is not changed by the fact that retransmissions are not simulated. In fact the inclusion of data retransmissions is only likely to render the uncontrolled case and the simple congestion control case more unattractive, as in both cases a large number of cell retransmissions would be required.

As discussed in conjunction with the voice and video traffic simulation results in section 8.1, the voice and video traffic are allowed to slow down their bit rates to 50% of the nominal bit rate values. Further evaluation and testing is needed to assess the audio and visual quality of voice and video traffic which is occasionally transmitted at 50% of the nominal rate.

In this thesis work, the capacity of the satellite link is fixed at 6 Mbps. In future work, the benefits of dynamic bandwidth allocation, which is of interest in satellite networks, should also be explored.

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