

# Development of a Congestion Control Scheme for ATM Networks

by

Adnan Ahmed Khan

A Thesis Presented to the

FACULTY OF THE COLLEGE OF GRADUATE STUDIES  
KING FAHD UNIVERSITY OF PETROLEUM & MINERALS  
DHAHRAN, SAUDI ARABIA

In Partial Fulfillment of the  
Requirements for the Degree of

**MASTER OF SCIENCE**

In

**COMPUTER ENGINEERING**

October, 1996

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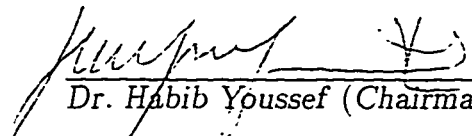
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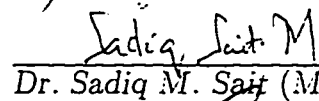
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DHAHRAN, SAUDI ARABIA**

**COLLEGE OF GRADUATE STUDIES**


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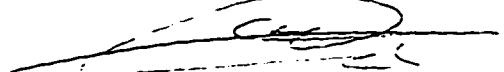
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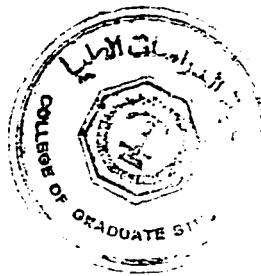
  
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Dr. M. S. T. Benten (Member)

  
Department Chairman

  
Dean, College of Graduate Studies

Date 7/10/1996



To

*my mother and my father*

## Acknowledgment

In the name of Allah, Most Gracious, Most Merciful. Read in the name of thy Lord and Cherisher, Who created. Created man from a {*leech-like*} clot. Read and thy Lord is Most Bountiful. He Who taught {the use of} the pen. Taught man that which he knew not. Nay, but man doth transgress all bounds. In that he looketh upon himself as self-sufficient. Verily, to thy Lord is the return {of all}.

(The Holy Quran, Surah 96)

First and foremost, all praise to Allah, *subhanahu-wa-ta'ala*, the Almighty, Who gave me an opportunity, courage and patience to carry out this work. I feel privileged to glorify His name in the sincerest way through this small accomplishment. I seek His mercy, favor, and forgiveness. And I ask Him to accept my little effort. May He, *subhanahu-wa-ta-Aaala*, guide us and the whole humanity to the right path (*Aameen*).

I thank my beloved mother for everything she has done. For me, she is the greatest living human. I owe so much to her.



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## Abstract

**Name:** Adnan Ahmed Khan  
**Title:** Development of a Congestion Control Scheme  
for ATM Networks  
**Major Field:** Computer Engineering  
**Date of Degree:** October, 1996

*Congestion control is of critical importance for the proper functioning of ATM networks.*

*The congestion control issue is difficult because of the large bandwidth-delay product of ATM networks.*

*In this thesis, a statistical approach of congestion control in ATM networks has been proposed. The statistical scheme operates differently from the previous congestion control schemes. It takes into account the statistical nature of the sources when reacting to congestion. It uses hypothesis testing to minimize the number of feedback throttle-down signals. Extensive experimentation has been carried out to relate the cause of congestion to the statistical nature of the traffic sources. The performance of the the proposed statistical scheme is measured through extensive simulation studies.*

Master of Science Degree

King Fahd University of Petroleum and Minerals

Dhahran, Saudi Arabia

October, 1996

## خلاصة الرسالة

الاسم: عدنان أحمد خان

العنوان: تطوير نظام التحكم في ازدحام الإشارات لشبكات التبادل غير المتزامن (ATM)

التخصص: هندسة الحاسب الآلي

تاريخ التخرج: أكتوبر ١٩٩٦ م

إن للتحكم في ازدحام الإشارات أهمية بالغة فيما يخص العمل الجيد لشبكات التبادل غير المتزامن. ويعتبر التحكم في ازدحام الإشارات مسألة صعبة بسبب قيمة ضرب سعة الموجة في الزمن التأخير لشبكات التبادل غير المتزامن.

في هذه الأطروحة يتم اقتراح طريقة مبنية على إحصاءات للتحكم في ازدحام الإشارات في شبكات التبادل غير المتزامن. يعمل نظام الإحصاءات بطريقة تختلف عن النظم المعروفة سابقاً في التحكم في ازدحام الإشارات، فهو يأخذ بعين الاعتبار طبيعة الإحصاءات المصادر حين تفاعلها للازدحام. ويستعمل فحص الافتراضيات لتقليل من عدد الإشارات المنخفضة بواسطة التغذية الخلفية. وقد أقيمت عدة تجارب من أجل الربط بين الازدحام وطبيعة الإحصاءات للإشارات الواردة من المصادر. إن أداء نظام الإحصاءات المقترح تم قياسه من خلال دراسة مكثفة للمحاكاة.

درجة الماجستير في العلوم

جامعة الملك فهد للبترول والمعادن

الظهران - المملكة العربية السعودية

أكتوبر ١٩٩٦ م

# Chapter 1

## ATM Networks

Broadband Integrated Services Digital Network (B-ISDN) is thought by many to be the network technology of the future. The target of B-ISDN is to merge the disparate set of networks that exist today, into a single unified infrastructure, capable of supporting all types of communication services. B-ISDN is based on a cell-based transport technology called Asynchronous Transfer Mode (ATM). The ATM standard defines a fixed cell size and structure into which all data is encapsulated. Cells are routed by identifiers present in their respective headers. ATM may be used to support connection-oriented as well as connectionless services. It allows statistical multiplexing to be used to aggregate traffic from many sources exhibiting different characteristics. Users may request connections with a certain Quality of Service (QoS) by specifying values for call attributes, such as cell delay and loss. Traffic sources may be constant bit rate (CBR) or variable bit rate (VBR). ATM

connections may support video, voice, and data traffic.

## 1.1 ATM Protocol Hierarchy

ATM networks have layered architectures. However, the functional layering in the B-ISDN protocol model does not follow the OSI model. The B-ISDN/ATM protocol hierarchy consists of the *physical medium-dependent layer* (PMD), the *ATM layer*, the *ATM adaptation layer* (AAL), and the *higher service layer(s)*. The PMD layer is responsible for the proper bit transmission and performs functions which are necessary to insert/extract the cell flow into/out-of a transmission frame. The ATM layer contains all the details of the ATM technique, and is common to all services. This layer is physical medium independent. The data unit of this layer is the cell and the ATM layer performs the function of delivering cells from the source to the destination. This layer also performs the cell-based multiplexing/demultiplexing and cell delineation. The AAL and the higher service layer(s) of the ATM protocol model are service-dependent. The boundary between the ATM layer and the AAL corresponds to the differences between functions applied to the cell header and functions applied to the information field. The AAL provides the higher service layer(s) with the necessary functions which are not provided by the ATM layer.

## 1.2 ATM Cell Structure

The ATM cell is 53 bytes long. The fixed cell size was chosen so that the processing at the switching nodes was made simpler. The cell size may seem to be too small, especially for data transmissions. However, a small cell size favors the transmission of real-time traffic. This is because of the fact that a small cell size results in lower delay jitters. The ATM cell structure is shown in Figure 1.1. The first five bytes

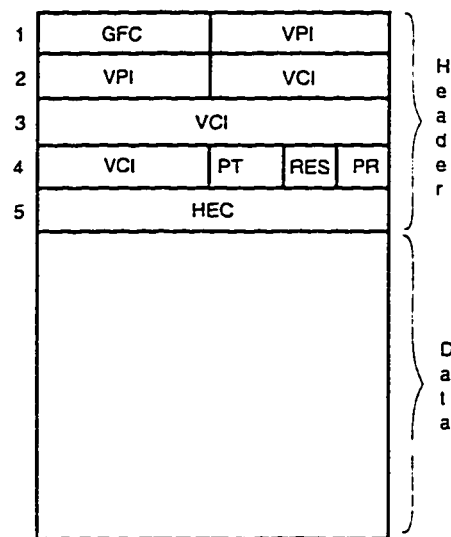


Figure 1.1: ATM cell structure.

of the cell constitute the header and contain control information. The remaining 48 bytes are used to carry data. This cell format has been approved by CCITT. The 4-bit *generic flow control* (GFC) field is used to assist the customer premises in controlling the flow of traffic for different qualities of service. The *virtual path*

*identifier* (VPI) field provides an explicit path identification for a cell, while the *virtual channel identifier* (VCI) field provides an explicit circuit identification for a cell. Also present in the header are the *payload type* (PT) field, a 1-bit *reserved* (RES) field and a 1-bit *priority* (PR) field.

### 1.3 VP/VC Concepts

ATM cells are routed based on the Virtual Path Identifier (VPI) and Virtual Circuit Identifier (VCI) fields present in the header. A unique identifier, VPI/VCI, is associated with each VP/VC at each local VP terminator/cross-connect. VCs sharing the same VP are distinguished by their VCI values, which remain constant throughout the same VP. A VP consists of a grouping of VCs. A VC consists of a concatenation of VPs. The segment of an end-to-end connection that extends between VC switching points is called a virtual channel link. A concatenation of virtual channel links constitutes a virtual channel connection. Similarly, the segment of an end-to-end connection that extends between VP switching points is called a virtual path link. A virtual path connection is a concatenation of VP links over which the VCI fields in each cell remains unchanged. These concepts are illustrated in Figure 1.2.



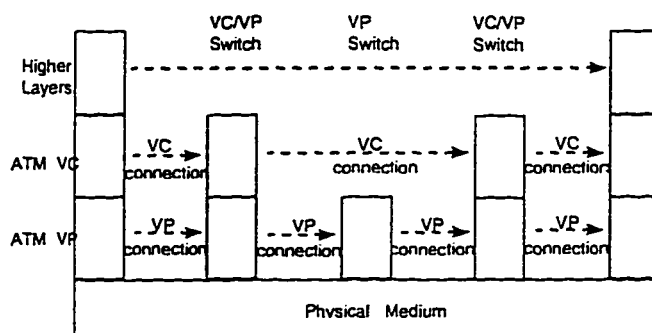


Figure 1.2: VP/VC concepts.

## 1.4 ATM Traffic Control Issues

ATM supports constant as well as variable bit-rate sources. Bandwidth allocation for CBR sources may be a trivial task, but that is not the case with the VBR sources. VBR sources are usually bursty in nature. That is, the VBR source would transmit at its peak rate for short periods and would remain dormant for the rest of its lifetime. Peak rate allocation could result in a severe under-utilization of the network. Therefore, a fundamental question is, how much bandwidth should be allocated to a variable bit rate source? Statistical multiplexing is done to get the maximum utilization of the network. As a result, the sum of the peak rates of all the sources present on a link could be greater than the link capacity. Thus, there exists a non-zero probability that all or many of the sources may start delivering at their peak rates resulting in unavoidable congestion. The goal of a good congestion control scheme is to minimize the damage caused by the congestion, while at the

same time it must allow the network to operate at reasonable utilization.

Congestion control, at present, is probably the most hotly debated issue in high speed networking. The issue has taken an unprecedented importance now with the universal proliferation of *ATM* [30]. What is congestion and what causes it? Congestion occurs when the network is unable to provide the service performance to the offered load. This may result in cells being lost, the throughput dropping and response time increasing. The cause of congestion is a well understood issue. However, there is no agreed upon congestion control strategy. It is generally argued that congestion can be controlled trivially by increasing the buffer size at the nodes, increasing the link capacities and using faster processors. These elements, no doubt, will help in easing congestion, but the actual problem is that of devising a suitable network architecture, and a set of protocols capable of handling the different scenarios that may occur during the operation of the network. Congestion is not a *static resource shortage* problem; rather, it is a *dynamic resource allocation* problem. Network is a dynamic entity, specially now as we encounter network of networks. Also, the offered load is very dynamic. This calls for an adaptive congestion control strategy that would take the characteristics of the sources into account while tackling congestion [4, 23].

The problem is exacerbated by the fact that in *B-ISDN*, which uses *ATM* as its transfer mechanism, every connection is guaranteed a minimum level of service, for example maximum delay and bandwidth. This creates a very conflicting and

multidimensional problem. With the advancement in technology, the bottleneck in high speed networking has shifted to the medium propagation delay. The channel transmission rates have become so high that a large number of cells may be travelling on the medium at a given time. If congestion develops, a large number of cells may be lost and positive feedback may occur, that is. the sources would retransmit the lost cells again and again, resulting in a complete system shutdown. *Preventive measures* are called for which prevent the conditions that would lead to the system getting congested [40]. But these schemes are inherently conservative and under-utilize the system resources. Further, their implementation is not trivial.

The *reactive schemes* on the other hand allow the network to operate at higher efficiencies but risk congestion. These schemes rely on some feedback being conveyed from the point of potential congestion to the controlling point in the network [4, 10, 23]. The mechanisms include:

- feedback messages, *chokes*, calling on the sender to reduce its transmission rate. But these choke cells themselves consume valuable processing power.
- *delay adaptive routing* as used in ARPANET. This is another alternative, but also increases the control traffic.
- a trivial mechanism rejecting the cells without sending any feedback. This is very simple to implement but could create positive feedback. That is, the sender would have to retransmit these dropped cells, thereby sending more

traffic on an already congested network and aggravating the situation.

- sending *probe cells* by the sources into the network and monitoring the delay incurred by these cells. The sources then adjust their transmission rate accordingly. But, this is a very approximate method as the information extracted from these cells is time-delayed and the occurrence of congestion in a high speed network can dramatically change the situation.
- the introduction of *feedback fields* in cells. But the performance of such scheme is not consistent, because there may or may not be traffic going the other way.

## 1.5 Desirable Properties of Congestion Control

### Mechanism

The difficulty of the congestion problem lies in the fact that there are conflicting requirements placed on a congestion control scheme [20, 23]. Some of the requirements are:

- the scheme should be fair. This is an important factor at high loads but being fair may degrade the system performance depending upon the definition of fairness.
- it should incur low overheads. This implies as little control as possible which may lead to the loss of contact between the control point and the point of

congestion.

- it must be such that it always tries to get the maximum out of the network while maintaining the system congestion free. This implies that the scheme asks the users to increase the traffic when network is under-utilized and vice versa. But that is a very imposing assumption given that the traffic may be real time and of heterogeneous nature,
- the control should be distributed. With a central controlling node, the control would be lost if that node goes down. Also, the effect of the control would depend on the distance from the controlling node. This would result in an asymmetrical network,
- the congestion control scheme must work in any combination of bad events, that is under congestion, high error rates and high cell losses, and
- it must be such that it has an overall view of the network while catering to the needs of the individual user.

Dealing with congestion in ATM networks is particularly difficult. This is due to the fact that *statistical multiplexing* is used in these networks to achieve better network utilization. If the sources are allocated their peak bit rates then the network may be highly under-utilized because of the fact that most of the sources are bursty in nature. This statistical multiplexing of the sources makes the congestion control

problem a highly non-trivial task. The probability of all the sources delivering at their peak rates is non-zero and in that case the network would certainly be in a state of congestion and if adequate measures are not taken then the promised *Quality of Service* cannot be guaranteed.

The important events occurring in the network can be classified as: users coming to and users leaving the network. These events, coupled with the fact that the requirements and the characteristics of the sources may vary widely, make the task of predicting the network behavior an awesome task. Therefore, many simpler techniques have been proposed that either allow the network to work only in a region where the *statistical multiplexing gain*<sup>1</sup> is close to one. That is, these techniques are congestion avoidance techniques. In other techniques, the network is allowed to enjoy higher statistical multiplexing gain and some feedback mechanism is used at the congestion site to signal the sources to somehow throttle-down.

Yet another class of congestion control has been suggested, which segregates the traffic into two types: real time and non-real time. The problem with these techniques is that they don't exploit the statistical nature of the different types of traffic present in the network. This exploitation is by no means an easy task, however, an effective control of congestion without it is difficult to imagine.

---

<sup>1</sup>The ratio of the capacity allocated to the network capacity.

## 1.6 A Detailed Look at Congestion and Related Issues

This section presents a detailed description of the possible types of congestion in an ATM network. This is followed by a discussion of the various mechanisms used for congestion control.

### 1.6.1 Types of Congestion

The types of congestion occurring in a computer network can be classified according to their duration of occurrence [4, 12, 23]:

- The increasing ratio of propagation delay to transmission time means that the switch can be suddenly overcrowded with a large number of cells for such a short time that there is no time for the feedback protocol to work and can result in a loss of a considerable number of cells. To quickly fight back this sort of congestion, efficient and *local measures* have to be taken.
- For longer durations, congestion feedback can help to inhibit the source and allow the switch to recover.
- If there are long persistent congestion periods then this indicates that new resources should be installed.

## 1.6.2 Policies Effecting Congestion

Network policies may determine the extent of congestion possible. These policies, which are often dictated by the commercial aspect of the network, are:

- *Cell queuing and service policies* in the switching nodes which affect resource allocation among users and may determine the extent of congestion possible.
- *Cell dropping*, if there is insufficient space in the buffer or even if there is space in the buffer but the deadline of the cell has passed.
- *Multipath routing*. This issue takes importance when the shortest source to destination path is over-utilized [5].
- *Lifetime control*. Some measure has to be taken to check that cells are not allowed to roam the network for arbitrarily long times .
- Estimation of delay and timeout interval. A good judgement could result in the network being guarded efficiently against congestion.
- *Buffer management policies* which affect the rate at which the cells can be accepted and directly influence congestion.
- *Congestion isolation*. The congestion control scheme must be such that it restricts and remedies the congestion without affecting other parts of the network.



- Multiplicity of the types of traffic sources present in ATM networks. In B-ISDN, voice, data and real time video are allowed and all have different delay and cell loss requirements. The network must be able to take a decision as to what is to be done in case of congestion, like which cells are to be dropped [4, 10, 17, 24, 39].

### 1.6.3 Admission Control

B-ISDN provides each connection a guaranteed *QoS* (quality of service). For this reason there has to be a reliable check before allowing a new connection into the network, because the protocol has to make sure that the *QoS* of the existing connections are not affected. Also it has to know what are the characteristics of the new connection so as to determine if the network is in a position to furnish the required services [4]. This task is far from trivial. The presence of a large number of sources having different and dynamic characteristics together with the high transmission rates make the problem very difficult, if not intractable. The mechanism which has to decide as to whether to allow an incoming call to go through or not has to be devised relating the network performance parameters with the traffic descriptors of the incoming call and predict the resulting network performance. Therefore, an accurate description of the state of the network as well as that of the characteristics of the incoming call is required so that a decision be made on the fly as to whether the call should be accepted or not. The *network parameters* that are considered to

be most descriptive of the network performance are:

- *Cell transmission delay*, giving the average transmission delay encountered by the cells . It depends on the buffering employed in the system,
- *Cell loss probability*, if the amount of buffering in the network is unlimited (say), then there would be no cell loss, but the delay encountered would be too prohibitive. On the other hand if transmission delays are to be minimized then the cell loss probability would increase, and
- *Variance in cell transmission time*, to provide the connection with promised QoS, variance in cell transmission time should be minimal.

Among the above mentioned metrics, the cell loss probability is considered to be the single most dominant indicator of the network performance. An important problem is how to accurately calculate the cell loss probability. ATM network with fiber-optic lines is too fast in case a time-averaged value of cell loss probability is used, and would result in inaccurate reflection of the state of the network. Cell loss probability could change drastically once congestion occurs and could remain high for sometime even after congestion ends. If the time averaged quantity is used, it would be too low as compared to the instantaneous loss probability at the time of congestion.

The variance in the transmission delay is also an important decision criteria. Its value is related to the buffering in the network, the greater the amount of buffering the higher will be the variance. These are the important network parameters.

Accurate traffic parameters of the incoming call have to be furnished also, so that the protocol can relate the network parameters with the traffic descriptors and decide about the fate of the call. This is an active research area and several traffic descriptors have been proposed. IEEE has described *burstiness* as descriptive of the nature of the call. But this term can have several mathematical interpretations. The most accepted are the ratio of peak to average bit rate and the average burst length. If just the peak rate is considered, it could give a very inaccurate description, because the bursts might be of very short length. A useful descriptor could be the *effective bit rate*, described as  $peakrate \times a$ , where  $a$  is a constant with a value less than or equal to 1. The value of this constant would be dependent on the nature of the call. For a constant bit rate connection this constant would be 1 [4, 23].

The simplest admission control scheme would be to allocate each incoming call a bandwidth equal to its peak bit rate. This is the simplest technique, but would result in a major waste of the resources if most of the incoming calls have variable bit rate. The advantage of statistical multiplexing would be obtained if a call is allocated a bandwidth less than its peak rate. What would be the bandwidth allocated and how is the protocol going to ensure that all calls are treated fairly and disproportionately high amount of bandwidth is not allocated to a single source-destination pair? One solution could be that based on the equivalent bit rate of a call. A call is allocated bandwidth if this allocation does not push the expected bandwidth usage for a link over the threshold of available bandwidth. Implementation of this scheme is simple,

but if a high bandwidth call is allocated its equivalent bandwidth then it would result in the excessive blocking of smaller calls. This scheme does not take into account the fact that an unfair amount of the available bandwidth might be allocated to a single source-node. One solution could be to give preference to the smaller bandwidth calls, causing the average blocking rate to come down, but would result in higher blocking rates for high-bandwidth calls. The concept of *virtual paths* (VP) is introduced, which is a logical link between a source-destination pair. Each VP can initially be allocated a small amount of bandwidth and as the demand of each increases, the respective bandwidths can be increased. The schemes can have several variations, but a major point is that the implementation of any scheme should be simple and time efficient, so that it is practical to be implemented. In any admission control scheme every node along the source-destination route has to run the admission control scheme and determine the status of its available resources. In case of call rejection or call acceptance, each node has to update its state. A variation is the use of pseudocells, which are processed only at the source nodes, and save the processing costs at the intermediate nodes. Yet another proposed alternative is to divide the available bandwidth into pools. Each pool accommodates calls requiring (almost) the same class of service. This would prevent low priority calls from being drowned by the high priority calls. The number of pools and the bandwidth allocated to each pool is an open research area [4, 10, 17, 37].

#### 1.6.4 Bandwidth Enforcement

Once allowed access inside the network, the state of each connection has to be continuously monitored since the connections may *misbehave* [4]. Several approaches have been suggested in this regard. The most prominent among them is the *Leaky Bucket* mechanism and its variations. Other schemes include Killing, Jumping, and Moving windows, or Exponentially weighted moving average [4]. All have comparable performance since their control mechanism is based on the estimation of the average bandwidth allocated to each connection. Bandwidth enforcement measures are needed for *each virtual connection* that exists. Their position in the network is at its edges. Once a decision is taken to allow the call to enter the network, a leaky bucket is allocated to the connection. The original leaky bucket mechanism consists of a pool of tokens that are generated based on the average bandwidth allocated to the connection. The tokens are generated periodically until a threshold value, at which the generation stops. The token generation rate is dictated by the characteristics of the traffic. For example, in case of a constant bit-rate source, the tokens would be generated at a constant rate, equal to the rate of transmission of the source. If the cells arrive at a higher rate than this, they are dropped or marked depending upon the implementation of the leaky bucket mechanism [20]. The scheme behaves very well with respect to delay but may result in a high cell loss. For enforcing peak bandwidth, the tokens can be generated at the corresponding peak rate.

A variation of this scheme is the *Buffered Leaky Bucket Scheme* in which there is an input queue at the bucket. When there are no more tokens available, cells are stored in the queue until it fills. Overflowing cells are dropped. Assuming a shared pool, the limiting case is that no cells will be dropped but the associated delay will be prohibitive. The mechanism is quite approximate in the sense that it may prove unfair to the sources that are transmitting cells under the average rate but suddenly send a burst of cells, the burst may exceed the capacity of the leaky bucket and cells may get dropped, even when the average rate is lower than what was decided at the call setup time. To alleviate this problem the size of the token pool may be increased but that would be like allowing the misbehaving sources to escape the check. As a result none of the proposed schemes seems to be satisfactory as they all are intrinsically unfair or incapable of providing a perfect policing mechanism.

This weak behavior can be remedied by combining the bandwidth enforcement with the congestion control mechanisms. This merger results in what is called as the *Virtual Leaky Bucket Mechanism*. In this scheme, the excess cells are not dropped but are allowed to enter the network with a *marking tag*. These marked cells move normally in the network unless congestion occurs, in which case the marked cells are dropped. Only when there are no more marked cells to be dropped that the unmarked cells are selected for dropping. This scheme allows for the optimum use of the network resources but is probably unfair for the connections behaving properly because of the fact that the marked cells will be the ones that cause congestion and

cause the *good sources* to suffer delay. Also some overhead processing at the nodes will be required to differentiate between the marked and the unmarked cells and to recognize that congestion has occurred, so that the marked cells can be dropped. But, it is generally believed that this processing load is not considerable as compared to the advantage that the scheme offers.

Another variation is the use of a *Buffered Virtual Leaky Bucket* approach in which the cells are marked when the queue length is beyond a certain threshold. The performance of the scheme is dependent on the value of the threshold. A *spacer* can also be used along with the bucket so that the maximum rate at which the cells enter into the network can be controlled [10, 17, 23]. A spacer is usually placed at a *User Network Interface (UNI)* and is used to limit the rate at which traffic can enter into the network. This helps in preventing a node from being suddenly overwhelmed by a large number of cells. A simple spacer can be implemented with the help of a larger buffer.

Another popular policing algorithm is the *Dangerous Bridge* or the sliding window type algorithm [34]. If it is assumed that the average cell rate be defined by parameters  $T$  and  $X$ , where  $T$  is the time period that defines the average cell rate, and  $X$  is the maximum number of cells that can be generated in any time period of length  $T$ . The Dangerous Bridge counts cells arriving in any phase of time period of  $T$ , and polices traffic so that the maximum number of cells never exceeds  $X$ . The algorithm requires memorization of cell arrival information for  $X$  cells, and

this requirement increases the hardware amount when  $T$  is large. Leaky bucket, on the other hand requires relatively smaller hardware, because it is a counting type algorithm. It counts *up* at cell arrival and counts *down* at the average rate. A comparison has been done between the two approaches in [34]. It is concluded that the Dangerous Bridge algorithm provides a more effective solution, while the Leaky Bucket algorithm is not effective in properly identifying the worst case traffic of the source.

An alternate method is the use of a *traffic smoothing* mechanism at the user-network interface. This is a *rate control mechanism* which limits the maximum rate at which the cells can enter the network. This helps to keep the network in a stable state but may result in increased delays and cell losses due to the buffering at the interface. A *priority shaper* is presented in [32]. This scheme is based on a network employing leaky bucket algorithm for bandwidth enforcement. Due to the failure of leaky bucket in accurately modeling the average behavior of the source, cells could get lost in case a burst of length greater than the number of tokens in the bucket occurs, even if the overall rate is within the average rate predicted at the time of call admission. To minimize this loss, a *ghost* policer is implanted at the source, which will smoothen the traffic entering into the network, thus avoiding bursts of excessive length. However, this would result in a higher average transmission delay. To minimize this delay, two classes of traffic are defined for the source. One class is the higher priority class, like real-time video which is sensitive to delay and to a lesser



extent to loss. The other class is the delay insensitive class, like data transfer. The shaper would allow the cells of higher priority class to go through but would hold the cells of non-real time calls, thus making an acceptable trade-off. The operation of smoothing can be mimicked in the leaky bucket through the use of the spacer. Even when the spacer is not used the leaky bucket approach is considered more pragmatic as it results in an efficient use of the network resources. Further, the leaky bucket mechanism is flexible because of its ability to pass the excess cells into the network, allowing an optimal use of the resources. From a system point of view the leaky bucket provides a good alternative. This is because of its ease of implementation. However, from a user point of view the leaky bucket approach is not so good. This is due to its approximate nature of operation [4, 20, 32].

### **1.6.5 Role of Routing in Congestion Control**

Routing is a major issue in any network design. It is a dynamic problem which depends upon the topology of the network and the different traffic types present. In ATM the network is very sensitive to any scheme that does not use the network resources in a balanced manner [5]. A bad scheme may cause congestion in any part of the network which may ripple throughout. Routing is intricately related to the link capacities, the location of the sources, the processing speeds of the nodes, the buffer capacities, and the backbone design.

There are a number of routing schemes that are proposed for use in ATM. It is

generally argued that the routing scheme be designed keeping congestion control in perspective. The two factors affecting the routing strategy should be the cell loss probability and delay. If a scheme is developed based on just the delay, then the routing algorithm always looks for the shortest path irrespective of how congested it is, which will result in an increased cell loss probability and congestion, while the other links may remain under-utilized. Some techniques use just the cell loss probability to calculate the source-destination path. Several schemes are proposed for routing, such as: shortest path, alternate path, random select, reservation and multipath routing schemes [5]. Studies have been done comparing the performance of these schemes and it is claimed that multipath routing provides the overall best results under all load conditions. The shortest path scheme is very simple to implement but may cause system congestion even if the other links are under-utilized. The alternate path scheme shows oscillations and results in the over-utilization of the shortest path resulting in a higher cell loss probability. Random select strategy wastes system resources and shows the poorest results. It tries to use the alternate paths even at light loads. The call blocking probability is the same as the alternate strategy. The reservation scheme shows some oscillations between shortest and alternate paths and over-utilizes the shortest path which results in cell loss. The call blocking probability is the same as multipath strategy. The multipath routing scheme is similar to the alternate routing scheme, but has important differences. It uses implicit trunk reservation, causes proportional distribution of load throughout

the network, and exhibits a natural load balancing property. Call blocking is lesser because of the use of alternate paths. It tries to use the alternate paths before the shortest path is over-utilized [5].

### 1.6.6 Priority Schemes

Priority schemes are an essential part of ATM as there are several different classes of traffic types supported by B-ISDN. Each type has different requirements of loss and delay. Assigning priorities to the cells for servicing is a difficult task. The simplest solution is the *first come first serve* (FCFS) scheme. However, application of this scheme may result in a degradation of service provided to the *delay-sensitive traffic*.

An effective static scheme is to give preference to delay-sensitive traffic. But this would result in severe degradation of *loss-sensitive traffic* if there is a sudden burst of delay-sensitive traffic.

To overcome this drawback two schemes are proposed. In one scheme, called *Maximum Laxity Threshold* (MLT), a counter is associated with every delay-sensitive cell which is decremented at regular intervals. If the count reaches zero it means that the deadline of the cell has expired and it is no longer important to forward this cell and is therefore dropped. Delay-sensitive cells bearing a count less than a threshold value are given preference, otherwise loss-sensitive cells are preferred. In the other scheme, *Queue Length Threshold* (QLT), priority is given to the loss-sensitive scheme if their number exceeds a threshold, otherwise delay-sensitive cells

are preferred. The QLT scheme is preferred because of its simplicity and reasonable performance [4, 20].

Within the delay and loss sensitive cells there would exist multiple classes, each with different requirements. This could be implemented with multiple queuing schemes in which the input buffer is divided into subsets each holding cells of different priority levels. Service is given to the cells of a given priority only if all the higher order queues are empty. In order not to delay a cell arbitrarily, the priority of a cell in a lower level is increased after a certain time interval so that it may be given a fair treatment [4].

### 1.6.7 Use of Priority in Congestion

Priority schemes can be applied to keep congestion in check. As discussed earlier, there are cells of different classes moving through the network. Real-time video is both delay and loss sensitive, while voice is delay-sensitive and data is generally loss-sensitive. Again, there may be multiple sub-classes in each class. If and when congestion occurs, a decision has to be taken as to which cells are to be dropped. Simulation studies suggest that least harm is done if the voice cells are dropped. This results in a high quality of service to the loss-sensitive class. Within each class, cells may be divided as more or less important ones, especially the voice cells class. However, the bandwidth occupied by voice connections is usually much less than that occupied by the video connections. Therefore, the proportion of the

bandwidth occupied by voice calls as compared to that occupied by the video calls in the network at any given time plays an important role in the effectiveness of these schemes [4, 20].

In another scheme, reported in [4], higher priority is given to the cells carrying relatively more important information. For example, in a video connection, higher priority may be given to the cells carrying the synchronization information. In another scheme, cells are numbered as even/odd and even samples are more important than their odd counterparts. In yet another scheme two classes of voice cells correspond to active and semiactive periods.

Another scheme suggests of going to the bit level in each cell and regarding some bits as more important than others, but this directly affects the cell size and offends the notion of transparency and increases the processing overhead.

The same schemes can also be applied to video traffic with some restraints [4, 20].

### **1.6.8 Effect of Error Control Schemes on Congestion Control**

ATM networks use optical fibers. These have a very low error rate, which can be as low as 1 in a billion! This coupled with the fact that the ratio of the propagation delay to the transmission time is getting higher strongly suggests for a different error control strategy than that used in conventional packet-switched networks. Of

course, each cell lost or damaged has to be retransmitted, but the important point is, which layer should be responsible for retransmission and should the strategy be implemented on a *link-by-link* or *end-to-end* basis.

End-to-end schemes are more appealing because of the fact that they help in simplifying the protocols which is a major issue in ATM networks. However, the strategy results in a loss of network resources if the cells encounter an error later in their path. A number of simulation studies have been done and it is generally believed that for low throughput connections there is no significant difference between the performance of the two strategies. As the throughput increases the end-to-end scheme works better because of the better use of buffers and higher saturation point. Also, when the error rates are increased, then at high throughput the end-to-end scheme works better as the buffer overflow problem link-by-link schemes becomes significant. Finally, when the number of intermediate nodes is increased then the performance of link-by-link starts to get better because of the inherent drawback of the end-to-end scheme as stated above. But the performance of end-to-end schemes is better than link-by-link for realistic number of hops. Further, as the link-by-link schemes put extra load on the processing resources and the high ratio of propagation delay to transmission time in ATM, the end-to-end control schemes are generally recommended.

As for the protocol to be used, intuitively one would opt for selective repeat because it saves on the bandwidth. However, it requires *reordering* buffers at the

receiver. In addition, since the error rates are low in optical fibres that the difference in the performance of the two schemes is not significant. However, selective repeat can gain advantage by sending acknowledgements of blocks instead of individual cells as shown by several studies. A cautious suggestion made is that a block acknowledgement scheme coupled with block-based selective repeat transmission protocol be executed on an edge-to-edge basis in high speed networks [4, 23].

The chapter presented an in depth view on the problem of congestion control in ATM networks. The next chapter gives the different traffic source models that have been proposed in the literature for ATM networks.

## Chapter 2

# Traffic Source Modelling

ATM networks must support various communications services, such as data, voice and video, each having different traffic characteristics. To evaluate the performance of such networks, accurate source modeling is required. This chapter examines several traffic source models proposed for data, voice and video sources. The formulation, evaluation, and selection of ATM source models constitutes a broad area of research, rather than a concise problem. This is because of the diverse nature of the traffic sources present in an ATM networks. There is no one model that best fits all the sources, each model has its advantages and disadvantages.



## 2.1 Traffic Source Characteristics

Traffic source characterization is necessary for the precise definition of the behavior of each particular source; it also provides network management with the ability to manipulate flexibly the various services in terms of connection acceptance, negotiation of QoS, congestion control, traffic enforcement and resource allocation. There is a general trend to visualize the cell generation process as a succession of active and silent periods. Cell generation occurs only during active periods. The most prominent paradigm of a source model exhibiting this behavior is the popular ON/OFF model, which is discussed in detail later.

According to ITU-T (formerly CCITT), the following parameters are important in source characterization:

- $p$ : peak arrival rate of the cells when the source is at the active state, or the maximum amount of network resource requested by the source. Alternatively, this parameter may be defined as the reciprocal of the minimum inter-arrival time between two consecutive cells belonging to the same connection.
- $m$ : average cell arrival rate, or the average amount of network resource requested by the source. Two parameters can characterize the average cell rate: the *true average cell rate*, the number of cells measured during the duration of a connection divided by this duration; the *estimated average cell rate*, the number of cells measured during a long time interval  $T$ , divided by  $T$ .

- $\beta$ : burstiness, defined as the ratio between the peak cell rate and the average cell rate. It can be viewed as a measure of the duration of the activity period of a connection.
- $t_{on}$ : the average duration of the active state (peak duration).

An equivalent set of parameters that can be used for the characterization of ATM traffic sources is:

- $R_p$ : peak value of cell rate.
- $N$ : average number of cells sent in a burst.
- $T_i$ : average interarrival time between two consecutive bursts.

An additional traffic parameter is defined as the sustained average cell transfer capacity,  $R_{sa}$ , which is the ratio of the maximum number of cells in a burst divided by the minimum interburst time.

## 2.2 Types of Traffic Sources

In this section, the three categories of traffic sources; data, voice and video are discussed.

### 2.2.1 Data Sources

The generation of data from a single data source can be characterized by a Poisson arrival process (continuous time case) or by a geometric interarrival process (discrete time case). For interactive data transmissions, a single cell may be generated at a time. For bulk data transmission, such as a file transfer, a large number of cells may be generated at a time (batch arrivals). In conventional packet transfer networks, packets could be either of variable or constant length. In ATM networks, however, the cell size is fixed. Due to the small cell size as compared to the length of the packets in existing networks, multiple cells may be created from one data packet. Data services are usually very sensitive to cell-loss, while their respective delay requirements are not so strict.

### 2.2.2 Voice Sources

Voice sources have different QoS requirements than that of the data sources. They are more sensitive to delay as compared to loss. The simplest possible model for a source generating packetized voice is CBR, in which cells are generated continuously at a constant pace. However, a substantial part of a voice conversation (up to 60 – 65%) involves no talk. The analysis of talkspurt periods is quite simple. It has been found in [18] that for a monologue, the talkspurt period can be assumed to be geometrically distributed. For the idle periods, the measured probability

density function (*pdf*) can be approximated by two properly balanced geometric *pdfs*. Furthermore, the traffic produced by multiple voice sources could be described as the superposition of the individual stochastic process characterizing each source. However, this process leads to rather complicated models involving related stochastic processes.

### 2.2.3 Video Source

Video sources are the most difficult to handle among all the traffic types present in ATM networks. These sources are very sensitive to delay and to a lesser extent to loss. The statistics of the traffic generated by a video source are greatly influenced by the nature of the pictures transmitted, by the relation among them, by the quality of the service produced, and by the coding technique adopted. There are two coding techniques used: *frame-buffered* and *non-frame-buffered*. The former encodes the differences between successive images, while the latter encodes such messages independently. A very important function for the analysis and modeling of video sources is the *autocorrelation* function of the traffic generating process. This function is influenced by the above mentioned factors. Due to this fact, there is no unique modeling approach for video sources.

## 2.3 Source Models

In this section a brief survey of the different traffic source models is presented. We begin with the ON/OFF source model, which is the most widely used model, because it offers the best trade-off between tractability and accuracy.

### 2.3.1 ON/OFF Model

In an ON/OFF source model, a single ATM source is modelled as a succession of active and silence periods. Cell generation occurs during the active periods, as opposed to the silence period, in which no cell generation occurs. Different ON and/or OFF periods are assumed independent of each other. Each period is taken as either an exponential or geometric random variable. An illustration of cell generation of an ON/OFF traffic source is given in Figure 2.1. The mean value of the active period is  $a^{-1}$ , while that of the idle period is denoted by  $b^{-1}$ . During the active period, one cell is generated every  $T$  time units. The ON/OFF model can also be described in terms of the traffic source parameters described earlier. The relations are:

$$p = \frac{1}{T}$$

$$m = \frac{a^{-1}}{(T(a^{-1} + b^{-1}))}$$

$$\beta = \frac{p}{m}$$

$$t_{on} = a^{-1}$$

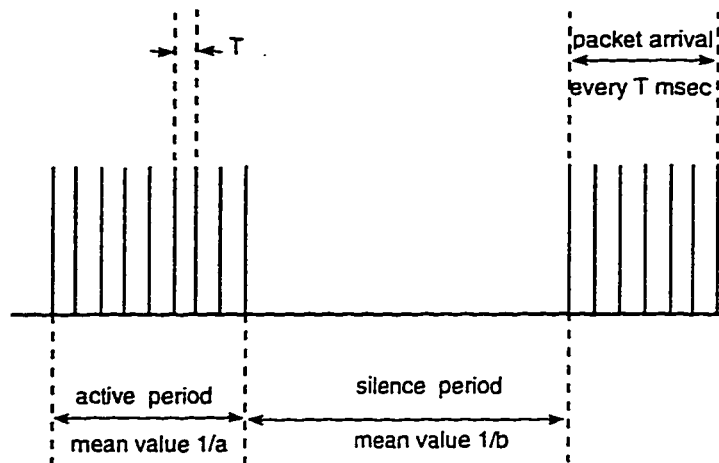


Figure 2.1: ON/OFF source model.

and

$$R_p = 1/T$$

$$N = a^{-1}/T$$

$$T_i = a^{-1} + b^{-1}$$

$$R_{sa} = R_p$$

Despite its simplicity, the ON/OFF model can be used for traffic modelling of B-ISDN services. Also, given its analytical tractability, the ON/OFF source model is by far the most popular in performance evaluation work.

### 2.3.2 Generally Modulated Deterministic Process Model

In a generally modulated deterministic process (GMDP) model, the source can be in one of  $N$  possible states. While in state  $j$ , cells are generated at constant rate  $\lambda_j$ . The time spent in a state is usually geometrically distributed, with a mean sojourn time  $t_j$ . When a state change occurs, the new state is determined according to a transition probability matrix,  $p_{ij}$ , as shown in Figure 2.2. When the GMDP has only two states and one of them has a zero cell-generation rate, it reduces to the slotted-time version of the ON/OFF model (ON/OFF model with geometrically distributed active and idle periods). The GMDP model has shown to be more accurate for data

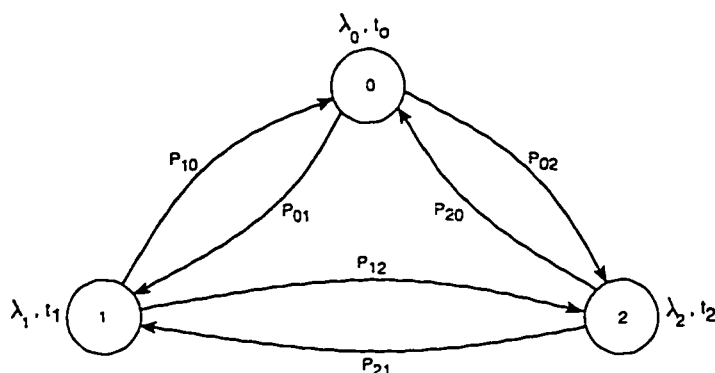


Figure 2.2: GMDP source model.

and voice sources as compared to the ON/OFF model. However, its mathematical tractability is much more difficult than that of the ON/OFF model [11]. It is also shown that the GMDP is not appropriate to characterize video sources.

### 2.3.3 Markov Modulated Poisson Process Model

The markov modulated Poisson process is a doubly stochastic Poisson process [19]. It is based on a continuous-time  $m$ -state Markov chain. The sojourn time for state  $j$  is exponentially distributed. When in state  $j$ , cells are generated according to a Poisson process with rate  $\lambda_j$  as shown in Figure 2.3. MMPP being more realistic, better models the voice and data connections present in an ATM network. However, its mathematical tractability is even more difficult than the GMDP model. This is because of its double-stochastic nature.

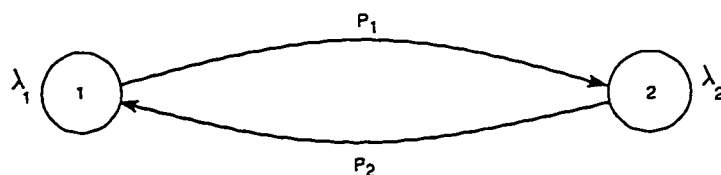


Figure 2.3: MMPP source model.

### 2.3.4 Fluid Flow Models

Fluid-flow models constitute a class of models in which cells are generated continuously, rather than at individual time instants. In the context of a communication system, fluid flow models are analysed by ignoring altogether the discrete nature of the cells. Thus, the analysis gives rise to differential equations with respect to the *pdf* of the system state. However, the fluid flow based analysis is inherently



approximate, since it is based on the assumption of continuous cell generation.

### 2.3.5 Models Specific to Video Sources

Extensive research has been carried out to come up with models that accurately represent the behavior of the video sources present in ATM networks. As discussed earlier, it is particularly hard to come up with an exact model for the video sources. This is because of the high degree of autocorrelation property present in the video data cells. Some models have been formulated, but the problem with these is that none of them is easy to tract. The video source models that have been suggested are:

- Binomial autoregressive model,
- First-order autoregressive model,
- Autoregressive moving average model,
- Discrete-time Markovian arrival process model, and
- Linear model for bit-rate prediction.

Details of these models can be found in [11, 18, 19, 38].

This chapter described the different traffic source models proposed in the literature for the performance evaluation of ATM networks. The next chapter gives a detailed review of the available literature on traffic control in ATM networks.

# Chapter 3

## Literature Review

This section provides an extensive literature review of the latest research work that has been carried out on congestion control in ATM networks. The review is divided into subsections. Each subsection deals with the work that has been done in a particular avenue of the congestion control problem. The different schemes proposed by the researchers are very varied and hard to compare. Some have adopted the analytical approach, while others have done simulations to test their strategies. Techniques presented in the literature are claimed to give good results.

### 3.1 Admission Control Techniques

A new Admission control strategy is presented in [1]. Instead of relying on the user to provide the traffic descriptors, the scheme determines these parameters by

monitoring and learning the user's cell-traffic flow. The traffic parameters are derived from congestion prediction based on Gaussian and Poisson distribution and a method determining their values is shown. In addition, a short-term congestion control strategy is also presented which relies on the selective discarding of low-priority traffic. Buffer construction and buffer control algorithm used in the approach are also presented. Simulation results are presented for VBR sources.

A different approach to admission control has been proposed in [13]. Instead of doing the bandwidth estimation on-the-fly, an effective bandwidth is allocated to the incoming call using a lookup table. Initially 32 traffic types are proposed which can be extended to about six thousand, and simulation results are reported for fewer still. In addition to the lookup table, the effective bandwidth is calculated using the global network condition. One of the concepts presented is to allocate bandwidth close to the average when link loads are low. As the load grows the bandwidths are adjusted to reflect a more accurate picture of actual multiplexed bandwidth needs in order to ensure the QoS promised. Simulation results are presented for video, voice and data traffic.

A two level framework for traffic control in an ATM network supporting multimedia connections is reported in [7]. At the higher layer, an admission control scheme decides about the connection acceptance and converts the end-to-end performance requirements of the connection to the performance requirements in each intermediate node of the routing path. At the lower layer, a round-robin/priority

mechanism is used to satisfy the specific performance requirements of connections in each network node. The admission control is completed in one round trip delay. During the first phase, a control message is sent to the destination. On its way the loss performance requirements at each intermediate node are calculated. On the return phase a reply message is sent from the destination to the source. During this phase, approximate studies about the delay performance are carried out. Since the optimal delay time assignment for maximizing the saturation load of the network is an NP-hard problem, two heuristics are employed. In the first, the target delay in each intermediate node is made as even as possible. In the second, the target delay in the middle nodes is made larger than those at the edges.

## 3.2 Congestion Avoidance Techniques

A congestion avoidance technique is reported in [9]. The parameters sought are network efficiency, fairness, and convergence. A distributed algorithm is presented with additive increase and multiplicative decrease of the user's load. A single bit feedback is considered for simplicity. The algorithm presented is such that the response is similar for all users, that is all of them either decrease or increase their load. This is a very severe approximation. In addition the algorithm does not take into consideration the VP/VC concepts of ATM.

A class-based congestion avoidance scheme is presented in [36]. Calls are divided

into classes based on their requirements. The CBR traffic is assured the required QoS and is multiplexed with the VBR traffic. Statistical multiplexing is used within each class. If there is unused bandwidth in a class it may be used by another class temporarily. A penalty scheme is also proposed for the violating sources. Simulation results are presented showing the cell loss ratios, the statistical gain and cell delay variation.

Simulation results are reported by Sykas [40] for a congestion avoidance scheme. An on-off source model is assumed and the multiplexer queue is modeled by M/D/1/K for low values of buffer capacity. For the large buffer region a burst scale cell model has been used. If the buffer size is small, it would provide a good delay performance. However, the cell loss performance could be unacceptable. In case of large buffer sizes, the variations in delay could be large. Larger buffers, however, provide a better cell loss performance, as they are capable of absorbing a sudden *burst of cells*. Allocation of equivalent bandwidth to the resources is advised based on their characteristics. However, during the simulation, all the sources are allocated equal equivalent bandwidth. The results show very little statistical gain for acceptable cell loss probability. An analytical formula for estimating the equivalent bandwidth is also derived.

A congestion avoidance scheme is proposed in [28], which is based on a bandwidth assignment mechanism. A bandwidth called equivalent bandwidth is allocated in order to ensure the cell transport objectives. A Definition is proposed for equivalent

bandwidth for CBR, bursty data, and variable bit rate video sources. The equivalent bandwidth values are evaluated by simulation considering both homogeneous and heterogeneous sources. In the heterogeneous mixes with different cell loss probability requirements the selective discarding mechanisms permit a more efficient utilization of the network resources.

An attempt to come up with a simple and effective congestion avoidance scheme is reported in [31]. The scheme uses minimum amount of feedback and tries to balance the tradeoff between significant queuing (high throughput) and increased idle time of the switch (lower response time). The scheme tries to operate the network slightly towards the left of the knee as shown in Figure 3.1. Policies are

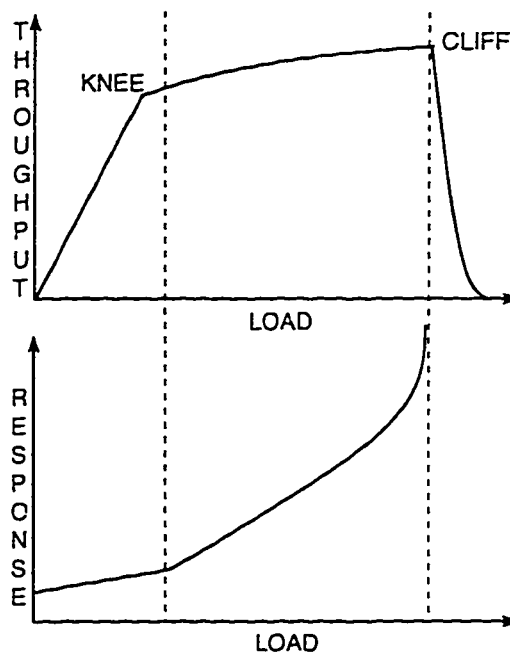


Figure 3.1: Network performance as a function of load.

when the average queue size goes beyond a certain threshold. The average queue length is determined based on the number of cells in the network switch that are queued and in service averaged over a certain interval. As soon as the average queue length exceeds the threshold, a feedback signal is sent to the sources. Upon the reception of the feedback signal, the sources immediately throttle-down. This allows the networks to operate to the left of the knee and ensures congestion avoidance. As for the sources, there is a policy for the time to update the window size, for choosing the significant feedback bits, for congestion signal filtering and for increasing and decreasing the window size. A simple multiplicative decrease and additive increase formula for the transmission rate is suggested. Other aspects of control in the scheme are efficiency and fairness.

### 3.3 Reactive Congestion Control Techniques

A dynamic approach to rate-based feedback traffic control in ATM networks is presented in [3]. Instead of using static thresholds to detect impending congestion, dynamic thresholds are set based on the short-term measurements of the arrival rate. Through these measurements, the number of simultaneous bursts in progress at each node is estimated, and is used to adjust the peak rate of the sources. The departure of this technique from the conventional methods is that it does not detect congestion by noticing the queue length, rather it detects congestion by observing the short-

term cell arrival rates. The scheme is reported to react faster and the requirement of buffers is also lower. This lower buffer requirement results in a tighter bound on the variance of delay. The performance of the scheme is compared with a static open-loop policy and the results are reported to be significantly better when the system load is high.

A very different approach to congestion control is presented in [5]. The scheme takes routing into consideration to deal with congestion. The concept of Multipath routing is introduced in which the shortest path from source to destination is considered if that path is lightly loaded. Otherwise, alternate paths are explored based on a weighed criterion of cell loss and delay. The performance of the scheme is compared with other proposed schemes, such as, Shortest path, Random, Alternate, and Reservation schemes. Simulation studies were done and a comparative analysis was provided under different load conditions. The Shortest path scheme congests the system early even if other lines are underutilized. The Random technique wastes system resources, tries to use alternate paths even at light loads and the call blocking probability is found to be the same as that for the Alternate Path Scheme. This scheme shows oscillations, overutilizes the shortest path and results in higher cell loss. The Reservation Scheme shows some oscillations between shortest and alternate paths, overutilizes shortest path, results in higher cell loss and the call blocking probability is found to be the same as for the Multipath routing scheme. Finally, the proposed Multipath routing scheme results in a proportional distribution of load



throughout the network and a natural load balancing. The call blocking probability is lesser because of the alternate paths and tries to use alternate paths before the shortest path is congested heavily. However, due to the nature of ATM, which favors simple and fast routines, the Shortest path method delivers the best overall results.

A dynamic traffic control mechanism is reported in [25]. Interrupted Bernoulli process is used to model the bursty sources. An (L,M,T) mechanism is proposed by the authors, which is implemented in two phases. L is the minimum inter-cell distance, M is the maximum number of cells that are allowed in the burst and T is the minimum distance that is allowed between two adjacent ON periods in an ON-OFF source model. In the first phase those cells that arrive at a rate greater than  $1/L$  are discarded. In the second phase, those cells that violate the (M,T) condition are discarded. Comparison of the technique with the leaky bucket mechanism is done and it is found that, if each session is policed separately, then the (L,M,T) scheme fares better for larger burst sizes. Finally, two schemes are presented that improve statistical multiplexing. In these schemes, the traffic control mechanisms are not used in isolation for each session, rather they are used on groups of sessions. In the first of these techniques, traffic control is performed on the aggregate traffic of a group, while in the second scheme, traffic control is performed on single sessions with information about the status of all the sessions of the group being available. The first one improves upon statistical multiplexing by controlling groups of sessions together and the second one improves statistical multiplexing by admitting more

cells when more sessions are off than when fewer are off. Analytical models and examples for both the schemes have been presented.

A congestion control scheme is proposed in [2] which requires simple protocols and provides guaranteed grade of service. It is suggested that the network provide two grades of services: express and first class with the latter being used for non-real time applications. The proposed scheme is applied at two levels: call and cell levels. The call level control takes place during the call setup in the form of (logical) bandwidth reservation for individual VCs. The cell level control is exercised across the network interface in the form of the choking/relieving of only first class VC traffic in the case of congestion.

In [15], congestion control is performed using a user-network policer. The technique consists of a service dependent user policer and a service independent network policer. Users are responsible for policing and marking their traffic appropriately before sending them into the network. The network is only responsible to verify the correctness of user policing and transporting cells transparently across the network.

A marking-based congestion control strategy is described in [29]. The approach consists of monitoring the traffic at the user-network interface. Cells exceeding the allowable bandwidth are marked as violating. These marked cells are allowed into the network. Under steady state operation, these cells are treated like normal cells. Marked cells are dropped at a node only if congestion is present at that node. Simulation results show that the marking method can improve the service quality

and network throughput by allowing cells exceeding the allocated bandwidth into the network instead of discarding them.

In [16], the leaky bucket algorithm has been proposed as an enforcer of the rate-based management of bandwidth in high performance, integrated cell communication networks. The efficiency of the unbuffered leaky bucket mechanism in protecting the quality of service offered to connections passing through a common internodal link queue is studied. It is shown that the extra traffic generated by a source can pass through the leaky bucket without being marked or discarded and cause unacceptable cell delay or loss to other connections sharing the queue. It is also shown that average bursts lengths and average data rates that are higher than those suggested at the time of connection establishment seriously deteriorate the queue length performance. This is so even when the leaky bucket mechanism discards cells based upon the negotiated QoS. It is suggested that the problem be alleviated by allocating network bandwidth above the leaky bucket permit rate and configuring the leaky bucket with a loss-marking ability.

In [22], an alternate control scheme resident at a higher layer is proposed that realizes the fact the a B-ISDN based on ATM must support several kinds of transport services with different traffic characteristics and service requirements. As there are neither link-by-link flow control nor error control in the ATM layer, different flow/error control can be performed at a higher layer (AAL or transport) for the different services. A flow control mechanism (rate control for end-to-end transport)

is proposed which shows acceptable performance when the average overload period is bounded by a certain time.

In [26], analysis of a rate based control scheme based on a buffered leaky bucket mechanism has been carried out. It is inferred that for sources with relatively large active periods, to insure an acceptable QoS at the input queue, the token generation rate should be chosen close to the peak rate of the source. Furthermore, increasing the bucket size of the leaky bucket doesn't have a considerable effect on the performance at the input queue.

A leaky bucket approach is proposed in [42] to meet the performance requirements of the different sources attached to the network. The performance is evaluated as a function of the source characteristics. The effect of statistical multiplexing is also investigated. The results indicate that the leaky-bucket scheme and statistical multiplexing are more effective for bursty traffic. This calls for a dynamic control mechanism to achieve optimal system utilization.

A different congestion control mechanism is suggested in [6]. The authors contend that using just one bit to assign priorities is not enough as there are two types of priorities that exist in ATM: service-oriented and congestion-oriented. They introduce a four level priority scheme using one bit (RES) along with the commonly used CLP bit. Also a Forgiving Leaky Bucket concept is presented, which in addition to marking excessive cells, will also unmark a fraction of incoming previous excessive marked cells. Simulation has been done and the behavior of the new leaky

bucket mechanism is found to be an improvement over the more conventional Virtual Buffered Leaky bucket.

An attempt to achieve high network utilization based on reactive control is proposed in [8]. Preventive measures to control congestion require a more precise knowledge of the traffic characteristics of the sources. The window based mechanisms do not require such precision. A variation of the sliding window control is presented here, called the Leaky window. Excess cells are marked and are allowed into the network if it is lightly loaded. The network load is determined by the acknowledgments received in a certain amount of time. The marked cells can be discarded in case of congestion at any node. The performance of the scheme was compared to the sliding window and was found to be better. Its performance as compared to virtual leaky bucket was better at high loads. However, under light loads, the virtual leaky bucket fared better. It is concluded that there is no harm in providing bandwidth on demand, but as the load increases the control should get more aggressive to deliver the agreed upon QoS, otherwise the marked cells would cause the network to go into congestion, which would result in frequent retransmissions.

Cooper in [10] has argued that an efficient congestion control strategy has to provide features for admission control, buffer and queue management, traffic enforcement and reactive control. It has also been shown that maximum statistical gain results from the presence of highly bursty sources. But the achievable utilization is only a fraction if all the sources are VBR type than what is achieved with

the CBR type sources. He has also argued against the tagging of cells, because it makes the enforcement the QoS agreement very difficult.

An adaptive rate based congestion control scheme is presented in [47], which is a variation of the open loop leaky bucket mechanism. A simple threshold policy is adopted to detect congestion at the multiplexers and a feedback signal is sent back. In response, the leaky buckets decrease the rate of generation of the tokens. A discrete system is considered where time is slotted and the cell arrival process in each slot is governed by a homogenous finite-state, aperiodic discrete-time binary Markov chain called the modulating Markov chain. The effectiveness of the feedback policy is studied by comparing it to a non-adaptive leaky bucket mechanism.

A congestion control technique based on dynamic evaluation of the transmission rates of the sources is reported in [35]. The problems encountered by the previous techniques are that all the VCs connected to a congested node have to decrease their rates. Then some require per-VC buffering which is difficult to implement. And in most schemes, there is just one bit in the feedback, which is considered by the authors as too little. They have proposed a dynamic congestion control scheme in which the congested node will estimate the optimal cell rate for every VC. A periodic cell is sent from the sources towards the destination carrying the present allowed cell rate and the maximum possible cell rate. A congested node will update the allowable cell rate for each VC by doing some computation. This information cell then reaches the destination and from there, it is sent back to the source so that

it can update its current cell rate. This current cell rate is the minimum cell rate allowed by any switch along the VC. After sending the information cell, the source continually decreases its cell rate until it receives the feedback from the destination about its new cell rate. This cell rate can never be higher than the maximum allowable cell rate for that particular source.

A congestion control strategy for multi-media services is presented in [43]. A framing congestion control strategy based on a cell admission policy at the edges of the network and a service discipline called Stop-and-Go queuing at the switching nodes is described. The strategy provides bounded end-to-end delay and a small controllable delay-jitter. The strategy is applicable to packet switching networks in general. This Stop-and-Go queuing discipline is recommended along with an admission control policy based on a time-framing concept.

A congestion control strategy for frame relay and X.25 networks is suggested in [45]. Attempt has been made to come up with a congestion control strategy that is applicable to both of these networks. Although, the two networks differ in operation, they share some common principles. The common elements are the detection of impending congestion within the network, carriage of notification of congestion to the periphery of the network, filters to provide the proper feedback, and throttle mechanisms to control the flow of traffic into the network. The performance of these mechanisms is analyzed and the results are presented. The basic working of the strategy revolves around a feedback signal sent from the destination to the

source after it has received an indication of congestion along a given virtual path. This indication is given by a congested switch by setting the Forward Congestion Indication (FCI) bit. The destination sets the Backward Congestion Indication (BCI) bit in a cell travelling towards the source requesting it to throttle down. A dynamic window algorithm for the X.25 and a rate adaptation strategy for the frame relay have been proposed by the authors.

### 3.4 Buffer Management Based Congestion Control Techniques

A traffic and congestion control strategy is presented in [14]. Stress is laid on the need for the traffic descriptors to be unambiguous and also for the explicit description of the performance expected from the network. It is suggested that explicit priorities should be used when tackling congestion. Higher priority traffic (CLP=0) is guaranteed the QoS requirements at the expense of lower priority (CLP=1). Additionally two traffic-monitoring modules (TMMs) are inserted. These can be leaky bucket mechanisms or something else. One of them monitors the CLP=0 traffic and extracts the violating cells and tags them (CLP=1). These tagged cells along with the other low-priority cells are fed to the other TMM, which selectively discards the lower priority traffic.

A state-dependent priority based buffer management scheme is proposed in [21].



Two separate buffers are used, one for delay-sensitive and the other for loss-sensitive traffic. Two schemes have been proposed for buffer management. The first scheme takes into account the arrival rates as well as the instantaneous size of the queues when generating a congestion indication, while the other scheme generates the congestion indication by considering only the current buffer sizes. The performance of the schemes is compared with the unprioritized buffer management schemes. The buffer requirements were found to be reduced for the priority based schemes, while the performance of the delay-sensitive traffic was found to be significantly better at the expense of some increase in the delay of the loss-sensitive traffic.

A priority-based congestion control strategy for CBR and VBR traffic is given in [48]. An explicit congestion notification (ECN) is fed back to the sources, which are then supposed to throttle down. A two level priority at the switching nodes is observed. Higher priority is enjoyed by the CBR sources. Selective discarding of the lower priority cells is observed in the lower priority queue depending upon the threshold set. Also VPI/VCI lookup tables are maintained at each node to determine the connection priority and whether a burst for that VBR connection is expected or not. This information is provided by the burst admission control. Burst admission control is used for those VBR sources whose traffic characteristics are hard to predict. Under normal circumstances, connection admission control will accept all such connections without allocating any bandwidth for them. The bandwidth will be allocated only to the bursts, according to their peak bandwidth

requirements, by the burst admission control prior to their transmission. If there is not enough bandwidth available from the pool of bandwidth reserved for this service category, the transmission of the burst will be blocked or delayed. In this way more than two priority classes can be implemented.

In [27], a congestion control scheme has been presented which is applied when the buffer queue length in an ATM switch exceeds a certain threshold. Cells from lower priority traffic such as voice are rejected. As a result higher priority traffic such as compressed video can be transferred with good quality. Using numerical examples, it is shown that the threshold value at which to trigger congestion control is an important parameter for effectively transferring multimedia traffic.

An interesting inference has been reached in [44]. Examples are presented to show that the buffer requirements for guaranteeing a loss-free service in an ATM network supporting heterogeneous traffic types could be extremely hard to fulfil and is very dynamic. The combination of multiplexers and demultiplexers makes buffering inevitable. This is because of the fact that very high rate sources may be preceded by low rate traffic on a multiplexer and when demultiplexing is done, cells from the high rate sources have to wait in the buffer. To alleviate this problem it is suggested that some sort of scheduling algorithm is needed for multiplexing VCs of different rates.

A priority assignment control scheme based on multiple QoS classes is reported in [41]. Presence of multiple classes is assumed, each one with a different cell loss

and delay requirements. Different control parameters are outlined, which are; priority assignment ratio, priority assignment period, and priority scheduling in each cell slot. To satisfy the delay bound of each class, an upper limit of the queue length corresponding to the minimum bandwidth allocated to that class is determined. Therefore, any cell which might eventually exceed its delay bound is discarded immediately due to buffer overflow. Four different priority classes are considered in the work. It is concluded that when the multiplexed QoS classes have very different requirements then the proposed method is more effective. Analytical analysis, supported by simulation, was performed to find out the effect of dynamic priority assignment ratio changing depending upon the load of each class. It is concluded that this change should be done after relatively long periods in order to ignore the short-term fluctuation in the traffic.

This chapter presented a detailed review of the available literature on traffic control in ATM networks. The next chapter discusses the development of a statistical congestion control scheme and the motivation for such a work.

# Chapter 4

## A New Statistical Congestion

## Control Scheme

This chapter describes the proposed congestion control scheme. An introduction to hypothesis testing, which is used in the proposed control mechanism, is given. After that, a detailed description of the experiments that have been conducted to develop the control mechanism is given. Finally, the proposed scheme is presented.

### 4.1 Motivation

As has already been described in detail, congestion is a very serious threat to the proper working of ATM networks. In order to develop an effective congestion control strategy, several issues of the scheme have to be decided. These issues are discussed

strategy, several issues of the scheme have to be decided. These issues are discussed below.

#### 4.1.1 Reactive Versus Preventive Control

In ATM networks, where the user is given a QoS guarantee, it is very important that congestion be prevented. One way to achieve this goal is to keep the load on the network below the knee of the curve, as shown in Figure 3.1. This low load will allow the network to enjoy the spare bandwidth, which would be used to accommodate the bursts of data that may arrive. This technique, though useful, will severely under-utilize the expensive network resources.

On the other hand, reactive congestion control schemes take advantage of statistical multiplexing and allow the network to enjoy higher resource utilization. Reactive control operates the network at a load between the knee and the cliff of the curve in Figure 3.1. IEEE has adopted reactive control for use in ATM networks. Our scheme, therefore, uses the reactive strategy. The network, in that case, would run the risk of being congested. However, our scheme allows the network to operate at higher utilization, while maintaining the guaranteed QoS, through the use of an efficient control.

### 4.1.2 Queue-length Based Congestion Detection

Due to the very high delay-bandwidth product of ATM networks, the reactive time available to a congestion control technique is very small. That is why we believe that the key to the *proper operation* of ATM networks is held in the *timely and proper detection* of congestion. Queue-length based congestion detection has been the most dominant method. A node declares congestion when the queue length crosses a certain threshold. The associated link is then declared as congested and appropriate actions are taken, depending upon the scheme that is being implemented. Its effectiveness depends on how the parameters of the scheme have been set. The parameters could be the queue length, threshold value, and/or the delays involved. However, the problem with this method is that it detects congestion only on the basis of the instantaneous queue length. But, violation of the threshold may be due to the arrival of bursts from several sources whose overall transmission rate may be below their mean-rate. In this case there would actually be no congestion in the network. The static detection mechanism, however, will send out congestion alarms. This would force the sources, using the effected path, to slow down their data rate. A two-fold effect would result from this scenario:

- Decreased resource utilization, and
- Overhead in processing the excessive feedback signals.

We believe that these problems result from the fact that this congestion detection methodology does not use the *statistical information* of the traffic sources. A more intelligent congestion detection scheme would relate the occurrence of congestion with the nature of the traffic sources. The development of this relationship, however, is not an easy one. The bursty and stochastic nature of the sources, coupled with the multiplicity in the types of sources, makes this job a very difficult one. It is this nature of the sources that causes congestion (in addition to the use of statistical multiplexing, of course). Therefore, a major part of this thesis work was spent on experiments that tried to get an effective relationship between the nature of the traffic sources and the occurrence of congestion.

The next section describes the network model that we used to develop and test our hypothesis testing based congestion control strategy.

## 4.2 ATM Switch Queuing Model

The network model assumed in this research work is as follows:

1. the network topology consists of an interconnection of ATM switching nodes.

A typical network model is shown in Figure 4.1. Switching nodes are interconnected through high-speed links. Local sources ( $x$  and  $y$ ) are attached directly to the switching nodes. The traffic from the local sources is accepted in its respective *source queue*, whose output is controlled by a leaky bucket

mechanism. In effect, admission control is being done to the sources at these points. Let some sources at nodes  $a$  and  $b$  be transmitting some traffic to some destination(s) at node  $d$ . Node  $i$  acts as the intermediate node. All the traffic for node  $d$  passing through node  $i$  is collected by a *transit queue* before getting transmitted on the outgoing link. Also, let some local sources at node  $i$  be transmitting some traffic to the same node  $d$  from their source queues. We assume that the multiplexer transmits from all these queues in a round robin fashion.

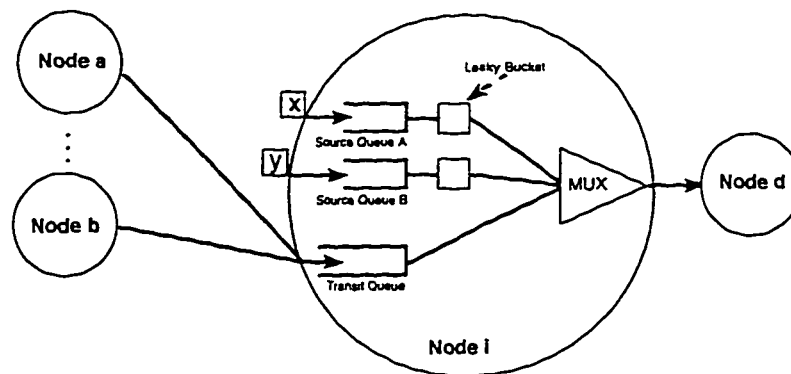


Figure 4.1: Network model.

2. each node can have an arbitrary number of traffic sources attached to it.
3. traffic emanating from a source is received by a *source queue*. The output of the queue is controlled by a *leaky bucket* mechanism. Overflowing cells are dropped and are declared as lost.



4. each outgoing link associated with any node is fed from a *transit queue*, in addition to the traffic from the local source queues. This transit queue gets traffic from up-stream nodes. The up-stream traffic to a transit queue may be from a source directly connected to the up-stream node, or from another transit queue.
5. a node would have as many transit queues as the number of outgoing links.
6. the output traffic from a transit queue is not controlled by a *leaky bucket mechanism*. However, if there are no local sources sharing the link with a transit queue, special control steps have to be taken as described later.
7. each cell coming to a node is examined to see whether that particular node is the destination node for that cell or not. If it is, the cell is not inserted in the transit queue.
8. the capacity of the interconnecting links may be different from one another.
9. 5% capacity of each link is reserved to carry control cells.
10. the number of nodes as well as links in the network is arbitrary.

This network model corresponds very well to the actual switch model that we have in practice. This is shown in Figure 4.2. Admission control for the local sources is done at the UNI (user network interface). The non-local sources have their traffic collected at the transit queues, from where their transmission is scheduled.

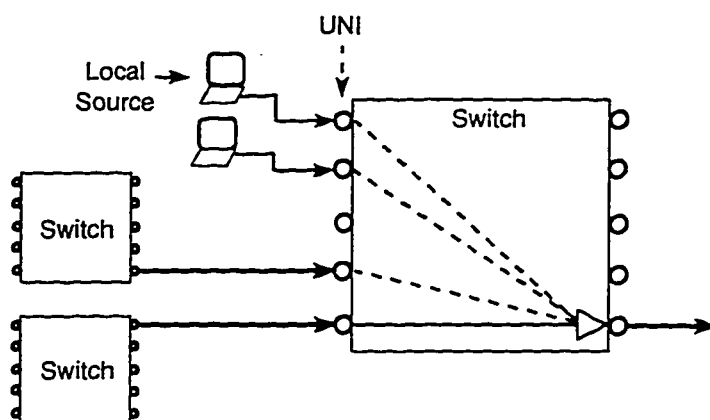


Figure 4.2: ATM switch interconnection.

### 4.2.1 Selection of Queue Sizes

The queue sizes for the source and the transit queues are taken from the work of [49].

- The size of a source queue has been calculated using the formula for  $M/M/1/K$  queuing model. The cell arrival process was approximated by a Markovian process. The service rate was chosen as exponential. Though the cell size is fixed in ATM networks, there may be multiple queues attached to a single outgoing link. The service rate, in that case would not be approximated well with a deterministic model. The queue capacity  $K$  for  $M/M/1/K$  is approximated as:

$$K = \frac{\log(p_i)}{\log(\rho_i)}$$

where,  $p_i$  is the cell loss tolerance of the traffic source,  $\rho_i$  is the maximum target utilization of the outgoing link by source  $i$ . Adjustment was made to take the burstiness of the source into consideration. The resulting formula for source  $i$  is as follows:

$$Q_i = \frac{1}{\beta_i} \times \frac{\log(p_i)}{\log(\rho_i)}$$

where,  $\beta_i$  is the burstiness of the traffic source  $i$ .

- The size of a transit queue has been calculated so that it should be large enough to hold simultaneous bursts from every traffic source using that queue.

The formula used is:

$$\sum_{i=1}^n N_i$$

where,  $N_i$  is the average burst length of a traffic source, and  $n$  is the total number of traffic sources using that transit queue.

Having described the network model, we now describe hypothesis testing. We also describe how and why have we used it in the congestion control structure.

### 4.3 Hypothesis Testing

Formation of *data based* decision procedure that can produce a *conclusion* about some scientific system is of a major concern to an engineer or a scientist. For example, an engineer might have to decide on the basis of sample data whether

there is a difference between the accuracy of two kinds of gauges. He would have to postulate or conjecture something about a system. In addition, he must involve the use of experimental data and decision making that is based on the data. Formally, the conjecture can be put in the form of a statistical hypothesis. Procedures that lead to the acceptance or rejection of statistical hypothesis comprise a major area of statistical inference. A formal definition of statistical hypothesis may be as follows [46]:

**statistical hypothesis** *is an assertion or conjecture concerning one or more populations.*

The truth or falsity of a statistical hypothesis is never known with absolute certainty unless we examine the entire population. This, of course, would be impractical in most situations. Instead, we take a random sample from the population of interest and use the data contained in this sample to provide evidence that either *supports* or *does not support* the hypothesis. Evidence from the sample that is inconsistent with the stated hypothesis leads to the rejection of the hypothesis, whereas, evidence supporting the hypothesis leads to its acceptance.

However, it must be kept in mind that the acceptance of a hypothesis merely implies that the data does not give sufficient evidence to refute it. On the other hand, rejection implies that the sample evidence refutes it. In other words, rejection means that there is a small probability of obtaining the sample information observed

when, in fact, the hypothesis is true.

The formal statement of the hypothesis is often influenced by the structure of the probability of a wrong conclusion. If one is *strongly supporting* a contention, he hopes to arrive at the contention in the form of rejection of a hypothesis.

### 4.3.1 The Null and Alternative Hypothesis

The structure of *hypothesis testing* will be formulated with the use of the term null hypothesis. This refers to any hypothesis we wish to test and is denoted by  $H_0$ . The rejection of  $H_0$  leads to the acceptance of an alternate hypothesis, denoted by  $H_1$ . A null hypothesis concerning a population parameter will usually be stated so as to specify an exact value of the parameter, whereas the alternative hypothesis allows for the possibility of several values. Hence, if  $H_0$  is the null hypothesis  $p = 0.5$  for a binomial population, the alternative hypothesis  $H_1$  would be one of the following:  $p > 0.5$ ,  $p < 0.5$ , or  $p \neq 0.5$ .

To illustrate the use of hypothesis testing, let us take an example where a new type of vaccine is compared against an old one. The effectiveness of the old vaccine is known to be 25% after a period of 2 years. 20 people are chosen for the experiment. If more than 8 of those receiving the new vaccine surpass the 2-year period, the new vaccine will be considered superior. The choice of number 8 is somewhat arbitrary, but appears reasonable because it represents a modest gain over the 25% efficiency of the older vaccine. We are essentially testing the null hypothesis that the new vaccine

is equally affective after a period of 2 years as the one already in use. The alternative hypothesis is that the new vaccine is, in fact, superior. This is equivalent to testing the hypothesis that the binomial parameter for the probability of a success on a given trial is  $p = 1/4$  against the alternative that  $p > 1/4$ . This is usually written as:

$$H_0 : p = 1/4,$$

$$H_1 : p > 1/4.$$

The test statistic on which we base our decision is  $X$ , the number of individuals in our test group who receive protection from the new vaccine for a period of at least two years. The possible values of  $X$ , from 0 to 20, are divided into two groups: those numbers less than or equal to 8 and those greater than 8. All possible scores greater than 8 constitute the **critical region**, and all possible scores less than or equal to 8 determine the **acceptance region**. The last number that we observe in passing from the acceptance region into the critical region is called the **critical value**. In this case, the critical value is the number 8. Therefore, if  $x > 8$ , we reject  $H_0$  in favor of the alternative hypothesis  $H_1$ . If  $x \leq 8$ , we accept  $H_0$ . This decision criterion is illustrated in Figure 4.3.

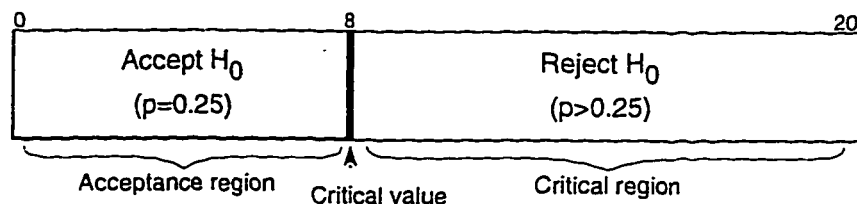


Figure 4.3: Decision criterion for testing  $p = 0.25$  versus  $p > 0.25$ .

### 4.3.2 Type I and Type II Errors

The decision procedure just described could lead to either of the two wrong conclusions. For instance, the new vaccine may be no better than the one now in use and, for this particular random selected group of individuals, more than 8 surpass the 2-year period successfully. We would be committing an error by rejecting  $H_0$  in favor of  $H_1$  when, in fact,  $H_0$  is true. Such an error is called a **type I error**. A second kind of error is committed if 8 or fewer of the group surpass the 2-year period successfully and we conclude that the new vaccine is no better when it actually is better. In this case we would accept  $H_0$  when it is false. This is called a **type II error**.

In testing any statistical hypothesis, there are four possible situations that determine whether our decision is correct or in error. These four situations are summarized in Table 4.1.

The probability of committing a type I error is also called the **level of significance**, denoted by  $\alpha$ . The level of significance is also called the **size** of the critical

	<i>H<sub>0</sub> is True</i>	<i>H<sub>0</sub> is False</i>
<i>Accept H<sub>0</sub></i>	<i>Correct decision</i>	<i>Type II error</i>
<i>Reject H<sub>0</sub></i>	<i>Type I error</i>	<i>Correct decision</i>

Table 4.1: Possible situations in testing a statistical hypothesis

region. The probability of committing a type II error is impossible to compute unless we have a specific alternative hypothesis. And, for a fixed sample size, a decrease in the probability of one error will usually result in an increase in the probability of the other error. However, the probability of committing both types of errors can be reduced by increasing the sample size.

### 4.3.3 Application of Hypothesis Testing to Congestion Detection

To control congestion, one has to detect it first. Such detection is a very difficult task because of the statistical nature of the traffic sources and the large delay-bandwidth product. Since the traffic sources are bursty, the existence of congestion at a particular node is a statistical event. A node can only test at a certain confidence level for the existence of such an event, and can never be 100% sure that congestion exists. Hence, it seems most appropriate to formulate the congestion detection problem as a hypothesis test. In this thesis we adopted such an approach. We have applied hypothesis testing to our congestion detection scheme. The structure of the hypoth-



esis testing allows us to identify, with a certain *level of confidence*, the occurrence of congestion at a certain node in the network. The higher the level of confidence (in percent), the more accurate will be our decision regarding the detection of congestion. Our aim is to minimize the number of false feedback congestion alarms to the sources, so that the sources are allowed the liberty of transmitting at their maximum rate. The source will only be forced to cut down when the hypothesis test tells with some confidence level that congestion is impending. The structure of our congestion detection scheme based on hypothesis test is as follows:

- If congestion is suspected,

*Run hypothesis test*

$$H_0 : Y = A$$

$$H_1 : Y > A$$

Where,  $Y$  is the parameter that is most intimately related to the occurrence of congestion. Whereas,  $A$  is the value of the parameter  $Y$  which is thought to be of critical importance.

If  $H_0$  turns out to be true then no control action is taken, otherwise, congestion is declared and back-pressure signals are sent.

In our case, the type-I error would correspond to the situation when congestion is declared by the control when actually there is none. On the other hand, type-II

error would correspond to the case when the control would ignore the congestion while there is one impending. Both of these are important and had to be taken care of.

In order to effectively formulate hypothesis test for congestion, we first need to identify the statistic  $Y$  and the parameter  $A$ .

## 4.4 Data Collection and Experimentation

The available literature does not contain any studies that relate the statistical information of the sources to the occurrence of congestion. In this thesis, extensive experimentation was conducted to study the relationship between the traffic source parameters and congestion.

There were many different source parameters and almost as many network parameters. Therefore, we had to perform several different sets of experiments. In each set of experiment we studied the relationship between some independent variables and a dependent variable. The independent variables represented both the statistical information of the sources as well as the parameters of the network. The dependent variable in each case was a parameter that was thought to be intimately related to the occurrence of congestion in the network. The next step was the identification of the different possible parameters.

#### 4.4.1 Identifying Important Variables

The independent variables that were used in our experiments are enumerated below:

1. Average mean rate of the traffic sources in Mbps ( $m$ ),
2. Average burst length of the traffic sources in cells ( $B$ ),
3. Average burstiness of the traffic sources ( $\beta$ )<sup>1</sup>,
4. Average peak rate of the traffic sources in Mbps ( $pr$ ),
5. The traffic intensity on a link ( $\rho$ )<sup>2</sup>, and
6. The transit queue-threshold in cells ( $th$ ).

The selection of independent variables for a particular set of experiments was based on the nature of the experiment.

Congestion is a phenomenon that affects mainly the transit queue. Therefore, the dependent variables investigated in the experiments are:

1. Probability of violation of threshold for a transit queue ( $P_{tv}$ ),
2. Rate of change of the transit queue ( $R_{cq}$ ), and
3. Average of transit queue length.

---

<sup>1</sup>The ratio of the source peak rate to the mean rate.

<sup>2</sup>The ratio of the sum of the mean-rates of the sources using the link divided by the capacity of the link.

## 4.5 Details of Experiments

This section gives the details of the experiments conducted to come up with an efficient predictor of congestion.

### 4.5.1 Experimental Environment

The experimentation was done using an ATM network simulator. The simulator, written in C, can work with any network topology and any number of traffic sources. For each run of an experiment, the simulator was fed an input file that contains the network topology, the traffic characteristics, and the length of the simulation run.

A typical network topology used in the series of experiments is illustrated in Figure 4.4. Traffic generated by the sources would be transmitted to the transit queue at *node m*. The state of this transit queue was observed in each experiment.

### 4.5.2 Classes of Experiments

Each experimental run investigated a relationship between an independent variable and a dependent variable. Other independent variables were kept constant. Several simulation runs were done for each experiment. Each run gave a record of the states of the transit queue at *node m*.

The files collected in one experiment were used to investigate any possible relationship between the independent and the dependent variable. This was done using

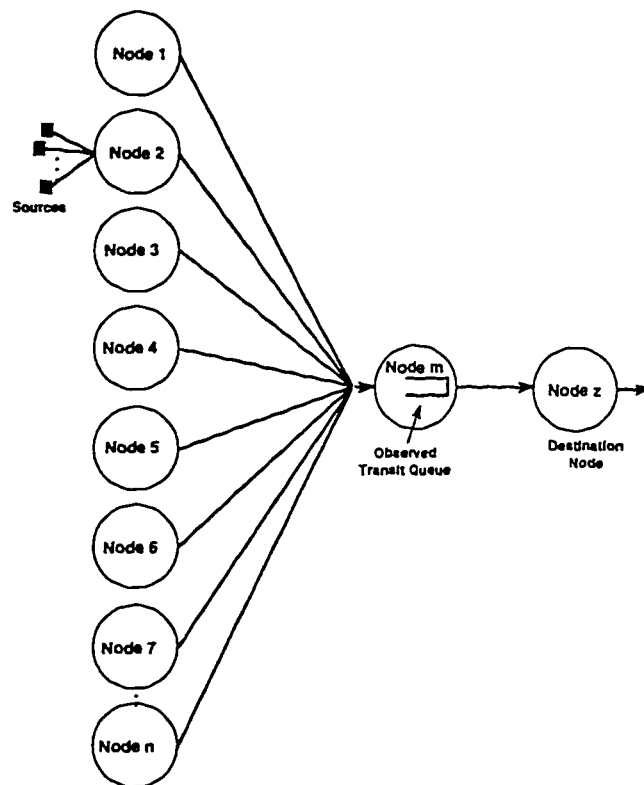


Figure 4.4: Experimental set-up.

SAS (Statistical Analysis System). SAS allows us to do **regression analysis** and **analysis of variance** (anova). We were interested in two of the fields returned by the SAS procedures. These were: the  $R^2$  and Mean-square-error (MSE) fields. A high value of  $R^2$  (greater than 0.7) suggests a strong linear relationship between the dependent and the independent quantities. For the MSE field, a value close to *zero* is desired. We also got scatter-plots of the data obtained. The details of the classes of experiments are discussed in the following subsections:

### 4.5.3 Effect of Traffic Sources on Probability of Loss

**Objective:** To investigate the relationship between the probability of violation of threshold ( $p_{tv}$ ) of the transit queue, and the characteristics of the traffic sources, i.e.,  $\beta$ ,  $B$ , and  $m$  of the sources using the transit queue.

A threshold was placed on each transit queue in the network. The probability of violation of threshold corresponds directly to the probability of cell loss, hence, to the occurrence of congestion. The experiment was run with varying values of  $th$ .

It was found that  $p_{tv}$  was very strongly related to every traffic source characteristic. The resulting graphs for some of the experimental runs are shown in Figures 4.5-4.7. Similar trend was observed for the rest of the experimental runs.

The strong visual correlation is also supported by the regression and anova analysis. Tables 4.2, 4.3 and 4.4 show the resulting values of  $R^2$  and  $MSE$  (Mean Square Error) for some of the experimental runs. The value of  $R^2$  is close to one in every

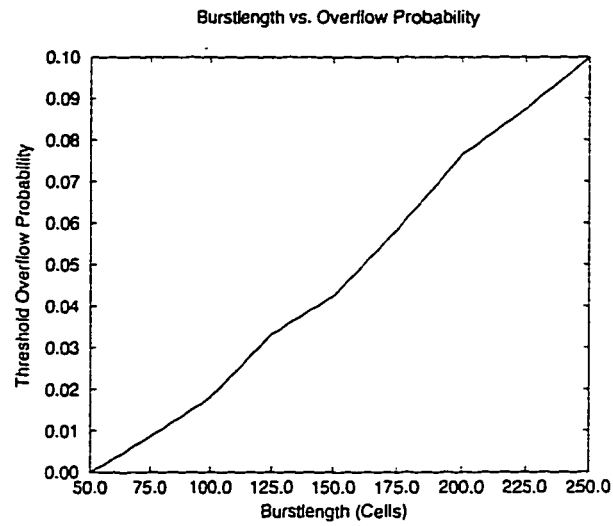


Figure 4.5: Probability of threshold violation versus burst length ( $m = 10$ ,  $\beta = 10$  and  $th = 250$ ).

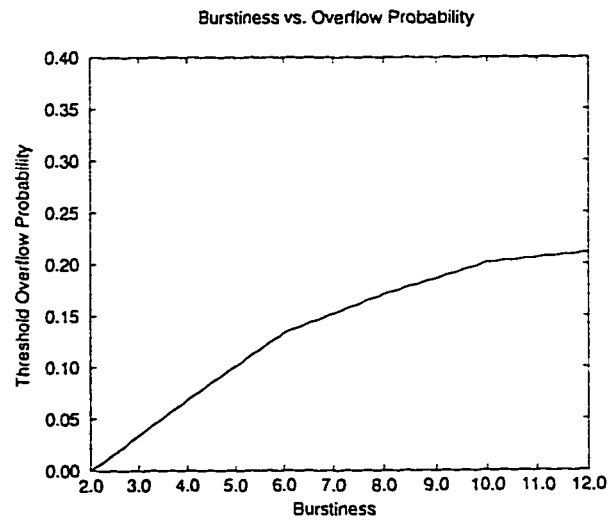


Figure 4.6: Probability of threshold violation versus burstiness ( $m = 15$ ,  $B = 100$  and  $th = 250$ ).

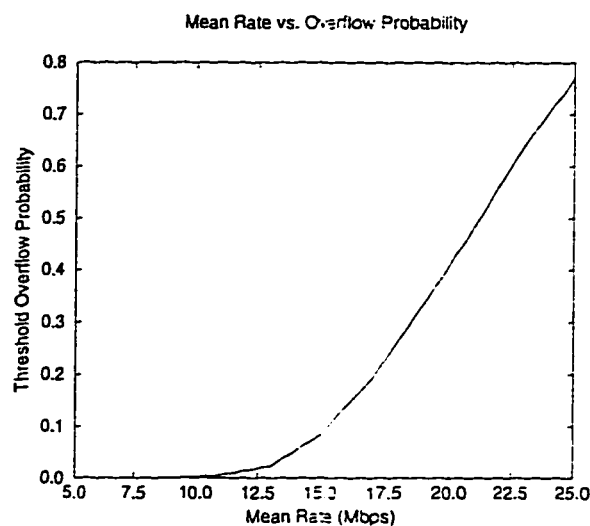


Figure 4.7: Probability of threshold violation versus mean rate ( $\beta = 5$ ,  $B = 100$  and  $th = 250$ ).

case, an indication that the linear model fits. A value of  $R^2$  greater than 0.7, in general, indicates meaningful correlation.

$B$	$m$	$\beta$	$th$	$R^2$	$MSE$
50 - 250	10	10	250	0.9929	0.00321
25 - 250	15	10	150	0.8957	0.04925
25 - 250	15	10	250	0.9685	0.02528
50 - 250	15	15	250	0.9844	0.01869

Table 4.2: Regression analysis of queue-threshold violation versus burst length.

This experiment shows that the overflow probability, i.e., the occurrence of congestion is strongly related to the characteristics of the sources. However, we were looking for a statistic that had a normal distribution. Unfortunately, this was not



$\beta$	$m$	$B$	$th$	$R^2$	$MSE$
2 - 15	10	100	150	0.9945	0.00464
2 - 12	15	100	250	0.9326	0.02397

Table 4.3: Regression analysis of queue-threshold violation versus burstiness.

$m$	$B$	$\beta$	$th$	$R^2$	$MSE$
5 - 25	100	5	250	0.8479	0.11710
5 - 25	100	10	250	0.9205	0.08166

Table 4.4: Regression analysis of queue-threshold violation versus mean-rate.

the case with the overflow probability.

#### 4.5.4 Effect of Traffic Sources on Queue Length Variation

**Objective:** To investigate the effect of the traffic characteristics on the average rate of change of the transit queue length over a fixed time interval.

The relationship was observed for various time intervals. However, it was found that there was no apparent relationship between the rate of change of the transit queue and the characteristics of the sources. This was true for every source characteristic. The resulting graphs are given in Figures 4.8, 4.9, and 4.10. The results of regression and anova also show the absence of a relationship. Table 4.5 shows the results of the statistical analysis.

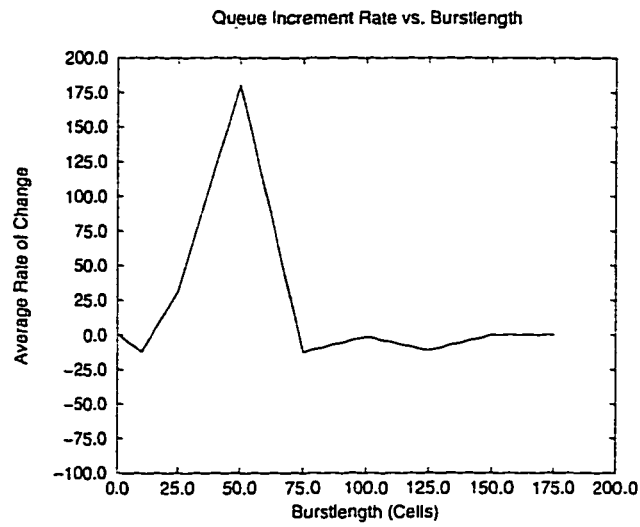


Figure 4.8: Queue length variation rate versus burst length ( $m = 15$  and  $\beta = 5$ ).

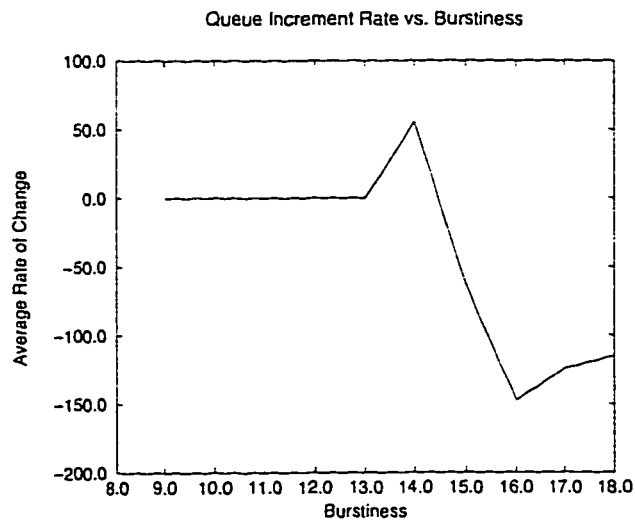


Figure 4.9: Queue length variation rate versus burstiness ( $m = 10$  and  $B = 50$ ).

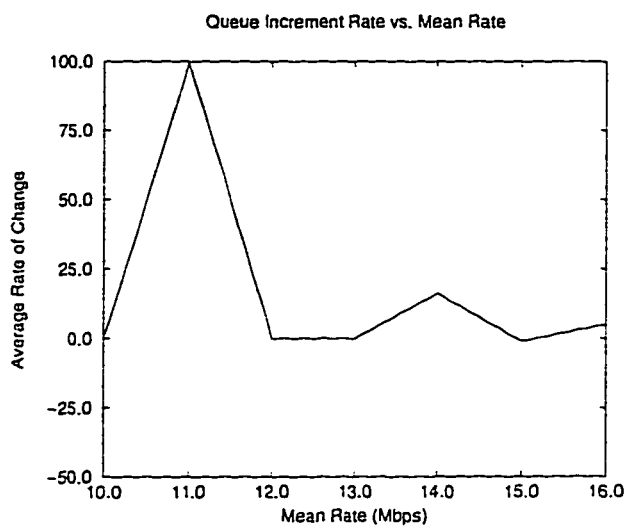


Figure 4.10: Queue length variation rate versus mean rate ( $B = 100$  and  $\beta = 5$ ).

$m$	$B$	$\beta$	$R^2$
10 - 17	50	10	0.0000
15	1 - 200	5	0.0000
10	50	9 - 20	0.0001

Table 4.5: Regression analysis of queue-increment-rate.

#### 4.5.5 Effect of Link Traffic Intensity ( $\rho$ ) on Overflow Probability ( $p_{tv}$ )

**Objective:** To investigate the relationship between  $p_{tv}$  and  $\rho$ .

Traffic intensity is defined as the ratio of the sum of the mean rates of the sources using the transit queue to the link capacity used by that queue. A value of  $\rho$  very close to 1 is undesirable, because it would make the traffic control almost impossible due to the bursty nature of the traffic.

The results show that  $p_{tv}$ , or the occurrence of congestion, is very intimately related to  $\rho$ . As can be seen in Figures 4.11 and 4.12,  $p_{tv}$  shoots-up towards 1 beyond a certain value of  $\rho$ . This is true for varying values of the threshold ( $th$ ). Thus,  $\rho$  is a very good indicator of congestion. However, its application to the hypothesis based congestion control structure is not easy. This is due to the fact that it is difficult to monitor the traffic intensity on all the links in the network. Doing this would require that each node has information of all the sources transmitting through it. This could result in significant computational overhead.

#### 4.5.6 Effect of Link Traffic Intensity ( $\rho$ ) on Transit Queue Length

**Objective:** To investigate the distribution of the transit queue length.

The experiment was repeated for different load conditions and different source

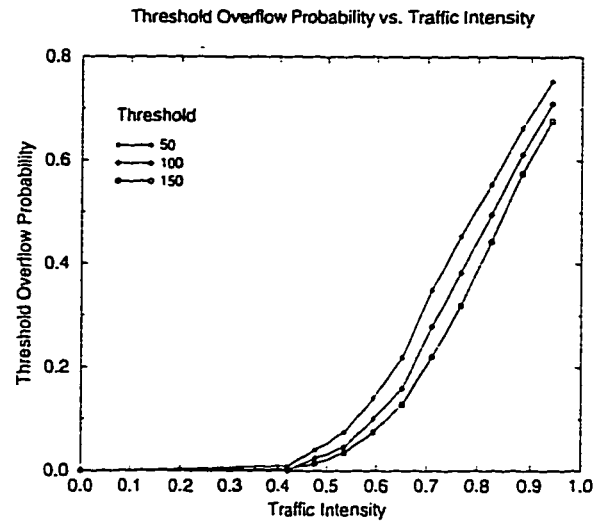


Figure 4.11: Threshold overflow probability versus traffic intensity ( $B = 200$  and  $\beta = 5$ ).

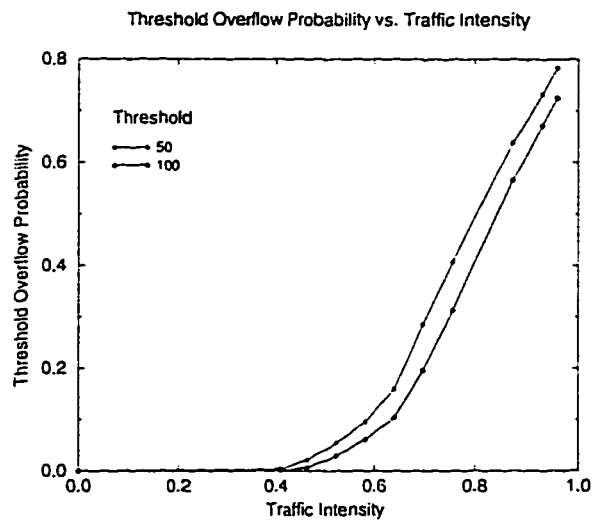


Figure 4.12: Threshold overflow probability versus traffic intensity ( $B = 100$  and  $\beta = 5$ ).

<i>load</i>	$m_{qt}$	$\sigma$
70%	344	130
75%	357	131
80%	386	129
85%	417	140

Table 4.6: Mean transit queue-length and standard deviation.

characteristics. The resulting graphs are shown in Figures 4.13, 4.14, 4.15, and 4.16. The transit queue size follows a normal distribution. Two-thirds of all the observations in each case fell within  $\pm\sigma$ , where  $\sigma$  is the standard deviation of the observed sample. Table 4.6 shows the mean queue-length ( $m_{qt}$ ) and the standard deviation ( $\sigma$ ) found for different load conditions.

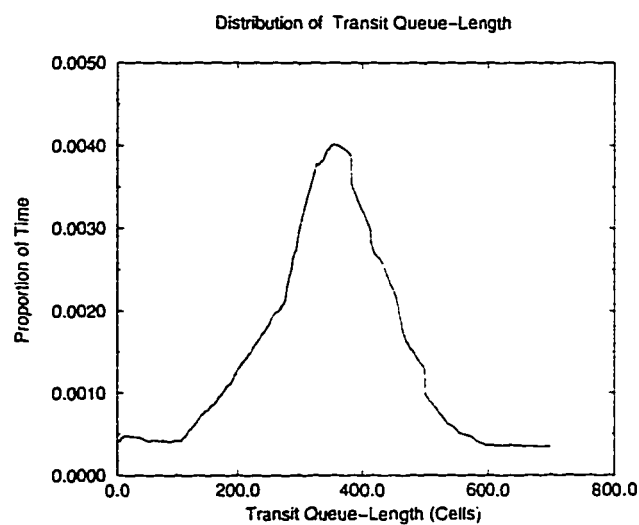


Figure 4.13: Effect of  $\rho$  on transit queue length ( $\rho = 0.7$ ).

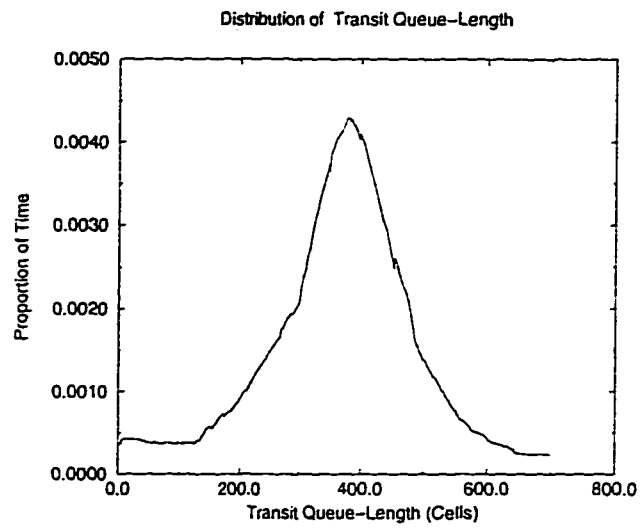


Figure 4.14: Effect of  $\rho$  on transit queue length ( $\rho = 0.75$ ).

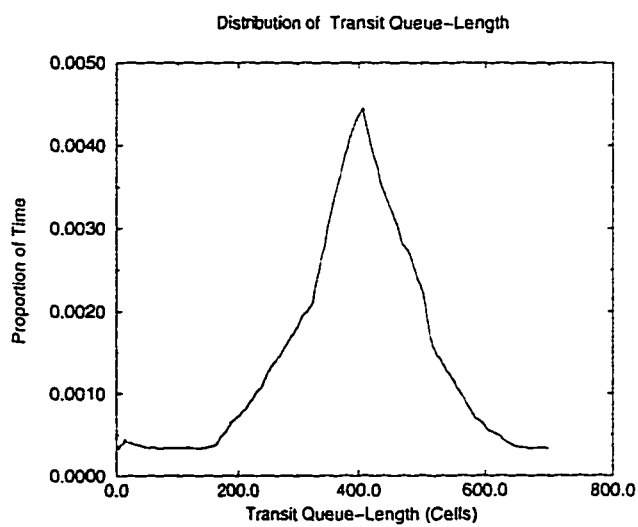


Figure 4.15: Effect of  $\rho$  on transit queue length ( $\rho = 0.8$ ).

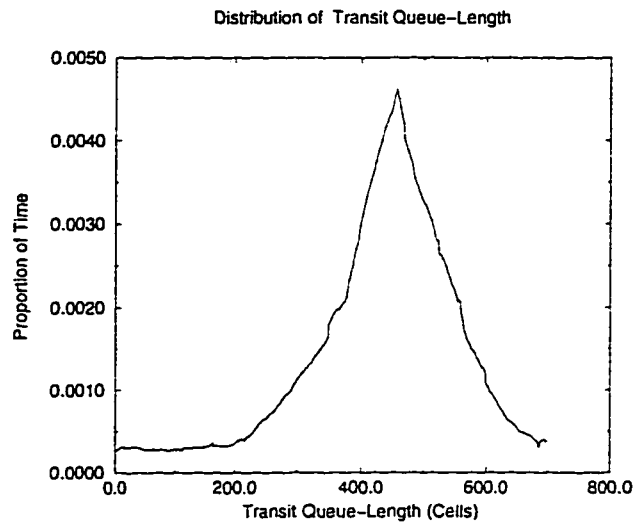


Figure 4.16: Effect of  $\rho$  on transit queue length ( $\rho = 0.85$ ).

## 4.6 Formulation of the Test

As a result of the extensive experimentation carried out, a strong correlation has been established between congestion and the link traffic intensity ( $\rho$ ). Also, a strong correlation between  $\rho$  and the mean length of the transit queue has been found. It was also found that the transit queue length is normally distributed.

On the basis of these observations, congestion control is performed as follows. We rely on hypothesis testing and use the transit queue length as a test statistic. The level of significance of the test,  $\alpha$ , is chosen to be 0.05. This is done to minimize the *typeII* error, that is, accepting  $H_0$  when it is not true, i.e., not invoking the control mechanism when there is a need. Minimizing the probability of a *typeII*



error would result in the minimization of cell loss.

A value of  $\alpha$  lower than 0.05 would help lower *type – I* error. In our case, the occurrence of a *type – I* error would occur when the control mechanism would be triggered even though there was no congestion. A decrease in the probability of one type of error results in an increase for the other. To help minimize both the type of errors, the population size of the test statistic should be reasonably large.

The population size is a function of the propagation delay. A low population size would result in an increase in the probability of the two types of errors. A large population size, on the other hand, would result in a slow control mechanism.

In this hypothesis test, we chose to work with the  $z$  values [46]. The value of  $z$  is determined as follows:

$$z = \frac{(M_{qt} - m_{qt}) \times \sqrt{n}}{\sigma}$$

where,

$M_{qt}$  = the observed mean transit queue-length.

$m_{qt}$  = the mean tested for.

$n$  = population size.

$\sigma$  = Standard deviation.

The value of  $z$  corresponding to  $\alpha = 0.05$  is 1.645. This served as our *critical value* ( $z_c$ ). A value of  $z$  from the hypothesis test that is greater than  $z_c$  would be enough evidence to reject  $H_0$  and to invoke the control process.

The value of  $m_{qt}$  to be used in the hypothesis test is decided as follows. It has

$B$	$\beta$	safe $ti$
100	10	0.65
200	5	0.67
100	5	0.66
200	10	0.63

Table 4.7: Search for a suitable traffic intensity.

to correspond to the maximum value of  $\rho$  at which the observed cell loss rate does not exceed the maximum cell loss rate to be guaranteed by the network.

For this purpose, another experiment was conducted which measured the value of  $\rho$  below which no cells are lost in the common queue. The simulations in this experiment didn't use any sort of congestion control. The results are shown in Table 4.7. The column safe  $ti$  refers to the value of the traffic intensity below which no cells are lost, when the traffic sources have the characteristics indicated in the first two columns of the table.

The results show that a value of  $\rho$  close to 0.70 would be reasonable to use in the test. From Table 4.6, the corresponding value of  $m_{qi}$  is about 345 (the value of the mean may change with different networks). The whole congestion control mechanism is now in place. The congestion control algorithm used is summarized in Figure 4.17.

In the congestion control algorithm, a sampling period is required to obtain the short term behavior of the network. To satisfy the central limit theorem, a minimum of 30 observations are required. After the sampling period, a check is made to see

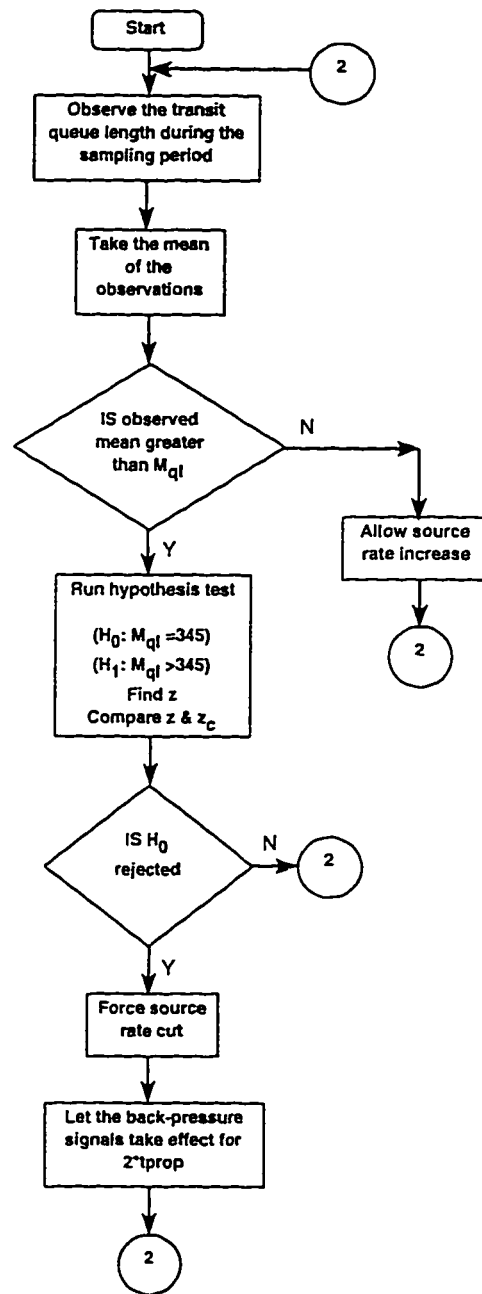


Figure 4.17: Flow chart of the statistical congestion control.

whether a hypothesis test is needed to be run or not. If the observed mean is too low, hypothesis test is not run and the sources are allowed to increase their rates, because there is no imminent threat of congestion. No action is taken when the hypothesis test is run and  $H_0$  is accepted. In this case, though the mean is greater than  $M_{qt}$ , the test says that this state is due to short lived bursts of cells. However, no rate increase is allowed, preventing the control from being too aggressive. Only when  $H_1$  is accepted, does a rate cut is forced and a time equal to  $2 \times t_{prop}$  is allowed to let the back-pressure signals take effect, otherwise the sources might be quenched by excessive back-pressure signals. This waiting time is not allowed when a rate increase is ordered, because, during that time bursts of cells might arrive, necessitating a run of the hypothesis test.

This chapter gave a detailed description of the steps that led to the development of the statistical congestion control scheme. The next chapter will provide the simulation results of the performance of the scheme.

## Chapter 5

# Simulation Results and Discussion

The performance of the proposed scheme, which we call the statistical scheme, was evaluated by extensive simulation. The queue delays, the cell loss, the link utilizations and the cell delay jitter were the observed performance metrics. The scheme was tested on various networks with different load conditions. The performance of the statistical congestion control scheme on these networks was reasonably similar. The difference in the performance was within a few percent (at most 5%). Therefore, the results for a typical network is presented in this chapter. The results of our scheme are compared with a scheme given in [49] which uses a queue-threshold based congestion detection mechanism and is referred to as the threshold based congestion control scheme. The scheme is similar to the BECN (Backward Explicit Congestion

Notification) scheme. A congestion notification signal is sent from the congested node to the up-stream nodes in both the schemes. However, unlike BECN, the congestion notification in the threshold based scheme does not travel beyond the immediate up-stream neighbor. The congestion notification does not travel all the way back to the traffic sources. This is done in order to reduce the control overhead and the reaction time. Recall that congestion occurs when the network is unable to provide the guaranteed service performance to the offered load. This may result in cells being lost, the throughput dropping and response time degrading.

The sensitivity analysis of the proposed scheme is also done. The purpose is to see if the performance of the scheme is too sensitive to its parameters. The analysis is accompanied by a comparison of the number of feedback throttle down signals sent by the statistical scheme and the threshold based scheme.

Finally, the performance of the statistical scheme is compared with BECN (Backward Explicit Congestion Notification) and FECN (Forward Explicit Congestion Notification) schemes.

The topology of the network is shown in Figure 5.1. The eight different load conditions are shown in Tables 5.1 and 5.2. There are ten different sources for each load condition. When the load is changed, the burst length of a source is kept constant (to keep the queue lengths from changing with the load), while its mean rate and burstiness are changed. For example, in Table 5.1, source number 1 has a mean rate of 5 Mbps for the first load and 7 Mbps for the second. The delay jitter

and cell loss tolerances are shown in Table 5.3. The minimum delay jitter tolerance is 10 msec. This means that the maximum observed delay jitter should not exceed 10 msec. The simulation time for each run was 10 minutes.

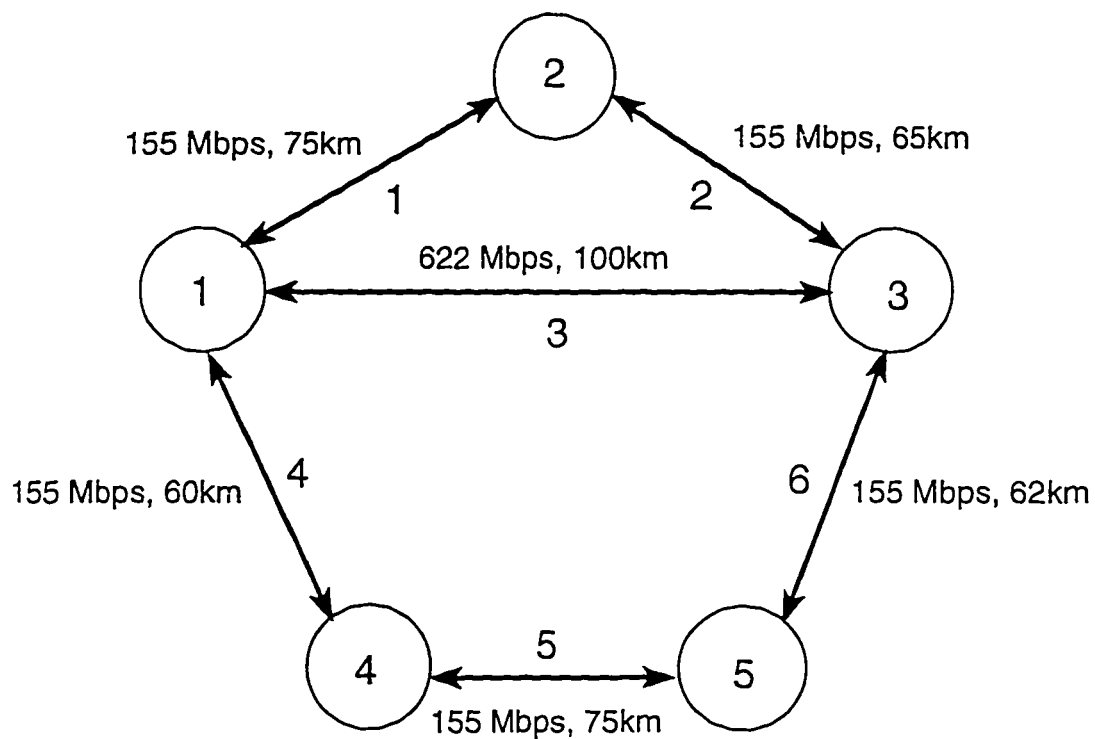


Figure 5.1: Network topology.

The states of all the transit and source queues were monitored during the simulation. Average delays per cell for the source queues for the threshold based and statistical control are shown in Tables 5.4 and 5.5, respectively. The tables show the average delay for each source queue under the different load conditions. The statistical control showed a slight increase in delay. This increase on average is

Source No.	Avg. Burst Length	Load 1		Load 2		Load 3		Load 4	
		Av. Rate (in Mbps)	$\beta$	Av. Rate (in Mbps)	$\beta$	Av. Rate (in Mbps)	$\beta$	Av. Rate (in Mbps)	$\beta$
1	1000	5	2	7	3	10	4	13	5
2	50	2	5	3	7	4	9	5	11
3	500	10	2	12	5	15	7	20	8
4	300	6	6	10	10	13	12	17	13
5	1200	15	3	20	5	20	7	24	7
6	200	5	8	8	10	10	10	15	15
7	800	10	5	13	8	15	9	18	10
8	1300	20	4	25	7	30	9	35	10
9	100	1	10	5	12	7	15	9	15
10	1500	10	3	11	6	14	7	17	8

Table 5.1: Traffic sources for different load conditions (continued).

Source	Load 5		Load 6		Load 7		Load 8	
	Av. Rate (in Mbps)	$\beta$	Av. Rate (in Mbps)	$\beta$	Av. Rate (in Mbps)	$\beta$	Av. Rate (in Mbps)	$\beta$
1	15	6	19	7	23	8	27	9
2	6	11	7	12	8	12	9	12
3	24	10	25	11	27	12	32	12
4	20	14	23	14	26	15	31	15
5	29	8	32	8	35	9	37	9
6	18	13	20	10	24	7	28	5
7	23	11	26	12	30	13	34	14
8	40	12	45	15	50	15	55	15
9	13	16	15	17	17	17	19	18
10	19	9	21	10	25	11	27	12

Table 5.2: Traffic sources for different load conditions.



Source No.	Delay Jitter Tolerance (in msec.)	Cell Loss Tolerance
1	70	$10^{-10}$
2	10	$10^{-6}$
3	10	$10^{-10}$
4	100	$10^{-9}$
5	10	$10^{-10}$
6	50	$10^{-9}$
7	80	$10^{-9}$
8	20	$10^{-10}$
9	10	$10^{-7}$
10	10	$10^{-10}$

Table 5.3: Delay jitter and cell loss tolerances.

about 4.5%. Table 5.6 shows a percentage increase in delays of source queues for the proposed scheme. This increase is due to the increased link utilizations, as will be shown later in this discussion.

A comparison of the average delays of transit queues is shown in Tables 5.7 and 5.8. Table 5.9 shows the percentage increase in delay for the transit queue for the statistical scheme. Again, an average increase of about 4.6% is observed.

The slight increase in queue delays for the proposed scheme is well compensated by an increase in link utilization. Table 5.10 shows the average link utilization for the threshold based control, while Table 5.11 shows the link utilization for the statistical control. Each row in the tables corresponds to the average link utilization for a single load condition. A reasonable increase in the link utilizations was observed for the

Source Queue	Average delay (in msec.) for different loads							
	1	2	3	4	5	6	7	8
1	2.80	3.18	3.92	4.21	4.76	5.02	5.62	6.11
2	3.13	3.81	4.23	4.72	5.19	5.64	6.28	6.87
3	2.01	2.46	2.98	3.59	4.02	4.51	4.87	5.29
4	3.15	3.62	4.09	4.69	5.45	6.02	6.81	7.52
5	2.91	3.38	3.95	4.61	5.09	5.52	6.12	6.62
6	3.39	3.85	4.38	4.89	5.69	6.20	7.01	7.69
7	3.22	3.71	4.16	4.81	5.49	6.11	6.71	7.58
8	2.72	3.05	3.69	4.32	4.91	5.48	6.15	6.71
9	3.11	3.67	4.13	4.65	5.06	5.54	6.12	6.81
10	2.73	3.31	3.79	4.46	5.21	5.71	6.29	6.89

Table 5.4: Average delays of source queues for threshold based control.

proposed scheme. The average increase was about 5.5%. The percentage increase in link utilization is shown in Table 5.12.

This increase in link utilization was achieved without degrading the cell loss behavior. Figure 5.2 shows the comparison of the cell loss for the two schemes. Both the curves overlap, showing that both the schemes perform equally well with respect to cell loss.

A comparison of the cell delay jitter shows that the proposed scheme results in a smaller delay jitter. Figure 5.3 shows the cell delay jitter comparison. This is due to the fact that the statistical scheme generates lesser number of control signals, thereby, resulting in a smoother operation. The comparison of the number of control signals generated is given later in the section. Both the schemes satisfied the minimum delay jitter requirement, which was set equal to 10 msec.

Source Queue	Average delay (in msec.) for different loads							
	1	2	3	4	5	6	7	8
1	2.94	3.34	4.09	4.41	4.95	5.26	5.87	6.41
2	3.30	3.95	4.41	4.91	5.42	5.91	6.58	7.19
3	2.12	2.58	3.11	3.76	4.19	4.73	5.11	5.54
4	3.29	3.79	4.25	4.86	5.70	6.31	7.14	7.89
5	3.06	3.51	4.10	4.78	5.31	5.79	6.41	6.94
6	3.55	3.96	4.57	5.11	5.90	6.50	7.34	8.07
7	3.35	3.87	4.31	4.97	5.71	6.39	6.99	7.94
8	2.85	3.15	3.82	4.49	5.09	5.72	6.41	7.03
9	3.24	3.79	4.27	4.81	5.28	5.81	6.40	7.14
10	2.86	3.44	3.92	4.62	5.45	5.98	6.58	7.23

Table 5.5: Average delays of source queues for statistical control.

Source Queue	Percentage change in delay for different loads								
	1	2	3	4	5	6	7	8	Avg.
1	0.05	0.05	0.04	0.05	0.04	0.05	0.04	0.05	0.046
2	0.05	0.04	0.04	0.04	0.04	0.05	0.05	0.05	0.045
3	0.05	0.05	0.04	0.05	0.04	0.05	0.05	0.05	0.047
4	0.04	0.05	0.04	0.04	0.05	0.05	0.05	0.05	0.044
5	0.05	0.04	0.04	0.04	0.04	0.05	0.05	0.05	0.044
6	0.05	0.03	0.04	0.04	0.04	0.05	0.05	0.05	0.043
7	0.04	0.04	0.04	0.03	0.04	0.05	0.04	0.05	0.041
8	0.05	0.03	0.04	0.04	0.04	0.04	0.04	0.05	0.041
9	0.04	0.03	0.03	0.03	0.04	0.05	0.05	0.05	0.041
10	0.05	0.04	0.03	0.04	0.05	0.05	0.05	0.05	0.043

Table 5.6: Percentage change in delays of source queues.

Transit Queue	Average delay (in msec.) for different loads							
	1	2	3	4	5	6	7	8
1	2.62	2.98	3.56	4.15	4.82	5.49	5.98	6.53
2	1.69	2.16	2.65	3.22	3.83	4.43	4.85	5.28
3	2.77	3.48	3.91	4.41	4.95	5.48	5.89	6.48
4	2.48	3.14	3.71	4.23	4.77	5.19	5.77	6.29
5	1.68	2.12	2.62	3.20	3.92	4.49	4.89	5.31
6	3.03	3.54	4.02	4.72	5.09	5.78	6.26	6.72
7	2.45	3.16	3.60	4.34	4.85	5.18	5.77	6.26
8	2.94	3.43	3.95	4.68	5.17	5.84	6.29	6.79
9	3.01	3.48	3.98	4.65	5.06	5.72	6.21	6.66
10	2.59	2.93	3.64	4.21	4.74	5.40	5.84	6.48
11	2.85	3.54	3.93	4.58	5.16	5.51	5.98	6.50
12	2.82	3.49	3.85	4.41	5.08	5.59	6.15	6.65

Table 5.7: Average delays of transit queues for threshold based control.

Transit Queue	Average delay (in msec.) for different loads							
	1	2	3	4	5	6	7	8
1	2.73	3.11	3.72	4.34	5.05	5.75	6.27	6.85
2	1.75	2.25	2.82	3.37	4.01	4.63	5.09	5.54
3	2.88	3.59	4.09	4.62	5.17	5.73	6.18	6.79
4	2.57	3.28	3.87	4.43	4.98	5.44	6.05	6.61
5	1.75	2.21	2.74	3.35	4.11	4.71	5.13	5.57
6	3.17	3.70	4.21	4.91	5.33	6.05	6.57	7.05
7	2.56	3.29	3.75	4.54	5.08	5.43	6.05	6.58
8	3.06	3.58	4.14	4.91	5.41	6.13	6.58	7.13
9	3.15	3.63	4.16	4.87	5.30	5.98	6.51	6.99
10	2.69	3.04	3.81	4.42	4.95	5.66	6.13	6.81
11	2.98	3.69	4.12	4.79	5.41	5.78	6.27	6.83
12	2.95	3.65	4.03	4.62	5.32	5.85	6.45	6.98

Table 5.8: Average delays of transit queues for statistical control.

Transit Queue	Percentage change in delay for different loads								
	1	2	3	4	5	6	7	8	Avg.
1	0.04	0.04	0.04	0.05	0.05	0.05	0.05	0.05	0.046
2	0.04	0.04	0.05	0.05	0.05	0.05	0.05	0.05	0.047
3	0.04	0.03	0.05	0.05	0.04	0.05	0.05	0.05	0.044
4	0.04	0.04	0.04	0.05	0.04	0.05	0.05	0.05	0.045
5	0.04	0.04	0.05	0.05	0.05	0.05	0.05	0.05	0.047
6	0.05	0.05	0.05	0.04	0.05	0.05	0.05	0.05	0.046
7	0.04	0.04	0.04	0.05	0.05	0.05	0.05	0.05	0.046
8	0.04	0.04	0.05	0.05	0.05	0.05	0.05	0.05	0.047
9	0.05	0.04	0.05	0.05	0.05	0.05	0.05	0.05	0.047
10	0.04	0.04	0.05	0.05	0.04	0.05	0.05	0.05	0.046
11	0.05	0.04	0.05	0.05	0.05	0.05	0.05	0.05	0.047
12	0.05	0.05	0.05	0.05	0.05	0.05	0.05	0.05	0.047

Table 5.9: Percentage change delays of transit queues.

Load No.	Link Utilization						Av. util
	Link 1	Link 2	Link 3	Link 4	Link 5	Link 6	
1	0.49	0.49	0.51	0.51	0.48	0.49	0.495
2	0.55	0.52	0.55	0.52	0.51	0.53	0.530
3	0.61	0.58	0.65	0.54	0.56	0.58	0.586
4	0.69	0.62	0.71	0.62	0.61	0.66	0.651
5	0.72	0.68	0.75	0.69	0.65	0.72	0.701
6	0.77	0.74	0.78	0.75	0.73	0.76	0.755
7	0.81	0.80	0.81	0.79	0.78	0.81	0.800
8	0.82	0.81	0.83	0.82	0.81	0.83	0.820

Table 5.10: Link utilizations for threshold based control.

Load No.	Link Utilization						Av. util
	Link 1	Link 2	Link 3	Link 4	Link 5	Link 6	
1	0.52	0.52	0.54	0.54	0.51	0.52	0.525
2	0.58	0.55	0.58	0.55	0.54	0.56	0.560
3	0.64	0.61	0.68	0.57	0.61	0.62	0.622
4	0.71	0.65	0.74	0.67	0.66	0.69	0.687
5	0.75	0.72	0.79	0.73	0.69	0.75	0.738
6	0.79	0.78	0.82	0.81	0.77	0.81	0.797
7	0.83	0.84	0.85	0.84	0.84	0.85	0.842
8	0.85	0.86	0.87	0.87	0.86	0.86	0.862

Table 5.11: Link utilizations for statistical control.

Load No.	Percentage change for different links						Av. change
	1	2	3	4	5	6	
1	0.061	0.061	0.059	0.059	0.063	0.061	0.061
2	0.055	0.058	0.055	0.058	0.059	0.057	0.057
3	0.049	0.052	0.046	0.056	0.089	0.069	0.060
4	0.029	0.048	0.042	0.081	0.082	0.045	0.054
5	0.042	0.059	0.053	0.058	0.062	0.042	0.052
6	0.026	0.054	0.051	0.080	0.055	0.066	0.055
7	0.025	0.050	0.049	0.063	0.077	0.049	0.052
8	0.037	0.062	0.048	0.061	0.062	0.036	0.051

Table 5.12: Percentage change in link utilizations.

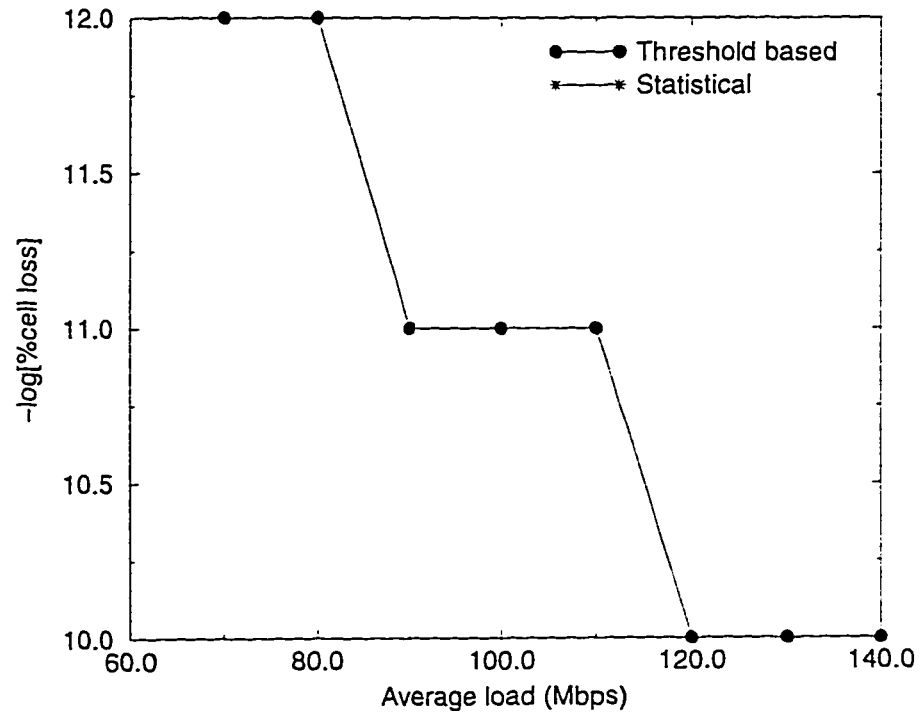


Figure 5.2: Percentage cell loss versus average load.

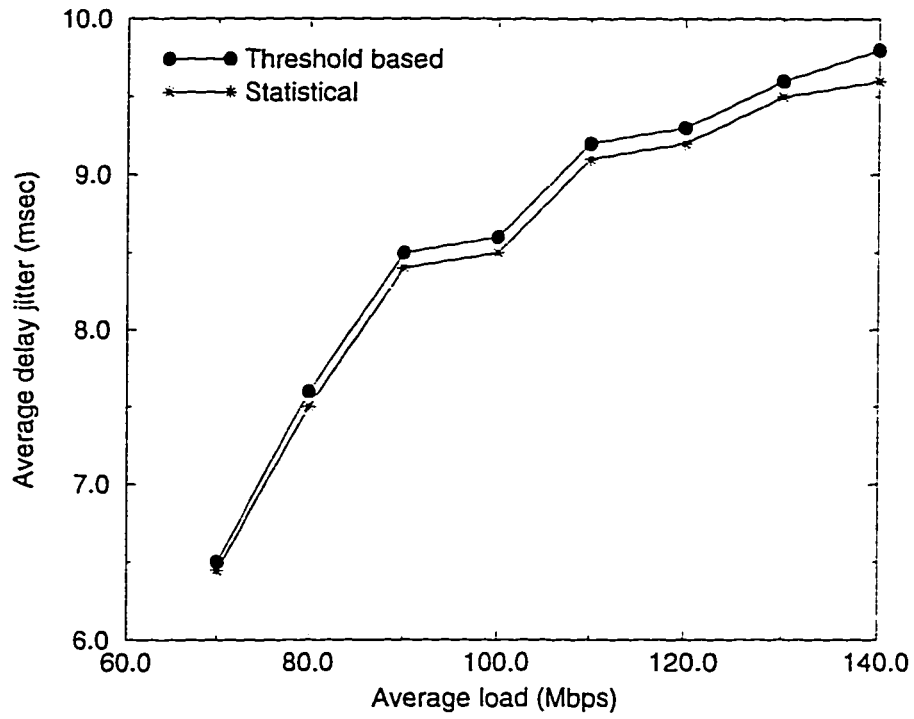


Figure 5.3: Average delay jitter versus average load.



Load	Sampling Period (statistical scheme)				Threshold based control	%diff within statistical	%diff from threshold based
	0.25tprop	0.5tprop	0.75tprop	1.0tprop			
70%	916	928	934	932	1000	1.9	-6.8
75%	929	910	927	925	1000	1.6	-7.5
80%	917	913	907	919	1000	1.3	-8.1
85%	902	901	909	913	1000	1.2	-9.5

Table 5.13: Sensitivity of the statistical scheme to the sampling period.

## 5.1 Results of Sensitivity Analysis

This section presents the results of the sensitivity analysis for the proposed statistical scheme. This was done to determine how sensitive the statistical scheme was to its parameters. The simulation time for each run was 10 minutes.

### Sensitivity to the Sampling Period

The results of the sensitivity analysis are given in Table 5.13. The analysis was done for four different load conditions. This load represents the load on the link to which the observed transit queue is attached. The table shows, for every load condition, the number of back-pressure signals sent by the statistical scheme for every 1000 back-pressure signals sent by the threshold based scheme. There are four observations, for each load condition, for the back-pressure signals sent by the statistical scheme. Each observation corresponds to a different sampling period. For example, for a load of 70%, the number of back-pressure signals sent by the statistical control (for

every 1000 signals sent by the threshold based scheme) is 916 when the sampling time is  $0.25 \times t_{prop}$  and is 928 when the sampling period is  $0.5t \times prop$ . Table 5.13 also shows the percentage difference in the number of back-pressure signals sent by the statistical scheme, for the same load condition, when the sampling period is changed. For example, this difference is 1.9% for a load of 70%. The last column of the table shows the percentage difference in the number of back-pressure signals sent by the threshold based scheme and the statistical scheme (with a sampling period of  $1.0 \times t_{prop}$ ). The observations made as a result of the sensitivity analysis are as follows:

- to minimize the probability of either of the *type-I* or *type-II* errors, the sampling period should be large enough to allow for a reasonable population size.
- sensitivity analysis showed that the control scheme is not very sensitive to the size of the sampling period. This is due to the fact that, even though the size is varied to 4 times the minimum sampling period, the smallest period had enough size to give a reasonably close performance, in terms of the number of back-pressure signals sent, to that of the largest period run.
- as can be seen in Table 5.13, the difference between the results of the four runs, i.e., the difference in the number of back-pressure signals sent gets lesser with an increase in the load. This is expected, because at a higher load, the traffic is so large that even a relatively small sample time gives a close performance

Load	Tested mean (statistical scheme)				% decrease in signals within statistical	Threshold based scheme
	345	355	365	375		
70%	932	928	919	904	1.9	1000
75%	925	922	914	900	1.6	1000
80%	919	915	912	898	1.3	1000
85%	913	910	905	896	1.2	1000

Table 5.14: Sensitivity of the statistical scheme to the mean.

to the larger one. (The probability of the occurrence of congestion is large at very high loads and the mean queue-length is higher at higher loads.)

- the difference between the threshold based and statistical schemes becomes larger as the load increases, because, the statistical scheme does the statistic analysis after  $2 \times t_{prop}$  before acting, while the threshold based scheme just reacts after  $2t_{prop}$ . The extra time spent in the statistic analysis is a time well spent, because, the new scheme gives a better link utilization at comparable cell loss. This comparison was done between the new scheme employing  $2 \times t_{prop}$  sampling time and the threshold based scheme.
- the performance of the schemes with the 4 different sampling times is expected to be very close. Therefore, only the results for one ( $t_{prop}$  sampling time) have been reported.

## Sensitivity to the Tested Mean

The results of the sensitivity analysis are given in Table 5.14. This analysis was also done for four different load conditions. The table shows, for every load condition, the number of back-pressure signals sent by the statistical scheme for every 1000 back-pressure signals sent by the threshold based scheme. There are four observations, for each load condition, for the back-pressure signals sent by the statistical scheme. Each observation corresponds to a different tested mean. For example, for a load of 70%, the number of back-pressure signals sent by the statistical scheme (for every 1000 signals sent by the threshold based scheme) is 932 when the tested mean is 345 and is 904 when the tested mean is 375. The table also shows the percentage decrease in the back-pressure signals sent when the tested mean is varied from 345 to 375. The observations made as a result of the sensitivity analysis are as follows:

- the difference in the number of back-pressure signals sent decreases with the increase in the load, because, at higher loads the mean queue-length increases.
- even at the lowest offered load, the difference in the number of back-pressure signals sent is at most 3% different. Therefore, the simulation results for only one have been reported (assumed mean = 345).

Load	$\alpha$	
	0.05	0.1
70%	1000	1096
75%	1000	1087
80%	1000	1080
85%	1000	1068

Table 5.15: Sensitivity of the statistical scheme to  $\alpha$ .

### Sensitivity to the Confidence Level of the Test

The results of the sensitivity analysis for the level of significance ( $\alpha$ ) are given in Table 5.15. The observations made are as follows:

- the sensitivity analysis was done for an  $\alpha$  of 0.05 and 0.1.
- the table shows the number of back-pressure signals sent by the control for  $\alpha = 0.1$ , for every 1000 back-pressure signals sent by the control with  $\alpha = 0.05$ .
- when  $\alpha$  is increased (an increase in the probability of making a *type - I* error), the average increase in the number of back-pressure signals is about 10%.
- the control with  $\alpha = 0.1$  turns out to be more conservative in dealing with congestion. This would result in a decreased resource utilization and increased overhead processing.

## Summary of Sensitivity Analysis

An in-depth sensitivity analysis of the proposed statistical congestion control mechanism was undertaken. The analysis was done at four load conditions for the sampling period, the tested mean and  $\alpha$ . It was found that the mechanism is not too sensitive to the sampling period. Similar inference was made for the supposed mean. However, the analysis for the  $\alpha$  showed that the control is reasonably sensitive to it. This was expected because a change of  $\alpha$  from 0.05 to 0.1 would mean a larger probability of making a *type – I* error. Hence, there would be more rejection of  $H_0$ , resulting in the generation of more back-pressure signals.

## 5.2 Comparison of Statistical Scheme with BECN and FECN

The proposed scheme was compared with BECN (Backward Explicit Congestion Notification) and FECN (Forward Explicit Congestion Notification) schemes. These two schemes are among the most widely prescribed schemes for congestion control in ATM networks [33]. Before giving the details of the comparison, a summary of the working of BECN and FECN is given.

## Working of BECN

Two thresholds are placed on each transit queue. The first (smaller) threshold is called the relief threshold. If the transit queue's length goes below this relief threshold, a congestion relief signal is sent "back" to the sources. The sources can increase their rates upon its reception. The second (larger) threshold is called the congestion threshold. A throttle down signal is sent "back" to the sources when the transit queue length reaches or goes beyond the congestion threshold. BECN control then waits for  $2 \times t_{prop}$  time before it can send another throttle-down signal if the the queue length is still beyond the congestion threshold.

## Working of FECN

The working of FECN is similar to BECN. However, the congested node sends the signals to the destination nodes instead of sending it back to the sources. The destination nodes then inform the source nodes. The waiting time after sending a throttle-down signal is equal to double the propagation delay from the source nodes to the destination nodes.

Instead of using the relief threshold procedure, we could have used another method in which the sources would increase their rates after not receiving a throttle-down signal during a certain time interval. This was not adopted due to two reasons:

- In case the network is highly congested the throttle-down signals might be lost

on their way to the sources. The sources, in this case, would increase their rates upon not receiving the throttle-down signal. This could result in network collapse.

- The determination of the time interval itself is not easy.

## Details of Comparison

The test network used for the comparison is shown in Figure 5.4. For the test network, BECN has double the propagation delay and FECN has four times the propagation delay than that of the statistical scheme. BECN and FECN may have much larger delays in actual networks. Comparison was done with respect to the relative cell loss as well as the number of throttle-down signals sent by the three schemes. Four different load conditions were used for the comparison.

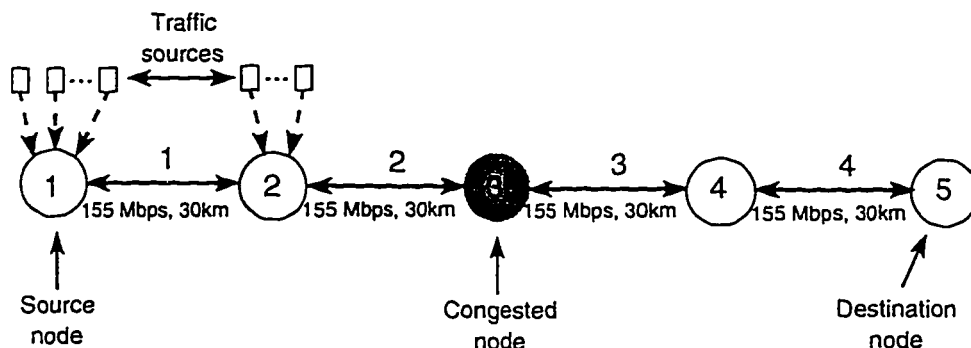


Figure 5.4: Test network.



Scheme	70% load	75% load	80% load	85% load
Statistical	1	1	1	1
BECN	15	19	34	55
FECN	24	32	58	97

Table 5.16: Cell loss comparison of statistical scheme against BECN and FECN.

### Comparison of Cell Loss

The cell loss comparison is summarized in the Table 5.16. The table shows the number of cells lost in BECN and FECN per single cell lost in the statistical scheme.

Following are the observations made regarding the comparison:

- The difference in cell loss appears to increase sharply with the increase in load.
- FECN losses the most number of cells, because of the much larger propagation delay involved.
- The cell loss in BECN/FECN is up to 97 times more than that in the statistical scheme. In some cases this difference in cell loss could effect the guaranteed QoS in BECN/FECN.
- The lower delays involved in the statistical scheme allows it do deal more effectively with cell loss, making it more attractive.

Scheme	70% load	75% load	80% load	85% load
Statistical	1000	1000	1000	1000
BECN	481	490	538	598
FECN	232	241	270	310

Table 5.17: Comparison of control signals sent by statistical scheme against BECN and FECN.

### Comparison of Throttle-down Signals Generated

The comparison of the throttle-down signals generated by the three schemes is shown in Table 5.17. The table shows the number of congestion signals sent by BECN/FECN per 1000 signals sent by the statistical scheme. Following are the observations made regarding the comparison:

- The statistical scheme sends more throttle-down signals than BECN/FECN.
- The difference in throttle-down signals gets smaller with increased load.
- The increased delay restricts the reaction of FECN/BECN and results in increased cell loss.

BECN and FECN show lower performance on cell loss, because of the inherent large delays involved. These delays are largely independent of the medium and are hard to deal with.

### 5.3 Discussion

This section discusses the performance of the statistical scheme. As discussed in Chapter 1, the performance of a congestion control scheme can be measured by its following characteristics: efficiency, fairness, distributedness, responsiveness, and convergence. The processing overhead in the proposed statistical scheme is somewhat greater than the threshold based scheme. This increased overhead is due to the hypothesis test. However, the statistical scheme tries to limit the overhead by checking whether it is necessary to run the hypothesis test or not. The statistical scheme, like the threshold scheme, lacks fairness because, it forces every source to cut its rate. Even the sources which are not violating their contract will be effected. This is the price that has to be paid to keep the control simple and responsive. The statistical scheme has a distributed control. There is no central control involved. Congestion is controlled locally with the interaction of the effected node and its neighboring nodes. The statistical congestion control scheme was also able to converge the network to its equilibrium state (normal operation) every time it was called upon to. The following subsections discuss the performance metrics (utilization, delay, jitter and loss).

## **Link Utilization**

From the analysis of the link utilization performed in the above section, it is evident that the statistical control scheme results in reasonable increase in link utilization. This is achieved through a more intelligent control which sends lesser feedback signals and allows more freedom to the traffic sources.

## **Queue Delays**

Due to the higher link utilization, increased queue occupancy was observed. This resulted in increased average queue delays. However, the increase, in general, was only about 4.5%. This increased delay was still well within the range of the QoS agreement.

## **Delay Jitter**

The delay jitter performance of the statistical scheme was better than that of the threshold based scheme. This was due to the fact that the statistical scheme sends lesser control signals. This results in a more smoother operation. Both the schemes, however, satisfied the minimum delay jitter requirement of the traffic sources.

## Cell Loss

Despite the fact that the statistical scheme results in higher link utilization and lesser delay jitter, its cell loss performance is comparable with that of the more conservative threshold based scheme. This shows the success of a more intelligent control mechanism, satisfying the QoS agreement and increasing resource utilization.

## Comparison with BECN and FECN

The performance of the statistical scheme is better than BECN/FECN with respect to cell loss. The lower cell loss allows the statistical scheme to provide the guaranteed QoS to the traffic sources. While, it may not be possible in some cases for BECN/FECN. However, the statistical scheme sends more throttle-down signals than BECN/FECN.

This chapter presented the simulation results of the performance of the proposed statistical congestion control scheme. The next chapter presents the conclusions that are made as a result of this thesis.

# Chapter 6

## Conclusion

In this thesis, a new approach to congestion control in ATM networks has been presented. The developed scheme is able to give a (short term) prediction of the state of the network. The scheme does this prediction by taking the statistical information of the traffic sources into account. Extensive simulations were conducted to design and evaluate the scheme.

Chapter 1 gives an introduction to ATM and B-ISDN. The need for traffic engineering and other related issues are discussed. The importance of an effective congestion control strategy is also explained. The choice of a traffic source model constitutes an important part of the development of a congestion control scheme. Chapter 2 describes the different traffic source models available. Chapter 3 gives an in-depth survey of the available literature related to traffic engineering, in particular congestion control. Chapter 4 describes, in detail, the steps that led to the develop-

ment of the congestion control scheme. It gives the description of hypothesis testing and why it is chosen in our congestion control structure. Extensive experimentation results are presented. The experiments are conducted to tune the control structure. The final control structure is also described. Chapter 5 gives the simulation results and discussion. The results of the proposed statistical scheme have been compared with a threshold based scheme.

The performance of the proposed statistical congestion control scheme has been shown to be better than the threshold based congestion control scheme. The results hold for all the different load conditions and test networks. The statistical scheme allows the network to enjoy higher resource utilization. This is made possible due to the more intelligent control mechanism, which takes the statistical information of the traffic sources into consideration. The cell loss performance of the statistical scheme is comparable with that of the threshold based scheme. The cell delay jitter gets even smaller with the statistical scheme. This is due to the fact that the statistical scheme generates lesser number of feedback signals, offering a smoother control.

The sensitivity analysis of the statistical scheme showed that the scheme is sensitive, but not extremely sensitive, to its parameters. The analysis for the sampling period shows that the sampling period should be large enough to satisfy the central limit theorem. The analysis for the mean-tested-for shows that as the mean is increased, the number of feedback signals decreases. Within a reasonable limit, corresponding to the experiments conducted, the variation in the mean-tested-for

does not result in a radical change in the behavior of the scheme. The analysis for  $\alpha$  shows that the scheme is sensitive to the change in the level of significance ( $\alpha$ ).

The performance of the statistical scheme is also compared with BECN and FECN schemes. The performance of the statistical scheme is better than BECN/FECN with respect to cell loss. However, the statistical scheme sends more throttle-down signals than BECN/FECN.

## 6.1 Future Work

The nature of the congestion control problem is such that there is always room for improvement. We think that there are still many areas where improvements can be made. The following improvements are suggested to enhance the performance of the scheme:

- Division of transit queue into multiple queues with different priority levels.
- Controlling the output rate of the transit queue upon congestion notification from downstream node.
- Prioritizing the *source* queues based upon the type of source.

Some of the improvements that can be made in the simulator are as follows:

- Developing a graphical user interface to make the simulator user friendly.



- Employing object-oriented paradigm so that addition of new functions to the simulator becomes easy.
- Developing a distributed simulator to reduce the simulation time.
- Using different traffic models to effectively model different types of traffic sources.

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# Vitae

- Adnan Ahmed Khan
- Received Bachelor of Engineering (B.E.) degree in Computer Systems Engineering from N.E.D. University of Engineering and Technology, Karachi, Pakistan in 1993.
- Worked for IBM at Karachi in 1993-94.
- Received Master of Science (M.S.) degree in Computer Engineering from KFUPM, Saudi Arabia in October, 1996.