# PACKET SWITCHING IN WIDE AREA BROADBAND PRIVATE NETWORKS

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**Doctor of Philosophy**

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## Summary

B-ISDN is a universal network which supports diverse mixes of service, applications and traffic. ATM has been accepted world-wide as the transport technique for future use in B-ISDN. ATM, being a simple packet oriented transfer technique, provides a flexible means for supporting a continuum of transport rates and is efficient due to possible statistical sharing of network resources by multiple users. In order to fully exploit the potential statistical gain, while at the same time provide diverse service and traffic mixes, an efficient traffic control must be designed. Traffic controls which include congestion and flow control are a fundamental necessity to the success and viability of future B-ISDN.

Congestion and flow control is difficult in the broadband environment due to the high speed link, the wide area distance, diverse service requirements and diverse traffic characteristics. Most congestion and flow control approaches in conventional packet switched networks are reactive in nature and are not applicable in the B-ISDN environment. In this research, traffic control procedures mainly based on preventive measures for a private ATM-based network are proposed and their performances evaluated.

The various traffic controls include CAC, traffic flow enforcement, priority control and an explicit feedback mechanism. These functions operate at call level and cell level. They are carried out distributively by the end terminals, the network access points and the internal elements of the network. During the connection set-up phase, the CAC decides the acceptance or denial of a connection request and allocates bandwidth to the new connection according to three schemes; peak bit rate, statistical rate and average bit rate. The statistical multiplexing rate is based on a 'bufferless fluid flow model' which is simple and robust. The allocation of an average bit rate to data traffic at the expense of delay obviously improves the network bandwidth utilisation.

During the data transfer phase, variable bit rate data traffic flow is enforced to its average bit rate using a window scheme. The window size and the respective window interval may be adaptively changed to meet the required delay limit. At the network nodes, a two level space priority mechanism is employed. The mechanism, based on partial buffer sharing with threshold, allows a maximum limited load of higher priority delay sensitive traffic to access the network. Non-delay sensitive traffic, however, is subject to flow control based on the threshold. This control mechanism is considered fair since the QOS for each bearer service class is guaranteed. In the the latter part of the work, the window scheme is extended to a flow control mechanism based on explicit feedback notification. The mechanism is basically a congestion avoidance strategy where the source is notified of the status of the internal network elements and is instructed to react according to the message in the acknowledgement .

The performance of the various traffic control capabilities is assessed against the high level goals; simplicity, flexibility, robustness and optimality and the results prove that they can be adequately employed in private broadband networks with some constraint on the network physical dimension.

**Key words:** ATM, Traffic control, Congestion control, CAC, Window flow control.

*Dedicated to my family*

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## ABBREVIATIONS

| | |
|---|---|
| ATM | : Asynchronous transfer mode |
| AAL | : ATM adaptation layer |
| B-ISDN | : Broadband integrated services digital network |
| CAC | : Connecttion admission control |
| CBR | : Continuous bit rate |
| CCITT | : Comite Consultatif International Telegraphique et Telephonique |
| CSDN | : Circuit switched data network |
| DQDB | : Distributed queue dual bus |
| FDDI | : Fiber distributed data interface |
| FIFO | : First-in first-out |
| ECN | : Explicit congestion notification |
| HDTV | : High defination television |
| LAN | : Local area network |
| MAN | : Metropolitan area network |
| MLT | : Maximum Laxity Threshold |
| NT1 | : Network termination 1 |
| NT2 | : Network termination 2 |
| NNI | : Network node interface |
| OSI | : Open system interconnection |
| PABX | : Public automatic branch exchange |
| PSDN | : Packet switched data network |
| QLT | : Queue length Threshold |
| SDH | : Synchronous Digital Hierarchy |
| SONET | : Synchronous Optical Network |
| STM | : Synchronous transfer mode |
| TA | : Terminal adaptor |
| UNI : | : User network interface |
| VBR | : Variable bit rate |
| VC | : Virtual channel |
| VCI | : Virtual channel identifier |
| VP | : Virtual path |
| VPI | : Virtual path identifier |
| VPN | : Virtual private network |
| UPC | : Usage parameter control |

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# CHAPTER 1

# INTRODUCTION

## 1.1 Trends in telecommunication technology

Current telecommunication networks are specialised in character. For example, a separate network exists to transport telex traffic and another exists to transport voice traffic (PSTN). Some networks transport data based on X.25 (PSDN) while others on X.21 (CSDN), and a different network exists to transport television signals (eg. cable television, direct broadcast satellite). Even narrowband ISDN is only suitable for a selection of services with bit-rates smaller than 2 Mbit/s. The narrowband services are not really integrated as there are two different bearer services; packet switched and circuit switched, resulting in two overlay services. Only access to the user is integrated.

The consequences of this specialisation is the existence of a number of independent, inflexible networks each requiring their own design phase, manufacturing and maintenance. The dimensioning of these networks is done per information type for its worst case traffic condition and is ill-suited for other applications. Therefore, resources which are still available in one network cannot be used in another network. This suggests the need for one universal network which is capable of transporting all teleservices by sharing all its available resources between the different teleservices in a flexible and service independent manner [1,2].

Besides the need to integrate the various services and networks, other main motivating forces behind further development of this universal network are

- user demands for broadband services largely seen in three sectors; data communication with high bit rates e.g. for interconnecting LANs, video communication with good image quality, e.g. for video conference and video telephony and distribution TV and HDTV via fibre network.

- the availability of high speed transmission, switching and signal processing technologies

- the advances in software application processing capabilities in the computer and telecommunication industries

- the improved data and image processing capabilities available to users

Two important characteristics of these broadband services are the high bandwidth and high burstiness requirements. To satisfy these needs the future universal network must suppport a wide variety of information transport needs and must be capable of dealing with rapid changes in the needs of a particular service over time.

### 1.1.1 B-ISDN

Broadband Integrated Services Digital Network (B-ISDN) is intended to meet the need for a universal network [1]. It is conceived as an all purpose digital network supporting a wide variety of applications in a flexible and cost-effective manner. The bandwidth spectrum ranges from a few bits per second up to 600 Mbit/s covering telemetry, voice, data, image and video such as Extended Quality (EQ) video and HDTV. In addition, the network allows distributive, interactive and multimedia communication. For the purpose of understanding the range of possible services and the capabalities needed, CCITT has classified teleservices into two catogeries [1] as shown in table 1.1;

Table 1.1 Two categories of teleservices

| Category | Teleservices |
|---|---|
| 1) Interactive services (bidirectional) | - conversational services (e.g. voice)<br>- retrieval services (e.g. image,video retrieval)<br>- messaging services (e.g. electronic mail) |
| 2) Distributive services | - services without user presentation control (e.g. conventional TV-distribution service)<br>- services with user presentation control (e.g. cabletext) |

B-ISDN with service independent capabilities will have many advantages. It will be flexible and future safe since it is capable of transporting all types of services and will be able to adapt to changing needs. While the available resources will be used very efficiently, the overall cost of the design, manufacturing, development, implementation, operation, and maintenance will be more economical since only one network will be used. Ideally the B-ISDN network should provide information transport as a basic resource, while remaining insensitive as to how the resource is used.

## 1.2 Integrated Network - Transport technology

Figure 1.1 depicts a continuum of techniques available for the transport of integrated services. In general, techniques towards the right of this continuum provide increasing flexibility to handle variable bit rate and burst information, while requiring more processing.

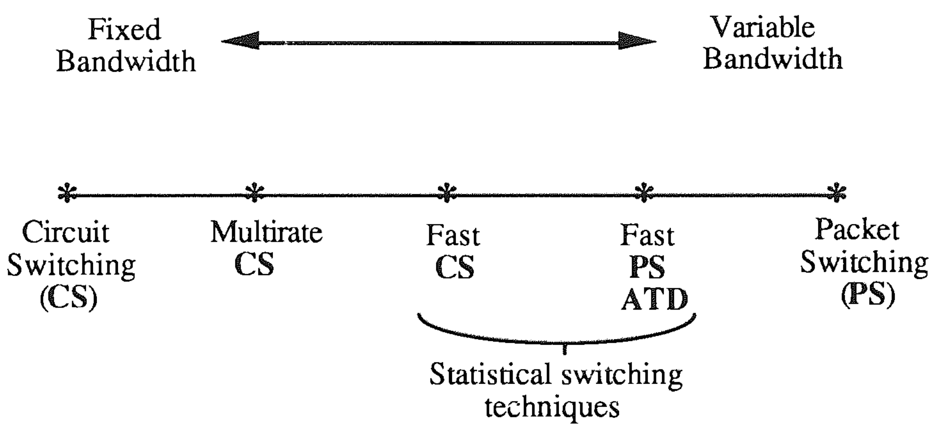


Figure 1.1 Switching technology continuum

Circuit switching has been used almost universally for speech transmission using a single transmission rate since telephony began. With the advent of digital switching, a multirate circuit switched (MCS) network based on Synchronous Transfer Mode (STM) has evolved to cater for the integration of voice and data transmission. Digital transmission hierarchies were then introduced into the ISDN with a 2B+D channel structure for the basic rate interface (B = 64 kb/s and D = 16 kb/s) and a nB+D for the primary rate interface (n = 24 in North America, n = 30 in Europe). An STM channel is identified by the position of its time slots within the synchronous structure. Initially it was assumed that STM would serve as the paradigm for B-ISDN [4]. The interface for a B-ISDN could be expressed as:

$$i\,H_4 + j\,H_3 + k\,H_2 + l\,H_1 + m\,H_0 + n\,B + D$$

where $H_0 \ldots H_4$ are high speed channels bit-rate oriented towards the bit-rate of the plesiochronous digital hierarchy (PDH). The coefficients $i.,\ldots n$ indicate the number of occurrences of a particular type of bearer channel on an interface. This structure is an extension of narrowband ISDN obtained by adding higher bit-rate channels to the previously standardized nB+D structures.

However, the suitability of the STM paradigm for B-ISDN was questionable [5, 8]. The main concern is its inflexibility to carry a dynamically changing mix of services at a variety of fixed channel rates. The question of how many combinations of H and B channels should be incorporated in the broadband interface cannot be resolved. If the channel rates are defined, the network must live with these rates for the rest of its existence. Another problem is in regard to the resource allocation. In a circuit switched network, physical resources are reserved as soon as a call is set-up. The channel is then allocated with the peak bit-rate of the traffic source. With bursty services the channel usage will be very inefficient.

Fast circuit switching (FCS) extends the concept of circuit switching by allowing the network to respond to bursty services [16]. With FCS, a circuit is not reserved when the customer makes a call, but is set up only when bandwidth is needed and torn down when there is no information to send, freeing the resource for other calls. Circuit set up and tear down can be carried out using a fast signalling mechanism such as Common Channel Inter-office Signalling (CCIS). This approach has also been called 'burst switching' [18].

However, a switch may not always be able to satisfy the request to activate a virtual circuit (VC), because bandwidth is not reserved for it. This results in the request being queued, which may cause delay due to buffering and information loss. In addition, as in CS, FCS does not allow the connection of users operating at different rates, which is a common need in data communication.

Packet switched networks, on the other hand, are the most flexible networks in terms of bandwidth requirements and the most efficient in terms of resources usage [129, 133]. No rate has to be specified in advance. However, it has problems in terms of delay and processing performance, especially in transporting continuous bit rate (CBR) services with stringent time constraints such as voice. Since packet-switching can only operate at medium-to-low speed the delay and the jitter on this delay will be quite large, making it impossible to reconstruct the CBR services and thus to guarantee the required time transparency.

An evolution in system concept for packet oriented networks has recently occurred. In 1983, an ideal mixture of both techniques (circuit switched and packet switched) was introduced. In the United States this technique was introduced by Turner [9] and referred to as Fast Packet Switching. In Europe it was called Asynchronous Time Division (ATD), and was introduced by Coudreuse [10].

The major idea behind this new concept is a combined use of the packet switching and time division multiplexing techniques, with all excess transport functions (semantic and time transparency) being removed to the edge of the network [7]. By doing this the transport functions are not allowed to be repeated in the network several times, while guaranteeing the required grade of service. This technique is known as asynchronous transfer mode (ATM) as opposed to synchronous transfer mode (STM) which is based on multirate circuit-switching.

### 1.2.1 Universal transport - Why Asynchronous Transfer Mode?

ATM has been accepted by CCITT Study Group XVIII as the ultimate information transfer mode for all types of services [3], the main reason being its universal flexibility to cope with the uncertainties of the future as described in [7, 8, 11,12].

ATM has been influenced by two important parameters; the progress in technology and the progress in system concepts. The considerable progress in the field of electronics and optical equipments has enhanced VLSI logic speed so as to provide fast switching techniques and greater communication potential running at high transmission rates. For example, future optical based networks will be capable of transmission at channel bit rates of up to 600 Mb/s in the user network interface (UNI) of B-ISDN.

Another key enabler of broadband capability is provided by the introduction of synchronous multiplexing at the UNI. The standardization of synchronous transmission and multiplexing techniques was pioneered in the United States under the name of SONET (Synchronous Optical Network) [13]. The same principles are now being incorporated into CCITT recommendations to produce a world wide standardisation. ATM has been proposed as the target protocol for carrying information within the SONET/SDH payload.

The merit of ATM as the future transfer mode in B-ISDN includes;

- **flexibility** in supporting a wide range of service types and mixes on demand basis

- **efficiency** gained by statistical multiplexing of bursty sources from different connections

- **simplicity** due to a simplified protocol to achieve high throughput

A further description of ATM concepts will be given in chapter 2.

## 1.3 Traffic engineering in B-ISDN

One of the most challenging problems in B-ISDN is how to handle the vast variety of different services with different traffic parameters and performance requirements. The network provider will of course wish to accept any requests for service since they are the source of revenue. However, the network provider also has the responsibility to ensure that the admission of the new connection will not adversely affect the performance of other connections sharing the same links in the network. Traffic controls will therefore be required in this network to ensure that sufficient bandwidth and satisfactory performance is provided to all multiplexed connections.

Where efficient utilisation of network resources such as bandwidth is not a concern, traffic controls are simple. Resources can be reserved to support each connection's peak bandwidth requirements and connections carried in a non-statistically multiplexed mode. However, to strive for more efficient resource utilisation through statistical multiplexing, the traffic control could potentially be very complex as it must limit the cell loss and delay caused by statistical fluctuations in the traffic levels at each multiplexing and switching point in the network.

Any traffic management schemes which are adopted will directly affect the complexity of network system hardware, and have an impact on a traffic structures. However, they are necessary if B-ISDN services are to be provided economically. The key challenge is the equitable and the efficient allocation of network resources, that is, to design effective traffic controls which allow a reasonable utilisation of the network resources while remaining both operationally simple and robust to the evolving ATM network and traffic environment.

### 1.3.1 Problems encountered

The new technologies for transmission and switching in an ATM environment give rise to many new practical and theoretical questions relating to traffic engineering such as

- traffic models for the different existing services

- suitable traffic models for the new services

- resource allocation strategies for the various services

- defining service quality parameters

In general, a series of assumptions must be made for the anticipated traffic behaviour whose justification will only become evident when B-ISDN can be monitored. This is because the main unpredictable factors in introducing B-ISDN are not so much in the technical field but are more related to the degree of successful user response.

## 1.4 ATM technology in advanced private networks

Private networks have traditionally been developed as customised networks, aiming at both high performance and economy. Hence, new technologies have often been adopted into private networks in advance of public ones. For example, PBXs offered digital interfaces and advanced data features prior to public ISDN services. High speed end-to-end digital transmission technology was also introduced first in into private networks resulting in wide area communication capabilities using leased lines.

As user demands for high speed multimedia communication increased, the following communication capabilities and facilities became necessary;

- high speed local area network (LAN) interconnection capability beyond local premises

- integrated multimedia communication facilities for local premises

- efficient inter-site multimedia communication facilities.

High speed LAN interconnections up to 100 Mb/s around a single site over a restricted distance of 100 km can be implemented by Fibre Distributed Data Interface (FDDI) [129]. FDDI uses a token ring protocol and runs over a fibre optic link. The second generation FDDI II offers both asynchronous and synchronous data transfer capability.

For inter-site interconnection these problems can be solved by Metropolitan Area Networks (MANs) [23]. The Distributed Queue, Dual Bus (DQDB) technology has been standardised as the IEEE 802.6 protocol for use in MAN. MANs are larger than LANs but smaller than

WANs. They use similar medium access control schemes as LANs and cover the area of a large city. The disadvantage of such a system is that it is limited in geographical coverage.

Another solution is to resort to virtual private networks (VPNs) based on the high speed public packet network. A VPN service such as the Frame Relay Service (FRS) is efficient as it takes advantage of statistical multiplexing, is able to support multiple logical connections across a single physical interface and offers possible savings. However, FRS is only appropriate for data communication users who are looking for a more cost effective alternative to private lines for LAN and host interconnections.

Alternatively, it would be possible to tackle these problems from the view point of private networks using high speed leased lines, as it is expected that fibre based wideband (50 Mb/s and/or 150 Mb/s) leased lines will be released in advance of their B-ISDN services [21, 22]. The network would span over a wide area featuring ATM technology. Figure 1.2 depicts an example of a private network configuration in a local premises connected to a wide area network.

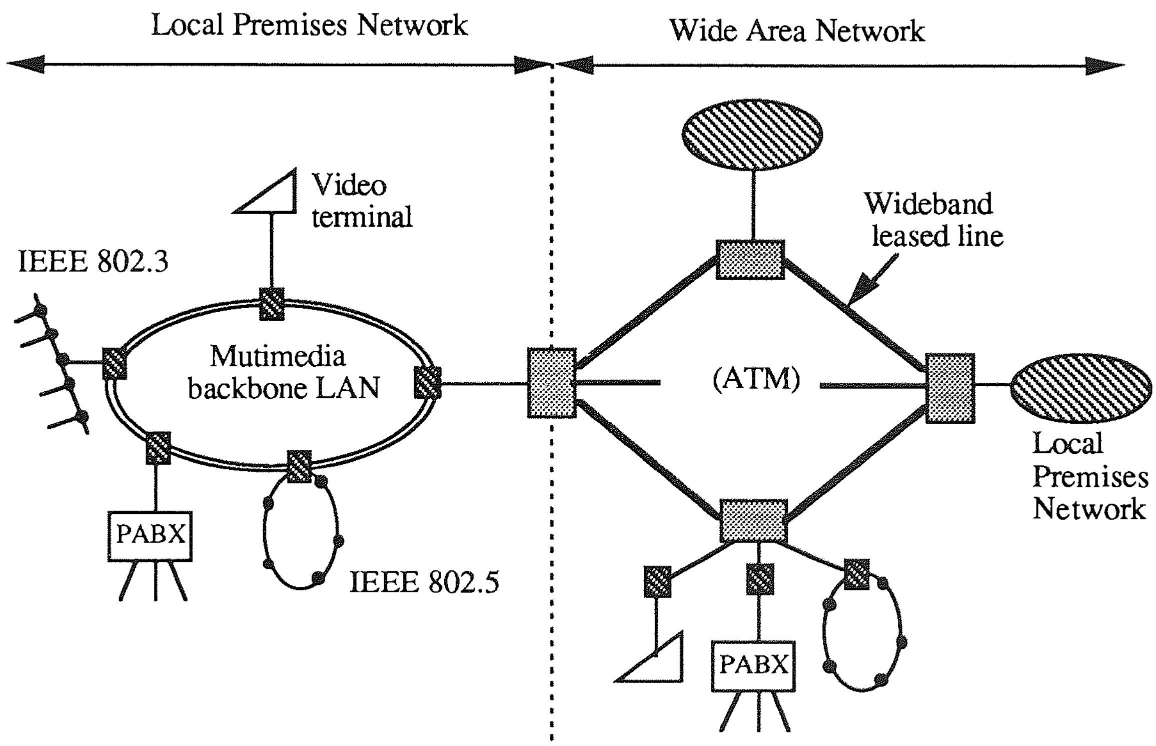


Figure 1.2 Private network configurations

Although the cost of using leased lines in the long run is lower than using public networks, it is still expensive to communicate freely between distance sites. Thus, efficient bandwidth allocation and multiplexing techniques for high speed leased lines utilisation are required. In such a wide area private network, ATM would be the most promising technology to satisfy such needs. Moreover, ATM would also be suitable for integrated multimedia networks in local premises.

Private networks are more flexible than public networks in that they are not bound by standards which have to be approved by CCITT. Since congestion and flow control is one of the most important and the most difficult problems in realising BISDN [30-36], these difficulties can be optimised and/or simplified in private networks due to their flexibility and perhaps limited applications.

## 1.5 Thesis outline

Due to the promising advantage of ATM technology, the basic concepts of ATM networks were studied for the implementation of a private broadband network. The objective of the research work was to investigate the use of an advanced packet switching technique, ATM, in wide area broadband private networks. In particular, the work concentrated on the congestion and flow control of the network. Congestion and flow control in a network of high link capacity involves several problems. The design of the network congestion and flow control must, therefore, take into account several factors;

- accurate source traffic modelling for stream and bursty types
- cell processing scheme must be performed at speeds comparable to the high switching speed.
- multiple quality of service (QOS) requirements must be handled effectively
- resources allocation for diverse traffic characteristics must be resolved especially when statistical multiplexing is involved
- congestion/flow control scheme must be based on the propagation delay-bandwidth product and cell processing scheme.

In this work preventive congestion and flow control mechanisms were proposed and their performance investigated in the light of the above factors. The preventive congestion/flow control is carried out at three levels, call, burst and cell level. The thesis is organised as follows.

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Chapter 2 begins with giving a brief scenario on the evolution of telecommunication systems towards the future B-ISDN. It goes on to give an overview of ATM that will be used in the future B-ISDN. The issues relating to traffic control in the broadband environment are discussed and the approach in tackling them is outlined.

Chapter 3 concentrates on one of the preventive congestion control methods, the connection admission control. It begins by differentiating the various problems encountered in the analysis of superposition of homogeneous traffic sources reported in the literature. A fluid flow mechanism is analysed and the bufferless fluid flow mechanism is selected to be used for connection admission control for statistical traffic based on the conservative measure and simplicity. Four categories of traffic are identified and they form the basis of resource allocation in the following work.

Chapter 4 introduces the various traffic flow enforcement mechanisms. A simple algorithm designed for category 2 services is presented. Besides enforcing the source traffic to the required parameters, the algorithm acts as a smoothing function to the bursty traffic source.

Chapter 5 deals with controlling quality of service in terms of cell loss and delay requirements through priority mechanisms. Two levels of priority control are identified. The respective qualities of service are maintained through a space priority scheme based on a modified partial buffer sharing with threshold.

Chapter 6 presents a feedback mechanism scheme based on an explicit notification. The window mechanism developed in chapter 4 is used to flow control the data flow at the access node based on the congestion status. The mechanism is designed as a form of congestion avoidance control.

Chapter 7 reviews the modelling process of the various strategies in the trafic control. The different source traffic models along with the assumptions made are presented.

Finally, chapter 8 concludes the thesis by assessing the various traffic control capabilities.

# CHAPTER 2

## ATM NETWORKS PROTOCOLS AND TRAFFIC CONTROL

### 2.1 Evolution scenario of B-ISDN/ATM networks

The development of B-ISDN is a challenging task for network providers and manufacturers. Despite the convincing advantages of being a universal network for all services, the high development and investment costs mean that B-ISDN cannot in the short term achieve a wide coverage. The attempt must therefore be to expand and add to the existing telecommunication networks in a market and demand oriented manner with suitable technical concepts in time phases and gradual transition. Any individual or intermediate solutions must be designed such that they can later be incorporated with the least possible expense in the integrated broadband networks.

The evolution towards B-ISDN differs from country to country depending on the existing telecommunication network structures and the customers' needs of each country. Current trends show that communication demands with respect to bandwidth largely comes from the business community [6, 20]. In the business community environment, logically and geographically homogeneous communication demands are served by LANs, PABXs, etc. These networks are mostly privately owned and operated.

LANs are primarily employed for data communications in the in-house area covering up to 10 km. with transmission speeds in the range of 1 - 16 Mb/s. LANs can be classified by their topology (bus, ring, star), the transmission medium (twisted pair, coax cable, optical fibre) or the medium access control procedure (carrier sense multiple access or token passing). The first step in coupling LANs was the interconnection via existing networks like the circuit switched public data network (CSDN), packet switched public data network (PSDN) or the ISDN. LAN interconnection can also be carried out by dedicated lines leased from the public telephone company. British Telecom offers leased links at speeds of 64 kb/s (Kilostream) and 2.048 Mb/s (Megastream). In the existing networks the transmission rates are very low (eg, 64 kb/s - 2.048 Mb/s) whereas LANs may require transmission rates up to 16 Mb/s [5]. Obviously, this interworking strategy does not provide good performance.

Recently, FDDI has been standardised by ANSI X3T9.5 committee for high speed LANs. FDDI can also provide LAN interconnection at speeds of up to 100 Mb/s. At the same time, there already exists a demand for high capacity backbone networks for high network throughput from applications requiring rapid transfer of large data objects such as high resolution interactive image transfer, video conferencing and LAN interconnection beyond the local area. This leads to the introduction of high speed MANs.

MANs based on DQDB consist of two buses with a multiplicity of nodes attached to these buses [23]. Initially MANs will provide only connectionless services and be used for LAN interconnection. In future MANs could be extended to support isochronous and connection oriented services. Increasing communication requirements will lead to the interconnection of MANs via dedicated links as shown in figure 2.1.

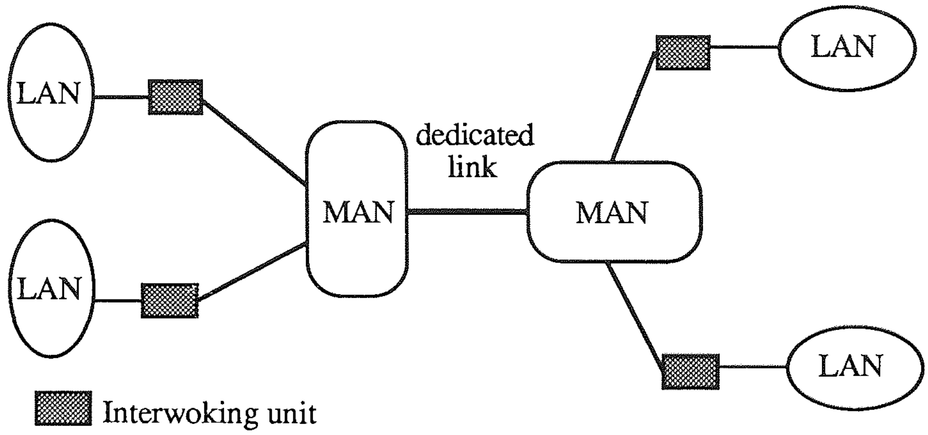



Figure 2.1 Interconnections of LANs via MAN

B-ISDN must in any case interwork with the existing services and networks such as the public data network (X.25), 64 kb/s ISDN, etc. The introduction of B-ISDN, however, will be compelled mainly by the need for interconnecting MANs [6]. This allows access to wider areas with more flexibility. With the emergence of B-ISDN, LANs, private MANs and WANs can be directly coupled via the B-ISDN. During this phase more individual users will be connected immediately through an adaptor to the B-ISDN without undergoing the evolutionary path. Simultaneously new broadband ATM terminals will be connected directly to either public or private B-ISDN. Figure 2.2 depicts the target B-ISDN/ATM architecture.

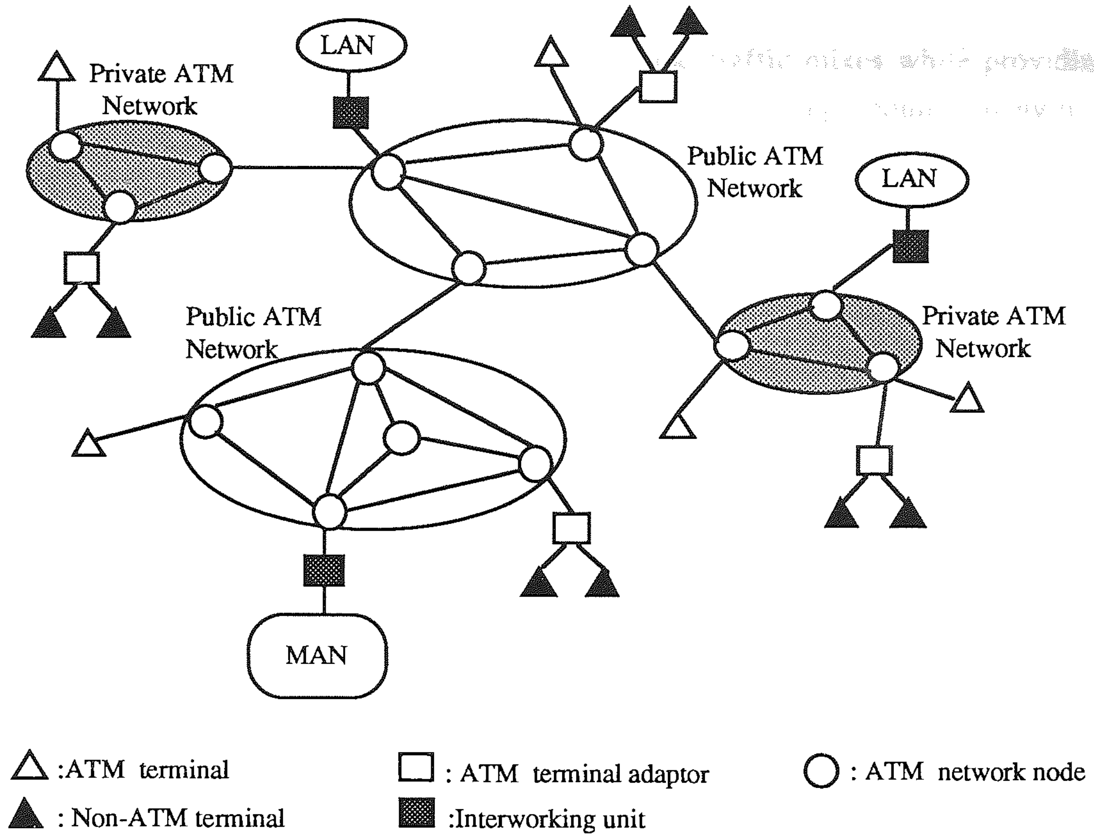



Figure 2.2 A B-ISDN/ATM target architecture

There are many possible evolutionary paths towards tomorrow's diverse and complex B-ISDN. In the ultimate B-ISDN scenario everything is supposed to be transmitted in ATM. The success of the realisation of this target, however, depends on the ability to resolve the following issues [35,36]:

* Interoperability of different ATM devices across ATM interfaces. This includes effective and complex ATM-based networking arrangements.

* Efficiency that can be achieved through statistical gain for more effective and economic network utilisation.

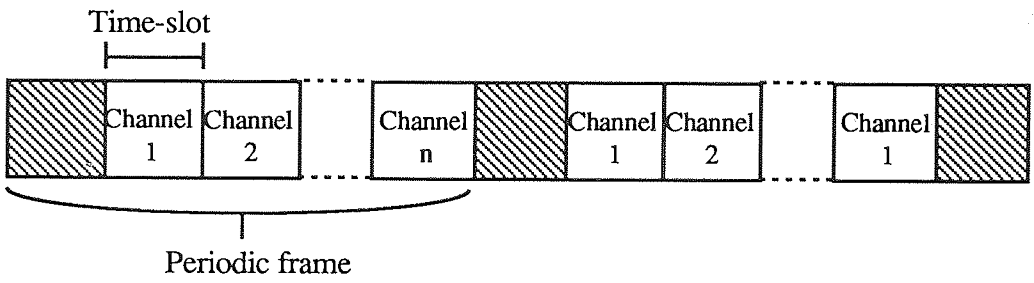
* Effectiveness of control structures and a robust means of supporting various services and applications through a hierarchy of control levels. Further elaboration on control level hierarchy is given in section 2.6. These control structures have an impact on the viability of ATM information transport.

B-ISDN/ATM goals are to support diverse service and traffic mixes while providing efficient network resource engineering. However, the ATM concept requires many new problems to be solved. For example, the impact of possible cell loss, cell transmission delay and cell delay variation on service quality requirements must be determined. These matters require the design of an effective traffic and congestion control management. In fact, the success and viability of B-ISDN/ATM is mainly influenced by the ability to meet this challenge. Before going into details on the subject of traffic and congestion control it is essential for the concept of ATM to be fully understood. In the next section, important features of ATM are elaborated.

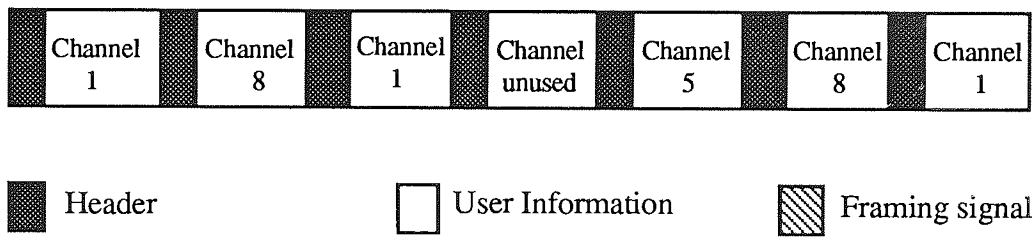
## 2.2 Asynchronous Transfer Mode (ATM)

ATM is a packet oriented transfer mode using an asynchronous time division multiplexing technique. The term transfer comprises of both transmission and switching aspects. Thus, a transfer mode is a specific way of transmitting and switching information in a network. The term asynchronous refers to the fact that cells allocated to the same connections may exhibit an irregular recurrence pattern as cells are assigned on demand depending on the source activity and the available resources. This is shown in figure 2.3(b).

In contrast to STM ,where a given channel is identified by its position in the transmission frame (see figure 2.3(a)), an ATM cell associated with a specific virtual channel may occur at essentially any position in the transmission frame. The flexibility of bit rate allocation to a connection in STM is therefore restricted due to predefined channel bit rates (eg. B, HO, etc.) and the rigid structure of the conventional transmission frame. This will not allow individual structuring of the payload and thus only a limited selection of channel mixes at subscription time is permitted at the corresponding interface.




(a) Synchronous Transfer Mode (STM)




(b) Asynchronous Transfer Mode (ATM)

Figure 2.3 STM and ATM principles

ATM is, by definition, a connection oriented technique. The connection oriented mode minimizes delay variation since cells belonging to the same call follow the same route. It also reduces the processing required to make the routing decisions. The cell sequence integrity is also preserved under normal fault free conditions.

The multiplexed information flow in ATM is organised in fixed sized blocks, called cells. A cell is made up of 53 bytes and consists of a header and a user information field as shown in figure 2.4(a). The primary role of the header is to identify the characteristics of the virtual channel on a multiplex link. All information to be transferred is identified and switched by means of labels in the header. In a way similar to packet switching, ATM can provide a communication with a bit rate that is individually tailored to the actual need, including time invariant bit rates. Since the multiplexing and switching of cells is independent of the actual application, ATM networks can in principle handle low bit-rate as well as high bit-rate connections, either of stream type or bursty in nature.

### 2.2.1 ATM cell format

The size of an ATM cell is designed small in order to reduce the degrading effect of packetisation delay at the source [29]. Furthermore, short cells coupled with the high transfer rate will result in transfer delays and delay variations which are sufficiently small for a wide range of applications including real-time services such as voice and video.

A cell is made up of 5 bytes header and 48 bytes information field. For the user network interface (UNI), the header consists of a 4-bit generic flow control (GFC), a 24-bit label field containing Virtual Path Identifier (VPI) and Virtual Circuit Identifier subfields, a 2-bit payload type (PT) field, a 1-bit reserved field, a 1-bit priority (PR) field and an 8-bit header error check (HEC) field. The cell header in the UNI differs from that at the network node interface (NNI) in the use of bits 5 - 8 of octet 1 for GFC as depicted in figure 2.4(b). At the NNI these bits are used for VPI, giving 12 bits altogether, thus providing enhanced routing capabilities.

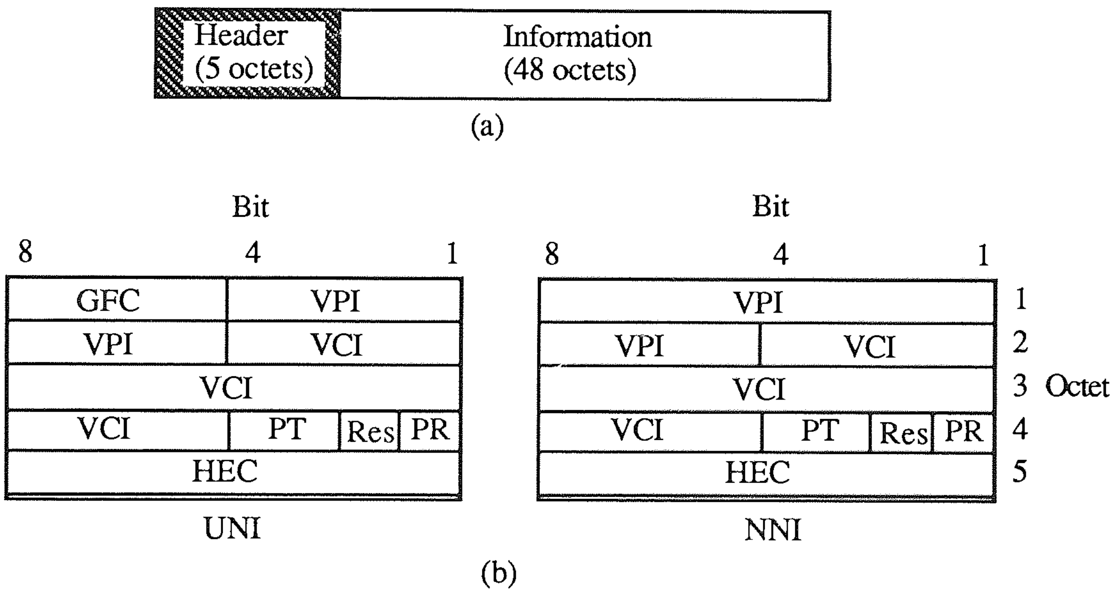



Figure 2.4 (a) ATM cell structure
(b) Cell header at the BISDN UNI and NNI

The header functionalities include sufficient functions to transport the information field through the ATM layer. These necessary functions are:-

- GFC appears only at the UNI. The GFC mechanism assists in the control of the traffic flow from ATM connections at the UNI. For example, it can be used for individual throttling of cell flow of each terminal to alleviate short-term overload situations.

- routing using VPI and VCI. The VPI provides an explicit path identification for a cell while the VCI provides an explicit circuit identification for a cell.

- PT field indicates whether a cell contains user information or network maintainance information.

- occupancy indication (unassign cells) to differentiate cells for the use of the ATM layer from those cells only used at the physical layer.

- priority/Quality Of Service indication for explicit priority indication. Cell loss priority is used to selectively discard cells when congestion occurs.

- header error check field (HEC) provides single-bit-error correction or multiple error detection on the cell header. The polynomial used to generate the header error check is $X^8 + X^2 + X + 1$. The HEC monitors errors for the entire header. The processing is done at the physical layer.

### 2.2.2 ATM Layer architecture and functions

A layered model is used to describe the architecture of the ATM protocol [8, 12] as shown in figure 2.5. The three layers, adaptation layer, ATM layer and physical media dependent (PMD) layer map different services to different implementations of switching, multiplexing or transmission applications.

The physical layer includes all the physical medium dependent (PMD) functions. The PMD will match the ATM cells to the particular network implementation. For example, to insert the cells in a Synchronous Digital Hierarchy (SDH) frame structure or to generate a Synchronous Optical Network (SONET) frame structure. Other functions include bit timing, transmission frame adaptation, cell delineation, HEC sequence generation and a comparison and cell rate decoupling mechanism.

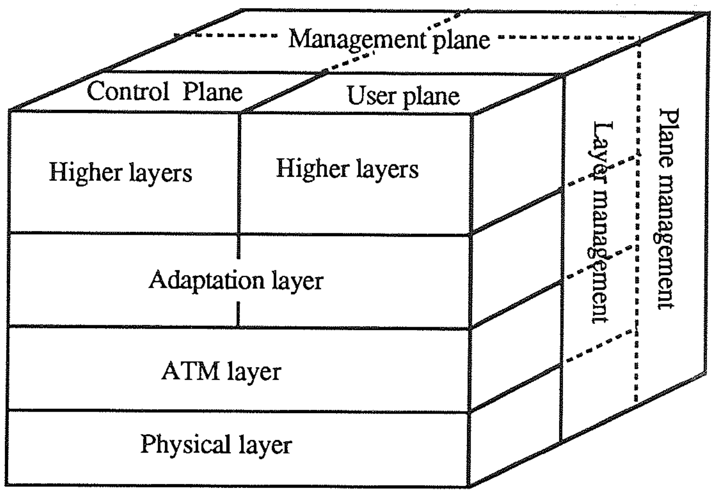



Figure 2.5 B-ISDN protocol reference model

The ATM layer contains the details of the ATM functions. It is where all the information is in the common form of ATM cells. This layer is physical medium independent. Cell header functions are performed in this layer. The major function of the cell header includes routing and error detection on the header. Cell based multiplexing/demultiplexing is also performed in this layer. The information field is transported transparently by the ATM layer without any processing such as error control being performed on the information field at the ATM layer. A more detailed description of the ATM layer protocol is given in [5, 140].

Since ATM offers a flexible transfer capability common to all services, the ATM adaptation layer (AAL) on top of the ATM layer is provided to accommodate various services. The AAL specific information is contained in the information field of the ATM cell. The AAL provides the higher service layer with the necessary functions which are not provided by the ATM layer such as preserving timing, data frame boundaries and source clock. To minimize the number of AAL protocols, four service classes specific to the AAL are proposed by CCITT (see Table 2.1). The classification of these services is based on the timing relation between source and destination, bit-rate (constant or variable) and connection mode.

Table 2.1 Service classification for AAL

| | Class A | Class B | Class C | Class D |
|---|---|---|---|---|
| Timing relation between source and destination | Required | | Not required | |
| Bit rate | Constant | Variable | | |
| Connection mode | Connection oriented | | | Connection -less |

Class A services correspond to the constant bit-rate (CBR) services. CBR voice and video belong to this class. Class B services are variable bit rate connection oriented services such as variable bit rate audio and video. Class C services include VBR connection oriented data services. Class D services correspond to VBR connection-less data services.

The AAL is divided into two sublayers, the Segmentation and Reassembly (SAR) Sublayer and the Convergence Sublayer (CS) These two sublayers provide different functions for each of the four service classes. Further details of their usage can be found in [5]. Until now, four AAL protocols types have been defined for the four AAL service classes. Figure 2.6 shows a model protocol for LAN interconnection. Note that segmentation and reassembling is done at the sending and receiving nodes only. These nodes could be an interworking unit (e.g. a bridge).

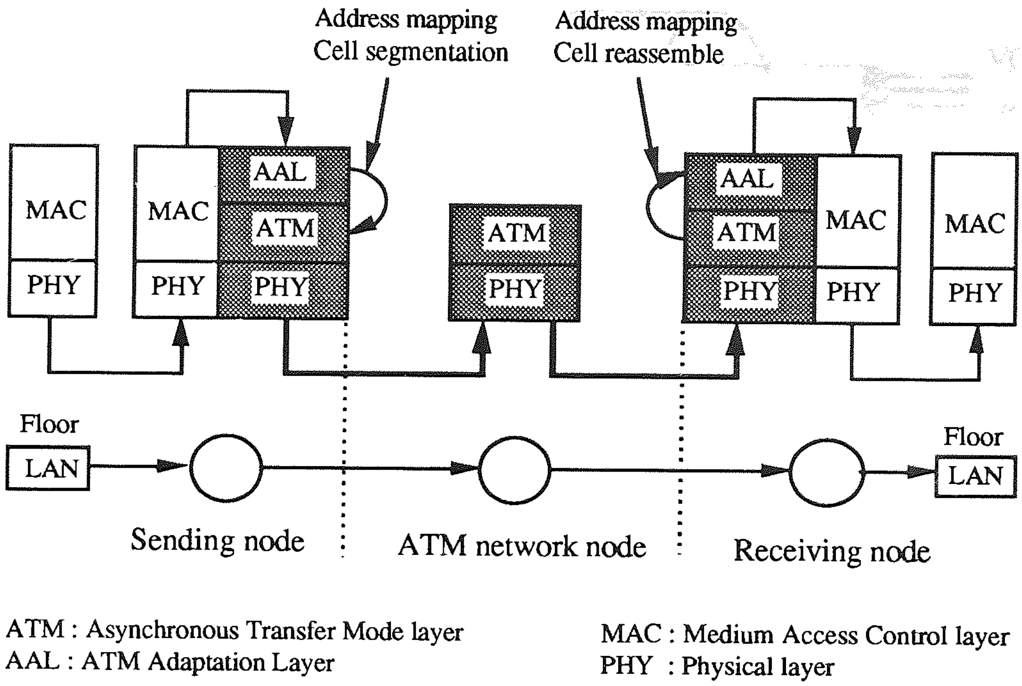



Figure 2.6 Protocol model for LAN interconnection

### 2.2.3 ATM layer connections

A basic component of the B-ISDN is a network for ATM switching of both constant bit-rate and variable bit-rate end-to-end connections. These connections, with information transfer rates up to 600 Mb/s, are carried by ATM at the B-ISDN UNI and at switching entities inside the network. An ATM connection is established by setting up translation tables at the switching and multiplexing points which map an incoming label into an outgoing link and label [14]. The connection so established is referred to as a virtual channel since no bandwidth is consumed by the connection unless traffic is actually being sent.

The basic ATM routing entity for switched services is the virtual channel (VC). Virtual channels are aggregated in virtual paths (VPs) which are routed as such through VP multiplexers/demultiplexers and VP switches (add-drop & cross-connect). At a given interface, the different VPs which are multiplexed inside a given transmission payload are identified by the VPI. The different VCs that are multiplexed inside a VP are identified by the VCI as shown in figure 2.7. A transmission path may comprise several VPs and each VP may carry several VCs.

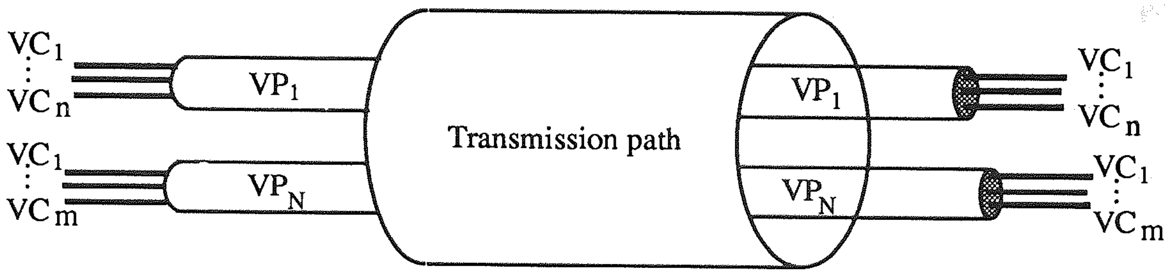



Figure 2.7 ATM connection identifiers and their relationship

### 2.2.3.1 Virtual path concepts

The hierarchy of connection using transmission path (link), VP and VC is a new concept described in [5, 24, 25]. A VP is a logical direct link between two nodes that carries a bundle of virtual channels A predefined route is provided with each virtual path in the physical network. Each virtual path has its own bandwidth which limits the number of virtual channels present in it. They are multiplexed in a physical link deterministically or statistically on a cell multiplexing basis. The node process for the cell follows the following procedure as described in [24]:-

- At the transit node through which the virtual path passes, the VPI in the incoming cell label is compared with the routing table and its outgoing link is found.

- At an end node terminating virtual path, the outgoing link or the destination terminal is found from the VCI.

As mentioned, the routing table of the transit node is referred to using only the VPI. It is not necessary to rewrite the routing table at call set-up [11]. The call set-up processing associated with the virtual channel is excluded from the transit node and executed only at the end node of the virtual path as illustrated in figure 2.8(b). Figure 2.8(a) shows the node process for each cell.

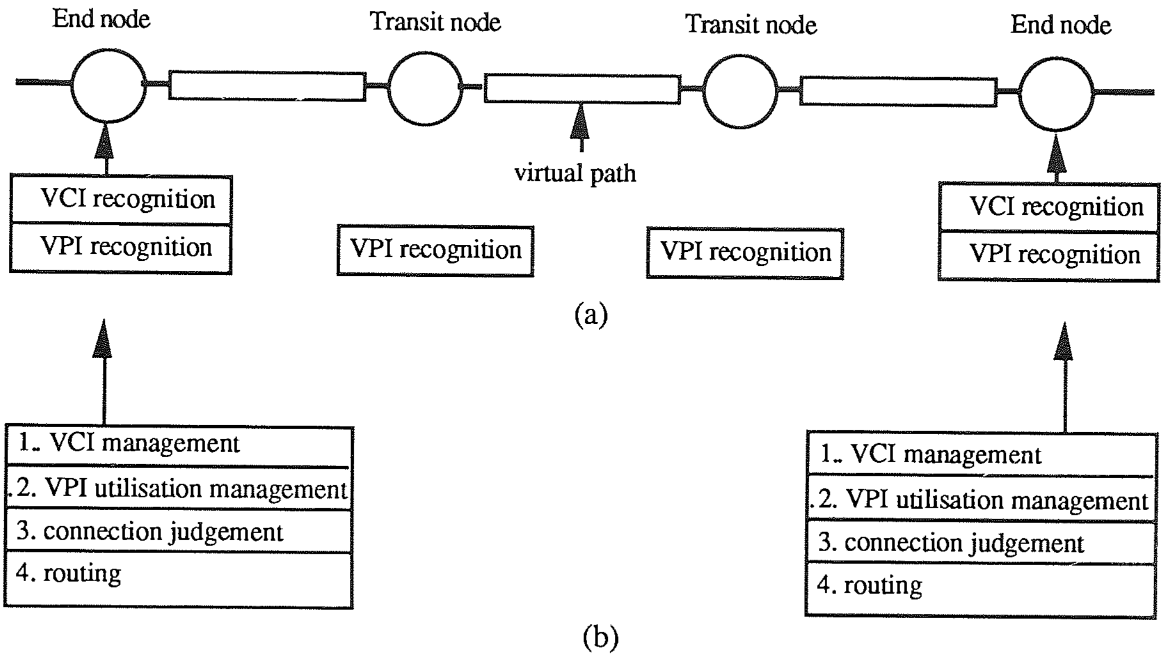



Figure 2.8 (a) Processes for each cell
(b) Processes for each call set up

### 2.2.4 ATM transmission structure at UNI

Two different UNI bit rates have been defined, 155.52 Mb/s and the other 622.08 Mb/s [3]. The lower rate will be used predominantly for interactive services like telephony, video-telephony and data services. The latter interface rate will be more suitable for higher traffic load such as simultaneous transmission of several TV programmes. These bit rates are identical to the two lowest bit rates of the SDH as defined in CCITT Recommendation G.707.

In principle, ATM cells can be transported on any transmission system as long as the bit sequence independence is guaranteed. Two options have been defined for the UNI, one based on SDH and the other on pure cell based multiplexing [140]. Figure 2.9 depicts the two possible interface structures.

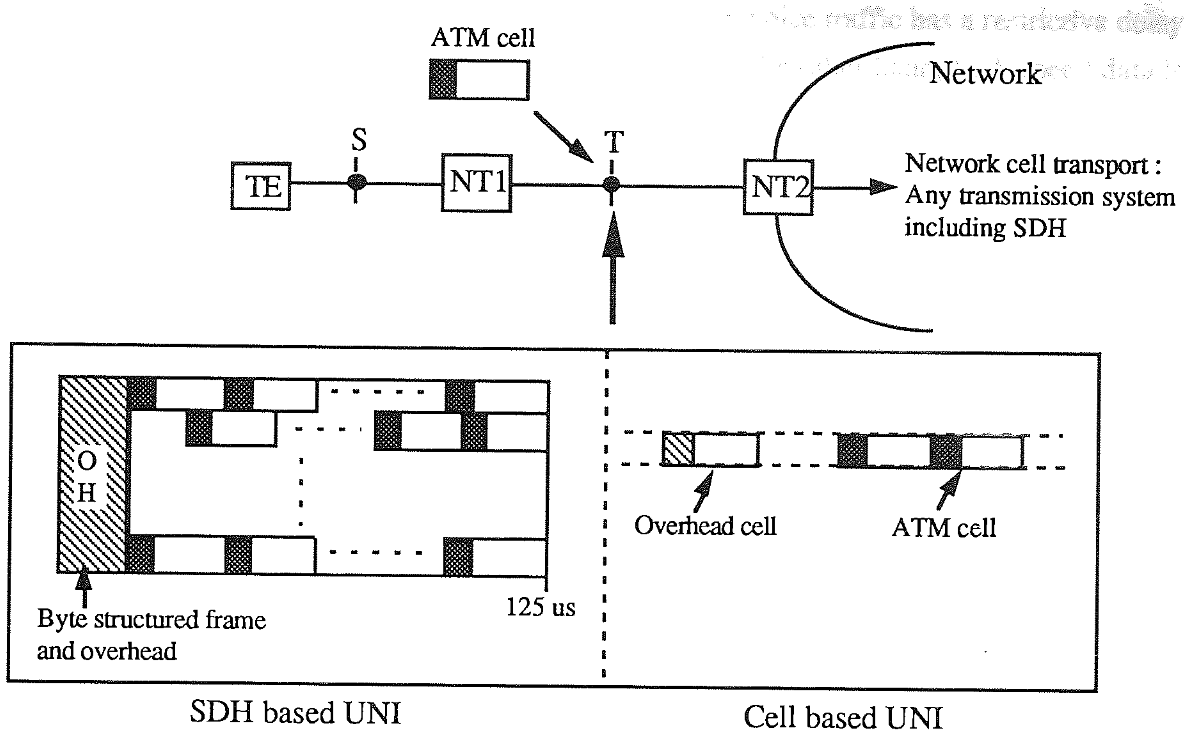



Figure 2.9 User network interface (UNI) transmission structure

SDH is a new transmission hierarchy based on the SONET concept [5,8]. Being a universal transmission concept, SDH allows full compatibility of the user network interface with the network node interface without the necessary conversion of signals that are sent from the user through the network. However, a large overhead capacity is required by SDH and the generation of byte structured SDH frame requires additional interface functionality. On the other hand, these would not be necessary if the interface was completely cell structured, hence the introduction of mere cell based multiplexing. However, the cell based interface has not yet been fully defined.

Once inside the network, the virtual path handler (VPH) or ATM cross-connect which is a VP switch, flexibly maps the incoming VPs on to outgoing VPs, thus enabling establishment of VPCs through the ATM network.

## 2.3 Quality Of Service (QOS) requirements

The wide range of multimedia services in a broadband network will have many different requirements of QOS [29, 30]. Figure 2.10 provides a rough illustration of the ranges of end-to-end traffic performance requirements in terms of maximum cell delay variation and cell loss probability due to buffer overflow. For example, voice traffic has a restrictive delay requirement but can tolerate moderate cell loss ratio. On the other hand, high speed data is very sensitive to cell loss but very delay tolerant. For services using coding techniques in which cell loss induces loss of synchronization and consequent loss of session, the absolute time between cell losses is more important than the loss ratio itself. To ensure network transparency with respect to the offered services, the QOS such as the probability a cell is lost should be extremely small typically in the order of $10^{-6}$ to $10^{-9}$ depending on the service requirements, and the end-to-end delay must satisfy real time service requirements.

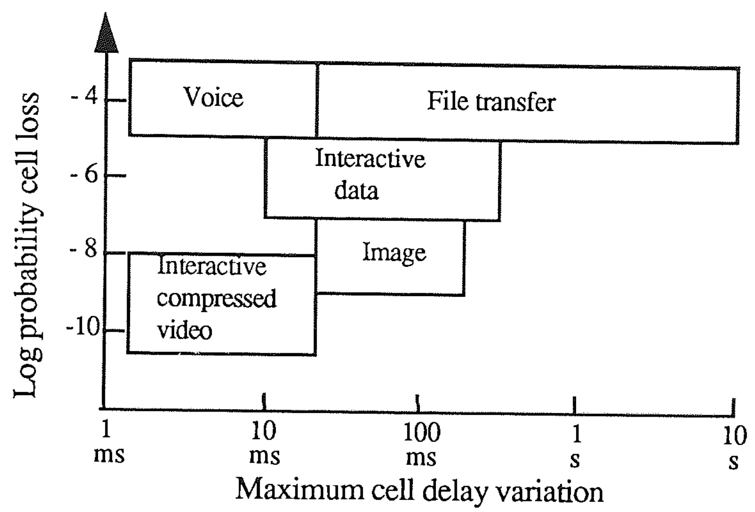



Figure 2.10 End-to-End traffic performance requirements

The end-to-end delay incurred in any packet switched, including ATM based, network is mainly due to packetisation delay, transmission delay, propagation delay, queueing delay and processing delay. Since the transmission link of a broadband network is made up of a high capacity link and fast switching techniques, the delay of transferring data across the network is small compared to the propagation delay and processing delay and can be assumed to be negligibly small. The queueing delay can be minimised by limiting the size of the buffer in the network. However, this must be compromised with the cell discarding rate.

Thus, the overall delay substantially depends on propagation delay and the packetisation delay which can be reduced to the lowest possible value since it is controllable. With statistical multiplexing of cells at a link rate of 150 Mb/s, it is possible to satisfy all delay requirements up to a certain maximum utilisation limit. This reduces the QOS determining the network performance mainly to that of cell loss rate.

### 2.3.1 Multiple traffic classes

Placing tight bounds on delay, delay jitter, cell loss and sizing of buffers at each node are important design issues. In practice, full integration of services implies that the most restrictive QOS requirements must be applied to all services . However, this approach does not allow flexible network utilisation. An alternative design is to segregate different traffic types with different QOS levels. Service segregation could be performed by using separate switches or by flexibly partitioning the ATM switch fabrics and trunk groups into separate parts where each part will comprise a number of physical links [63].

Another approach is to employ a layered system architecture as in the virtual path concept [24-26]. This approach allows less stringent service segregation, since different classes of service are multiplexed on the same ATM links at the expense of additional switching and multiplexing facilities at a higher layer. Both of the these options allow flexible support for future QOS demand.

Another means of providing the different classes of service with different levels of QOS is by having priority schemes. The priority mechanism will guarantee a certain QOS level to the service class of higher priority but will need a more complicated means to ensure the QOS level of the lower priority classes. Two different priority classes can be distinguished by an ATM cell through an explicit cell loss priority bit in the header. The details of this priority control and the need for additional priority mechanisms are an open issue and require further study.

Whatever method is used to take into account the variety of QOS requirements, it can only be considered the most cost effective if the increased complexity in implementation and control is justified by the gain in network utilization.

## 2.4 Traffic Control

Traffic control is required to ensure that the desired network performance is met. Traffic control procedures for ATM network are currently not standardised within CCITT [5]. It has been envisaged that it may not be fully standardised in order to compromise between the network providers' wish and the users' needs. For example, the network provider may want to have flexible tools so as to be able to react adequately to customers' needs, which speaks against standardisation. On the other hand, settled network standards are indispensable for the benefit of the users, especially the end terminal manufacturers.

The topic of traffic control in B-ISDN/ATM has been under research fairly extensively for the past several years [27-37]. The goal in designing a traffic and congestion control should be motivated towards achieving the following control attributes [35, 36];

Simplicity - simple control algorithms are more likely to be economically implementable. The target is to achieve high resource efficiency while maintaining a simple control structure. Moreover, simplicity will be essential when a large number of ATM-based entities are required to interwork.

Flexibility - it is of paramount important that the traffic control should have the flexibility to adapt as needed to new situations that arise. Some of the services and the associated traffic types can already be anticipated but many more will surface following the extensive deployment of B-ISDN.

Robustness - the traffic control should be relatively insensitive to the imperfect assumptions regarding the services, applications, traffic characteristics, performance objectives and the direction B-ISDN/ATM standards may take, which are subject to changes. Also it should not be critically dependent on specific behaviour of system components.

Controllability - traffic should be adequately controlled while ensuring efficient network resource utilisation without paying a penalty in performance. The aim is controllability rather than optimality since simplicity, flexibility and robustness would likely have to be sacrificed to achieve optimality.

### 2.4.1 Issues in BISDN/ATM traffic control

ATM shares much in common with conventional packet switching at lower rates in which the subject of traffic and congestion control has been studied extensively [28, 129, 133]. Several workable techniques addressed at different OSI levels include preallocation of buffers to each virtual circuit that has been set up, packet discarding and retransmission, limiting the number of packets in the network through isarithmic control with permits, using the various flow control mechanism to eliminate congestion, sending choke packets back to the source when congestion occurs and deadlock prevention mechanisms.

However, most of these conventional approaches do not scale well to the extremely high speeds associated with B-ISDN/ATM. It is apparent that the control implementability of such processing schemes must be compatible with the broadband speed and within the constraints imposed by the B-ISDN/ATM. This implies limited allowable control algorithm complexity and limited options for coordination within a system of individual controls.

Also, due to the wide area distances, the increase in bandwidth delay product is unavoidable. Thus, the amount of traffic that can be in transit during a propagation delay time would be extremely large. For instance, a 600 Mb/s link with a propagation delay of 20 ms will have 24 Mb already in the transmission pipe during a round trip delay after which the source receives an acknowledgement to react. A significant amount of buffer is therefore required to prevent loss leading to an intolerable delay. This suggests the potential of ineffectiveness and instability of employing a purely reactive type of control.

Another issue is how to effectively handle the multiple service performance requirement as illustrated in figure 2.10. It would be highly complex and undesirable to attempt to provide such diverse performance levels. The objective should be to manage the traffic in as simple an integrated a fashion as possible. Traffic control is also affected by the diverse traffic characteristics. The problem of characterising bursty traffic and controlling it to take advantage of statistical multiplexing must be also resolved. The subject of traffic characterisation is discussed below.

### 2.2.2 Traffic characterisation

Adequate traffic characterization at specific reference points of a broadband network is required to properly design and operate the network. The different traffic characteristics of each of the multi-media services, the wide range of bit rate, the statistical profile of the information flow and the variety of connection configurations such as point-to-point, multipoint and asymmetrical connections make the traffic characterization more complex than in narrowband ISDN.

In previous work, the generation of data packets from a single data source has been accurately characterised by a Poisson arrival process for the continuous time case and geometric interarrival process for the discrete time case. The mathematical models have been well developed for both cases [131]. However, either of these assumptions may not adequately describe the generation of cells from bursty sources such as voice and video. Their arrival processes are fairly complex due to a strong correlation between arrivals. An alternative means of modelling traffic sources in an ATM network therefore needs to be studied.

The concept of call pattern and connection pattern and its effects on the lower layers, introduced by CCITT as a sequence of layer events generated by a call attempt, can be extended to an ATM environment. Traffic destined for transport on a cell basis in an ATM network can be viewed as three levels of behaviour [62-64, 70, 82] corresponding to a certain resolution in time regardless of the service, as shown in figure 2.11.

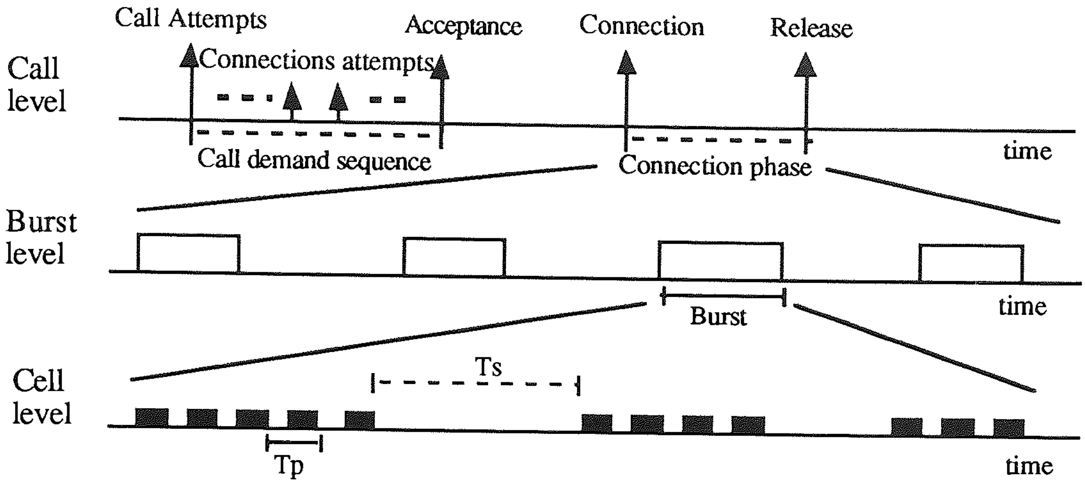



Figure 2.11 Traffic characterisation model

The connection or call level describes the behaviour of a traffic source on the virtual connection basis. The call set up and call clear events, which delimit a call duration, are the most macroscopic behaviour of a stationary traffic source at the connection level. The durations of calls are in the order of magnitude of minutes to hours. The burst level describes the behaviour of an active sender, that is its statistical behaviour including the autocorrelation due to the internal variation of the source.The on/off characteristics of the cell generation process are modelled in this level. Durations of the active and idle period are in the order of magnitude of milliseconds. Finally, the cell level describes the behaviour of cell generation during the active period. The duration of the intercell interval is in the range of several microseconds to milliseconds.

These levels have different impacts on network implementation. For example, cell level analysis can be used for buffer dimensioning [54], the burst level characteristics influence the connection admission strategies [69], and the call level consideration relate to ATM link dimensioning [75]. Based on this model, mathematical traffic characterisations have been defined by many authors with different levels of abstraction. These include the switched Poisson process (SPP) or interrupted Poisson process (IPP) [139], Markov-modulated Poisson process (MMPP) [44], generally-modulated deterministic process (GMDP) [138] geometrically distributed burst length [70, 92]. These detailed source traffic models are very useful in performance evaluation as the QOS parameters (e.g. loss probability, jitter) are sensitive to the assumed source characteristics.

The choice of traffic control algorithm directly impacts on the network resource allocation strategy. The subject of resource allocation will be addressed in a later section after we have more insight on the subject of congestion control in the broadband environment.

## 2.5 Congestion control

Congestion control is a means of protecting a network from traffic overload or control resource overload that would prevent the network being able to guarantee the negotiated quality of service to the already established connections on a short term basis. The possible causes of congestion are the unpredictable statistical fluctuation of traffic flows or sudden faulty conditions in the network.

In the high speed multimedia environment, transport performance level is the critical factor in retaining the integrity of connections. In order to maximise information transfer speed, traffic is best controlled at the access points to the ATM transport network [30]. There are two types of access control used in minimising congestion as summarised in [29], reactive and preventive congestion control.

Reactive congestion control is invoked at the onset of congestion, based on the current traffic level within the network, by regulating the traffic flow at the access points. A feedback mechanism with enough lead time to react is therefore required. As mentioned above and elaborated in more detail in [27], the reactive congestion control approach is not appropriate for broadband multimedia networks. Reactive controls are more suitable for private, localised networks carrying homogeneous traffic, where transport performance is not critical to the integrity of the connection and where all end terminals can be throttled back in a similar manner.

In contrast, preventive congestion control tries to prevent the network from reaching an unacceptable level of congestion. This approach is especially effective for ATM networks because of its connection oriented transport in which a decision to admit a new connection can be made based on the knowledge of the state of the route of the new connection.[34]. A connection would only be accepted if the network can guarantee a certain QOS level during the connection. This leads to two levels of preventive congestion control as agreed by CCITT; connection admission control and usage parameter control. A brief description of these control actions is given in the following section.

### 2.5.1 Connection admission control (CAC)

The connection admission control decides at the call set-up phase (or during the call renegotiation phase) whether a new connection request can be established on each link on the route through the network without degrading the QOS of the existing connections. The decision whether to accept or block the request connection on a specific link is based on the available network resources and the required QOS calculated from a set of parameters for each existing connection on a link and a corresponding set of parameters for the new connection.

Thus, two classes of parameters (connection descriptors) are forseen to support an admission control algorithm;

- a set of parameters describing the source traffic characteristics. The source traffic parameter can be characterised by its
    * peak bit rate
    * average bit rate
    * burstiness
    * peak duration.

- a set of parameters to identify the required QOS

Burstiness is an important parameter in an ATM network where traffic sources are highly bursty. It describes how densely or sparsely cell arrivals occur and it plays a critical role in determining network performance. Burstiness is generally characterised by the ratio of the source peak and average bit rate [69, 74-76, 78]. However, this may not give an accurate measure. Other possible definitions of burstiness proposed by different authors include;

- the average burst length [84]
- cell jitter ratio = Var[cell interarrival times]/E[cell interarrival times] [141]
- the squared coefficient of variation of the interarrival times = Var[cell interarrival times]/$E^2$[cell interarrival times] [43]
- peakedness defined as the variance to mean ratio of the number of busy servers in a fictitious infinite server group [42]

Due to the difficult task of defining the exact measure of burstiness, the definition of burstiness is still a pending issue in CCITT. The subject of CAC is covered in more detail in chapter 3.

### 2.5.2 Usage parameter control (UPC)

UPC is the term that has been chosen by CCITT to represent an enforcement function on traffic flow. It monitors and controls user traffic in terms of volume and cell routing validity. It protects the network resources from malicious as well as unintentional misbehaviour which can affect the QOS of other already established connections by detecting violation of negotiated parameters. The parameters subject to monitoring and control may be the same as the parameters that characterised the source traffic described in the previous section. However, this is subject to the connection admission control algorithm and further studies are required. The subject related to traffic monitoring will be further discussed in chapter 4.

## 2.6 Resource allocation

In an ATM network the issue of resource handling must be considered very carefully in order to enable the network to meet the most stringent requirements and to show an acceptable level of robustness. Resource management functionalities can be allocated in three different control levels [17]; network control level, call control level and cell control level. Accordingly, the resource allocation functions may be classified into three categories on different time scales;

1) control functions during the ***data transfer phase*** comprising functions performed to control the load in the network by managing the flow of ATM cells ( eg. policing, buffer management)

2) control functions during the ***connection set-up phase*** including functions performed when a request for a connection between two or more users is granted or blocked due to lack of resources (eg. call acceptance, overload control)

3) control functions to perform network management, including functions performed to allocate and reallocate network resources on a long term basis (eg. VCI and VPI routing, connection management and recovery).

These functionalities ensure that the cell losses and delay remain at a satisfactory level for all users. An outline of the first two categories of these functions is given in the following paragraph.

Integrated call services may consist of, for example, mixed voice and data connections or integration of voice and video connections. In order to control these multimedia call services it is wise to separate call control functions from connection control functions in the implementation of the control entities as illustrated in figure 2.12. The call control entity performs functions related to call establishment which coordinates the individual connections associated with the call. For example, user to user signalling, user to network signalling, charging aspects related to the call etc. Whereas the connection control entity performs functions which are directly related to a connection such as connection establishment during the connection set-up phase, flow conditioning performs functions during data transfer phase and instigates metering to facilitate charging and QOS supervision.

Figure 2.13 illustrates a resource allocation framework proposed by Kallberg and Stavenov [63]. Based on this framework, issues related to resource allocation in the connection set-up phase will be further elaborated in the next section.

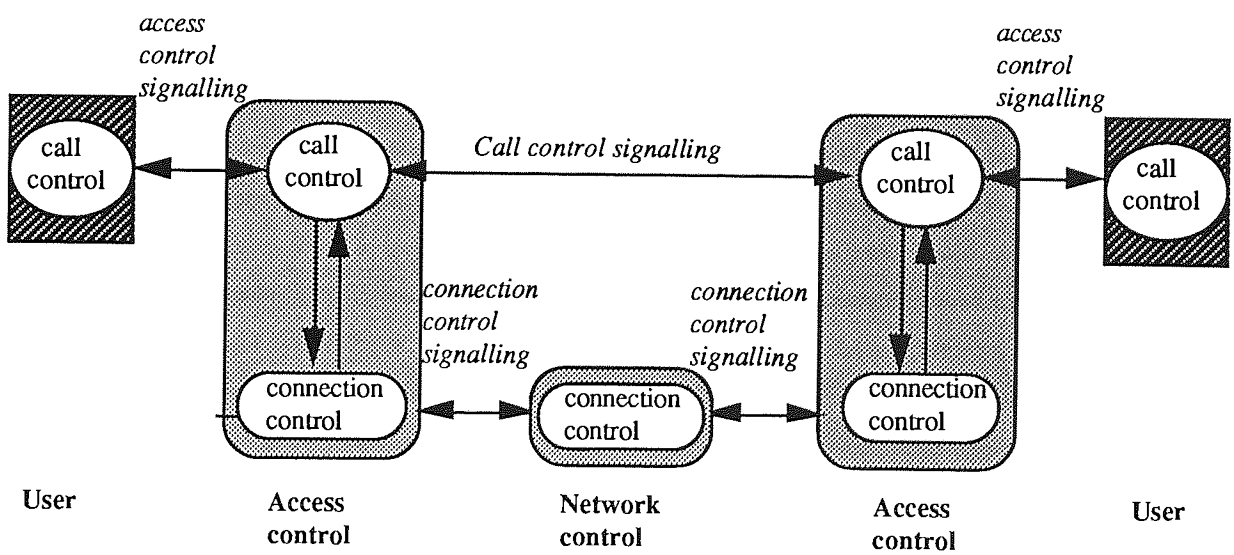



Figure 2.12 Network control and signalling

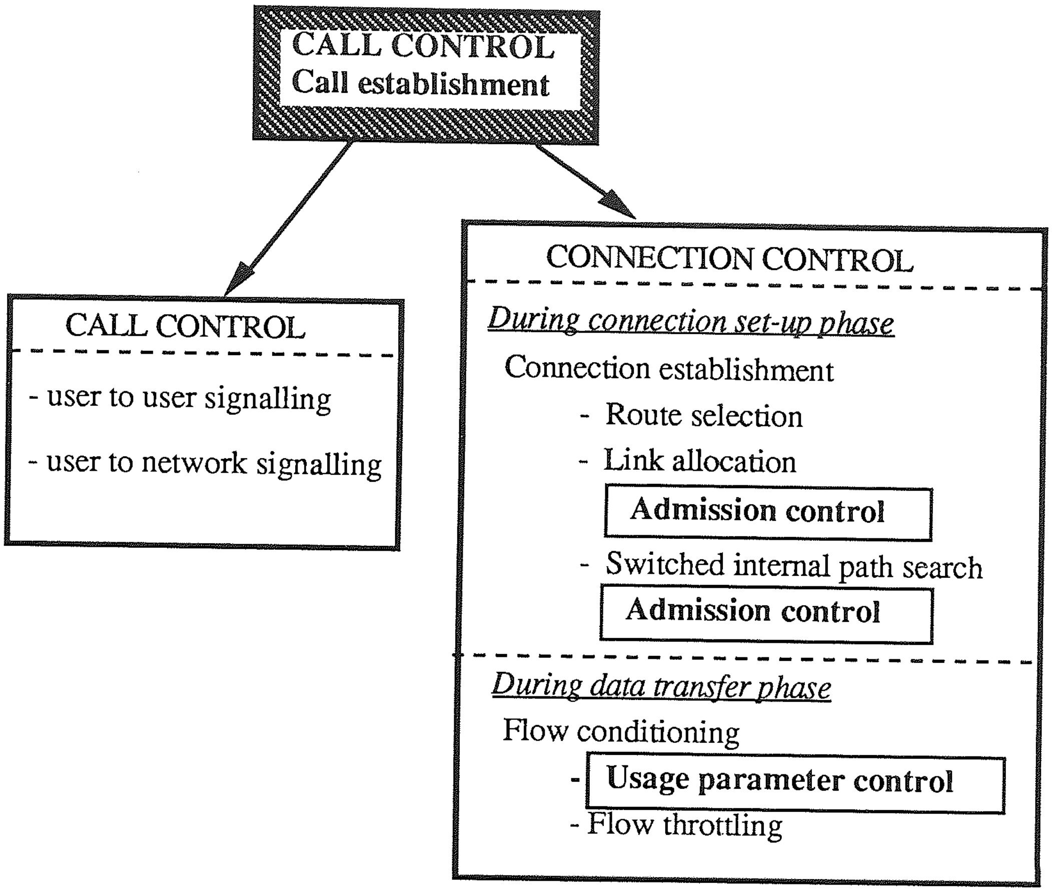



Figure 2.13 A resource allocation framework

### 2.6.1 Connection set-up phase

Connection establishment is defined as the process of establishing a connection between any pair of origin and destination nodes. In an ATM network a connection is always unidirectional and no rearranging capability is provided. According to [63] the connection establishment algorithm can be partitioned into four parts:-

1) Route selection

- selects an overall route between a pair of nodes in the network. The route is specified in terms of a number of trunk groups connecting the source and the destination switching nodes.

2) Link allocation

- when the route has been selected, a link within the trunk group is allocated to the connection. There are two options regarding the resources of a trunk, either a connection can be carried on different links or on a single link. The latter option is more straightforward and does not need resequencing but it requires bandwidth allocation and trunk group dimensioning.

3) <u>Switch internal path search</u>

- a tractable path for interconnection between an input port and an output port of a switching module must be provided to connections carried in a link. The design of the switch fabrics, the different switch topologies and the service requirements such point-to-multipoint affects the algorithm.

4) <u>Admission control</u>

- determines if the new connection with certain characteristics could be admitted on a specified link and decides on the allocation of the amount of bandwidth based on the QOS carried on the link (virtual path). This function resides in the link allocation and the switch internal path search algorithms.

### 2.6.2 Data transfer phase

Once the connection set-up is successful, data transfer will take place through the successive links/paths. A flow conditioning mechanism is required to make sure that the data flow over the selected virtual paths is in accordance with the characteristics of the requested connection declared during connection set-up [85]. This function controls the load by managing the flow of cells into the network. The flow conditioning concept as illustrated in figure 2.14 may include two types of functions:-

1) *Usage parameter control* - concerned with supervising an individual connection at the receiving entity at the edge of the network;

2) *flow throttling function* - concerned with smoothing of data flow in a connection at the corresponding peer sending entity.

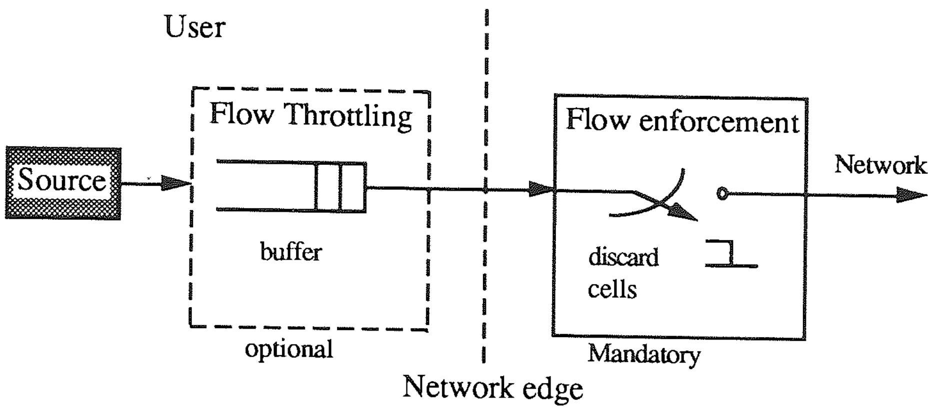



Figure 2.14 Basic flow conditioning functions

The UPC function identifies and takes action against any connection not used in accordance with the agreements. This function must be inserted at each location where the ATM layer is offered a data flow from another party (eg. at the traffic source). The flow throttling function assures that the characteristics of data flow are in accordance with the agreement. This function could act as a shaper or a smoothing mechanism. Through this function the user adapts to the traffic characteristic within the network and achieves better utilization of the allocated bandwidth.

## 2.7 Summary

ATM is a promising transfer technique for B-ISDN due to its flexibility, efficiency and simplicity. In this chapter, several topics related to the ATM concept and network aspects have been reviewed. More specifically, the subject related to traffic control for B-ISDN-ATM have been dealt with in some detail. Proper traffic control and management is a very challenging task for the viability and success of the future B-ISDN.

The various issues in traffic control at broadband speed require more accurate and appropriate traffic characterisation and control strategies. The control architecture should be designed towards achieving the goals listed in section 2.4. Two types of congestion control have been identified. The preventive congestion control methods are more appropriate for connection oriented networks such as an ATM network.

The resource allocation conceptual framework gives an insight into the hierarchy of the control structure and the design of traffic controls at different levels. Based on this framework, the two main resources bandwidth and buffer memory can be assigned properly at the respective control levels. This research, concentrates on controls at call and cell level for a private ATM-based network. The call control level is dealt with in detail in chapter 3, while chapters 4 and 5 are related to controls at cell level.

# CHAPTER 3

## CONNECTION ADMISSION CONTROL

### 3.1 Introduction

The objective of preventive congestion control described in chapter 2 is to guarantee a certain QOS-level during the data transfer phase of each connection. At call level, this is done by only accepting new connections when an acceptable QOS-level can still be achieved for all connections concerned, including the new ones, with the probability arbitrarily close to 1. This approach makes the peventive congestion control concept similar to the blocking in the traditional circuit switched concept but with the additional capability to handle connections of a more general type [34, 63].

The connection admission control (CAC) addresses those control procedures which lead to a decision whether or not a new request can be accepted based on the traffic characteristics of the connections already established. The main difficulty faced by CAC is making the right decision based on the parameters (traffic descriptors) declared by the established connections. The determination of an appropriate minimum set of traffic descriptors addressed in [69, 74, 76, 83] still requires further study. Often at the time of connection establishment negotiation, the terminal hardly knows the exact statistics of the traffic it will generate. Even if the terminals do know in advance their traffic statistics, the performance of the network elements (multiplexers, switches etc) depends on these statistics in a rather complicated manner. The uncertainty of the traffic statistics and, in turn, the traffic descriptors (parameters) raise severe obstacles for on-line computations within the limited time of the CAC operations.

In this chapter, the approach towards a simple, flexible and robust CAC is addressed. The initial simulation study on the performance of superposition of input traffic in a multiplexer in section 3.3 gives an insight into the behaviour of a multiplexer. Performance evaluation on this behaviour form the basis on which CAC is developed. The evaluation through mathematical analysis is reviewed in section 3.4 based on the fluid flow mechanism.

## 3.2 Review of network performance evaluation methods

The multiplexing of a large number of traffic sources of different characteristics on a high capacity link of 150 Mb/s according to FIFO is equivalent to a single server with a deterministic service time queueing system of a particular nature. Queues of cells may arise, for example, at the output of a switching node as shown in figure 3.1, mainly due to over allocating the available capacity to gain statistical multiplexing or coincidence of cell arrivals from different sources. It is assumed all the cells arriving at an input port are destined to an output port and that the switch is fast enough so that queueing of cells occurs only at the output port. This assumption enables an output port buffer to be analysed independent of the others [100].

To ensure network transparency with respect to the offered services, the QOS, such as the probability a cell is lost due to such queues exceeding buffer capacity, should be extremely small, typically in the order of $10^{-6}$ to $10^{-9}$, depending on the service requirements and the end-to-end delay must satisfy the real time service requirement [15, 16].

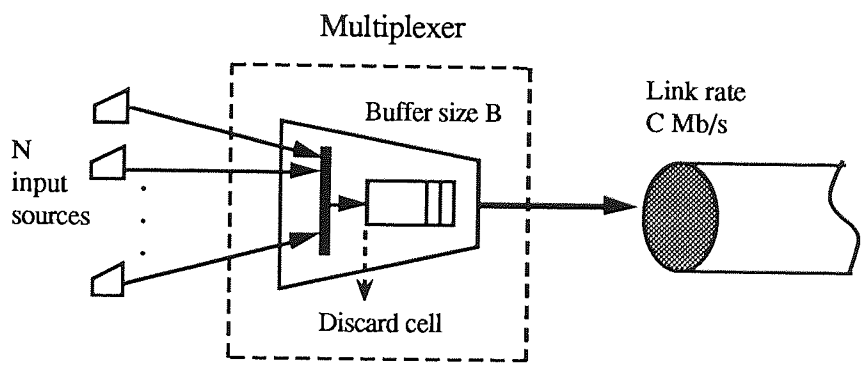



Figure 3.1 A node model with high capacity link

The purpose of performance evaluation here is to enable the development of simple resource reservation rules which will work effectively over any mix of connection traffic types within a given operating region, rather than the optimisation of the resource utilisation for a given traffic mix. In general, there are three techniques to evaluate the multiplexer node performance in figure 3.1: analytical, computer simulation and empirical measurement with hardware testbed.

Analytical techniques are generally used for performance evaluation since they provide reasonable computation requirements and accuracy. In order that the system can be sized to satisfy the required quality of service objectives, a sound understanding of the behaviour of the queue of the node model in figure 3.1 is needed. The behaviour of the queue can be studied analytically through queueing models having an input arrival process determined from the traffic descriptors [125] or in terms of the average and the variance of the number of arriving cells during a fixed interval [126].

Various analytical approaches representing the superposition of homogeneous on/off sources proposed in the literature may be classified according to whether they take account of correlation effects. Approaches which take into account burst level correlation are typified by a number of analyses of packetized speech in [44, 46, 47, 50] and analysis of superposition of variable bit rate video sources in [48]. Correlation effects are ignored in the simple M/D/1 model which has been suggested for output buffer dimensioning by Hui [62] and Yeh [127]. The Poisson assumption in the M/D/1 model does not account for the positive correlations between the number of cell arrivals in successive burst intervals. The negative correlation between successive cell interarrival times due to the periodic emission of cells within each burst is also ignored in the Poisson assumption [54, 56]. Eckberg [128], demonstrated that the M/D/1 model can significantly overestimate the buffer requirements. The difference between the Poisson and the correlated input arrival models is mainly apparent at high load [47, 50]. However, the M/D/1 model has the advantage of simplicity and gives a good approximation for the superposition of bursty sources when the multiplex load is low.

Every analytical model has certain limitations with respect to the approximation stated, such as the inability to represent general input traffic flows and heterogeneous inputs. Various analytical approaches that consider correlation effects such as in [52, 57] are almost accurate and thus give an exact solution as they explicitly take into account the individual contribution of each source. In practice, however, these techniques experience huge numerical and hence computational complexity. The matching method approach in [44] is well known as the Markov Modulated Poisson Process (MMPP), and is based on the approximation of the superposed stream with a suitably chosen simple arrival process for which an exact queueing analysis is possible. Although its complexity is slightly less, it does not provide a satisfactory result accuracy for a large variety of real situations [50]. Another approach by [45, 46] resorts to a fluid flow approximation technique which is much simpler and is excellent for predicting the performance of the multiplexer from an engineering point of view. However, it still requires some computational effort.

Simulation methods remain the most flexible for determining node operation and performance under a variety of conditions, and are also a necessity for validating analytical approximations. However, computation time to obtain statistically significant estimates of the probability distribution tails can be prohibitive. The computational time can be reduced by several orders of magnitude through heuristic approaches based on the assumption of an either/or combination of geometric probability distribution tails, least-square fit and extrapolation from simulation results [49].

Finally, a hardware testbed provides the most detailed system representation and can run in real time. The testbed serves as a means to validate the results estimated by analytical or simulation methods. However, this method requires the development of a real system which may not necessarily be available.

The simulation extrapolation approach and simple analysis methods have been chosen as the means to explore the sensitivity of access node control performance for certain input traffic parameters. Through simulation, it is possible to make general observations about the traffic operating region neccessary to meet the proposed traffic control/management design principle. Within the operating region, further analytical and simulation methods could be used to determine the node performance as accurately as possible.

## 3.3 Simulation results on the superposition of homogeneous sources

The simulation results of the superposition of homogeneous sources are based on assumptions of the node model in figure 3.1. Homogeneous sources have been used since this allows better analysis and simplifies the search for the appropriate parameters. It was shown that when the offered traffic is fixed, the saturation probability varies with the buffer size as depicted in figure 3.2.

Here there are two distinct effects occurring within different time scales corresponding to the congestion phenomena, referred to as cell scale congestion and burst scale congestion as observed in [51-53, 56] . For short buffer size, cell scale congestion occurs even when the overall cell arrival rate due to active sources is less than the multiplex capacity due to the simultaneous arrivals of cells from independent sources. These short term variations determine the queueing behaviour which is usually characterised by cell loss and cell delay. The time scale of the short term fluctuations is in the order of the minimum cell interarrival time Tp. Within this period, a negative correlation of the cell arrival process is observed. This means that if many cells arrive within a given time slot, fewer cells will arrive during the following time slot because of the minimum spacing between consecutive cells of a given connection.

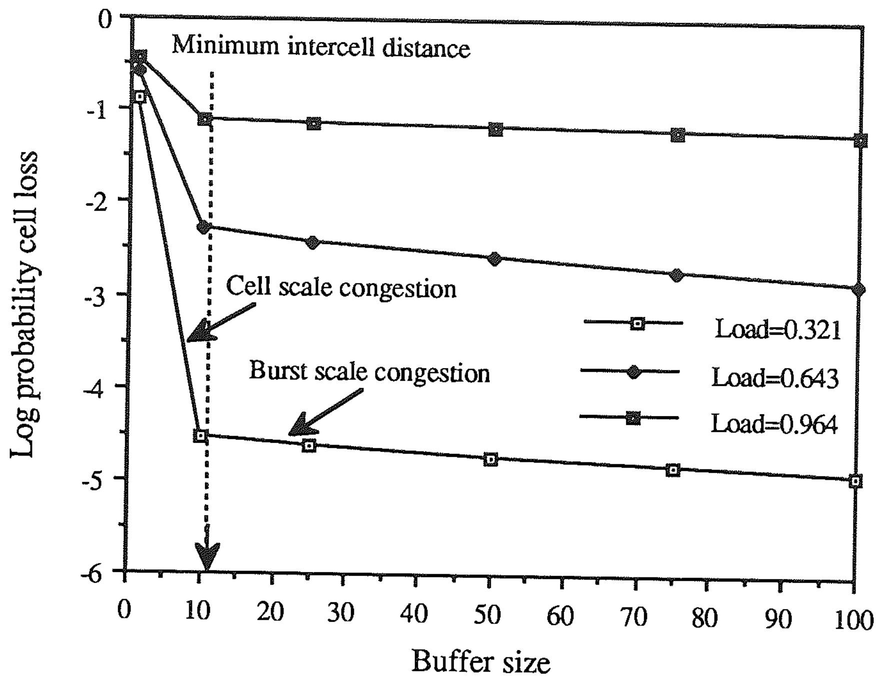



Figure 3.2 Probability of cell loss against buffer size for homogeneous sources of various load (Bp=10 MB/s, Bm=2 Mb/s and L=100 cells)

On the other hand, for increasing buffer size, burst scale congestion occurs when the overall cell arrival rate is momentarily greater than the multiplex capacity. The long term fluctuation within the cell arrival process which is generated by alternation of burst and silence periods of the independent sources become more significant. The durations of the burst and silence phases are much longer than the minimum inter-cell distance Tp. Hence, within this time scale, the cell arrival process has a positive correlation. This means that if many cells arrive during a cycle of successive time slots, then many cells will arrive during the next cycle due to many sources being in an active state.

When the offered load is varied, the effect on the probability of saturation is shown in figure 3.2. The slope of the cell component changes, but the most significant variation with increasing load is the vertical translation of the burst component. When the load is low the queueing behaviour is dominated by the short term fluctuations.

Figure 3.3 shows the effect of the mean burst length on cell loss probability. The slope of the burst component is proportional to the mean burst length and tends to zero if the burst length approaches infinity (where the coefficient of correlation for the cell arrivals in consecutive cycles approaches 1). Meanwhile the cell component remains unchanged. A very short burst, ie a burst containing only a single cell, may lead to a geometric cell interarrival time distribution for each of the superimposed traffic streams [52, 53]. This is because the burst effect diminishes and the behaviour of the cell arrivals is characterised by the cell component.

Figure 3.4 shows the saturation probability for varying burstiness of the traffic streams where the number of sources, the minimum cell interarrival time and the burst length are fixed. It was observed that the saturation probability becomes higher as the burstiness becomes smaller due to an increase in the traffic load.

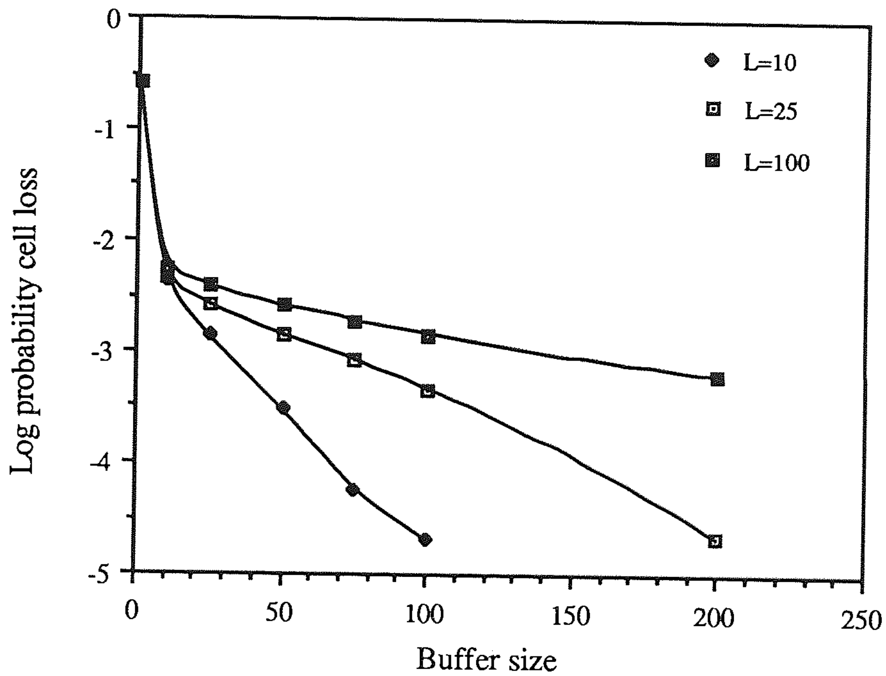



Figure 3.3 Variation of cell loss against buffer size for various mean burst length (Bp=10 Mb/s, Bm=2 Mb/s, Load=0.642)

Obviously more connections with fixed peak bit rate can be multiplexed at a saturation probability level as the burstiness increases. If a constant offered traffic load of a particular burstiness and mean number of cells within a burst are assumed, the variation of cell loss probability for different peak bit rate is as illustrated in figure 3.5. The results indicate that the achievable multiplexing gain increases as the peak bit-rate decreases. Since the curves in the burst component are almost parallel, the long term fluctuation dominates the queueing behaviour. However, the burst level effect depends on the minimum spacing of cells $T_p$ (maximum bit rate) originating from the connection.

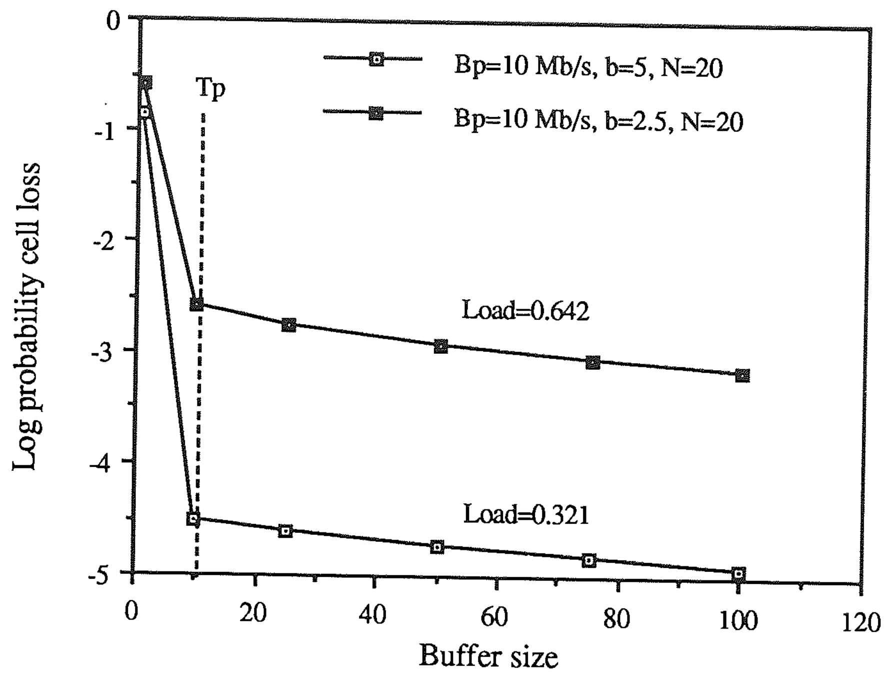



Figure 3.4 Variation of cell loss against buffer size for sources of different burstiness but of similar peak bit rate

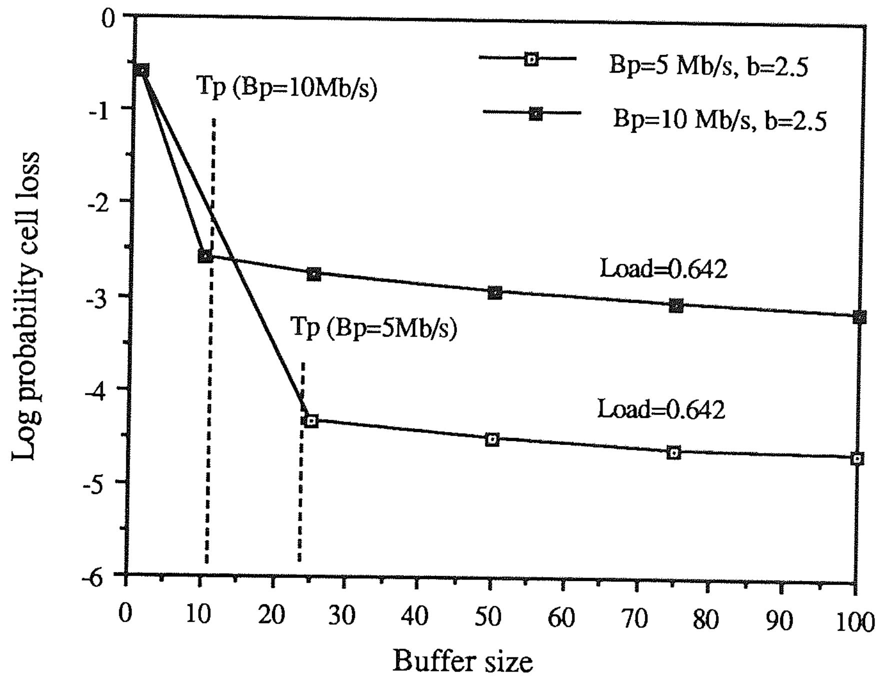



Figure 3.5 Variation of cell loss probability against buffer size for sources of different peak bit rate but if of equal burstiness

The simulation results on the superposition of on/off homogeneous sources indicate that the burstiness b is the most important parameter that determines the statistical multiplexing behaviour. The mean burst length is also a sensitive parameter if its length is comparable to the buffer size. However, in a real case the mean burst length L is usually greater than the buffer size for most services and thus the sensivity to parameter L is low and can be disregarded.

### 3.3.1 Buffer requirements

The simulation results above show that the amount of buffer required to prevent congestion due to the variation at cell level is quite small. The queueing behaviour at cell level can be well approximated by simple queueing models such as M/D/1, GEO/D/1 and (N.D)/D/1 [52, 53, 54, 58, 61]. Figure 5.9 (see section 5.3.3) shows the congestion level at buffer limit for increasing load according to the M/D/1 queueing model. A buffer size of about 100 cells would provide a cell loss probability of $10^{-9}$. Such a small buffer has the advantage of introducing only small delay, small delay jitter, and small changes in the characteristics of the cell streams. The disadvantage is a potentially small network efficiency due to potential cell loss.

Larger buffer size must be used in order to absorb congestion at high burst levels as shown in figure 3.3. However, large buffer size will introduce intolerable delay for real time services such as voice and video services. Also the increase in network efficiency which can be obtained by increasing the buffer size for large mean burst length is small as shown in figure 3.3. The buffer size at the output port of the switch must therefore be dimensioned to prevent congestion at cell level and a trade off between the allowable delay due to the buffer size and the burst level congestion must be determined.

## 3.4 Analysis using Fluid Flow model

Qualitative performance considerations on the superposition of homogeneous on/off sources derived from a correlated input model of a multiplexer with particular offered load has been carried out by simulation in section 3.3. In this section, the performance analysis based on a fluid flow mechanism was reviewed. This model has been proved by [45, 46, 53, 56] to be a versatile representation of superposition of bursty sources with good accuracy and simplicity.

The fluid flow model or uniform arrival and service model in [46] approximates the multiplexer by supposing that the input sources generate a constant, state-dependent flow of information or cells, with a rate equal to the mean rate of cell generation while in that state and the server removes the cells from the queue in the same manner at a different rate. The sojourn time spent in this state is usually assumed to be exponential. This assumption removes the concept of packetisation from the multiplexer. This is the main drawback of the model as it does not take into account the increase of the short-term queue when two or more cells arrive almost simultaneously. However, this inaccuracy can be reduced if the queue limit is small and sufficient buffer is provided.

In this model the number of active sources are modeled by a continuous-time birth-death process[131] as in figure 3.6. If a burst length is defined as $1/\mu$, the silence or idle period as $1/\lambda$ and the transition rate from state i to states j as p(i,j), the birth and death rates are given by

$$
\begin{aligned}
& = i\mu && j=i-1 \quad i \neq N \\
p(i,j) & = 1 - \{(N-i)\lambda + i\mu\} && j=i \\
& = (N-i)\lambda && j=i+1 \quad i \neq 0 \\
& = 0 && \text{otherwise}
\end{aligned}
$$

The total flow rate out of state i is denoted by $p_o(i)$ where

$$
\begin{aligned}
p_o(i) & = p(i,i+1) + p(i,i-1) \\
& = 1 - p(i,i)
\end{aligned}
$$

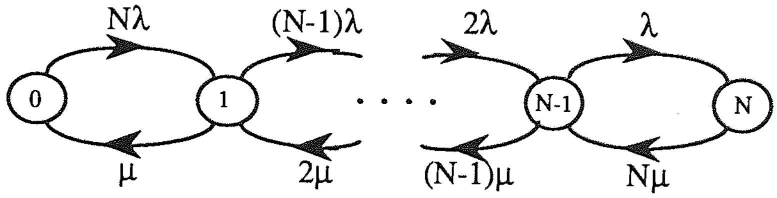



Figure 3.6 State transition-rate diagram

The mean traffic intensity contributed by the $i^{th}$ source is

$$\overline{\rho_i} = \frac{\lambda_i/\mu_i}{\left(1 + \lambda_i/\mu_i\right)} \frac{B_p}{C}$$

and the total load is

$$\rho = \sum_{i=1}^{N} \overline{\rho_i} \tag{3.1}$$

Let $P_i(t,x)$, $0 \le i \ge N$, $x \ge 0$, be the probability that at time t, i input sources are on and the buffer content does not exceed x. Two elementary events can occur during the next small time interval $\Delta t$ from time t; a new source can start or a source can turn off.

The mathematical derivations of the state equations are described in further detail in [45]. It has been shown that the equilibrium probability that there are i input sources active and the buffer content does not exceed x is given by $F_i(x)$ where

$$(i - c)\frac{d F_i}{dx} = p(i-1, i)\, F_{i-1}(x) - p_o(i)\, F_i(x) + p(i+1, i)\, F_{i+1}(x) \qquad x > 0 \tag{3.2}$$

and c is the ratio of the output capacity of the multiplexer to the source transmission rate during the active period. The above equation (3.2) can be be written in matrix form

$$\mathbf{D}\ d\mathbf{F}(x)/dx = \mathbf{M}\ \mathbf{F}(x) \qquad 0 < x < m \tag{3.3}$$

where D = diag{-c,1 - c, 2 - c, . . ., N - c), $c = \alpha C$ and $\alpha<1$, and

$$M = \left| \begin{array}{cccccc} -p_o(0) & p(1,0) & & & & \\ p(0,1) & -p_o(1) & p(2,1) & & & \\ & p(1,2) & -p_o(2) & p(3,2) & & \\ & & \cdot & \cdot & & \\ & & & \cdot & \cdot & \\ & & & p(N\text{-}2, N\text{-}1) & -p_o(N\text{-}1) & p(N,N\text{-}1) \\ & & & & p(N\text{-}1,N) & -p_o(N) \end{array} \right|$$

The solution to (3.3) is in the form

$$F(x) = \sum_{k=0}^{N} \exp(z_k x)\, a_k\, \phi_k \qquad x > 0 \tag{3.4}$$

where $z_k$ is the eigenvalue of $\mathbf{D}^{-1}\,\mathbf{M}$, $\phi_k$ is the corresponding right eigenvector and $a_k$ the coefficient that must be found by defining and solving boundary equations and eigenvectors.

The equilibrium probability of i input sources being active simultaneously $\pi_{oi}$ is given by the binomial distribution

$$F_i(\infty) = \pi_{0i} = \binom{N}{i} \frac{(\lambda/\mu)^i}{(1+\lambda/\mu)^N} \tag{3.5}$$

Obviously $$F(\infty) = \sum_{i=0}^{N} F_i(\infty) = 1 \tag{3.6}$$

Equation (3.4) has been derived not taking into account the probability $u_i$ of the queue being held at its upper limit when $x = m$. Defining $F_i(m-)$ as $\lim_{x \to m}$, therefore

$$u_i = \pi_{oi} - F_i(m-).$$

$F_i(m-)$ can be found by setting $x = m$ in (3.4) giving

$$u_i = \pi_{oi} - F_i(m) \tag{3.7}$$

<u>At the boundary limits:</u>

1) $i > c$, the queue is always increasing, so the queue length cannot be zero. Therefore, from equation. (3.4)

$$F_i(0) = \lim_{x \to 0} F_i(x) = 0$$

$$\sum_{k=0}^{N} a_k \{\phi_k\}_i = 0 \qquad c < i \le N \tag{3.8}$$

2) $i < c$, the queue length is always decreasing, so the queue would never be on its limit. Therefore, $u_i = 0$ and $F_i(m-) = \pi_{oi}$. Hence

$$\sum_{k=0}^{N} a_k \{\phi_k\}_i \exp(z_k m) = \pi_{0i} \qquad 0 \le i < c \qquad (3.9)$$

Equations (3.8) and (3.9) can be solved to yield $a_k$ and $\{\phi_k\}_i$ is the $i^{th}$ element of $\phi_k$.

If the probability of overflow beyond x for large values of x is G(x) = 1 - F(x), from equation (3.5) and (3.7) we have

$$F(x) = F(\infty) + \sum_{i=0}^{N-|c|-1} a_i \phi_i \exp(z_i x)$$

Hence

$$G(x) = -\sum_{i=0}^{N-|c|-1} a_i \phi_i \exp(z_i x) \qquad (3.10)$$

$|c|$ is the integer of c.

The solutions of F(x) and hence G(x) requires the computation of all the eigenvalues and the associated eigenvectors of the system. If there are N independent sources, each with K(i) states, i=1,..N, the system will compromise S number of states where

$$S = \prod_{i=1}^{N} K(i)$$

This will consume an unacceptably high computational load for any realistic number of sources and thus it is not possible to use it for on-line Network Control Operation.

A more easily obtainable approximation of the performance of the multiplexer can be found in [69] based on its asymptotic performance. The asymtotic behaviour of the system on a semilogarithmatic scale gives the linear asymptote of the CPDF. The slope of this asymptote $z_1$ can be explicitly computed from the dominating term in G(x) with the largest negative eigenvalue [45].

$$z_1 = \frac{(1 + \lambda/\mu)}{L} \frac{(1 - \rho)}{(1 - C/NB_p)} \qquad \lambda = \lambda_i, \; L=L_i, \; Bp = Bp_i$$

where L is the mean burst volume. For a different source having the same parameters L and $\lambda$, but different peak bit rate rate $B_p$ the asymtotic slope becomes

$$z_1 = \frac{(1+\lambda/\mu)}{L} \cdot \frac{(1-\rho)}{\left(1 - C/\sum_{i=1}^{N} B_{pi}\right)}$$

Assuming a common L for all sources the approximate initial position of the asymptote is log $a_1$ [69], given by d

$$d = \log\left(\rho\left(\frac{1}{b_{out}} - \frac{1}{b_{in}}\right)\frac{1+\lambda/\mu}{(1-1/Nb_{in})}\right)$$

where $b_{in}$ and $b_{out}$ are the normalised input and output rates respectively of each connection within a burst: $b_{in}$ = Bp/C, $b_{out}$= $c_{out}$/C and

$$c_{out} = \frac{B_p}{\rho(N-1)/N + Bp + \sum_{i=1}^{\lfloor c \rfloor} \{B_p + (i+1)B_p - 1\}\, \pi_{oi}}$$

The QOS criterion is the fixed probability G(x) of the buffer content exceeding a certain value x. By tailoring the asymptote to the QOS criterion, an upper and lower bound of the maximum number of sources allowed in the network can be found from the performance analysis in an inverse way. Although this approximation is accurate enough and requires less computation, it is limited to statistical multiplexing of identical bursty traffic of common mean burst length, L values only.

A more relaxed estimation of the QOS is obtained by assuming the system is at its limit x=0, and the mean burst length tends to infinity. The cell loss probability P(loss) expressed as a fraction of the total load can determined from equation (3.5)

$$P(\text{loss}) = 1/\beta N \sum_{i=\lfloor c \rfloor}^{N} \pi_{oi}\,(i - c) \tag{3.11}$$

where $\beta = \lambda(\mu + \lambda)$, the average fraction of number of sources being in the active state. On-line calculation is posssible in this case. However, the accuracy may not represent the real situation.

## 3.5 Review of Connection Admission Control (CAC) methods

Several connection admission control schemes for ATM based networks have been proposed in the literature [67-70, 72, 81,82]. The simplest approach is to allocate the declared peak bit rate of a connection for the whole call duration. The other scheme is to allocate an appropriate bandwidth lower than the peak bit-rate and thus taking advantage of statistical multiplexing gain at the expense of cell loss and delay. Basically, the latter approach can be classified into direct methods, indirect methods and methods based on the traffic mesurements.

The direct method requires on-line calculation of the QOS for the new state that will be achieved if a new call request is accepted [81, 82]. Accordingly, online calculation of cell level QOS function may become impractical as the number of classes of service increases and if enormous computation time and memory is required to solve a complex function. It may also be difficult to control the blocking probability among different classes.

On the other hand, the indirect methods allow cell level QOS and some parameters to be used for call admission control to be calculated off-line and stored in a look-up table, hence significantly reducing the amount of on-line calculation. There are two approaches to indirect methods of admission control; the virtual bandwidth method [26, 67,68, 70-74] and the virtual trunk method [77]. Most of the literature proposes the virtual bandwidth method which may be impractical in a large multiclass environment. However, approximation can be used to simplify calculation.

The methods based on measurement use traffic parameters such as cell loss rate, cell delay, buffer occupancy and and link occupancy as a complement or a substitute for the QOS function calculation. The accuracy may be improved in real time by using learning control methods [130].

The admission control algorithms proposed in this work may be used as direct or indirect methods and using virtual bandwidth and virtual link allocation depending on the service type and QOS requirements.

### 3.5.1 Bandwidth allocation

1) Non statistical bandwidth allocation

Continuous bit-rate (CBR) services will be allocated according to their peak bit rate as in the STM network. There is no advantage to be gained by cell multiplexing. The maximum load and hence the maximum number of sources allowed into the network without cell loss depends on the buffer size. The buffer size must be at least the length of the minimum cell interarrival distance as shown in figure 3.7. Sufficient buffer size must therefore be provided to avoid cell scale congestion. A new connection of peak bandwidth $Wp_N$ is accepted only if

$$\sum_{j=1}^{N} Wp_j \leq \alpha C$$

where $\alpha$ is a reduction factor of the link capacity and $\alpha < 1$.

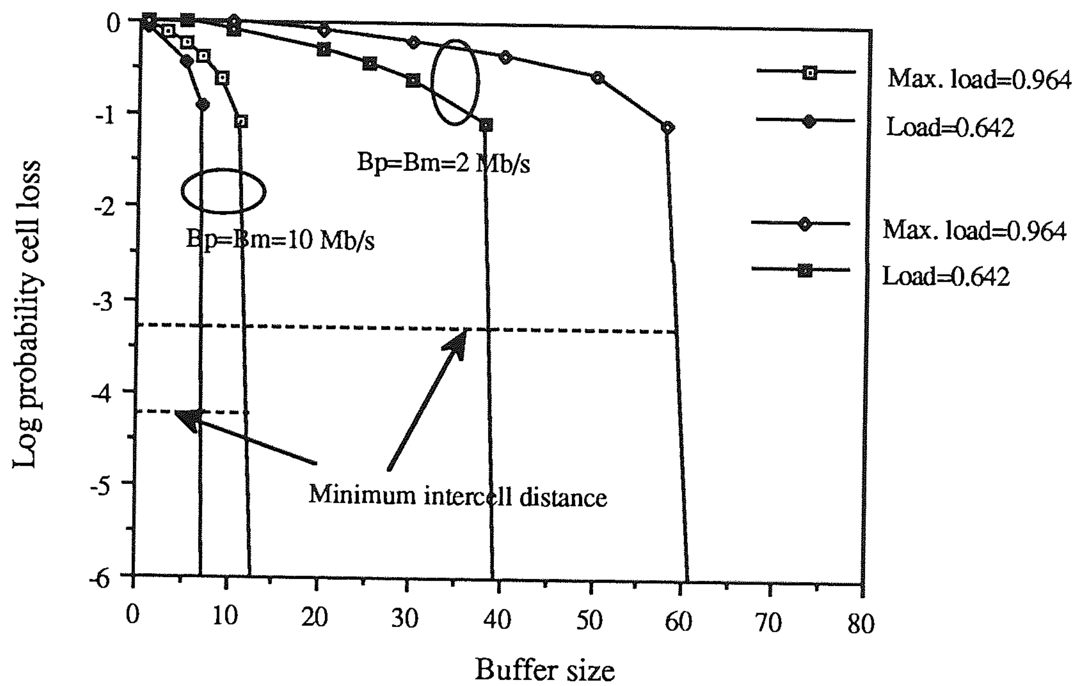



Figure 3.7 Probability of cell loss for CBR traffic of different load against buffer size.

2) Statistical bandwidth allocation

The statistical approach to bandwidth allocation is more appropriate for bursty traffic sources. In order to take advantage of statistical multiplexing gain the call acceptance algorithm must define the amount of bandwidth $Wv_j$ to be reserved for a connection. The amount depends on the characteristics of the source traffic descriptors and also on the traffic currently carried by the network so that

$$Wv_N + \sum_{j=1}^{N-1} Wv_j \leq \alpha C$$

In reality, packet switched or ATM based networks will have buffers in each switch and the performance evaluation measure can be calculated from the queueing model (see section 3.3). The bandwidth assignment mechanisms then guarantee a given level of QOS that accounts for cell and burst level congestion. In those assignment methods [69, 72, 74-76, 83] the calculations of the individual QOS are rather complicated and the queueing model used is sensitive to mean burst length. Also, the required parameters to be declared at the connection set-up are the peak rate, mean rate and the mean burst length which are still unknown for some present and future services.

A simpler method is to adopt a quality measure based on a bufferless model [68, 78, 79]. In this model the virtual cell loss probability is based on a link overflow bufferless fluid flow model, as shown in figure 3.8. In the following sections the derivation of the quality measure is presented for homogeneous traffic and extended to the heterogeneous environment. This measure is used by the CAC in the decision making process.

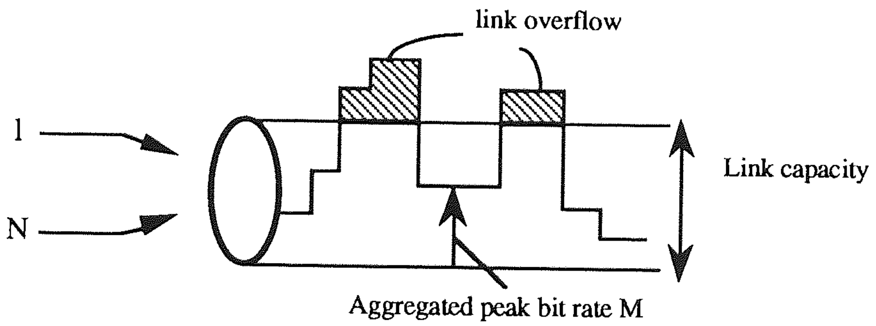



Figure 3.8 Link overflow model

### 3.5.2 Link overflow estimation - Homogeneous traffic

According to equation (3.5) and (3.7) in the fluid flow model analysis, when the system is at limits (ie. $n = i > c$), $F_n(0)=0$ and hence $u_n = \pi_{on}$. Therefore, the excess traffic OT in the link when there is no buffer in the system is

$$OT = \sum_{n=\lfloor M \rfloor}^{N} p(n)\,(n.Bp - C) \qquad n=i=M,\ldots N$$

where M is the aggregate peak bit rate, $M \in (n.Bp - C) \geq 0$ and $p(n) = \pi_{on}$ is as in equation (3.5).

$$p(n) = \binom{N}{n} (AF)^n \, (1- AF)^{N-n}$$

$AF = \lambda/(\lambda+\mu) = Bm/Bp$ is the activity factor. From equation (3.1) the traffic load is

$$\rho = \frac{N\lambda Bp}{(\mu + \lambda)C} = NBm/C$$

The virtual cell loss probability Po in the model is the ratio of the normalised overflow traffic and the traffic load given by

$$Po = \frac{OT/C}{\rho} = \frac{1}{N\,Bm} \sum_{n=|M|}^{N} p(n)\,(n.Bp - C) \tag{3.12}$$

The main advantage of this model is that the mean and peak bit rate can be easily declared and monitored. Cell loss is assumed never to occur when the aggregate peak rate M (M=nBp) is smaller than the link capacity. Nevertheless, cell loss does occur due to short term load fluctuation caused by simultaneous cell arrivals from different connections. Therefore in real systems appropriate buffer capacity must be allocated to prevent short term cell level congestion (see section 3.3.1 ).

In practice, the actual cell loss probability will always be less than the virtual cell loss due to savings by the buffer in a real system as shown in figure 3.9. The results show that the virtual cell loss probability is robust to the mean burst length variation and is a conservative quality measure.

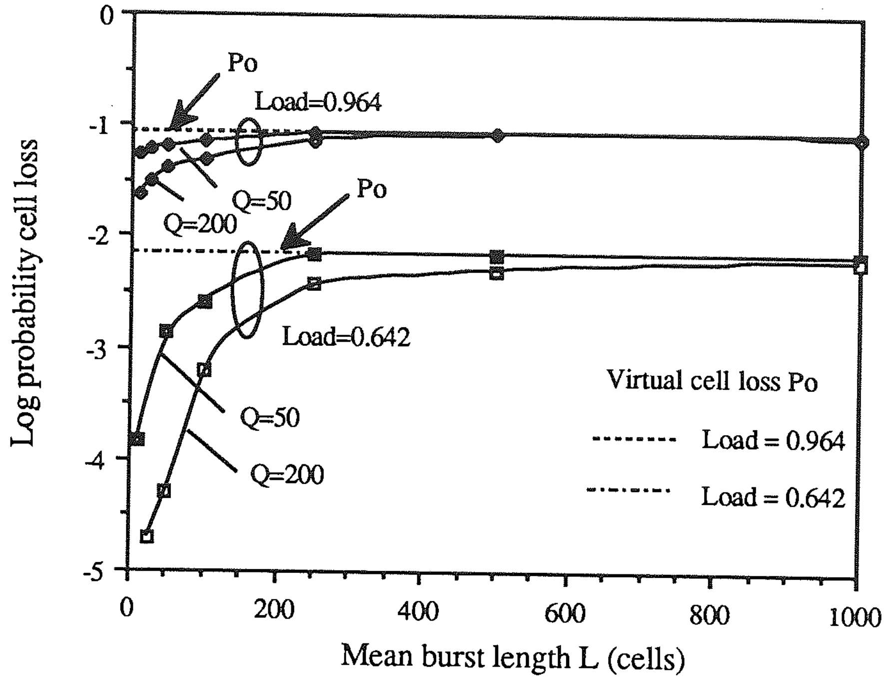



Figure 3.9 Probability of actual cell loss for buffer size Q=50 and Q=200 and the virtual cell loss as the mean burst length is varied

### 3.5.3 Heterogeneous traffic environment

The virtual cell loss probability for homogeneous traffic can be extended to the heterogeneous environment [68]. Assuming that traffic is categorised into K types according to their peak bit-rate $Bp_j$ and mean bit-rate $Bm_j$ and there are $N_j$ calls for type (j=1,...K). The average virtual cell loss probability PO for heterogeneous traffic is defined by

$$PO = \frac{1}{\rho_T} \sum_{n_j}^{n_1=N_1} \cdots \sum^{n_k=N_k} \left[ \prod_{j=1}^{K} p_j(n_j) \left( \left( \sum_{j=1}^{K} n_j.B_p \right) - C \right) \right]$$

$$\text{where} \quad n_j \in \left\{ \left( \sum_{j=1}^{K} n_j\, Bp_j - C \right) \geq 0 \right\},$$

$$\text{and} \quad p_j(n_j) = \binom{N_j}{n_j} (AF_j)^{n_j} (1 - AF_j)^{N_j - n_j},$$

and $\rho_T = \sum_{j=1}^{K} N_j \, Bm_j$ .

PO is the average virtual cell loss probability for K types of traffic. In reality, when different types of traffic are multiplexed, the cell loss probabilities observed for each individual type are usually different from the average virtual cell loss probability even if their required QOS (cell loss rate) are the same [74, 76]. It is therefore necessary that the admission control should provide a guarantee of a specific QOS for every type of service according to their individual cell loss probability.

The individual virtual cell loss probability among K types of traffic as denoted by $PO_j$ can be calculated based on the concept of the individual link overflow model as illustrated in figure 3.10.

The individual virtual cell loss probability for type j among K types of traffic is

$$POj = \frac{OT_j/C}{\rho_j}$$

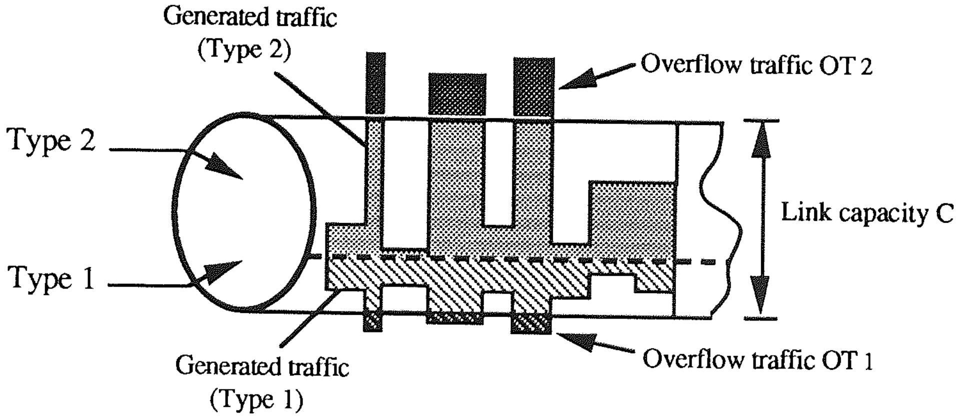



Figure 3.10 Individual virtual cell loss probability for two types of traffic

where $OT_j = \sum_{n_i}^{n_i = N_1} \cdots \sum^{n_K = N_K} \left[ \prod_{i=1}^{K} p_i\,(n_i) \left( \left( \sum_{i=1}^{K} n_i\, Bp_i \right) - C \right) n_j\, Bp_j \right) \Big/ \left( \sum_{i=1}^{K} n_i\, Bp_i \right) \right]$

and $\rho_j = N_j \, Bm_j / C$

$OT_j$ corresponds to the individual load that instantaneously exceeds the link capacity. Consider two types of traffic, Type 1 (Bp=10 Mb/s, Bm=2 Mb/s, b=5) and Type 2 (Bp=20 Mb/s, Bm=2Mb/s, b=10) where Type 2 traffic is more bursty than Type 1. The individual cell loss for each type is compared with the average virtual cell loss in figure 3.11. It is observed that the individual virtual cell loss and real cell loss for Type 2 traffic, which is more bursty, is relatively larger than those of Type 1. The significant difference is due to congestion by the more bursty traffic itself and hence causing cells to be discarded more frequently. The same characteristics hold true for more than two types of traffic. It is therefore important that the admission control scheme be based on the individual cell loss probability in order to guarantee a specific QOS for individual traffic.

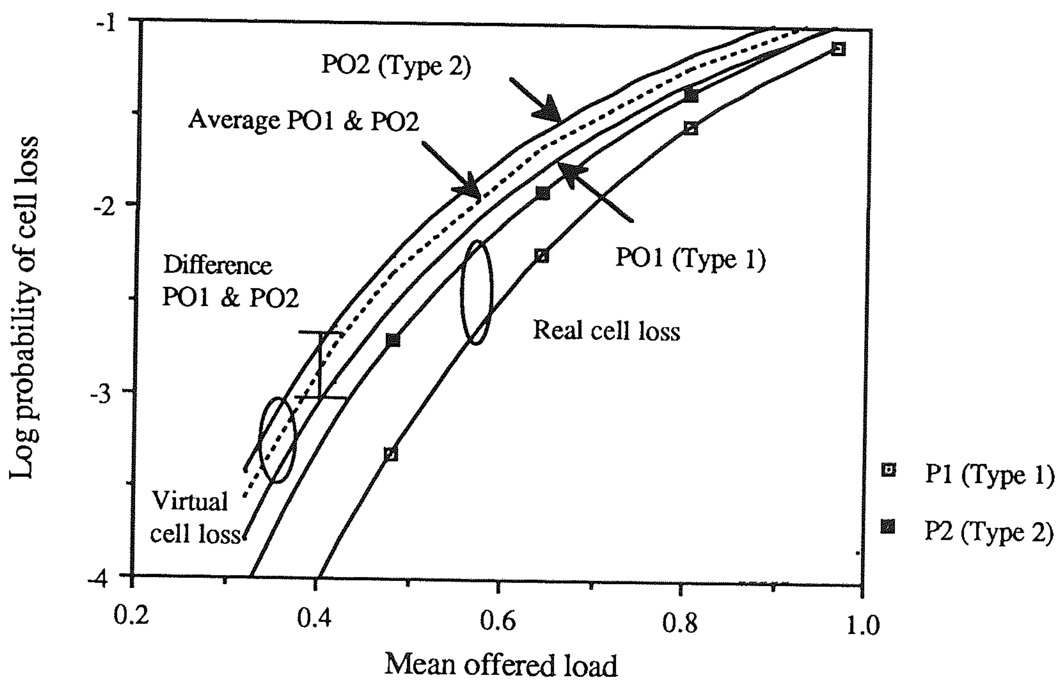



Figure 3.11 Individual virtual cell loss for two types of traffic
(Type 1: $Bp_1$=10 Mb/s, $Bm_1$=2 Mb/s, $\rho_1=\rho_2$)
(Type 2: $Bp_2$=20 Mb/s, $Bm_2$=2 Mb/s, $\rho_1=\rho_2$)

### 3.5.4 Virtual bandwidth allocation

A virtual bandwidth Wv of the required measure of QOS must be assigned to a new connection that has been accepted. The virtual bandwidth is defined as the effective bandwidth required by a call that will guarantee a specific cell loss probability in a homogeneous traffic environment. The virtual bandwidth of a particular QOS depends very much on the load and hence the number of connections in the system, as can be seen from figure 3.12. For example, in order to maintain a virtual cell loss probability of at least $10^{-6}$, the normalised mean offered load must be limited to 0.3 (N=19) and a virtual bandwidth of about 6.55 Mb/s or more must be allocated to the connection (for traffic of type Bp=10 Mb/s and Bm=2 Mb/s). When the cell loss probability is small, the maximum utilisation of the network is almost equal to the normalised mean offered load. The virtual bandwidth can be calculated as

$$W_v = \frac{Bm}{\rho_{max}/C}$$

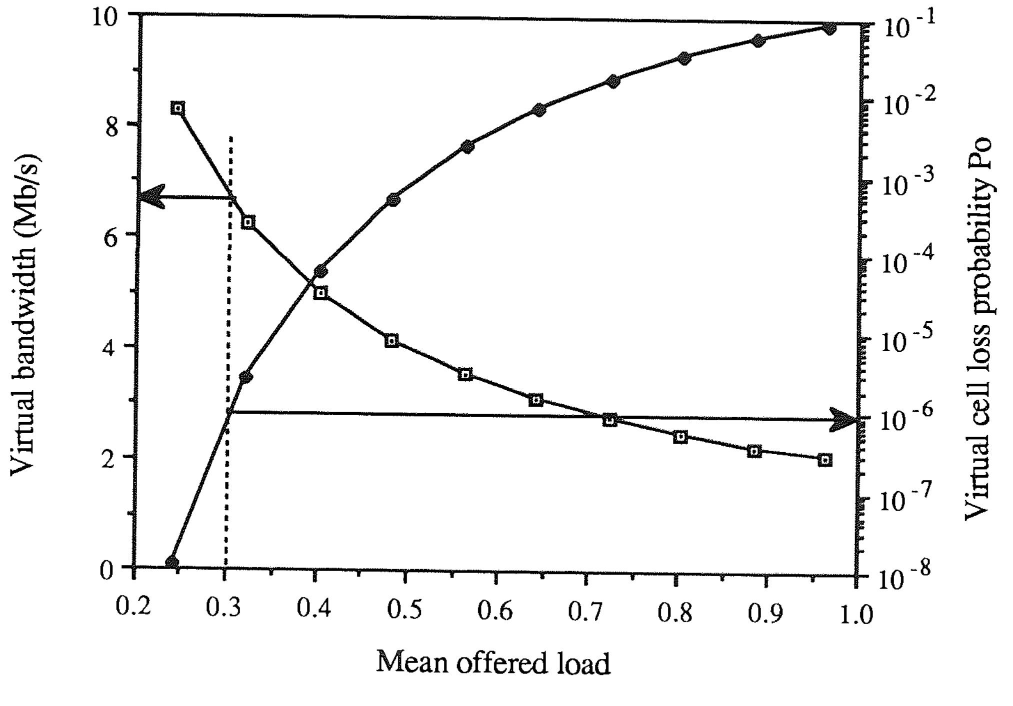



Figure 3.12 Virtual bandwidth and probability of virtual cell loss for homogeneous sources at various normalised mean offered load (Bp=10 Mb/s, Bm=2 Mb/s, L=100 cells)

Figure 3.13 and figure 3.14 show the call admissible region for different pairs of two types of traffic, corresponding to an individual virtual cell loss probability of $10^{-6}$ based on virtual bandwidth allocation. The $PO_1$ and $PO_2$ region are the exact admissible regions for type 1 and type 2 traffic with guaranteed cell loss probability of $10^{-6}$ (or $10^{-9}$). It is more concave than the Wv region in figure 3.14. If calls within the Wv region but above the $PO_1$ or $PO_2$

region in figure 3.14 are accepted, cell loss probability larger than $10^{-6}$ will occur. This is due to interference which causes the reduction in the statistical gain.

It was found, as explained by [30, 49, 78], that traffic sources having peak to link rate ratio near to or greater than 0.1 and traffic sources that are of a nearly continuous bit rate oriented will experienceno multiplexing gain and are prone to large interference. Such traffic requires virtual bandwidth as much as its peak bit rate (Wv = Bp) if strict cell loss probability (e.g. $10^{-9}$) is to be guaranteed.

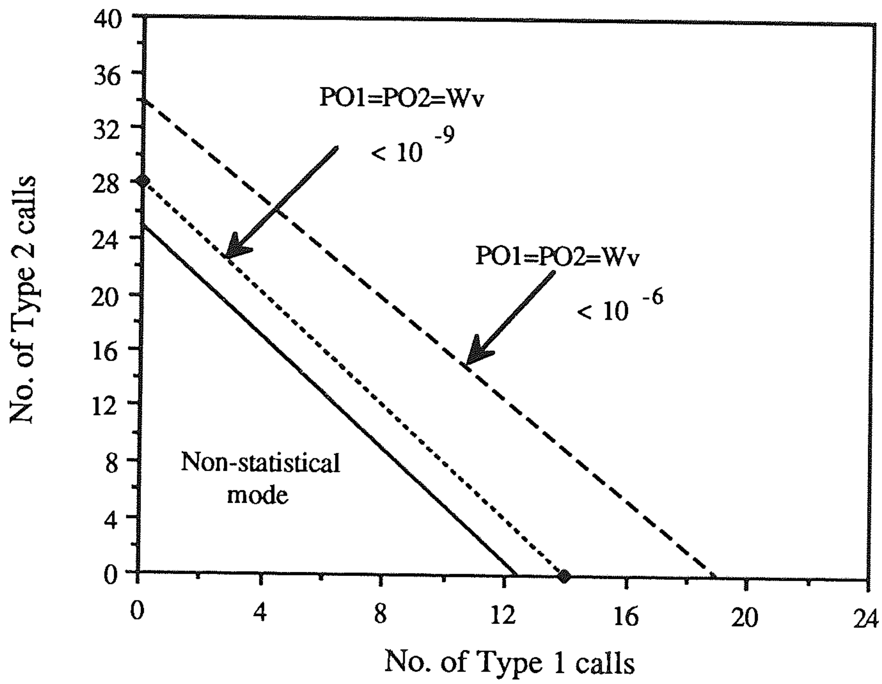



Figure 3.13 Admissible call region for two types of traffic
(Type 1: Bp=10 Mb/s, Bm=2 Mb/s)
(Type 2: Bp= 5 Mb/s, Bm=2 Mb/s)

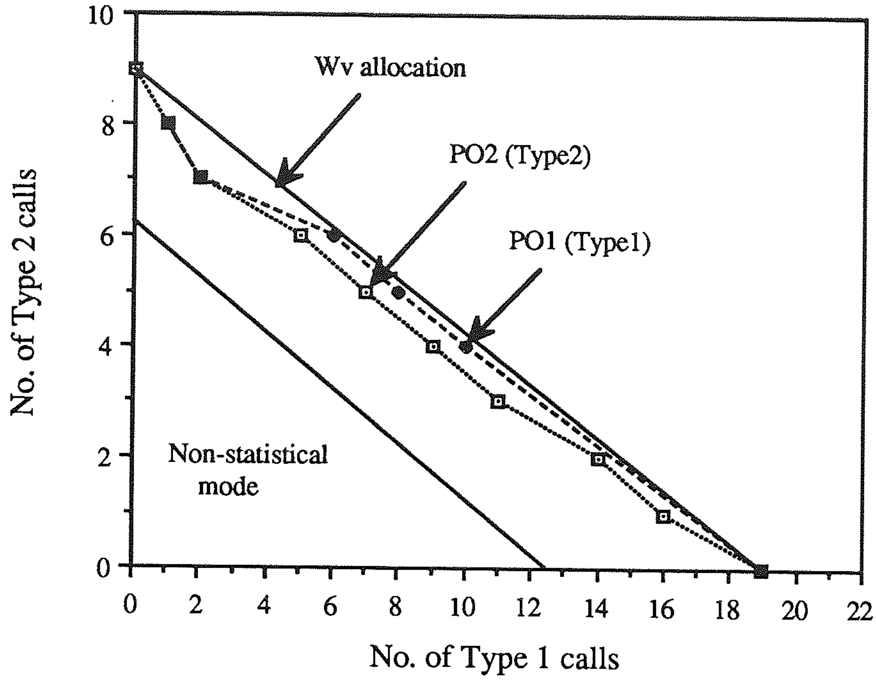



Figure 3.14 Admissible call region for two types of traffic
(Type 1: Bp=10 Mb/s, Bm=2 Mb/s)
(Type 2: Bp=20 Mb/s, Bm=2 Mb/s)

Although the virtual bandwidth allocation method is simple and practical, it encounters two main problems. Since it is based on a homogeneous traffic situation, specific cell loss probability cannot be guaranteed during statistical multiplexing of heterogeneous traffic, as already explained. Furthermore, the relation between the QOS and the allocated virtual bandwidth is not clear in the heterogenous environment. In order to use the virtual bandwidth allocation method effectively, it is proposed that the VBR traffic be divided into three categories as illustrated in table 3.1.

Category 1 traffic consists of large interference traffic whose peak bit rate to link rate ratio is near to or greater 0.1 and/or whose burstiness is almost 1.0. On the other hand, category 2 traffic is that traffic which is rather similar to category 1 traffic but with unrestricted burstiness and is delay tolerant. Category 3 traffic is made up low bit rate, very bursty and intolerable delay traffic that has small interference effect.

Table 3.1 Categories of variable bit-rate bursty traffic

| Category | Traffic source characteristics | | | |
|---|---|---|---|---|
| | Interference | Ratio Bp/C | Burstiness | Delay requirement |
| Category 1 | Large | > 0.1 | Near to or > 1.0 | Intolerable |
| Category 2 | Large | > 0.1 | No restriction | Tolerable |
| Category 3 | Small | < 0.1 | >> 1.0 | Intolerable |

If statistical multiplexing is adopted for all catogeries of service the uncertainty of cell loss probability can degrade the network performance. This can be avoided by allocating the virtual bandwidth according to the virtual division of the link capacity based on the traffic category. The required QOS provided by the virtual bandwidth for each type of call in separate virtual links can be maintained by assigning a priority mechanism.

## 3.6 Virtual link allocation

In order to achieve the greatest access flexibility while realizing good bandwidth utilisation, the total link bandwidth C is virtually split into two portions $C_d$ and $C_s$ where $C=C_d+C_s$ as shown in figure 3.15 with two QOS classes. Thus the admission control would assume that a total link bandwidth $C_d$ is reserved for non-statistical (deterministic) traffic and the other portion $C_s$ is available for statistical traffic.

The maximum limit of the link capacity for the deterministic traffic class is predetermined, while the statistical traffic may use link capacity up to $C-C_d$ available bandwidth. The link capacity has no explicit relation to any physical resources, such as dedicated slots within a frame, or specific buffers. Bandwidth is, therefore, virtually allocated within a given level of QOS but in each separate link capacity. For simplicity in control it is assumed that the values of $C_s$ and $C_d$ are controlled on a much longer time scale than connection requests.

The deterministic class implies an individual virtual cell loss probability of zero. CBR traffic will be assigned to the deterministic link capacity. The link capacity for the statistical class traffic $C_s$ is divided as illustrated in table 3.2.

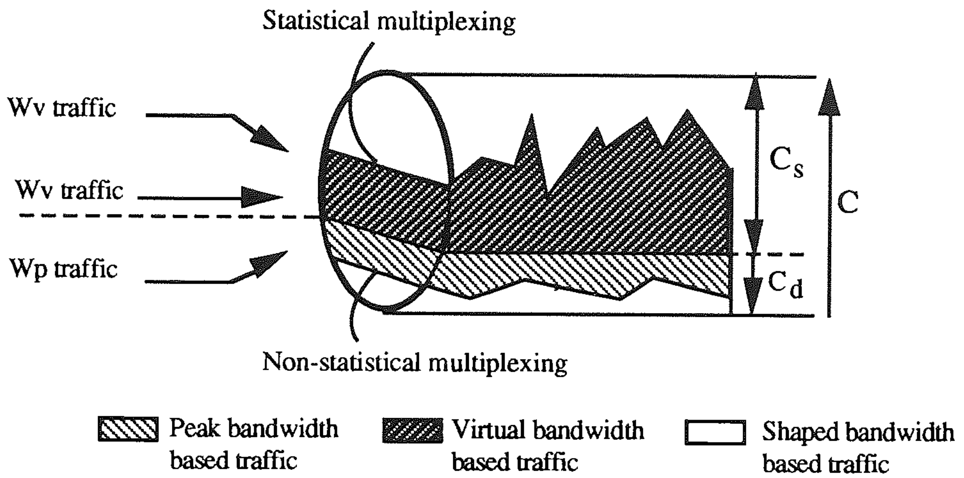



Figure 3.15 Virtual link capacity for heterogeneous traffic

Table 3.2 Bandwidth and link allocation

| Quality class | Category | Interference | Priority (Call level) | Reserved Bandwidth | Virtual link capacity |
|---|---|---|---|---|---|
| Deterministic | CBR | - | High | Wp=Bp | $C_d=\Sigma$ Bp |
| | Category 1 | Large | High | Wv=Bp | $C_{sd}=\Sigma$ Bp |
| Statistical | Category 2 | Large | Low | Wv=Bm | $C_{sv}=\Sigma$ Bm |
| (POj $<10^{-x}$) | Category 3 | Small | High | Bm<Wv<Bp | $C_{sv}=\Sigma$ Wv |

The QOS for statistical traffic depends on the individual traffic category requirement. Category 2 traffic is similar to that in category 1 but can tolerate larger delay such as data from a LAN. In this case more economical use of the network resource can be gained by a traffic shaping mechanism to reduce its peak bit rate to an appropriate value Bm. Alternatively, the traffic can be smoothed out by buffering and injecting cells into the network when there is available capacity.

Large interference traffic of category 1 is allocated in subcapacity $C_{sd}$ and category 2 in $C_{sv}$ using virtual bandwidth Bp and Bm respectively. For small interference traffic, its virtual bandwidth has to be calculated before it is assigned to subcapacity $C_{sv}$.

Each divided capacity is determined according to the forcast traffic intensity which is initially fixed but can be adaptively changed. The virtual bandwidth can be calculated on line or off line before call request. If the latter approach is adopted, the parameter sets such as (Wv:Bp,Bs,Cs), can be stored in a look-up table. This provides fast and convenient decision procedures.

### 3.7 Summary

In this chapter, performance analysis and simulation of the superposition of homogeneous on/off traffic based on the fluid flow model has been reviewed and carried out. The simulation results indicate that systems experience two congestion phenomena, the cell and burst level congestion. The fluid flow queueing model analysis may give quite accurate results but requires some complex calculation which must also be based on the declared parameter the mean burst length besides the peak and mean bit rate. The bufferless fluid flow model is a more flexible and robust approximation of the superposition of homogeneous sources. The analysis based on this model can be extended to a heterogeneous environmment and QOS (cell loss rate) can be based on individual virtual cell loss.

In a heterogeneous environment it was discovered that traffic sources that are more bursty experience more cell loss than less bursty ones. Bursty sources having peak bit rate ratio to link capacity near to or greater than 0.1 can cause large interference and do not produce statistical multiplexing gain. In the admission control the bandwidth allocation is based on the virtual bandwidth of individual traffic types.

However, in the heterogeneous environment, where statistical multiplexing gain is the prime aim, the bursty traffic with large interference will degrade the system performance. This situation is improved by two means of allocating bandwidth to such sources, depending on their delay requirements. For delay tolerant traffic the bandwidth can be shaped or smoothed out depending on the maximum tolerable delay. For delay intolerant traffic, a peak bit rate must be allocated to each source as in the deterministic class.

Traffic control based on preventive measure at call level has been dealt with in great detail in this chapter. The various bandwidth allocation methods in the connection admission control improves the network utilisation. However, this requires that statistical traffic does not violate the agreed parameters and a priority mechanism is carried out at the cell level in order to guarantee the different QOS requirements. Traffic controls at the cell level will be dicussed in the following chapters.

# CHAPTER 4

## INPUT RATE REGULATION

### 4.1 Review of problems in high speed networks

Any packet switched networks, including ATM based networks must have congestion management mechanisms due to their statistical nature. In practice, unacceptable delay and cell loss may arise, not simply from unpredictability of cell-based traffic, but also from contention, multiplexed time correlated traffic and cell clustering effects at switches. The characteristics of the ATM based network interface and the fact that a virtual connection can in principle exceed the negotiated throughput parameter up to the maximum capacity of the UNI may result in the degradation of the QOS for all connections sharing the same network resources in a statististical manner.

Most conventional packet switched networks carry only non-real-time data traffic which can be flow controlled in the case of network congestion. Since each user VC is allowed to accomodate more VC load than its capacity, the link/node will become congested if all VCs transmit at their peak rates simultaneously. Thus, congestion control must be invoked to control the input flow of each VC. Examples of congestion control schemes used in conventional packet switched networks are those based on end-to-end or hop-by-hop window flow control and back pressure with choking [129, 130]. These schemes are not appropriate in ATM based networks since;

1) the majority of traffic, for example voice and video, is not flow controllable. Such real time traffic has an intrinsic rate determined by external factors that are out of the control of the network.

2) the mechanisms rely on exchange of end-to-end control messages as feedback in order to regulate traffic flow. In a high speed link, the feedback is slow due to drastically reduced transmission time (less than $10^{-6}$) compared to the intrinsic propagation delays across the network (5 μs/km). Thus any action taken due to the outdated feedback is too late to resolve buffer overflows and avoid congestion.

The above factors argue for mechanisms that do not heavily rely on network feedback and suggest, for simple congestion control mechanisms that operate at the speed of the communication link. It is also desirable for the mechanism to use knowledge of the extrinsic parameters associated with the connection and control the source by forcing it to conform to these parameters. In this chapter, various traffic flow enforcement mechanisms are briefly outlined and a mechanism based on a window scheme is proposed. The various problems associated with it and the improvement measure are also discussed.

## 4.2 Traffic flow enforcement

Since the CAC algorithm relies on the contract parameter values, it can only perform correctly if all bandwidth contracts are respected by all connections already accepted. Therefore, an input rate regulation function is required to monitor the traffic generated by a subscriber and intervene when the negotiated parameters are violated. This function has been given many names in the literature, such as policing [86, 88, 89], bandwidth enforcement [74] and traffic flow enforcement [79]. CCITT has defined it as Usage Parameter Control (UPC) [3]. Traffic flow enforcement is used to describe the input rate regulation in this work.

The traffic flow enforcement function may be performed on virtual circuits, or virtual path or on the total traffic volume on access links within components such as concentrators and local exchange. In order to obtain a flexible and future proof network, the basic requirements of the traffic flow enforcement function are:-

- there should not be any action on the well behaving source (those who respect the contract).

- there should be fast detection and intervention to contract violation.

- to be cost effective, simple and easy to implement.

- it should be service independent, meaning it will only use different parameter settings for different services.

The ideal location of the traffic flow enforcement function would be to install directly in the subscriber terminal so that any traffic in excess of the negotiated parameter can be eliminated before it deteriorates the QOS of the other connections supported by the network. On the other hand, it has to remain under the control of the network to prevent technically experienced customers from manipulating the device to their advantage. Thus, it should be located as close as possible to the user network interface (UNI) but out of reach of the customers, for example, at the edge of the local exchange as shown in figure 4.1.

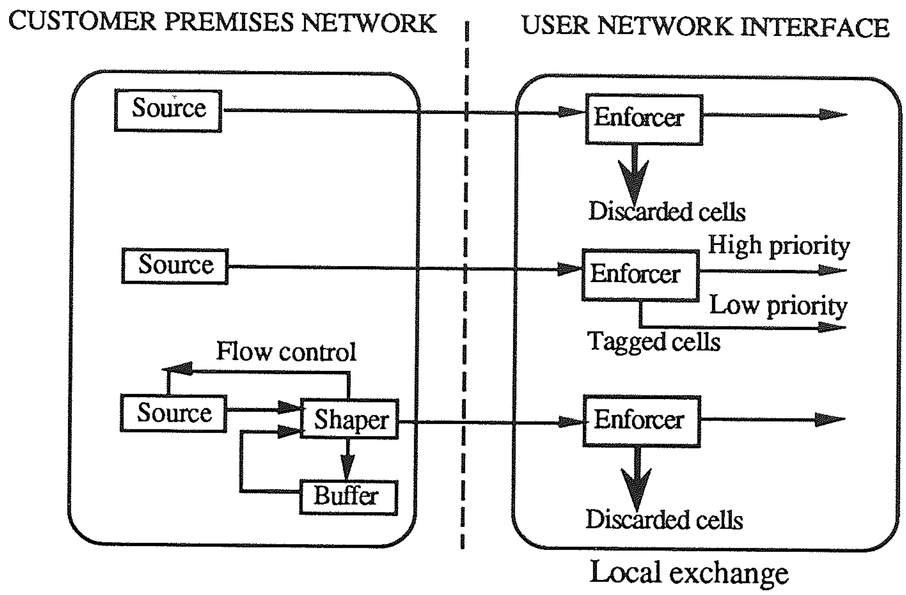



Figure 4.1 Location of traffic flow enforcement functions

The traffic flow enforcement function algorithms should consist of a measuring algorithm that provides the policing parameters and action to be taken when a threshold is reached. The most obvious action that can be taken after detecting a violation of the contract parameter is to discard cells. Alternatively, the violating cell can be marked or tagged for preferred deletion in case of congestion within the network.

In some applications where cell loss must be avoided, the user may employ a shaper. A shaper is similar to the enforcing unit but, instead of discarding cells, it buffers them and throttles the source in order to prevent cell loss. In effect it shapes the traffic to the desired characteristics so that the cell loss imposed by the enforcer within the network can be avoided. A shaper may be placed in the customer premises network as illustrated in figure 4.1.

The traffic flow enforcement function must cooperate with other traffic related functions as indicated in figure 4.2. The agreement is established by the CAC function and comprises of contract elements such as flow enforcement contract P, a set of QOS parameters Q, charging principle C and flow control scheme F.

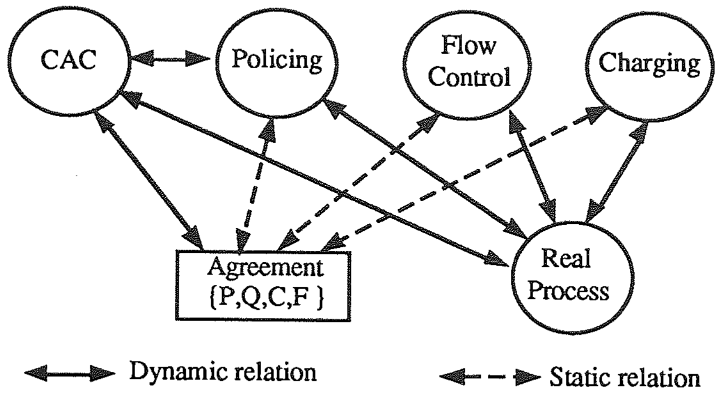



Figure 4.2 Policing and other traffic related functions

In the CAC described in chapter 3 the two main parameters in a contract are the peak and the mean cell rate. Enforcing the peak bit rate can easily be done by using any of the algorithms described in section 4.2.2.

The main problem with enforcing a statistical source close to its mean rate is that a large time constant (buffer) is needed to avoid excessive cell loss. For example, policing the mean value of a still picture source [97] (Bp=2 Mb/s, Bm=87 kb/s), with violation probability in the range $10^{-10}$ using leaky bucket, requires a buffer limit of approximately 400000 cells. This is unrealistic and implies inefficient control of the traffic flow. Proper dimensioning of the traffic flow enforcement mechanism is therefore ctritical.

### 4.2.1 Traffic flow enforcement mechanisms

In the literature, traffic flow enforcement has been classified into three catogeries; peak cell rate enforcement , mean cell rate enforcement and distribution based enforcement. Peak cell rate and mean cell rate enforcement can be carried out using one of the mechanisms described in the following section.

### 4.2.2 Cell rate based enforcement

The Leaky bucket scheme [89, 93-94, 96-98] is one of the well known traffic flow enforcement algorithms described in the literature. In this scheme, a cell is accepted into the network if it can draw a token from the token pool as shown in figure 4.3. In the original leaky bucket [9], the cell is discarded when the token pool is empty since there is no buffer.

However, if a buffer of size Q is employed, cells are allowed to queue on finding the token pool empty and will only be discarded if the buffer size Q is exceeded. The token pool is generated at rate Vm which must be close to the average bit rate of the source Bm and stored in the token pool of finite size. Evaluating the quantitative relation between Vm and Q is equivalent to estimating the maximum queue length when bursty traffic is input to a single server with service rate of Vm. The G/D/1-Q delay loss system is an exact model for the violation probability of the leaky bucket mechanism. Basically, the size of the bucket (ie. the buffer Q) decides the amount of credits given to the connection.

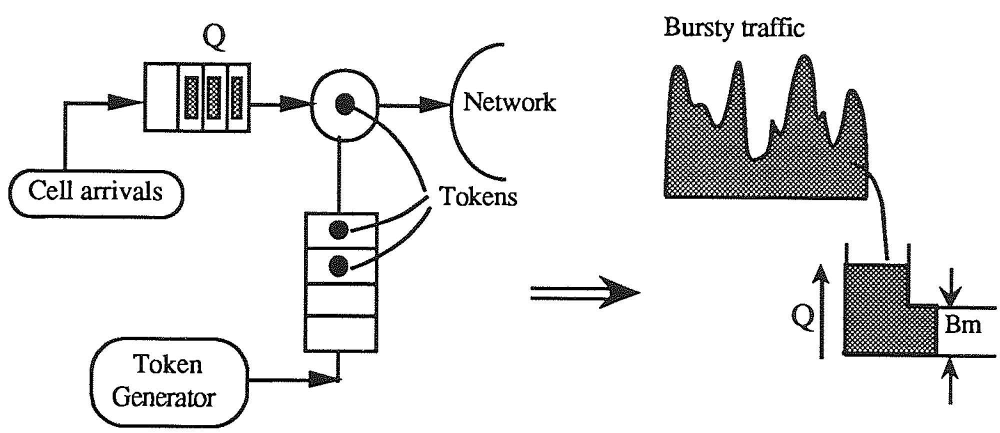



Figure 4.3 Leaky bucket traffic flow enforcement scheme

An alternative means to ensure that the source does not exceed the allocated parameters is the use of window mechanisms. The various window mechanisms mentioned in the literature are the jumping window, stepping window, exponentially weighted moving average and moving window mechanisms [92, 94]. The jumping window limits the maximum number of cells that can be accepted from a source within a fixed time interval T (window). When the time interval ends, the counters are reset and a new interval starts at the end of the preceeding interval. Alternatively, if a new window is triggered when the next cell arrives, this method is called stepping window.

The exponentially weighted moving average mechanism uses fixed consecutive time windows as in the jumping window, but the maximum number of cells accepted in the $i^{th}$ window is a function of the allowed mean number of cells per interval and an exponentially weighted sum of the number of accepted cells in the preceeding interval. The moving window mechanism is similar to the jumping window in which the maximum number of cells allowed in an interval T is limited. However, each cell is remembered for exactly one window width and a counter is incremented on arrival of each cell. Exactly T time interval after the arrival of the accepted cell, the counter is decremented by one again. This mechanism can viewed as a window which is steadily moving along the time axis.

### 4.2.3 Distribution based enforcement

In many cases, a VBR source such as videotelephony load state distribution cannot be defined by a known mathematical model. The Garbarit policing [88] approach is based on enveloping the source's distribution with a mathematically defined distribution such as the Gaussian distribution. The Gaussian envelope is described by its first two moments; mean and standard deviation. Since it is not feasible to police the Gaussian envelope continuously, the Gabarit enforcement function will monitor a set of cell rates of the source's Gaussian envelope denoted as quantiles, at fixed intervals of T cell time. The policing of the negotiated envelope at the selected quantiles results in an approximation of the Gaussian envelope by a staircase distribution denoted as the garbarit of the policed VC.

## 4.3 The proposed window mechanism

The dimensioning of the source traffic flow enforcement function requires realistic models for the traffic sources reflecting their main characteristics. However, exact traffic characteristics for most of the existing (eg. video) and future services are still unknown. In this work the signal sources are assumed to be categorised into two types:

1) signal sources whose statistical characteristics can be estimated in advance, for example, ouput of standard speech and video codecs.

2) signal sources whose statistical characteristics vary according to the terminal equipment and kinds of data to be transmitted.

For type 1 sources, the parameters already known in advance are used in the contract. On the other hand, type 2 signal sources have to declare the parameters at the call set-up phase and the traffic flow enforcement mechanism must monitor the signal source according to the declared parameters. The proposed traffic flow enforcement is for the latter type of signal source. Its mechanism is somewhat similar to the window mechanism in [24] where the maximum number of cells allowed in an interval $T_W$ is limited to M.

Unlike the leaky bucket, in which the users must declare the effective parameters which must be estimated from the relationship between Vm and Q, the method proposed here only requires a simple set of declaration parameters; TO, $T_W$, and M. TO is the minimum interarrival time between cells on the same VC, $T_W$ is the window time interval in terms of number of TOs and M is the maximum number of cells sent during the period $T_W$. Figure 4.4 shows the block diagram of the proposed mechanism.

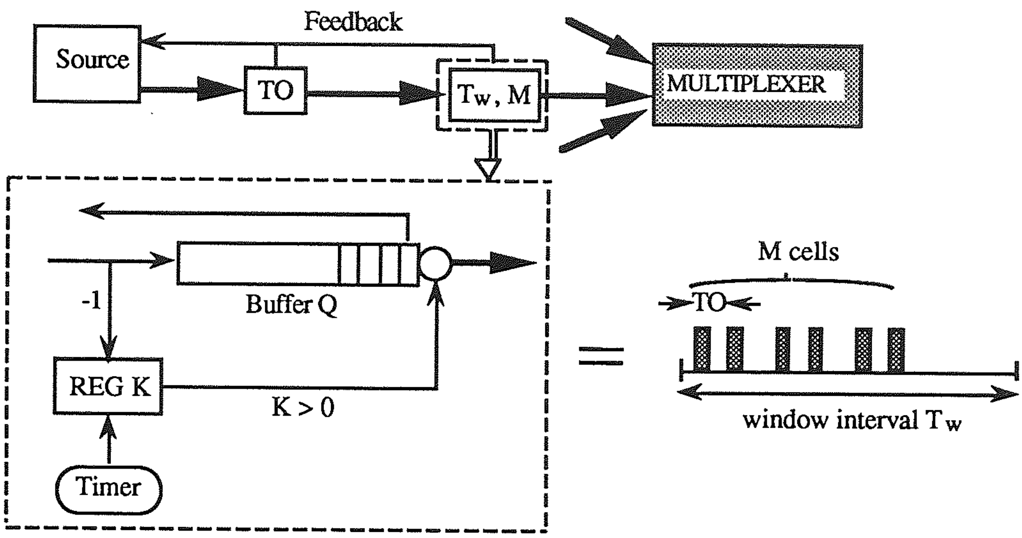



Figure 4.4 Block diagram of proposed traffic flow enforcement mechanism

The are two functional blocks in this mechanism, one will monitor the parameter TO and the other will enforce $T_W$ and M. The timer will update the period $T_W$ periodically. At each period $T_W$, the register K is set to the value M. Whenever a cell arrives, the register K is decremented by one before buffering the cell. The register is checked every TO time and if K is greater than zero then a cell is released. Cells that arrive when register K is equal to zero will be either discarded or transferred in the next time period when the register K is reset to M. This limits the maximum number of cells than can be transferred in an interval $T_W$ to M. When the number of cells held in the queueing buffer exceeds a certain threshold value, cells are discarded or a signal that suppresses cell generation is fed back to the signal source.

### 4.3.1 Virtual circuit traffic flow enforcement

In the literature, the performance analysis of many of the mentioned traffic flow enforcement mechanisms can be realised for a general renewal arrival process (eg. Poisson arrival process) through traditional teletraffic analysis or linear signal processing [93, 135]. For non-renewal arrival processes similar techniques may be used but more complex cell arrival properties have to be solved. The analytical evaluation of the proposed mechanism with two phase burst and silence sources can be estimated using the counting process for the cell arrivals which characterises the number of cell arrivals over an arbitrary time interval (0,t).

Assuming that the counter limit is M, the probability that a violation is detected by the mechanism is given by the mean number of policed (discarded) cells divided by the mean number of arriving cells within the interval $T_W$. In general this is equivalent to ;

$$p_{vio} = \frac{\sum_{i=1}^{\infty} i\, x_{M+i}(Tw)}{\sum_{i=1}^{\infty} i\, x_i(Tw)} \tag{4.1}$$

where $x_n(t)$ is the probability distribution of the counting process for n cell arrivals in an arbitrary interval $T_W$. In the case of a two phase on/off source, Rathgeb [92] computed the probability of violated cells of window length $T_W$ and maximum counter limit M using equation (4.2)

$$p_{vio} = \frac{Bm\left[\sum_{k=0}^{M} \binom{M}{k} (1-\alpha\, TO)^{M-k} (\alpha TO)^k \left(h - \frac{1}{\beta} \sum_{i=1}^{k} \left(1 - e^{-\beta h}\left(\sum_{j=0}^{i-1} \frac{(\beta h)^j}{j!}\right)\right)\right)\right] \sigma(h)}{Bm\, T_w} \tag{4.2}$$

where $h = (t - M.TO)$, $t = T_W$ and $\sigma(h)$ is the unit step function

$\alpha^{-1}$ = the mean burst length

$\beta^{-1}$ = mean silence length

Table 4.1 shows the violation cell loss probability for enforcing the mean bit-rate of a well behaved statistical traffic source with Bp = 10 Mb/s, Bm = 2Mb/s , $T_W$ = 17.6 ms and L = 100 cells. The results indicate that a non-deterministic traffic source will violate the policing criterion with a certain probability due to its short term statistical fluctuation, even if they respect the long term average. The long term average cell rate monitored by the mechanism is given by the ratio of M, the maximum number of cells accepted per interval and window width $T_W$.

This probability can be reduced by allowing more accepted cells (greater M) within a longer window interval while keeping the long term average bit rate constant (see Table 4.1). However, increasing the value M will increase the window length and hence the reaction time of the mechanism. A large reaction time implies inefficient control of the traffic flow.

Table 4.1 Probability cell loss for different counter limit

| Conter limit M | Window $T_W$ (TO) | Prob. of cell loss $P_{vio}$ |
|---|---|---|
| 100 | 500 | $4.14 \times 10^{-1}$ |
| 50 | 250 | $5.36 \times 10^{-1}$ |
| 25 | 125 | $6.41 \times 10^{-1}$ |
| 20 | 100 | $6.66 \times 10^{-1}$ |
| 10 | 50 | $7.29 \times 10^{-1}$ |

Although discarding cells may be the best action to control the source traffic stream, it implies strict requirements with respect to violation probabilities for well behaving sources. A more relaxed action is to buffer the cells and mark them for preferred deletion inside the network or inject them into the network at a later time. In this work the latter softer action has been opted, at the expense of lower QOS in terms of delay.

#### 4.3.1.1 Effect buffering cells

Assuming that the source (Bp=10 Mb/s, Bm=2Mb/s and L=100) does not violate the contract parameters, the influence of the counter limit on the queue length, waiting time and throughput was carried out while enforcing the mean cell rate. The negotiated peak cell rate was also enforced to prevent immediate flooding of the buffer. Table 4.2 shows the simulation results of the proposed mechanism based on the above assumptions.

The results show that larger counter limit and window interval will allow more cells to pass through with smaller average waiting time in the buffer. Enforcing the source in a smaller window interval increased the average waiting time of cells in the buffer and hence reduced the throughput measured within a certain time limit. However, smaller window interval reduces the number of cells in a burst which will have a great effect during multiplexing (see section 4.4).

Table 4.2 Effect of counter limit

| Counter limit M | Window $T_w$ (TO) | Ave. queue length (cells) | Ave. waiting time (ms) | Throughput % |
|---|---|---|---|---|
| 100 | 500 | 1176.5 | 203 | 93.15 |
| 50 | 250 | 1222.0 | 211 | 93.04 |
| 20 | 100 | 1252.4 | 216 | 92.96 |
| 10 | 50 | 1256.7 | 217 | 92.93 |

The main drawback of marking or buffering cells is that no distinction can be made between cells from a well behaved source and those from sources really violating the traffic contract. Another disadvantage is the large delay due to cells waiting in the buffer.

### 4.3.1.2 Overdimensioning

An alternative way of decreasing the delay incurred while keeping the probability of a non-deterministic, well behaved source violating the enforcing criteron low, is by introducing an overdimensioning factor $D \geq 1$. D is the ratio of the enforced cell rate $V_m$ to the actual mean cell rate Bm. Thus, the relationship between the counter limit and and the window length is given by

$$T_W = \frac{M_{OD}}{D\,Bm} = \frac{M_{OD}}{V_m} \qquad M_{OD} \geq M$$

Overdimensioning allows more than M cells to be transfered in any window interval $T_w$ and, therefore, the enforced mean cell rate is raised D times higher than the actual mean cell rate of the source. Figure 4.5 shows the simulation results of overdimensioning when enforcing the mean cell rate. Overdimensioning the actual mean cell rate allowed more cells to be delivered without having to wait long in the buffer. Hence the effect of queueing delay was very much reduced.

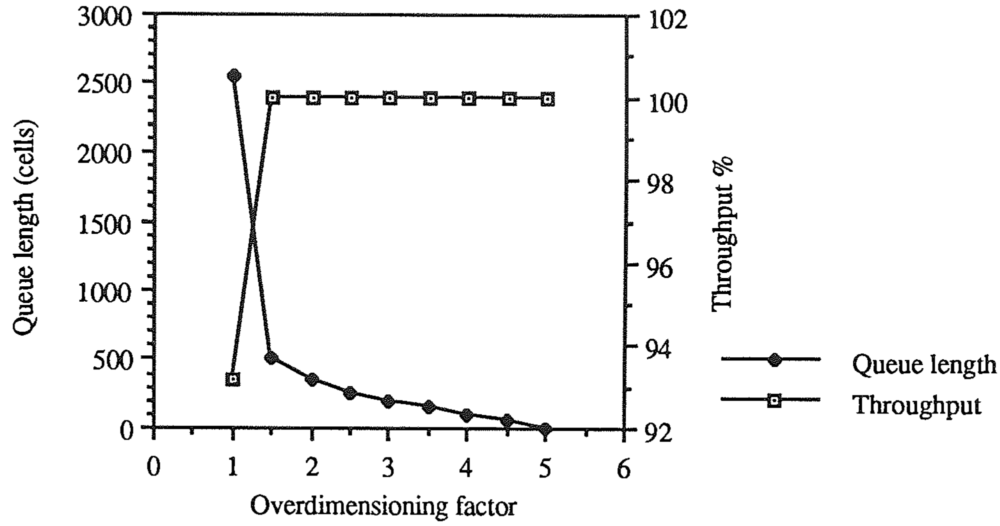



Figure 4.5 Effect of overdimensioning in an enforcement unit
(Bp = 10 Mb/s, Bm = 2 Mb/s and L = 100 cells)

Figure 4.6 illustrates the minimum required overdimensioning factor for different mean cell rates enforcement, assuming the peak cell rate is constant. It was found that with an overdimensioning factor $D \geq 1.5$, almost all cells were accepted within a smaller buffer size requirement and thus the delay experienced by cells before being released into the network is very much reduced. For example, enforcing a source characterised by peak bit rate 10 Mb/s, mean bit rate 2 Mb/s and mean burst length 100 cells to an enforced bit rate of 1.5 times higher than the actual mean bit rate reduced the mean cell waiting time in the buffer by almost 0.35 sec. which is a tremendous reduction.

It was also observed that the traffic burstiness had a great influence on the performance of the enforcement function. Enforcing very bursty traffic to its actual mean bit rate requires a lot of cells to be buffered, hence large queueing delay. Overdimensioning helps to reduce the buffering effect as shown in figure 4.6.

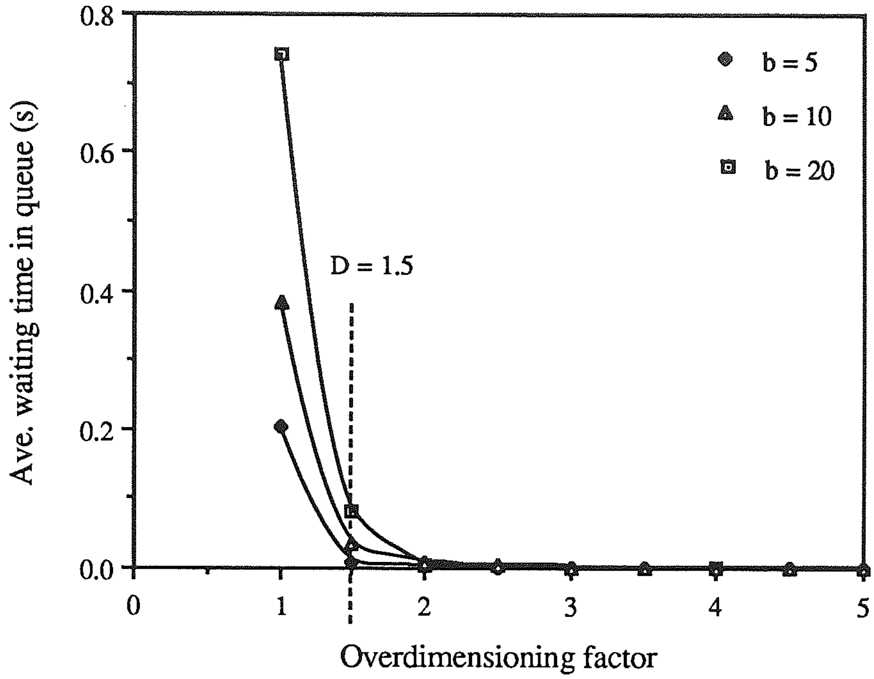



Figure 4.6 Overdimensioning effect for sources of different burstiness (Bp=10 Mb/s, L=100, Bm varies)

The main drawback of overdimensioning is that it will decrease the ability of the mechanism to detect real long term parameter violation. Users might take advantage by sending cells at the enforced bit rate which is D times higher than the actual cell rate without being detected.

The above arguments on dimensioning and actions taken on violating the enforcement criteria of the proposed traffic monitoring mechanism suggest that enforcing bursty sources to their mean bit rate cannot be performed efficiently and accurately to satisfy the most stringent QOS. There must be some tradeoffs between dimensioning and the accuracy in distinguishing between excessive and non-excessive traffic. In certain cases it is necessary to restrict the users to a virtual mean bit rate.The virtual bit rate is normally greater than the actual mean bit rate. However, the virtual mean bit rate must be less than the virtual bandwidth allocated by the CAC. Some kind of charging penalty must also be employed to discourage users from sending excessive traffic.

### 4.3.2 Virtual link traffic flow enforcement

Chapter 3 discussed the admission control assuming that the traffic sources do not violate the contract parameters and we do not consider the sources being enforced. This section considers the effect of statistical multiplexing the traffic streams coming out of the proposed policing function. Although the traffic flow emforcement function ensures that the source agrees to its contract parameters, it may alter the physical characteristics of the traffic source due to buffering. For example, the cell arrival process may differ from its original state and yet does not violate the contract parameters. Ideally, there should not be any buffer in the enforcement unit [9].

#### 4.3.2.1 Worst case traffic

To facilitate the study of the effect of multiplexing the enforced traffic streams, it is assumed that an infinite buffer is provided in the enforcement function. Generally, when the active and idle intervals are constant, the longer the active interval, the greater the multiplexing delay. The worst cell arrival pattern is that which produces the greatest multiplexing delay. In this case, the worst cell arrival pattern out of the police function is when two bursts of M cells, at the end of one interval and at the beginning of the next interval straddle two successive periods as shown in figure 4.7. The maximum burst length is doubled and the idle period is $\{2T_W - (2M-1)\}TO$.

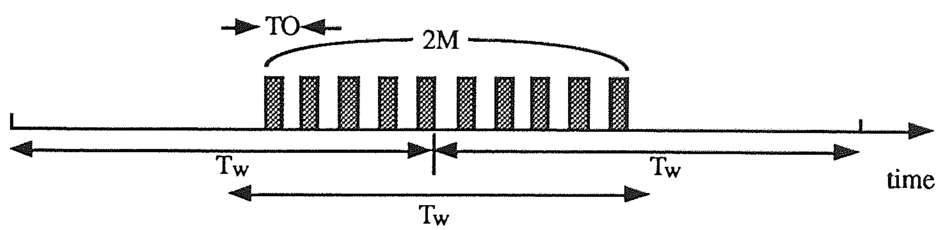



Figure 4.7 Worst cell arrival pattern

The probability density function (pdf) of interarrival time for the worst cell arrival process has been estimated by [26]. The approximation assumed that the cell interval for the first 2M-1 cells of burst 2M is TO and for the last cell is $\{2T_W - (2M - 1)\}TO$. The Laplace transform of the approximate probability density function of cell interval time is given by

$$\tilde{f}(s) = \left(1 - \frac{1}{2M}\right) e^{-TO\,s} + \left(\frac{1}{2M}\right) e^{-(2T_w - (2M - 1))TO\,s}$$

This equation can be used to estimate the counting of the cell arrivals process to represent the signal sources and the multiplexed state.

#### 4.3.2.2 Multiplexing worst case homogeneous traffic

The simulation results of multiplexing such worst case arrival patterns from homogeneous sources in a limited buffer system is shown in figure 4.8. The multiplexed state of independent arrival processes with worst case arrival pattern was a correlated and non-renewal process as described in chapter 3, section 3.3. Moreover, two bursts of M cells would concatenate in an arbitrary window interval $T_W$ resulting in short term average bit rate of $2M/T_W$ cell/s instead of $M/T_W$ cells/s. Although the long term average bit rate was not violated, the short term cell arrival fluctuations were increased and thus higher cell loss probability was observed (compare figure 3.2 and figure 4.8) if small buffers are allocated. A large buffer size was needed in order to overcome high cell loss probability. It was observed that a maximum load of $\rho = 0.964$ required a maximum buffer size of nearly $K = 6000$ cells to guarantee negligible cell loss.

The dotted lines in figure 4.8 show the cell loss probability limit for the bufferless model at different loads. If the CAC accepts a connection according to the bufferless model without considering the effect of the cell arrival process after the proposed enforcing unit, a huge cell loss will occur.

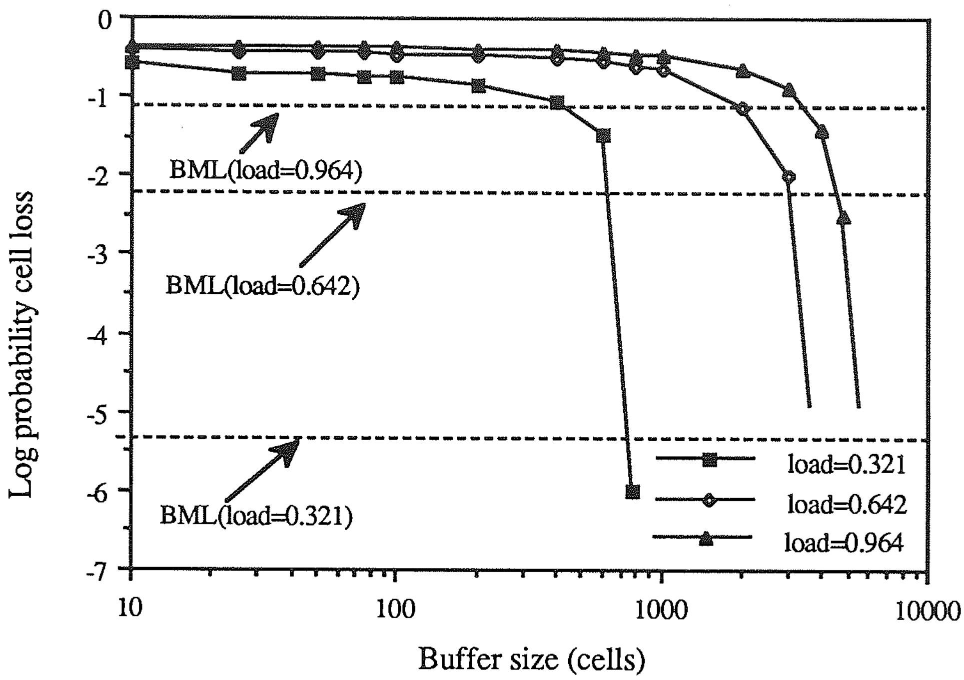



Figure 4.8 Multiplexing worst case traffic pattern at various load
(Bp = 10 Mb/s, Bm = 2Mb/s, L = 100 cells)

Figure 4.9 and figure 4.10 show the effect of multiplexing homogeneous sources at different loads using variable window length and counter limit M. The results indicate that in order to police the mean bit rate while guaranteeing negligible cell loss, the required maximum buffer size depends very much on the enforced window interval and hence the counter limit M. The cell arrival fluctuations at burst level can almost be suppressed by dimensioning the appropriate buffer size for the corresponding window interval. Obviously, buffer requirement for low load is very much less than for high load as shown in figure 4.9 and figure 4.10.

Dimensioning the network while considering the worst case arrival pattern will ensure a conservative QOS to the services. Assuming that the admission control accepts a connection of a well behaved source based on the declared parameter (peak and mean bit rate) and allocates a bandwidth equal to the actual mean bit-rate, and the enforcement function enforces the source traffic to satisfy the two negotiated contract parameters, the required QOS may be satisfied by dimensioning the buffer size such that the multiplexer will experience negligible cell loss. However, the delay incurred may be too large for real time services traffic.

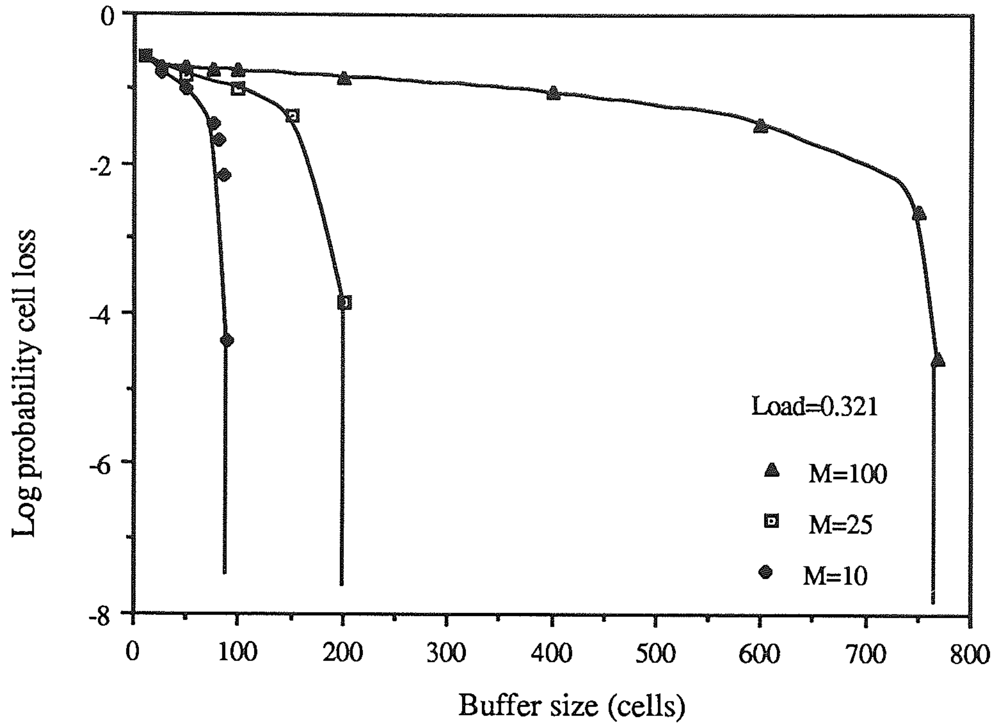



Figure 4.9 Multiplexing worst case homogeneous traffic at different policed counter limit and window length (Load = 0.321)

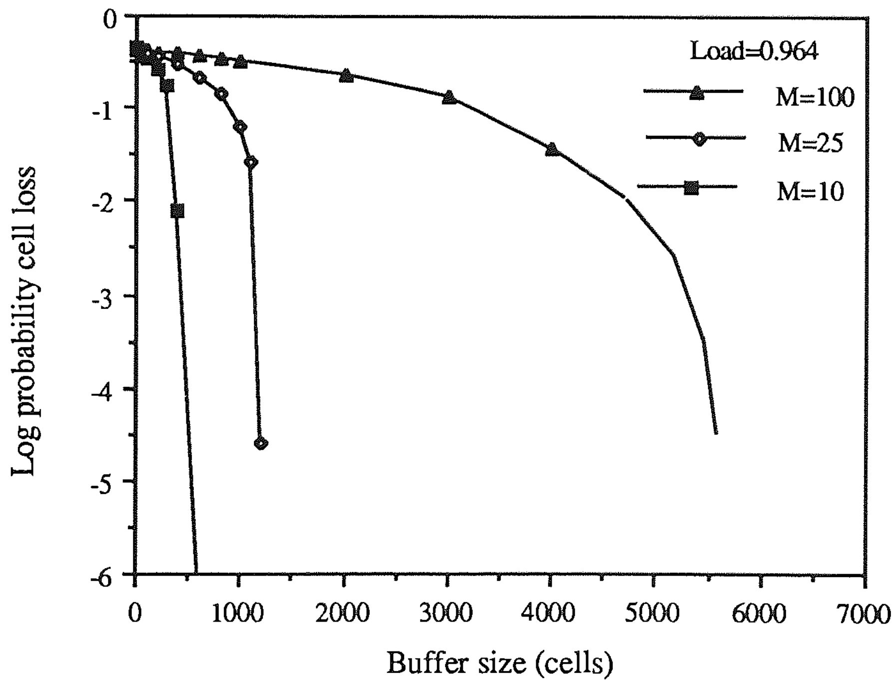



Figure 4.10 Multiplexing worst case homogeneous traffic at different policed counter limit and window length (Load = 0.964)

In order to reduce the delay, the admission control must allocate a virtual bandwidth higher than the actual mean bit rate and an overdimensioning factor can be introduced in the enforcement function. Figure 4.11 shows the effect of delay against the overdimensioning factor. Smaller delay is observed as the overdimensioning factor is increased. Therefore, the proposed mechanism is suitable for delay tolerable data services with stringent loss requirement.

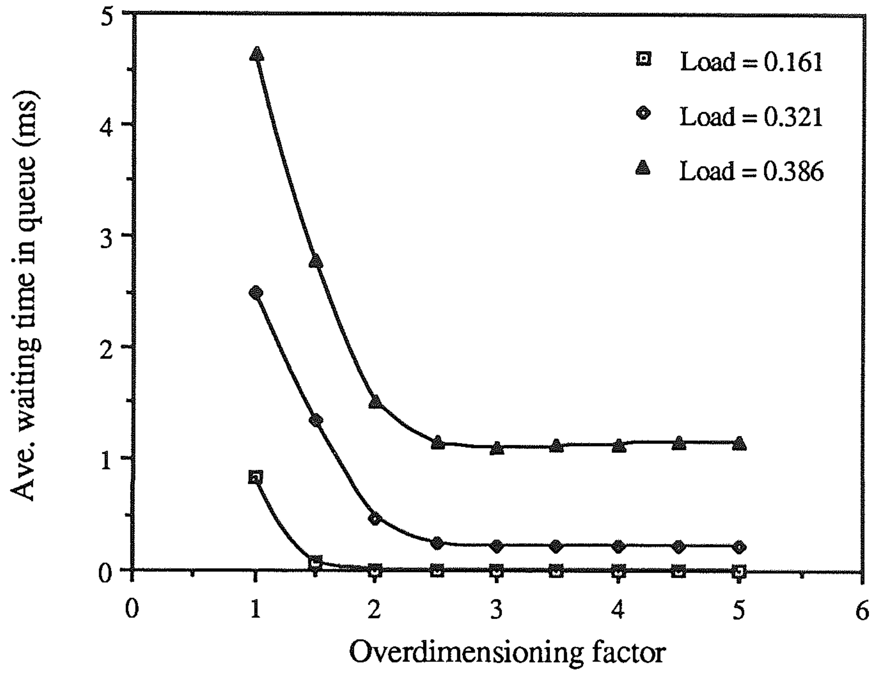



Figure 4.11 Reducing the multiplexing delay by introducing overdimensioning factor for different data load

## 4.4 Traffic smoothing

The window scheme ensures that the enforced bit rate is below its limit within the limited time interval. When several enforced traffic source are multiplexed, it was found that the multiplexing delay depends very much on the window size. The input traffic can be smoothed out to reduce the multiplexing delay in the node. The effect of smoothing is illustrated in table 4.3 for sources of different mean bit rates but similar peak bit rate and burst length.

A smaller window interval spreads out the worst case traffic pattern into several window intervals with the correspondingly smaller number of cells in a window. Thus, the effect of many cells clustering in a long burst is reduced and hence the delay due to queueing in the node buffer. However, there is a limit to the length of smoothing window interval as shown in table 4.3. Reducing the window interval to ≤ 50.TO (≤ 1.76 ms) will dramatically increase the mean delay in the node buffer. Similar results has been reported in [118]. It has been suggested that the smoothing window interval should be chosen reasonably small (greater than 1 ms for 150 Mb/s) depending on the requested throughput.

Table 4.3 Effect of reducing window size

| Window size M (cells) | Window interval $T_W$ (TO)( ms) | Mean queue length (cells) | Mean delay in node queue (ms) | Std. Deviation mean delay (cell time) |
|---|---|---|---|---|
| 200 | 1000 (35.2) | 1604.67 | 10.754 | 1204.2 |
| 100 | 500 (17.6) | 883.88 | 5.815 | 641.6 |
| 50 | 250 (8.80) | 490.28 | 3.19 | 282.97 |
| 10 | 50 (1.76) | 911.28 | 5.76 | 1404.4 |
| 8 | 40 (1.41) | 1476.2 | 9.33 | 2214.8 |
| 6 | 30 (1.06) | 2945.1 | 18.58 | 3705.66 |

### 5.0 Summary

Throughout the work, it was assumed that the traffic source did not violate their peak bit rate and thus enforcing traffic at the peak bit rate was not considered. A traffic flow enforcement mechanism has been proposed for signal sources whose statistical characteristics vary depending on the kinds of data. The mechanism is appropriate for data sources that require stringent cell loss probability and can tolerate delay. For other than this type of signal sources, it is suggested a leaky bucket mechanism is more adequate and easier to implement.

The proposed mechanism is simple and the parameters required are easy to declare. Excess traffic is either buffered or throttled and dishonest users will experience longer delay and some kind of charging penalty. With the appropriate window dimensioning, the multiplexing delay can be reduced to a minimum limit and the effect of cell clustering at a node can be minimised. The window mechanism can be extended to a flow control using a dynamic and adaptive window mechanism as described later in chapter 6.

# CHAPTER 5

## CELL LEVEL PRIORITY CONTROL

### 5.0 Bearer service and priorities

As mentioned in chapter 2, an ATM network must support a diversity of quality of service (QOS) and performance requirements. In principle it is possible to introduce only one unique bearer service that can satisfy the requirements of all services. However, this approach limits the flexibility and cost efficiency of the network. In order to achieve comparable flexibilities in QOS and bandwidth requirements, it has been proposed to introduce a second bearer service with a lower transfer quality and priority [102 - 108]. The two bearer capabilities could be offered either at the call level or cell level. Chapter 3 has dealt with the possibility of offering two bearer services at call level. In this chapter, the merit of using priority mechanisms with two bearer services at cell level will be discussed.

If two bearer capabilities are offered at the cell level, each cell of a given call may either be vital or ordinary. A vital cell must reach its destination if the adaptation layer is to retrieve the signal. On the contrary, the loss of an ordinary cell does not matter since its payload can be retrieved by the adaptation layer. The consequence is that there are now more than one cell loss requirements. This must be paid for by priority marking in the terminals and a more complex buffer management logic in the switches and multiplexers. Recently CCITT has reserved a cell loss priority bit (CLP) in the cell header for an explicit marking of cell loss priority [3,5]. The CLP bit has made it possible to assign the loss priority on a cell individual basis. Alternatively, a reserve bit in the cell header could be used.

### 5.1 Priority scheme

There are two different ways of implementing priorities; one way is to use a priority mechanism in the queueing discipline [29]. In practice, for a finite buffer system, the queueing strategy is composed of the service discipline which is concerned with the rule of selecting the next customer for service and the buffer access contol discipline which deals with the method of accepting cells into the buffer.

Another way of implementing the priority mechanism is as a means of controlling local congestion. Through this scheme different cell loss requirements can be satisfied by selectively discarding low priority cells when congestion occurs.

### 5.1.1 Queueing discipline scheme

Priority can be categorised according to a queueing strategy based on

1) service or time/delay priority which gives preferential delay treatment , or
2) buffer access or space priority which gives preferential loss treatment.

There are two further refinements possible in the priority queueing discipline, namely pre-emption and nonpre-emption. In pre-emptive cases, the customer with the highest priority is allowed to get service immediately even if another with the lower (service) priority is already present in service, or to push out another with lower (buffer access) priority from the queue.

The priority level can be differentiated by the delay sensitivity and loss sensitivity of each class of service. For example, voice is sensitive to delay but insensitive to loss. On the other hand, data is loss sensitive but delay insensitive. Priority can also be assigned in multiple classes of traffic within delay sensitive traffic each having their own delay requirement. In the following subsections, further elaboration of the different priority schemes is given.

#### 5.1.1.1 Scheduling between delay and loss sensitive traffic

Different methods of scheduling cells can be used at the switching node in the network. The simplest priority scheme is a static (fixed) priority. The push-out mechanism is an example of a static priority scheme [109]. In this scheme, if priority is given to the delay sensitive class, the delay sensitive class is always scheduled for service before the loss sensitive traffic. This scheme may cause high losses to the loss sensitive traffic while guaranteeing relatively low delay to the delay sensitive traffic. If a large portion of the network load consists of high priority traffic, then the performance of the low priority traffic will be severely degraded. The performance degradation of the low priority traffic can be reduced by the following dynamic priority schemes; the time priority Maximum Laxity Threshold (MLT) scheme [136] and the space priority Queue Length Threshold (QLT) [102, 103].

In the MLT scheme, a cell remains in the queue until either the cell is transmitted or its laxity reaches zero. The laxity of a cell in the MLT scheme is defined as the number of remaining slots in the buffer before its deadline expires. When the laxity reaches zero, the cell is discarded and considered lost. Priority is given to the delay sensitive cells if there are any delay sensitive cells in the queue, otherwise priority is given to the loss sensitive traffic. In this scheme, the laxity of each real-time cell needs to be updated in every timeslot, and each queue needs to be searched to find the minimum laxity. This discipline may involve heavy processing at each switching node. The number of queued cells at the switching node must therefore be kept small in order to minimised the processing time.

On the other hand, in the QLT discipline, priority is given to the loss sensitive traffic when the number of loss sensitive cells in the queue exceeds the threshold value, otherwise priority is given to the delay sensitive traffic. In both of these dynamic priority schemes, an appropriate threshold value must be chosen in order to achieve a desired performance level for each of the high and low priority classes.

The performance of MLT and QLT has been examined by Chipalkatti et. al [110] who showed that there is little difference in the performance trade offs between the two disciplines. They concluded that QLT is more practical than MLT due to its simpler implementation. Furthermore, queueing delay in a wide area broadband network is negligible compared to the propagation delay. The improvement in performance due to reduction of queueing delay from time priority is therefore of limited use for very high speed networks. Nevertheless, time priority may be used to reduce the delay jitter or variance of real-time constant bit-rate traffic.

#### 5.1.1 2 Scheduling between multiple delay sensitive traffic

The need may arise to provide some of the traffic such as signalling data and real-time control information with faster network transfer. In the literature different delay requirements within delay sensitive traffic have been identified and are differentiated by time priorities schemes such as Head-of-the-Line with Priority Jumps (HOL-PJ) [137]. The usual HOL [131] mechanism offers different delay characteristics for different classes in a limited buffer but cell loss is independent of HOL priority classes.

### 5.1.2 Local congestion scheme

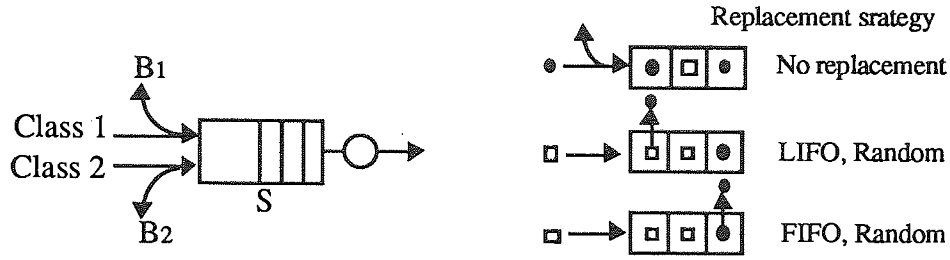
Priority schemes can also be used as local congestion control to satisfy the different cell loss requirements of different classes of traffic. When congestion is detected, priority is given to loss sensitive traffic such as data, over loss insensitive traffic such as voice, and cells from lower priority classes are discarded first. In discarding voice cells, an improvement can be obtained by selectively discarding voice cells containing less important information and whose loss will have the least effect in the reconstructed voice signal [138]. Similar treatment can also be applied to video traffic with embedded coding.

## 5.2 Space priority

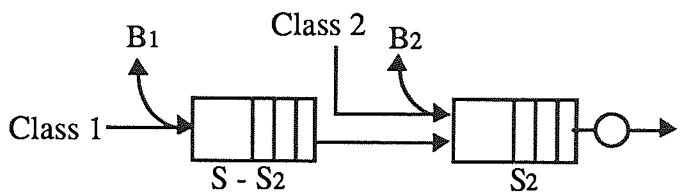
Priorities for space in the buffer include those mechanisms which offer dedicated access for each class of traffic as shown in figure 5.1. The push-out mechanism was introduced by Sumita [103] and Hubertere et. al [107] to increase the network utilisation. In this mechanism, the arriving cell can replace a cell with lower priority in the buffer S. The selection of the cell to be discarded depends on the replacement strategy. Since the cell sequence has to be preserved, this implies a complicated buffer management.

Kroner [102, 106] has proposed a partial buffer sharing mechanism which combines good performance with simple buffer management. This mechanism limits the number of buffers that can be shared by both classes of cells. If the number of cells in the buffer exceeds the threshold $S_2$, then class 2 arrivals are discarded, whereas class 1 arrivals are accepted as long as there is a free buffer. The threshold value $S_2$ can be adjusted to adapt the system to various load situations.

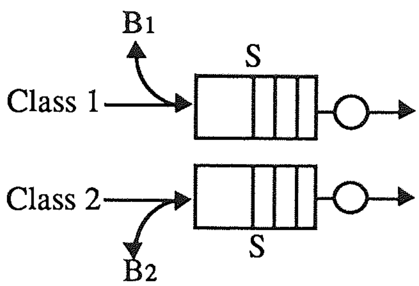
The simplest method to establish space priority capabilities within an ATM network is the mechanism which uses separate routes for different classes of traffic. In this approach the priorities are assigned on a per connection basis and remain unchanged for the duration of the connection in order to avoid cell mis-sequencing caused by a high priority cell overtaking a low priority one.




a) Push-out scheme with a common buffer and the different replacement strategies




b) Partial buffer sharing




c) Separate route for each class of traffic

Figure 5.1 Space priorities mechanisms for selective cell discarding

In the literature much work has been reported on the evaluation of the performance of time priority mechanisms. There is very little work reported related to space priority mechanisms. In this work a space priority queueing discipline based on the partial buffer sharing (PBS) has been adopted for a private ATM network. However, the queueing discipline differs from the PBS described in [102, 104, 106] in terms of the queue length threshold and the buffer size and the way arrival cells are accepted into the main buffer.

## 5.3 Proposed queue length threshold method

The proposed queueing strategy differentiates between loss sensitive and delay sensitive traffic. The priority level for a two bearer service is as shown in table 5.1

Table 5.1 Bearer service class and priority level

| Bearer service class | Sensitivity | | Priority level |
|---|---|---|---|
| | Loss | Delay | |
| Class 1 | No | Yes | 1 (High) |
| Class 2 | Yes | No | 0 (Low) |

It is assumed that loss sensitive traffic is delay tolerant and loss tolerable traffic requires a stringent delay limit. Class 1 traffic is traffic from the deterministic quality class and the statistical quality class of categories 1 and 3 as described in section 3.6. Class 2 traffic is mainly data traffic of category 2 that can tolerate delay.

The different loss probability requirements for various types of traffic in class 1 can be satisfied through the acceptance control strategy described in chapter 3. The proposed space priority mechanism offers only one delay characteristic for the different types of traffic in class 2. The main aim of the proposed mechanism is increased network utilisation which makes full use of the available bandwidth since leased lines in private networks are expensive.

### 5.3.1 Buffer access discipline

The buffer access discipline described below differs from those studied by Kroner [102] in the way the priority and non-priority cells are accepted into the buffer. This mechanism is shown in figure 5.2. The priority cells of class 1 are accepted as long as the buffer $S_1$ is not full. Arriving class 2 cells will be buffered in buffer $S_2$ and will only be transferred to buffer $S_1$ when the available queue length of buffer $S_1$ is greater than $S_1$-$S_T$, where $S_T$ is the threshold queue value. Arriving cells of any particular class will be discarded on finding the buffer full.

The threshold queue value $S_T$ limits the maximum load of class 1 traffic into the network while the buffer size $S_1$ limits the cell level congestion of the total load into buffer $S_1$. The buffer length of $S_1$ should be reasonably short in order to satisfy small delay and delay jitter limits required by class 1 traffic. On the other hand, buffer size $S_2$ is long and class 2 traffic on transit in this queue may experience large delay. The probability of cell loss for class 2 traffic is minimised to almost zero.

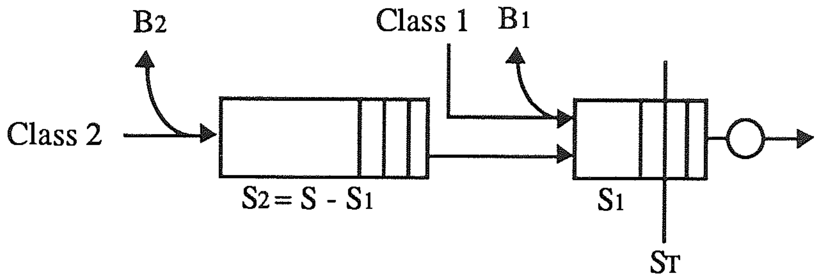



Figure 5.2 Partial buffer sharing with zero loss

The traffic model for class 1 traffic is as described in chapter 7. The arrival process to the buffer $S_1$ is governed by an independent process. The service time of the server is independent and deterministically distributed. The service discipline is FIFO in order to guarantee cell sequence intregity and the buffer has finite waiting room.

In a discrete-time system such as an ATM system, the interarrival time between successive arrival instants is geometrically distributed but the limitimg case of a discrete time GEO/G/1 queueing system is the M/G/1 queueing system [60]. The analysis of the queueing behaviour, therefore, can be approximated by a memoryless process. Moreover, the short-term behaviour of a statistical multiplexer can be characterised using this simple arrival process [56].

### 5.3.2 Preliminary study of the queue length distribution of buffer $S_1$

A preliminary study of the queue length distribution of the superposition of homogeneous and heterogeneous traffic was carried out in an infinite buffer system. Three categories of input traffic were considered in the simulation work for class 1 bearer service, CBR, voice and video traffic. The queue length distribution for each type of traffic and the combinations of the above traffic load were determined. Figures 5.3, 5.4 and 5.5 show the results of the queue length distribution obtained by simulation for homogeneous CBR, voice and video load respectively.

The queue length distribution for CBR traffic load is reasonably small (< 50) if the total peak bandwidth does not exceed the link capacity. Category 3 bursty voice traffic experienced statistical multiplexing gain and thus used a small buffer for multiplexing a large number of sources. However, bursty video sources of high peak bit rate requires large buffer size to take advantage of statistical multiplexing gain, which will introduce intolerable delay. Thus, if the total number of video sources is limited to a maximum of 13, the buffer must reserve a maximum buffer size of about 50 cells, which implies almost near to peak bit rate allocation.

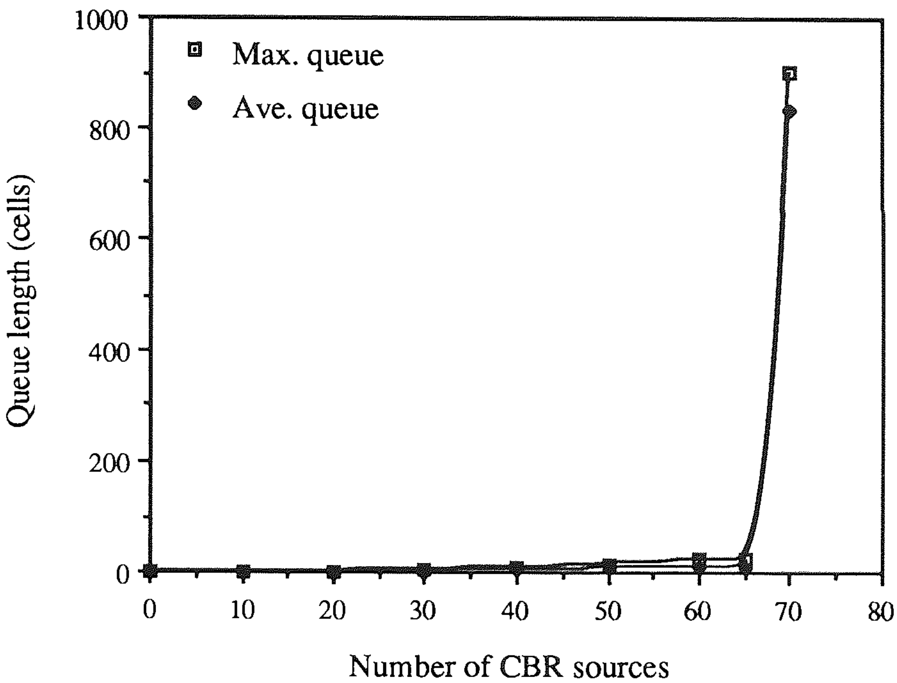



Figure 5.3 Queue length distribution for CBR source
(Bp = Bm = 2Mb/s)

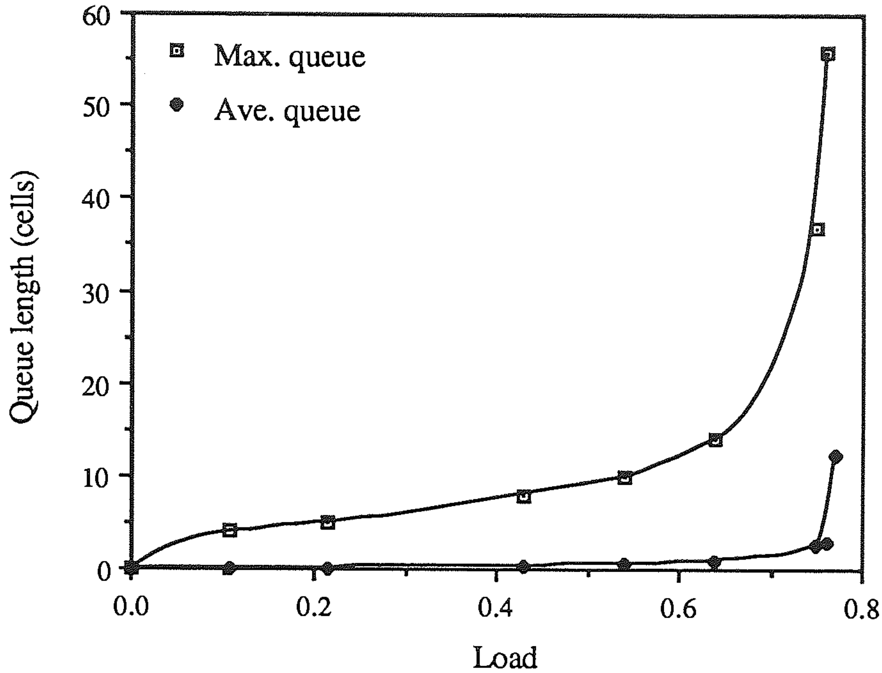



Figure 5.4 Queue length distribution for voice sources of various load
(Bp = 64 kb/s, Bm = 28 kb/s)

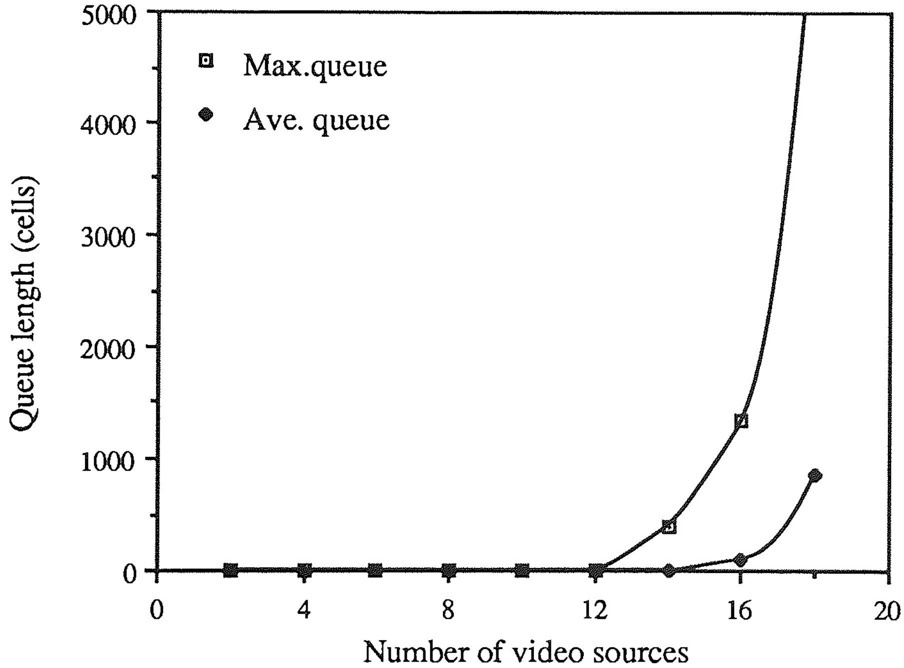



Figure 5.5 Queue length distribution for video sources
(Bp = 10.575 Mb/s, Bm = 3.9 Mb/s)

In all cases, the mean queue length is very much smaller than the instantaneous maximum queue length except for CBR sources. This exception is mainly due to equal interarrival distance of cells from each source. The maximum queue length is mainly attributed to the simultaneous arrivals of cells from different input sources. It is an important measure since it determines the probability of cell loss.

#### 5.3.2.1 Queue length distribution for heterogeneous traffic

A mixture of two types of traffic are considered. The queue length distribution of mixed traffic, for example, fixed CBR and variable video source, is shown in figure 5.6. The maximum queue length for a total load of up to 0.8 is reasonably small. The queue length suddenly rises when the total offered load exceeds this proportion of the link capacity.

In figure 5.7 the maximum queue length for mixed CBR and voice sources of variable load combinations shows very similar results as in figure 5.6. For example, the maximum queue length for mixed CBR and video sources is 10 cell slots for a total load of about 0.6. The same result is observed in mixed CBR and voice sources. Varying the load combination does not affect the queueing behaviour for a buffer size of about 50 cells. This implies that the queue length distribution mainly depends on the total load in the network for small buffer size.

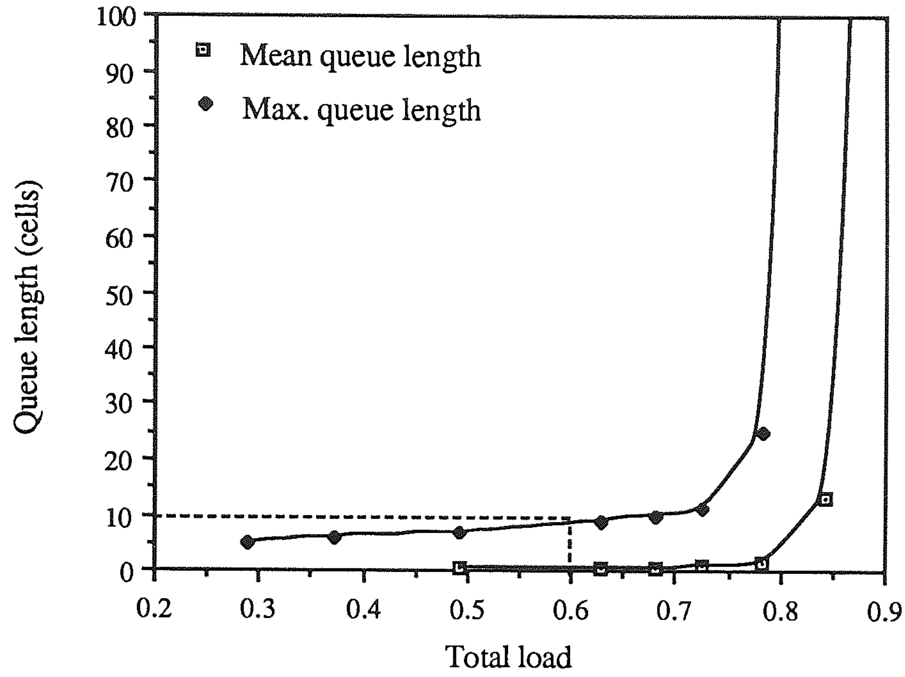



Figure 5.6 Queue length distribution for mixed CBR and video traffic
(625 CBR sources of 64 Kb/s and variable video load)

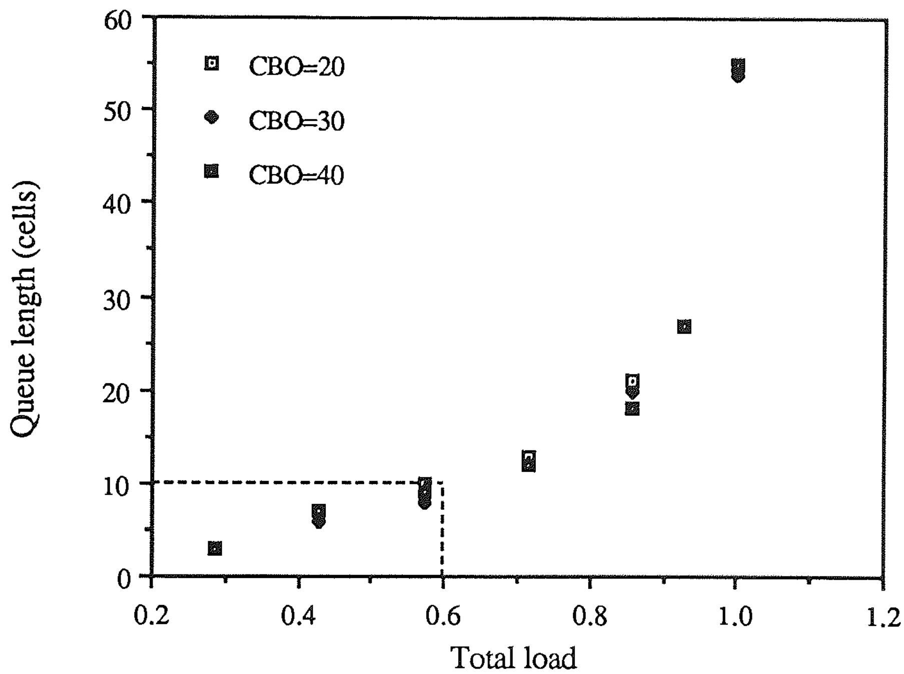



Figure 5.7 Maximum queue length distribution for mixed CBR and voice source of various combinations of load
(CBR sources - 2 Mb/s, and voice - Bp = 64 Kb/s, Bm = 28 Kb/s).

### 5.3.3 Dimensioning buffer $S_1$ and the threshold value $S_T$

The analysis of the above queueing behaviour can be approximated by the queueing system at the output queue of a switch [100]. Assume that cell arrivals on N input traffic sources are independent and identical Bernoulli processes and ρ is the probability of a cell arriving in any given cell slot in a queue. In such a queueing system, let the random variable A be defined as the number of cell arrivals destined for the output in a given cell slot. The probability of k arrivals has the binomial distribution

$$a_k \equiv P[A = k] = \binom{N}{k} (\rho/N)^k (1 - \rho/N)^{N-k} \qquad (5.1)$$

$k = 0, 1, 2, \ldots, N$, with probability generating function

$$A(z) = \sum_{k=0}^{N} z^i P[A = k] = \left(1 - \frac{\rho}{N} + z\frac{\rho}{N}\right)^N \qquad (5.2)$$

For $N = \infty$, $a_k$ tends to a Poisson process

$$a_k \equiv P[A = k] = \frac{\rho^k e^{-\rho}}{k!} \qquad k = 0, 1, 2, \ldots N = \infty \qquad (5.3)$$

with probability generating function

$$A(z) = \sum_{k=0}^{N} z^i P[A = k] = e^{-\rho(1-z)} \qquad (5.4)$$

Let the number of cells in the queue at the end of the m<sup>th</sup> cell slot be $Q_m$ and the number of cell arrivals during the m<sup>th</sup> cell slot be $A_m$. The queue length $Q_m$ is

$$Q_m = \min\{\max(0, Q_{m-1} + A_m - 1\}$$

When $Q_{m-1} = 0$ and $A_m > 0$, one of the arriving cells will be transmitted during the $m^{th}$ cell slot without delay. For finite N and buffer size, $Q_m$ can be determined from a finite-state discrete time Markov chain [100] with the state transition diagram illustrated in figure 5.8.

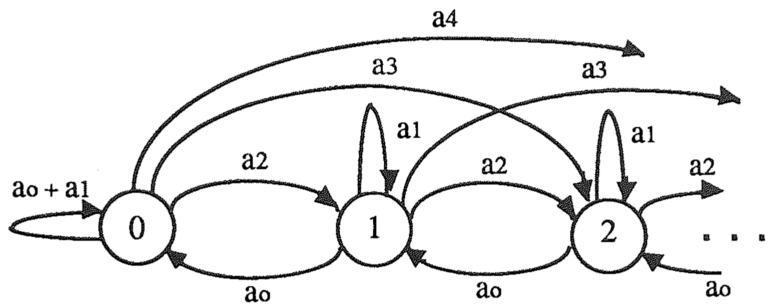



Figure 5.8 State transition diagram

The probability generating function for the steady state queue size is

$$Q(z) = \frac{(1 - \rho)(1 - z)}{A(z) - z} \tag{5.5}$$

Substituting equation 5.2 into 5.5 gives

$$Q(z) = \frac{(1 - \rho)(1 - z)}{\left(1 - \frac{\rho}{N} - z\frac{\rho}{N}\right)^N - z} \tag{5.6}$$

Taking the limit as N tends to $\infty$ on both sides of equation 5.6 we have

$$\lim_{N\to\infty} Q(z) = \frac{(1 - \rho)(1 - z)}{e^{-\rho(1 - z)} - z} \tag{5.7}$$

which corresponds to the probability generating function for the steady state queue size of an M/D/1 queue. Expanding (9) in a Maclaurin series gives the asymptotic ($N\to\infty$) queue size probabilities

$$P(Q = 0) = (1 - \rho)e^{\rho}$$

$$P(Q = 1) = (1 - \rho)e^{\rho}(e^{\rho} - 1 - \rho)$$

$$P(Q = n) = (1 - \rho)\sum_{j=1}^{n+1} (-1)^{n+1-j} e^{j\rho}\left[\frac{(j\rho)^{n+1-j}}{(n+1-j)!} + \frac{(j\rho)^{n-j}}{(n-j)!}\right] \qquad (n > 1)$$

The probability of cell loss is given by

$$P(\text{cell loss}) = 1 - P(Q = n) \qquad n = 0, 1, \ldots S$$

Figure 5.9 shows the probability of cell loss according to the M/D/1-S queue approximation for various loads. At an offered load of 80 percent and buffer size of 50 cells, the probability of cell loss is less than $10^{-9}$. In practice, with bandwidth allocation to bursty sources according to the bufferless fluid flow model the actual cell loss will be even less.

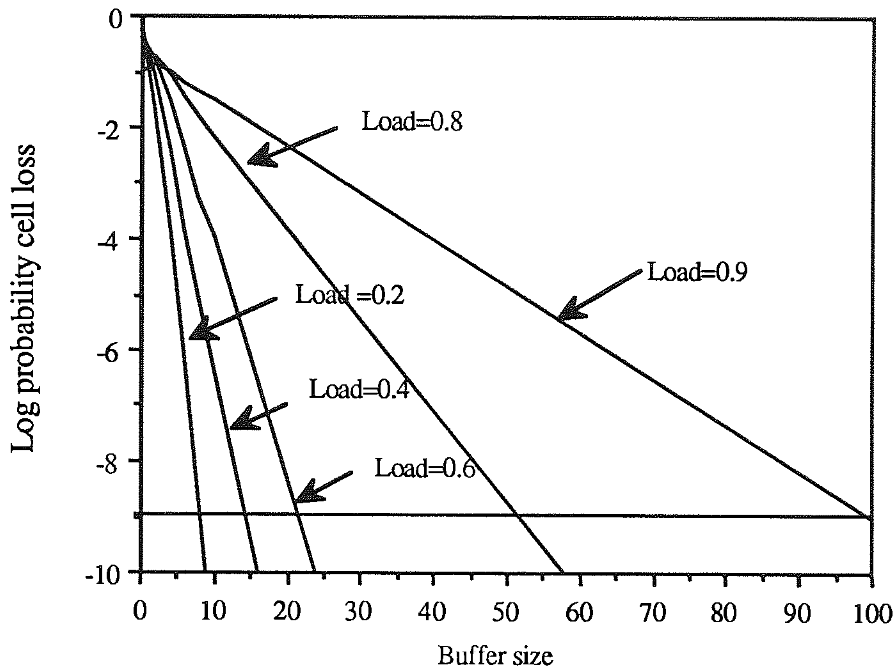



Figure 5.9 The cell loss probability as a function of buffer size for various offered load based on M/D/1 queue

The input to the buffer 1 is limited to a maximum load of 0.8 in order that class 2 bearer service may have some access to the network. This prevents bearer service 1 from dominating the network while limiting the cell level congestion to less than $10^{-9}$. In practice, class 1 traffic may not require all its capacity. Class 2 traffic can therefore take advantage of the unused capacity. Supposing that the total load of class 1 traffic consisting of CBR and voice traffic is 0.6, the maximum queue length used is 10 cell slots.

If the threshold queue $S_T$ is fixed to 10, then cells from bearer service class 2 on transit in buffer $S_2$ will be transfered to $S_1$ whenever the total free cell slots in buffer $S_1$ is greater than 40 cell slots. This allows class 1 cells of the particular load to have full access to the buffer $S_1$. The maximum queueing delay and jitter of class 1 traffic is thus limited to 50 cell slots. The threshold queue $S_T$ can be dynamically changed depending on the total load (< 0.8) of class 1 traffic.

### 5.3.4 Dimensioning buffer size S2

Class 2 traffic mainly consists of data traffic with a mean bit rate allocation. The buffer S2 is purposely chosen to be long to take advantage of statistical multiplexing among data sources at the expense of delay. All data traffic will have to go through the enforcement function described in chapter 4 before it is imposed onto buffer S2. The worst case traffic (see section 4.3.2.1) is considered when dimensioning the buffer size S2. In fact, the size of S2 depends very much on the maximum end-to-end delay limit of data service. In practice, data cell delays of up to several seconds may be acceptable, depending on the application.

Consider worst case data traffic of declared peak and mean bit rate of 10 Mb/s and 2 Mb/s. It was shown in chapter 4 that in order to keep the cell loss probability negligible when multiplexing the worst case traffic with mean bit rate bandwidth allocation, the burst level congestion must be totally suppressed by reserving a large buffer. Supposing that there is no load in bearer service class 1 traffic, then data traffic can make full use of the available total load in the network.

If the above data traffic has a mean burst length of 100 cells, then the buffer must allocate to each data source buffer cell slots equivalent to the mean burst length. Since the total load acceptable into the network is 0.83 (number of information bytes for data, i.e. 44 divided by the total length of the cell i.e., 53 bytes), then the maximum number of data sources allowed in the network is 62. Hence the buffer must reserve 6200 cell slots, which implies a maximum queueing delay of about 17.6 ms.

This is a conservative way of dimensioning the buffer size since in practice data will very rarely use up the whole network capacity. Figure 5.10 shows the average queueing delay experienced by data traffic of different mean burst length. Obviously, for a particular offered load, the average waiting time is higher for data sources with longer mean burst length.

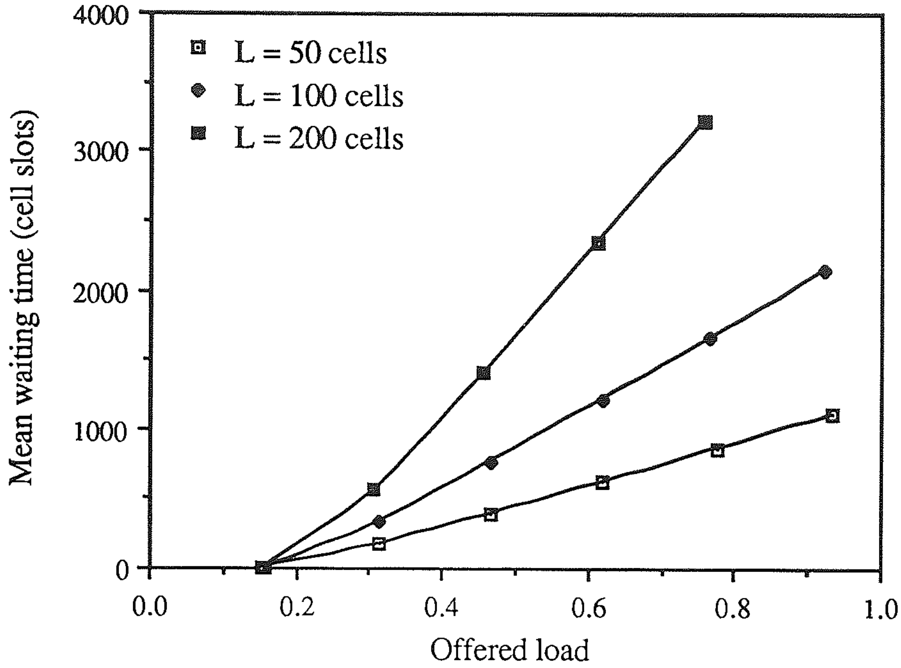



Figure 5.10 Average waiting time of data cells as a function of offered load (Bp = 10 Mb/s, Bm = 2 Mb/s and mean burst length L = 50, 100 and 200 cells)

## 5.4 Simulation results on the PBS with threshold

Assume an infinite buffer size $S_2$. Class 1 bearer service consisting of CBR and voice sources carries a fixed traffic load of 0.6 and uses up a maximum buffer size $S_1$ of 10 cell slots. A variable number of data sources are accepted into the network and the delays incurred are observed based on the queue length distribution. In figure 5.11, two types of data are considered. As the offered load is increased, the maximum and mean queue lengths increase proportionally. When the total offered load exceeds the link capacity, a sharp queue rise is observed. For a maximum of 93 percent of total offered load, the required buffer size is nearly 2500 cell slots, which is equivalent to 7.0 ms of maximum queueing delay (per node).

The maximum queueing delay which gives the delay jitter is not very important in data transfer, the mean queueing time or the average waiting time in the queue has a more meaningful value. However, in this case the cell average waiting time is only slightly less (on average about 0.5 ms) than the maximum queueing delay as shown in table 5.2. This is mainly due to cells having to wait in the buffer $S_2$ for some time as they are only transferred to buffer $S_1$ when the number of free buffer slots in $S_1$ is greater than $S_1 - S_T$ (> 40 ).

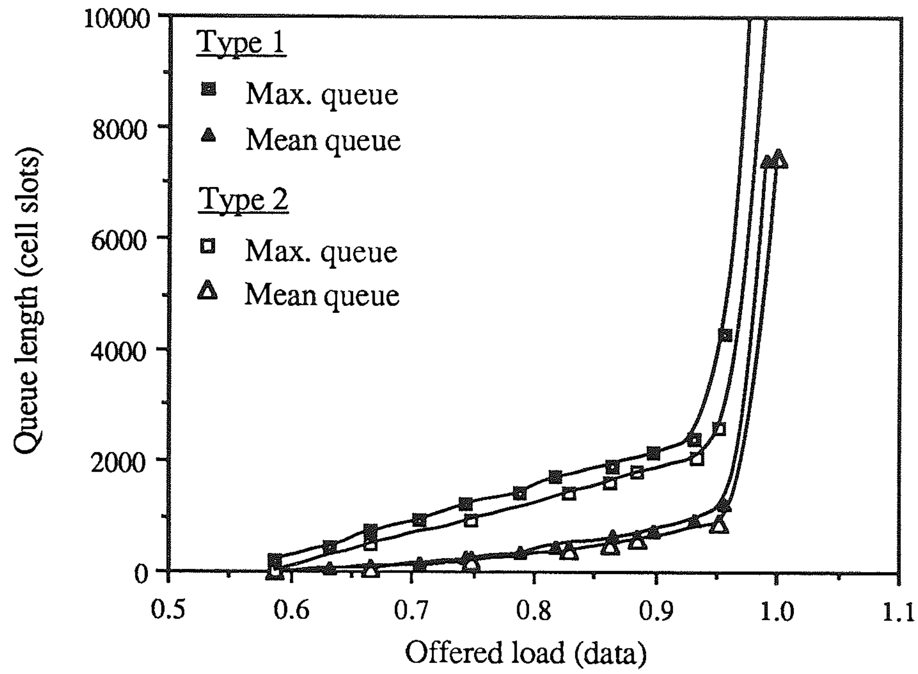



Figure 5.11 Queue length against offered data load for two types of data
Class 1 load=0.6, Type1: Bp = 10 Mb/s, Bm = 2 Mb/s, L=100 cells
Type2: Bp = 10 Mb/s, Bm = 1 Mb/s, L = 100 cells

Table 5.2 Data cell waiting time in queue (Bp=10 Mb/s, Bm=1 Mb/s, L=100)

| Total offered load | Maximum waiting time (ms) | Average waiting time (ms) |
|---|---|---|
| 0.63 | 1.27 | 0.94 |
| 0.67 | 2.01 | 1.58 |
| 0.74 | 3.41 | 2.78 |
| 0.82 | 4.81 | 4.28 |
| 0.90 | 6.01 | 5.53 |

### 5.4.1 Performance comparison

The performance of the proposed PBS queueing discipline is now compared with other methods of priority queueing, in particular, separate route with a) priority queue control and b) static bandwidth division. The separate route is chosen for comparison since it is the simplest space priority mechanism.

The separate route with a priority queue control mechanism described in [102, 104] differs from the method considered here in that three separate buffers were considered. This mechanism has only two separate buffers $S_1$ and $S_2$ and is meant for two classes of bearer service. The priority of the queue is determined by the number of cells that can be served successfully. In this method, the server polls the high priority queue $S_1$ first before serving the lower priority queue according to a limiting service discipline. The service method is expressed in terms of index of priority as M:N, where M and N determines the maximum number of cells that can be served successively in queue $S_1$ and $S_2$ respectively. If M is large, then $S_1$ will be served exhaustively before $S_2$ has its turn. Figure 5.12 shows the processing algorithm for the separate route with M:N limiting service discipline (SP-M:N).

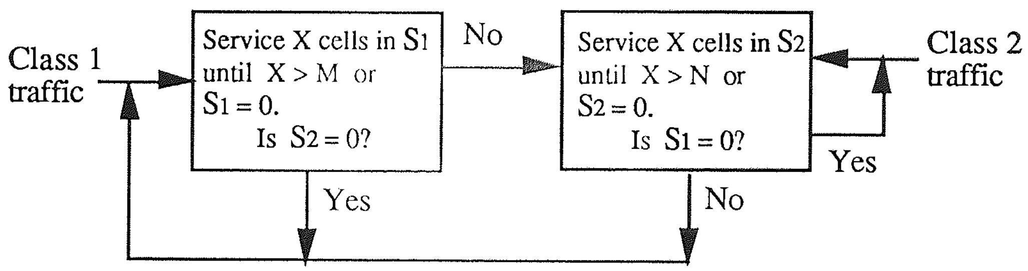



Figure 5.12 Processing algorithm for M:N limiting service discipline

The separate route with bandwidth division method (SP-BWD) allocates the semi-fixed bandwidths for each bearer service beforehand, for example at the time of the call admission. The semi-fixed bandwidth may correspond to a virtual path or virtual link capacity used by cells from one medium.

The performance of the priority mechanisms described above is compared in terms of the delay experienced by data cells and maximum acceptable load and the network utilisation. In all cases, the cell loss probability for bearer service class 1 is kept below $10^{-9}$ by reserving an appropriate buffer size for the specified acceptable load. Figure 5.13 compares the average cell waiting time for the three different queueing strategies as the number of data sources is increased.

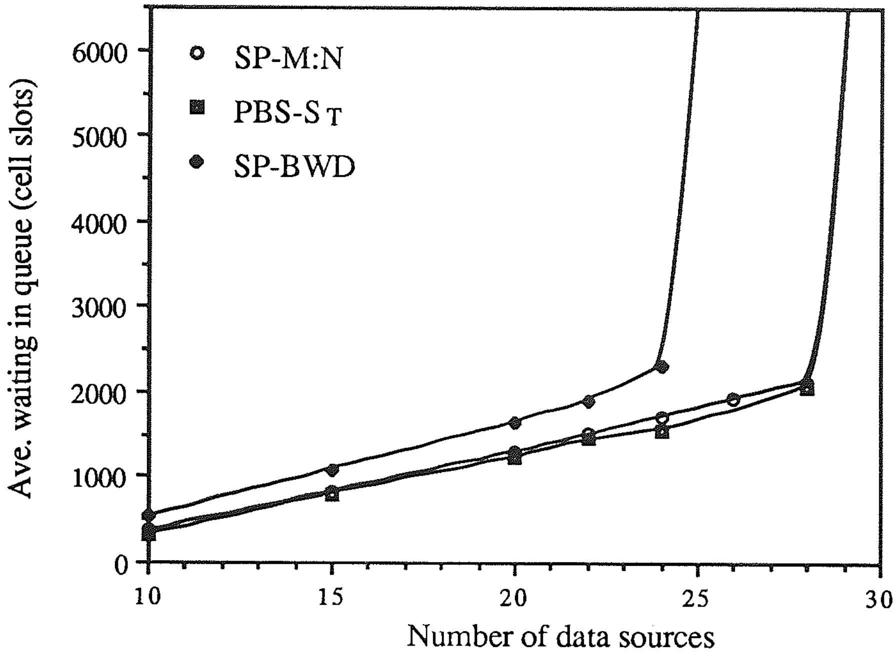



Figure 5.13 Average cell waiting time in queue for three different queueing strategies (Class 1 load = 0.6, Class 2 - Bp = 10 Mb/s, Bm = 2 Mb/s, L =100 cells)

The average cell delay incurred in both PBS-ST and SP-3:1 is almost the same as shown in figure 5.13. For both queueing disciplines, the network utilisation is nearly 0.95 when the maximum data load acceptable into the network experienced data cell average queueing delay of about 2500 cell slots (7 ms). In practice, both queueing strategies are easy to implement but the SP-M:N mechanism requires more processing to regularly check the two queues and change the index of priority to prevent the high priority queue from dominating the network. On the other hand, the PBS-ST mechanism requires only the threshold value ST to be observed regularly and this value can be dynamically changed since its relationship with the load is almost linear up to a certain load.

The bandwidth division method, however, experienced larger delay for the same number of data sources into the network (see figure 5.13). The maximum network utilisation achieved as shown figure 5.14 is less than the other two methods since the fixed allocated bandwidth to the high priority cells cannot be used by the lower priority cells when it is free to be used.

Furthermore, this method is not flexible since changing the bandwidth requires complicated calculations and control actions [101].

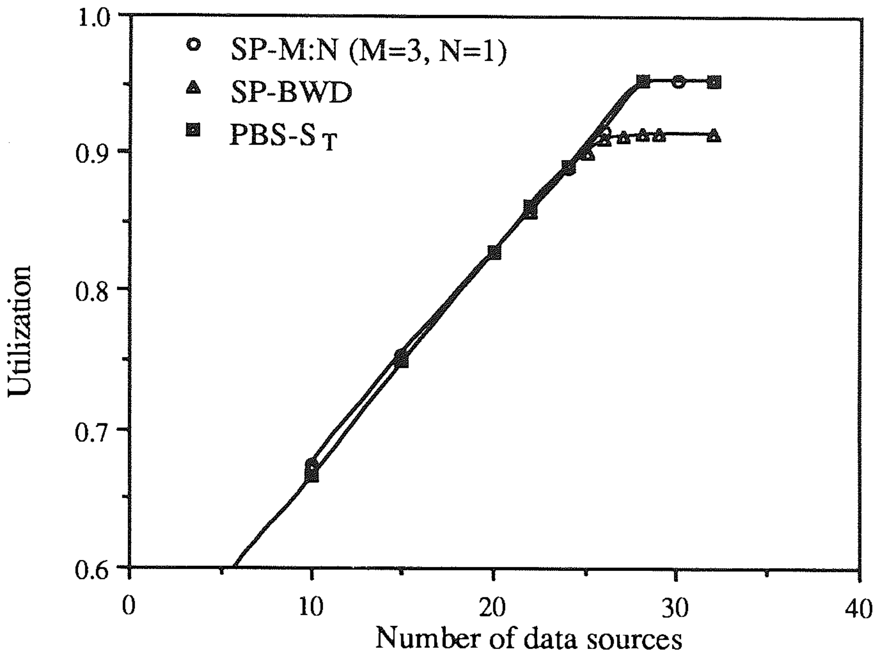



Figure 5.14 Network utilisation vs number of data sources
(Class1 load =0.6, Class 2 : Bp = 10 Mb/s, Bm = 2 Mb/s, L = 100 cells)

## 5.5 Summary

In this chapter two bearer services are considered for two classes of traffic, one requiring a stringent cell delay limit and the other requiring a strict loss probability. The proposed queueing strategy (PBS-ST) is based on a partial buffer sharing mechanism with the additional property of having a threshold load. The mechanism ensures cell loss probability at cell level for class 1 traffic to be less than $10^{-9}$ and almost negligible cell loss for class 2 traffic.

The maximum queueing delay per node for class 1 traffic is limited to 50 cell slots which restricts the maximum delay jitter to about 142 us. On the other hand, the delay experienced by class 2 data traffic is large. However, this delay can be reduced by either accepting less load into the network, which implies less network utilisation, or by reducing the window size of the enforced data. Further work on the latter option is described in the next chapter.

The performance comparison shows that PBS-ST works as good as the other two methods of queueing discipline with the advantage of being simpler and more flexible.

# CHAPTER 6

## CONGESTION AND FLOW CONTROL FOR DATA TRAFFIC

### 6.1 Review of traffic congestion control in packet switched networks

Congestion occurs when part of the network (subnet) begins to lose packets due to too many packets being dumped into the subnet as traffic increases, until the node may no longer be able to cope. Congestion control must therefore be invoked to make sure that the subnet is able to carry the offered traffic. Flow control, in contrast, is concerned with the point-to-point traffic flow between a given sender and a receiver. It ensures that a fast sender does not continually transmit data faster than the receiver can absorb it even if the transmission is error-free. Flow control mechanisms normally involve some direct feedback from the receiver to the sender to inform the sender of the state of the network at the other end.

In conventional packet switched networks (eg. ARPANET), flow control mechanisms have been used to eliminate congestion. Each user VC carrying bursty traffic is allowed to accomodate more VC load than its capacity so as to utilise the statistical fluctuations of each VC traffic and thus to save bandwidth. However, if all the VCs transmit at their peak rates simultaneously, the nodes may become congested. An example of a flow control scheme used to reduce congestion control in conventional packet switched network is the window flow control [28, 129]. The window flow control schemes keeps track of the number of packets pending in the network for each VC using some acknowlegdement procedures.

The preventive congestion control strategy described in chapter 4 is an open loop scheme where the control action is taken locally irrespective of the state of the global network. Thus the excess traffic will be dropped upon the onset of congestion. For data applications, lost information will have to be retransmitted by the traffic sources which may have the effect of reducing the network throughput. In order to reduce the amount of retransmitted traffic, it is desirable not to allow new cells into the network unless the network is able to carry them to their destination with some certainty. This requires the traffic sources to have some means of acquiring knowledge on the current state of the network and adjust their rates for better resource utilisation which leads to a closed loop control scheme.

It has been argued in chapter 4 that the window flow control scheme is not appropriate for a high speed broadband network such as in an ATM network environment since the majority of the traffic, such and voice and video is not flow-controllable. The increase in network speed also poses critical constraint on the complexity of the control mechanism at the link and network layers. However, the window flow control scheme as a form of closed loop scheme could be modified and used in controlling congestion for flow-cotrollable data services [114, 116, 118, 119]. Ramamurthy et. al [118, 119] has defined and an adaptive end-to-end window protocol whose complexity is concentrated at the user-network interfaces. The traffic sources can be informed about the network congestion status either through implicit or explicit feedback notification.

In this chapter, a modified window flow control scheme for data services of category 2 is proposed and its performance is investigated. The proposed flow control scheme is a form of congestion control for traffic types that require stringent cell loss probability but can tolerate delay, such as file transfer, LAN-to-LAN interconnection, image and document retrievals, etc. Unlike in [116], fairness is achieved by serving VCs according to priority levels based on the proposed partial buffer sharing with threshold as described in chapter 5. Fairness among category 2 users is maintained by providing distributed source control.

## 6.2 Window flow control

It is assumed that a window scheme proposed in chapter 4 is used to flow control each VC. The window size is the amount of data that a VC is allowed to have in transit at any given time. Note that although different notations are used to name the variables such as the window size and window interval, they bear the same meaning as those in chapter 4.

If the transmitter and receiver are linked by a transmission path of speed $C_L$ with a round trip propagation delay of $T_R$, then in order to maintain continuous transmission on an otherwise idle path, the window size must be at least as large as the full-speed round-trip window $W_R$ such that

$$W_R = \alpha C_L T_R, \qquad \text{where } 0 < \alpha < 1.$$

In principle, if a VC has a window of a given size and buffer overflow is to be rigorously avoided, buffer space adequate to store the entire window must be available at every queueing point (see chapter 4, section 4.3.2.1). On a lightly loaded network, a VC usually does not occupy much buffer space, and there can be statistical sharing of buffers among different users with negligible probability of overflow. On a congested network, the probability of shared buffer overflow rises. Losses due to overflow may trigger retransmission which can lead to instability. The following work is initiated from the well known fact that congestion instability due to data loss does not occur in a VC if windows are used and a full window of buffer memory is preallocated to each VC at each node.

Consider a source having several windows worth of information negotiating to transmit at an average throughput. The average throughput rate must be agreed by the enforcement function. If the originating source negotiates with the network node for an end-to-end window of $W_R$, and $T_R$ is the average round trip delay, then the maximum throughput attained by the connection is $W_R/T_R$. The node must reserve $W_R$ buffers for the connection.

Transmitting at a rate higher than the average throughput would not reduce the total transmission time since the source has to wait a round trip delay $T_R$ before transmitting the next window. On the other hand, if the source is allowed to transmit at its average throughput (mean bit rate), then the total transmission time would at most increase by one round trip delay. The number of active sources that are allowed to transmit simultaneously is $N_{active} = \alpha C_L/Bm$ and for $Bm << \alpha C_L$, then the number of $N_{active}$ sources can be very large.

In a wide area network with a high transmission link rate, the round trip delay is mainly dominated by the propagation delay and therefore high throughput can be obtained just by increasing the end-to-end window size. However, if there are N such connections using the given link then the node must allocate $NW_R$ memory in the node buffer. For example, if $T_R$ is 50 ms, a VC requiring 34 Mb/s maximum throughput would need a buffer memory of 4010 cells at every node. With three such VCs in a 150 Mb/s system, then the total required buffer memory would be impractical. In the following section, means of reducing the buffer requirement are discussed.

## 6.3 Input rate based control

The proposed window flow control scheme can be used as an input rate based control, or used in conjunction with higher layer end-to-end control as in [119]. It is implemented at the network access layer as a feed forward control monitored by individual users. The congestion control function inside the network is carried out with respect to the time constants of the nodes. These are determined by the delay requirements of the traffic types and the amount of memory or buffering S available inside the network node.

In this scheme the problem of large buffer memory is minimised by adjusting the window size so that each user has a window large enough to support a fair share of network bandwidth but not necessarily a full round-trip window. Assume that $T_a$ is in the order of the network time constant. If each source is allowed to transmit $W_a$ cells during every non-overlapping period of $T_a$, such that

$$\frac{W_a}{T_a} = \frac{W_R}{T_R} = \bar{\delta}$$

where $\bar{\delta}$ is the average end-to-end throughput,

then this ensures that the arrival rate averages over the two intervals $T_a$ and $T_R$ are the same, although $T_a$ is much smaller than the average round trip delay $T_R$. $T_a$ acts as the smoothing interval since the total number of cells in an end-to-end window $W_R$ is equally spread out into several intervals of $T_a$. Within the interval $T_a$, the source is allowed to transmit $W_a$ cells back-to-back meaning that the arrival epoch of these $W_a$ cells need not be equally spaced over the interval $T_a$. Such a scheme is completely feed forward and can co-exist with higher level protocols like end-to-end protocols as in [115, 117].

### 6.3.1 Window sizing

Once $T_a$ is fixed, the minimum buffer size at the output queue must be chosen such that the time to empty all the cells in the buffer is at least as large as the interval $T_a$. Therefore, the size of the buffer $S_2$ at each node should be

$$S_2 = T_a \, \alpha \, C_L$$

Assuming that each $VC_i$ (i = 1, 2,..,N) has an end-to-end window of $W_{Ri}$ with the round-trip delay $T_{Ri}$, then the average bandwidth for $VC_i$ is $W_{Ri}/T_{Ri}$ cells/sec. However, if the smoothing interval $T_{ai}$ is used, then the desired window must be dimensioned such that

$$W_{ai} = W_{Ri} \frac{T_a}{T_{Ri}}$$

In order to ensure that the probability of cell loss due to buffer overflow is small, two constraints must be satisfied:

1) the sum of the average bandwidth requirement for all VCs using the link must be less than the available link capacity such that

$$\sum \bar{\delta_i} = \sum \frac{W_{ai}}{T_{ai}} = \sum \frac{W_{Ri}}{T_{Ri}} < \alpha C_L$$

where $0 < \alpha < 1$

2) the sum of the maximum number of active windows must be less than the available buffer size at the node, that is

$$\sum W_{ai} < T_a \alpha C_L = S_2$$

Sometimes the exact round trip propagation delay is not known. In such a case, a conservative estimate of the $T_{Ri}$ must be made. Although this may lead to a larger window $W_a$, the average throughput will still be maintained. Thus, overestimating the smoothing window size will assure each VC with its expected average throughput while the number of VCs that can be active simultaneously will be reduced.

An accurate estimation of the round trip propagation delay is very critical when the physical path length is large, since the propagation delays would dominate over queueing delays. The largest national connection recommended by CCITT in recommendation G.104 is less than 2500 km with maximum round trip delay of less than 25 ms. If a private WAN is assumed to run over such a distance, it should experience a similar maximum round trip delay.

### 6.3.2 The effect of queueing delay

It was assumed that the mean round trip propagation delay $T_{Ri}$ is in the order of tens of ms. A call with peak bit rate of Bp Mb/s and mean bit rate of Bm Mb/s negotiates at the time of call setup for an average throughput equal to its mean bit rate. The initial parameters $W_{Ri}$ and $T_{ai}$ of the $VC_i$ are computed at the access node. The access node will change the window size to $W_{ai}$ depending on the available bandwidth in the access node and convey the reservation request to the other nodes along the chosen path of the $VC_i$. The intermediate nodes would accept the $VC_i$ if $W_{ai}$ can be guaranteed.

The effect of the queueing delay at a node is investigated by varying the smoothing interval and hence the corresponding window size. Figure 6.1 shows the relative queueing delay as the window size is reduced with respect to the end-to-end window of, in this case, 35.2 ms. As the smoothing interval decreases, the corresponding window size also decreases, resulting in a smaller batch of cell arrivals and thus a smaller variance in performance in terms of delays. The smaller interval $T_a$ effectively smooths out the incoming traffic and thereby reduces the mean and the variance of cell waiting time in the queue which leads to a lower buffer requirement.

Figure 6.2 shows the mean cell waiting time in the buffer as the number of data sources are increased for different window size. The queueing delay experienced by data traffic of a particular load is reduced as the window size gets smaller. This effect is greater for larger data load (number of data sources). The reduction in the queueing delay and hence the buffer requirement does not affect the flow of class1 bearer service traffic since it has higher priority in the partial buffer sharing mechanism with threshold as described in chapter 5, section 5.3.

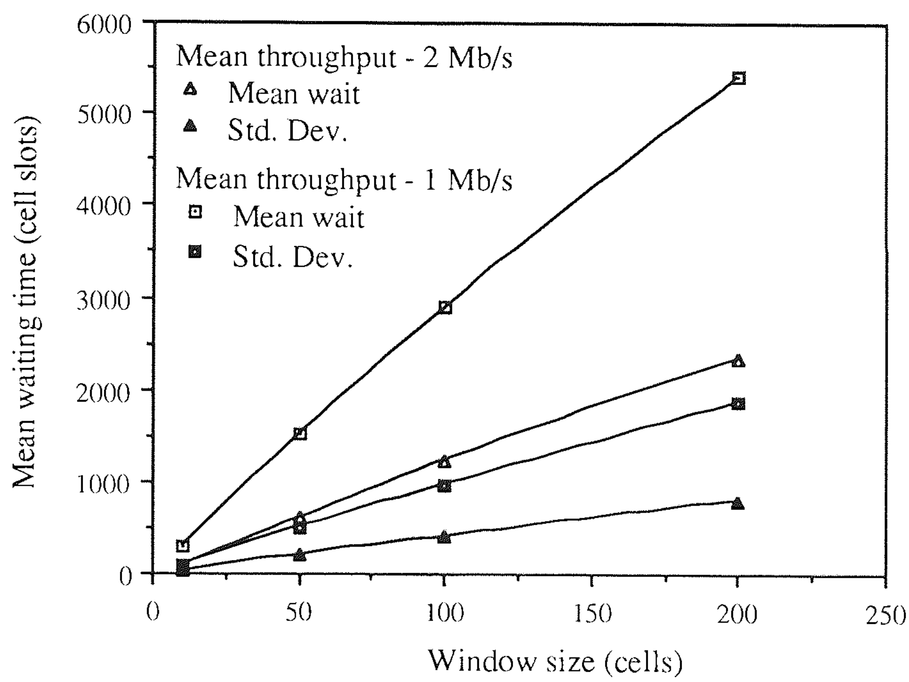



Figure 6.1 Mean queueing delay against window size
(Type 1: $Bp_i = 10$ Mb/s, $Bm_i = 2$ Mb/s, $\bar{\delta} = 2$ Mb/s)
(Type 2: $Bp_i = 10$ Mb/s, $Bm_i = 1$ Mb/s, $\bar{\delta} = 1$ Mb/s)

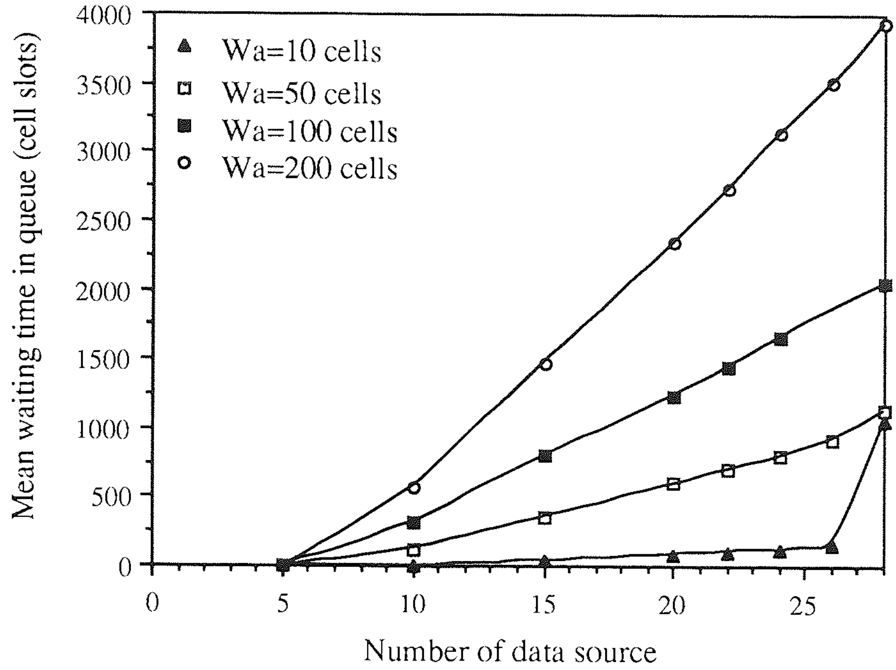



Figure 6.2 Mean cell queueing time as the window size is varied for variable number of data sources (Bp=10Mb/s, Bm=2Mb/s, L=100 cells)

## 6.4 Dynamic window flow control on congestion

The window flow control mechanism has been extended to operate a closed loop congestion control scheme based on explicit feedback notification. With explicit notification, the network node determines its congestion state by monitoring its utilisation or the length of its associated queue over some observation interval [115]. We consider a network node model shown in figure 6.3 with N identical data sources fed into the network access node.

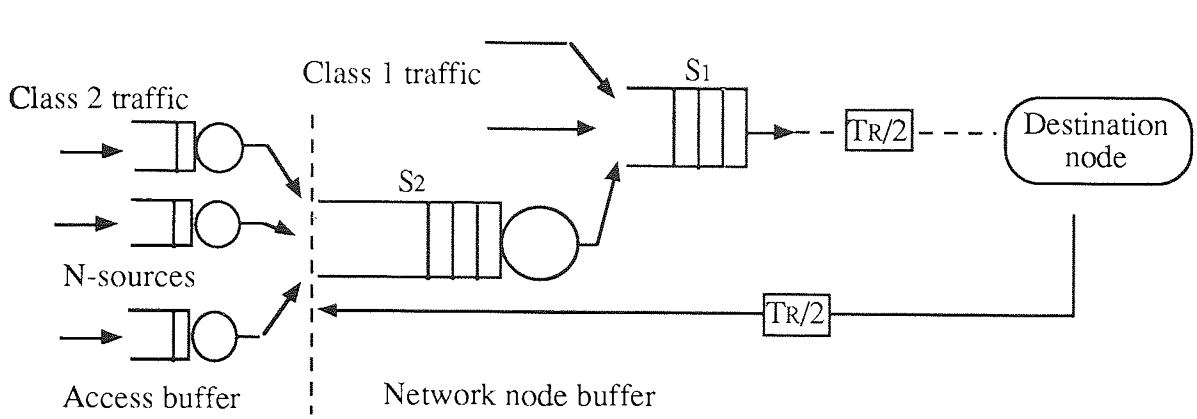



Figure 6.3 Network node model for window control congestion avoidance

The congestion control approach here is basically a congestion avoidance strategy. Besides allowing a certain number of cells in a window interval, the network is able to signal the end points that congestion is about to occur. The window flow control is characterised by the parameters $W_e$, the default or declared window size, $W_a$, the effective window size and $W_R$ the end-to-end window size. Cells generated by each source are admitted into the network queue as long as the total number of cells is within the declared window size. The access queue will hold those cells that are not instantaneously admitted into the network upon their generation. The access buffers are assumed to be of infinite length and therefore cell loss due to overflow at the access queue does not occur.

Cells that are admitted into the network may have to flow over several nodes which may be congested. The node that is congested sets on a congestion indication field in the cell that is flowing towards the destination. This is in contrast to the source quench scheme [28] where additional cells are selectively sent to sources causing congestion. At the destination this information is copied onto the header of the acknowledgement cell which will be transmitted to the source. Thus each cell is assumed to contain a field in its header known as the Explicit Congestion Notification (ECN) field to convey congestion status. This field is made up of only one bit in the header. On receiving the acknowledgement, the source is required to adjust its traffic on the network based on the interpretation of the congestion indication by adjusting its window size.

### 6.4.1 Feedback control system

The feedback control system has two sets of policies for controlling the traffic placed in the network. One set of policies is controlled by the network node and the other by the source or the user. Both of these policies do not necessarily have to be synchronised with each other. The network node policy specifically consists of congestion detection and feedback filter while the source node policies include the decision fuction and the decision frequency.

#### 6.4.1.1 Network node policies

Congeston detection

The congestion detection mechanism detects the congestion status and sets the ECN bit in the cell header based on the state of the average queue length at the node. The mechanism differs slightly from the approach reported in [114]. In their work the dynamic window was used to prevent congestion collapse for frame relay networks with transmission rates of 1.5 Mb/s and 45 Mb/s . In this work, the approach is extended to transmission of cells at higher transmission speeds of up to 150 Mb/s.

The average queue length $E(S_2)$ is compared with the ECN threshold queue Tq every one round trip delay (assuming that the round trip delay for every VC is the same). If the $E(S_2)$ is greater than the Tq, a bit in the ECN field is set to one, otherwise the bit will be left with its current value. The information will travel to its destination in the cell header. At the destination this bit will be transmitted to the source after encountering a total delay of one round trip delay $T_R$.

Feedback filter

The average queue length $E(S_2)$ is used as a congestion indication. Also, in contrast to [115], the duration over which the average queue length is calculated is a fixed interval of a round trip delay $T_R$ as shown in figure 6.4. An interval close to one round trip delay was chosen since [116, 117] found that the feedback signal generated and passed to the users is consistent and results in the fair allocation of the node and link resources. The computation of the average queue length $E(S_2)$ depends on the total queue length over and interval $T_R$;

$$E(S_2) = \frac{Q_1 + Q_2}{T_R}$$

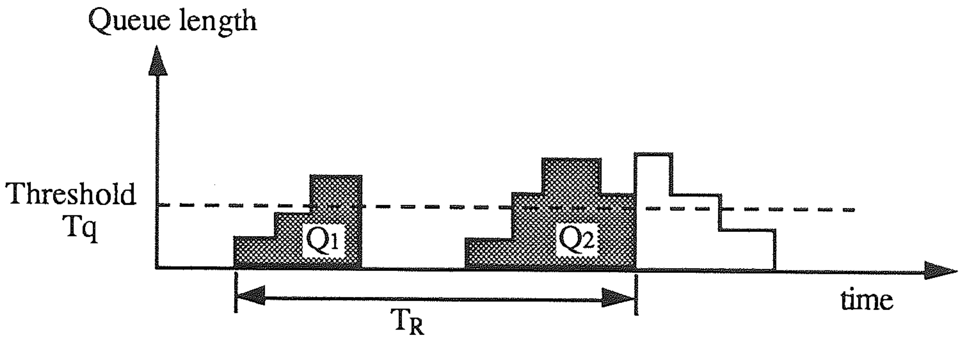



Figure 6.4 The average queue length as congestion indication

#### 6.4.1.2 Source node policies

Decision function

On receiving the congestion notification, the source will react according to the following algorithm;

> *{The node processor check the average queue length every $T_R$}*
> **If** *( ( $E(S_2)$ > Threshold queue) then*
>     *{Deal with congestion}*
>     *Reduce the window size by factor R*
> **Else**
>     *{Open window exponentially }*
>     *Increase window size by 1 cell*
> **End If**

The offered load is adjusted in terms of the window size. At the onset of congestion, the window size is multiplicatively reduced by a small margin say by a factor R. On the other hand, if the network is not congested, the window size is increased additively by one cell every window's turn. Basically, the change in the window size is governed by

$$1 < W_a < W_R$$

and $W_a$ is the default window size.

The additive increase and mulitiplicative decrease was adopted since it will quickly converge the network congestion state to a stable state and provide the fairest treatment to the users [115, 117].

Decision frequency

Since the node queue is examined every interval of one round trip delay, the user will receive an acknowledgement every such interval. Each acknowledgement indicates the status of the node queue of the previous interval. Therefore, the decision whether to increase or decrease the window size is updated every end-to-end window interval. It is assumed that the communication of the feedback information in the forward and backward directions is error free.

### 6.4.2 Determining the threshold queue

The performance of the node queue has been considered for different values of the system parameters such as the ECN threshold queue, the window size and the window reduction factor. The performance measures considered here are the probability of overflow, the mean queue length of the network queue and the mean delay to transfer data cells from the source to the destination.

The ECN threshold queue is an important parameter since it determines the congestion state of the network before any reaction is taken by the source. In the following simulation work it is assumed that a class 1 traffic source is kept constant at an offered load of 0.6. Class 2 traffic consists of a homogeneous data source input (Bp = 10 Mb/s, Bm = 2Mb/s, N = 28) with end-to-end window size $W_R$ of 100 cells requiring a minimum throughput equal to its average mean bit rate. Table 6.1 shows the effect of varying the ECN threshold queue on the network node queue for a window reduction factor of 1, that is, reverting the effective window size back to its end-to-end window value on the onset of congestion.

The results show that, as the ECN threshold was increased, the network node utilisation remained unchanged until the threshold queue was greater than 1200 cell slots. Cell loss due to buffer overflow was observed for a large ECN threshold due to less frequent window throttling. The obvious observation here is that the threshold queue must be dimensioned arbitrarily small for excellent network queue performance in terms of cell loss probability. The small ECN threshold queue causes frequent congestion acknowledgements to the source to reduce the window size and hence smaller average network node queue length and mean cell queueing time.

Table 6.1 Effect of increasing the the ECN threshold queue

| ECN Threshold (cell slots) | Network utilisation | Probability of cell overflow | Mean queue length (cell slots) | Mean wait in queue (ms) |
|---|---|---|---|---|
| 400 | 0.952 | - | 884.7 | 5.79 |
| 800 | 0.952 | - | 1006.6 | 6.57 |
| 1200 | 0.952 | - | 1203.9 | 7.85 |
| 1600 | 0.968 | $2.02 \times 10^{-5}$ | 1400.8 | 9.94 |
| 2000 | 0.979 | $3.88 \times 10^{-4}$ | 1768.0 | 11.1 |

The above simulation results represent the performance of the dynamic window control with ECN feedback without reducing the window size less than the end-to-end window. For this particular case, the ECN threshold must be set to less than 1400 cell slots for almost congestion free network performance. The determination of the ECN threshold in this range depends on the maximum end-to-end delay that the data source can tolerate.

Figure 6.5 shows the mean total delay observed by data cells as the threshold varies. It is found that the improvement of the queue length behaviour at small threshold queue has been achieved at the expense of the end-to-end data delay. This is due to the fact that packets have to queue more often at the access node as the effective window size is throttled back to its default value.

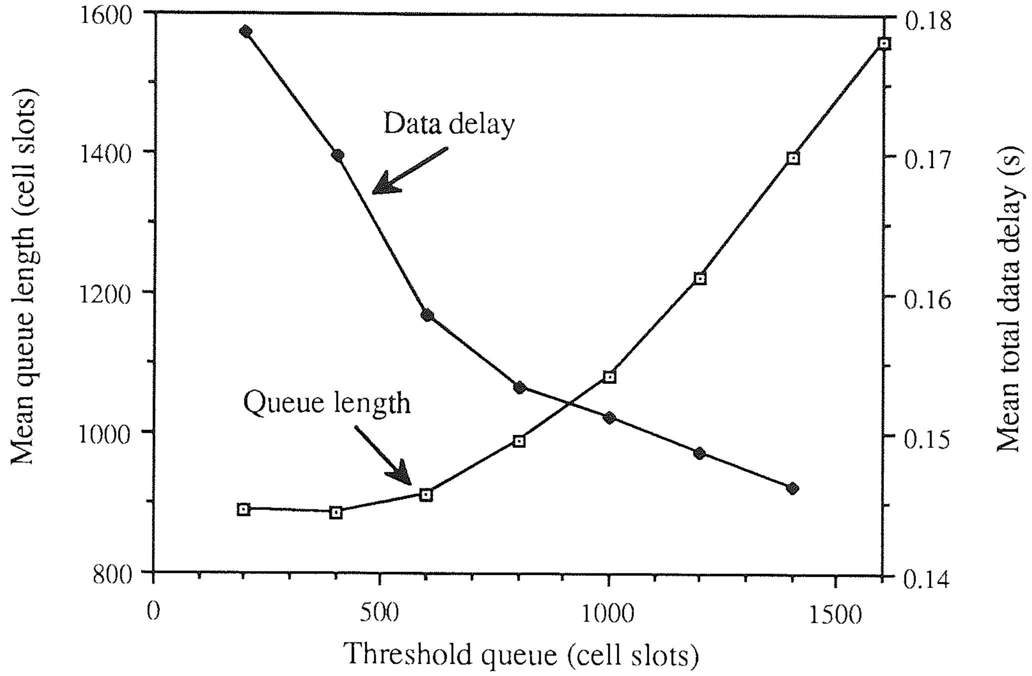



Figure 6.5 Variation of mean queue length and total data delay with ECN threshold queue

### 6.4.3 Effect of introducing the reduction factor

The effect on the performance of the network node of reducing the window size by a reduction factor in terms of cell loss and end-to-end data delay is depicted in figure 6.6 and figure 6.7 respectively for a network node with finite buffer size of one round trip interval (ie. 6220 cell slots). With the reduction factor on, the window size is reduced by a factor R when the node receives an acknowledgement. The value of the reduction factor R should be in the range $0 < R < 1$. If $R = 1$, then the effective window size is reduced to its default value $W_R$.

It was observed that a small reduction factor achieves better performance in terms of cell loss especially when the ECN threshold is small. However, this achievement has to be paid for by larger end-to-end delay, as shown in figure 6.7. The smaller reduction factor shortened the window size, which translated to throttling a larger total load and hence more frequent queueing at the access node, and therefore less network utilisation as shown in figure 6.8.

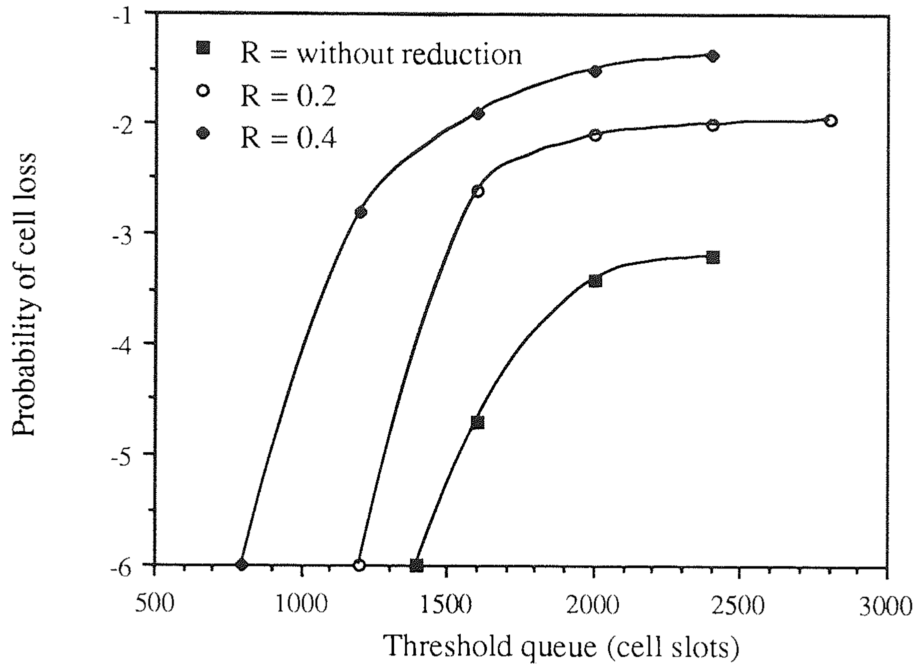



Figure 6.6 Probability of cell loss due to buffer overflow as the threshold queue is increased for different window reduction value.

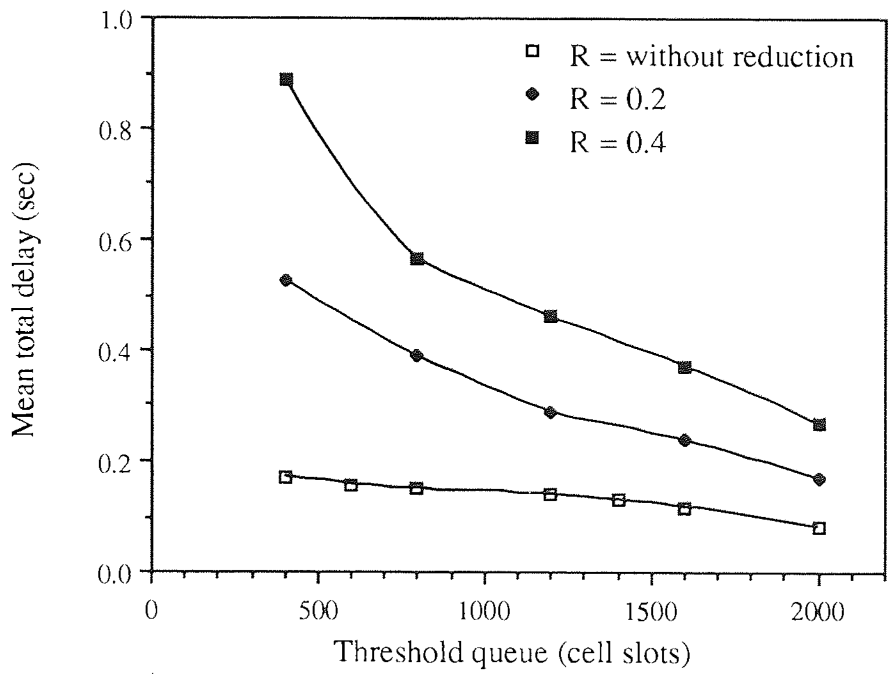



Figure 6.7 Variation of mean total data cell delay as the ECN threshold varies for different window reduction interval.

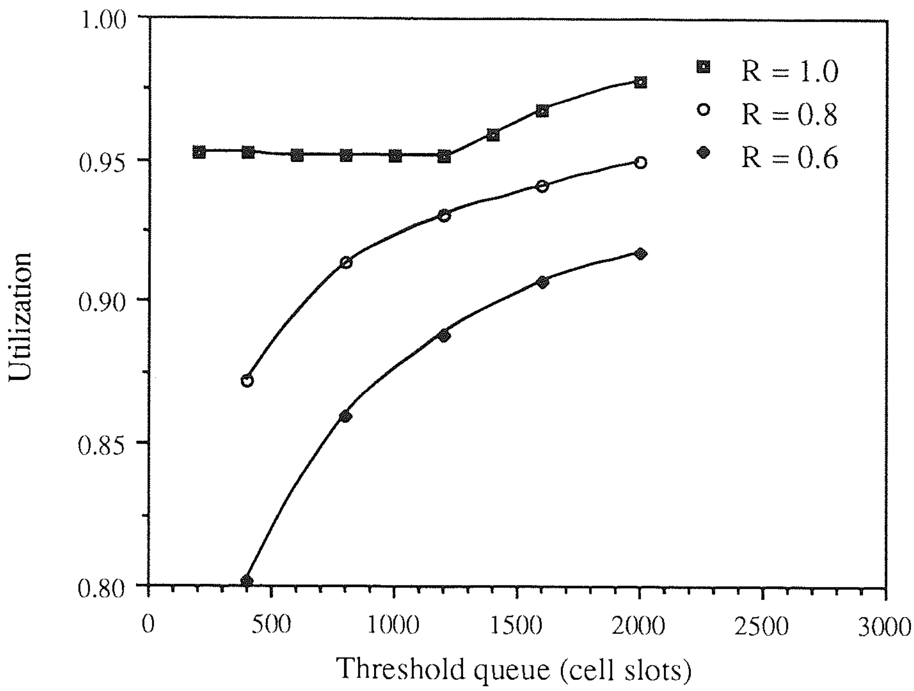



Figure 6.8 Network utilisation as the ECN threshold queue varies

In the folllowing paragraphs, the effect of varying the ECN threshold and the effect of reducing the window size by a factor R at the onset of congestion is examined as the buffer size is varied. This is to investigate the performance of the network at a buffer size smaller than the magnitude of the round trip delay. Figure 6.9 shows the variations of the cell loss probability due to buffer overflow as the ECN threshold changes by a small reduction factor. It is obvious that an improved network queue performance can be achieved by setting the ECN threshold value arbitrarily small. In this case, the probability of cell loss approaches $10^{-5}$ at a buffer level equal to 2500 cell slots for Tq = 800. A higher ECN threshold value Tq requires much larger buffer size to achieve a similar performance.

The probability of cell loss for two different values of the window reduction factor R is shown in figure 6.10. A small reduction factor provides a dramatic performance improvement especially at large values of buffer occupancy. Further improvement can be achieved at the expense of delay by choosing the correct threshold value.

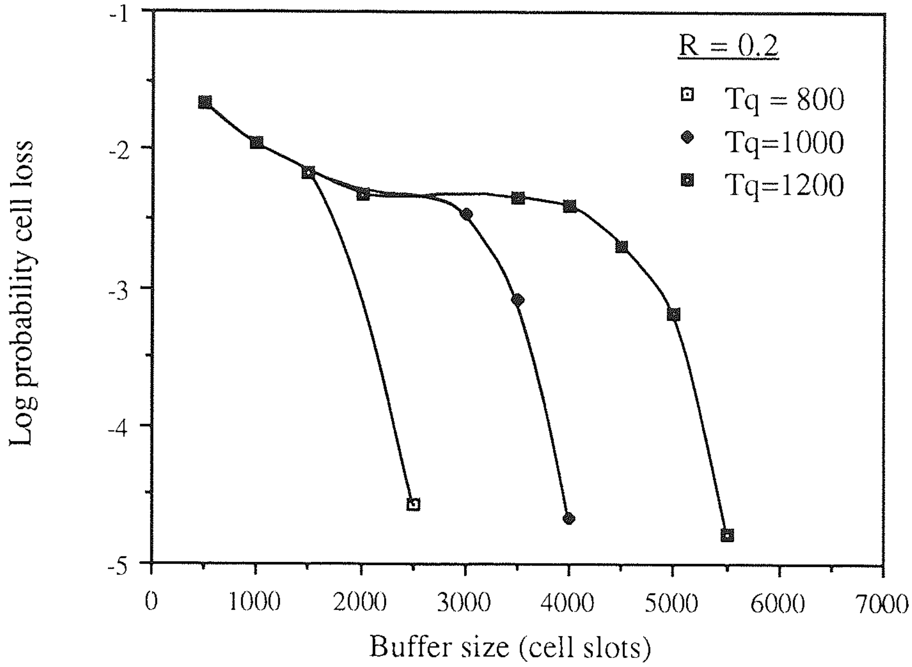



Figure 6.9 The effect of ECN threshold on the performance of the overflow probability at window reduction factor of 0.2.

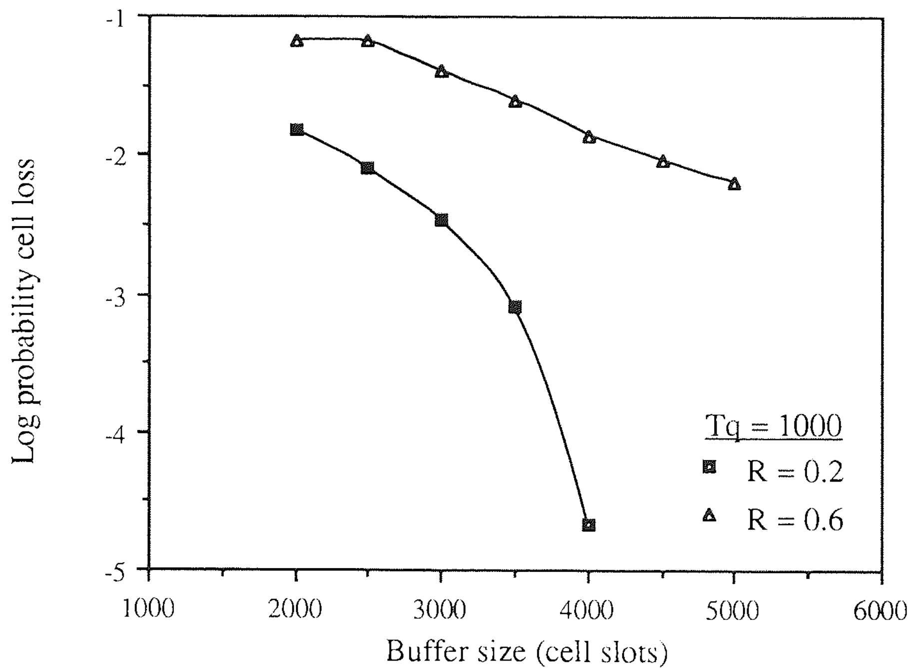



Figure 6.10 Probability of cell overflow for different values of window reduction factor at ECN threshold queue of 1000 cell slots.

## 6.5 Smoothing window

From the above results and discussion it can be seen that a the maximum end-to-end window size $W_R$ is required in order to allow full use of the network capacity. In this section the default effective window size is reduced to a window size relatively smaller than the end-to-end window size as in section 6.3 to take advantage of the smoothing effect. The reduced window size and the respective window interval is employed in the dynamic window control where an explicit feedback is effectively used to avoid congestion. As before, the average queue length is calculated over every one round trip interval independent of the window generation time and interval. Figure 6.11 reflects the hypothetical behaviour of a network node queue length;

a) for default window equal to an end-to-end window size, $W_R$ and

b) for default window equal to half of the end-to-end window size, $W_R/2$

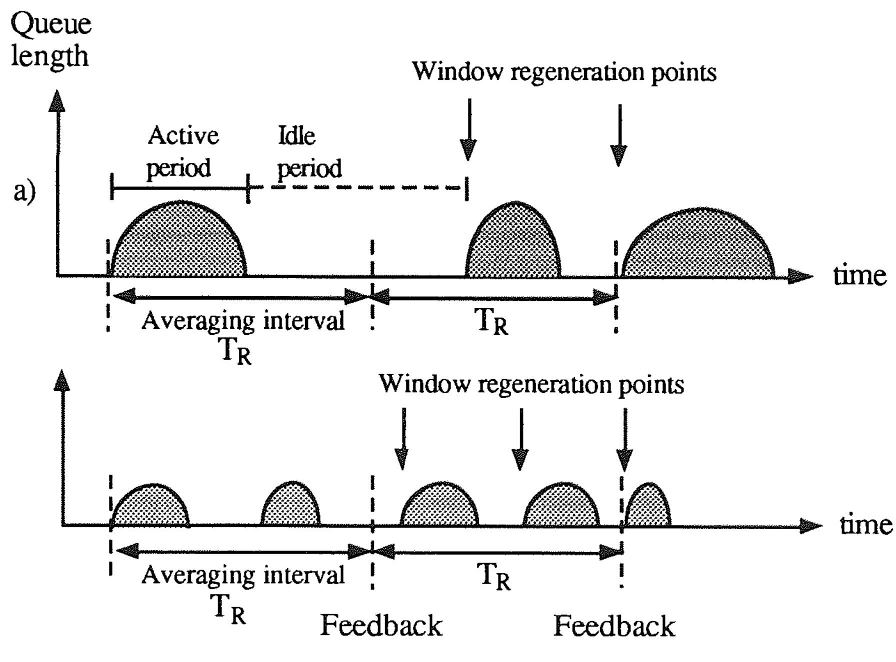



Figure 6.11 Hypothetical behaviour of network node queue length

The average queue length is computed by considering the integral (area under the curve) of the queue length during one round trip delay independent of the cell arrivals in a window interval. The feedback signal that reaches the source indicates the average queue length status of the previous round trip interval. If the node queue is not congested, the window size is increased by one. Since there are two windows in case b), the total number of cells in the associated round trip interval is raised by two. For n windows, we have

$$W_a(n) = W_e + n \qquad\qquad n = 1, 2, . .$$

$W_a$ is the value of the current window size after the acknowledment has been received.

The number of cells added to the network depends on the number of windows in an interval $T_R$. Thus, as the window size gets smaller, a greater number of windows can be accomodated in an interval $T_R$ and therefore more cells are allowed into the network which implies a larger step load increase.

Similar steps were taken by the network node and the users (i.e. sources) at the onset of congestion except that the reduced window size is much smaller than before and is even less if the reduction factor is effective. The performance of the network is investigated as the window size is varied for a particular threshold queue and reduction factor. Figure 6.12 shows that at a threshold queue of 400 cell slots and a reduction factor of 0.8, the smaller window size encountered higher loss. This is because the smaller window size experienced larger step load increase during uncongested network status but the total load is reduced by the same amount in an interval $T_R$ due to the multiplicative reduction. However, the smaller window size reaches a lower saturation probability of cell loss at a larger buffer size compared to the larger window size. Hence, if the buffer size is appropriately dimensioned, the smaller default window will give better network performance in terms of loss. However, this must be paid for by a larger mean delay as shown in figure 6.13.

The saturation probability for cell loss is observed to occur when the system has almost reached its maximum network utilisation as indicated by figure 6.14. At this state, the network performance will not be improved further by increasing the network resources. Increasing the buffer size will only lengthen the queueing delay without much reduction in cell loss (see figure 6.15).

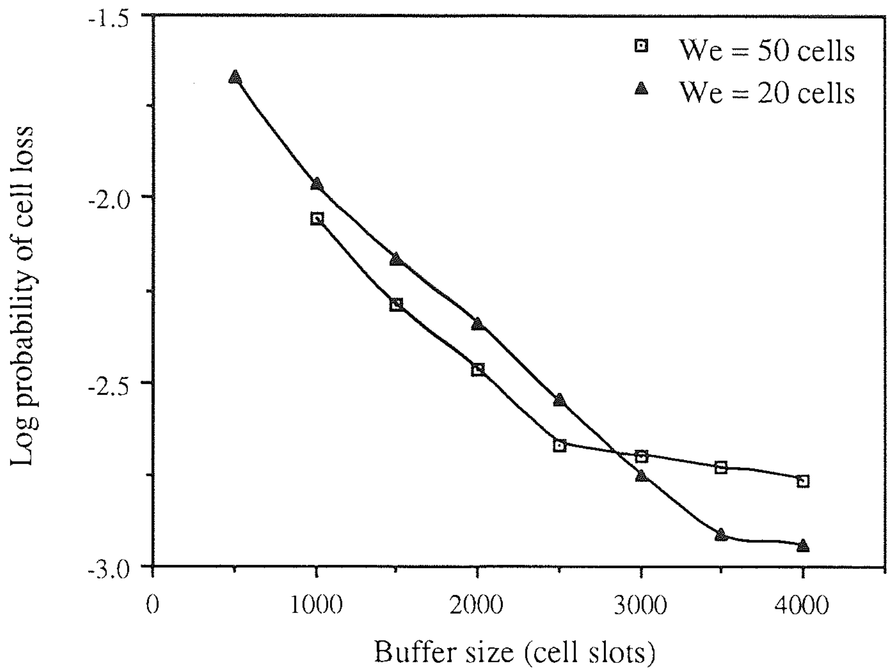



Figure 6.12 Probability of cell loss as the buffer size is increased for two different window size. (Tq = 400 cell slots, R =0.8)

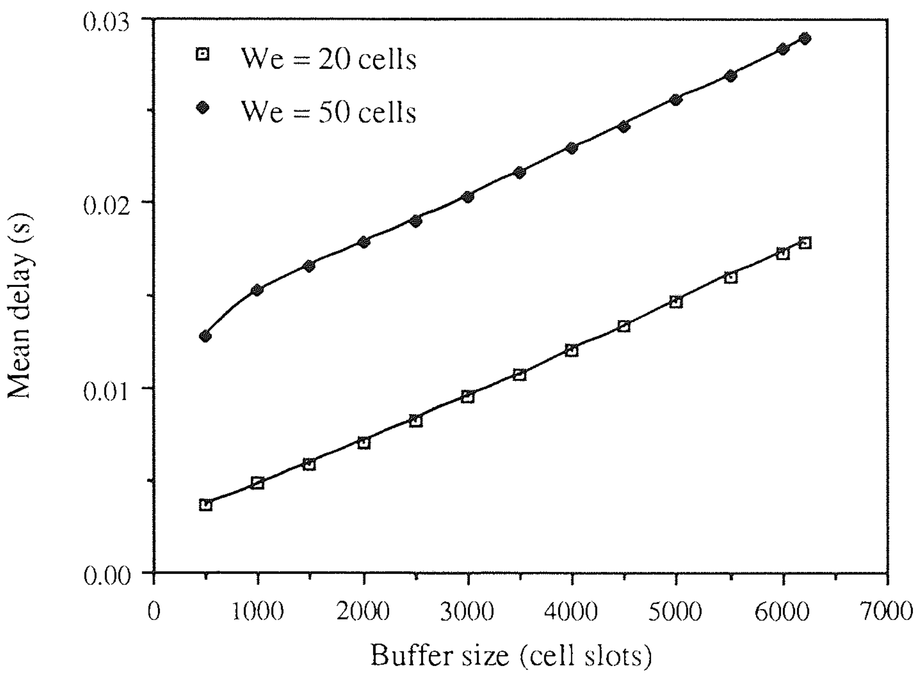



Figure 6.13 Mean delay experienced by data cells as buffer size are varied for two different default window size (Tq = 400 cell slots, R = 0.8)

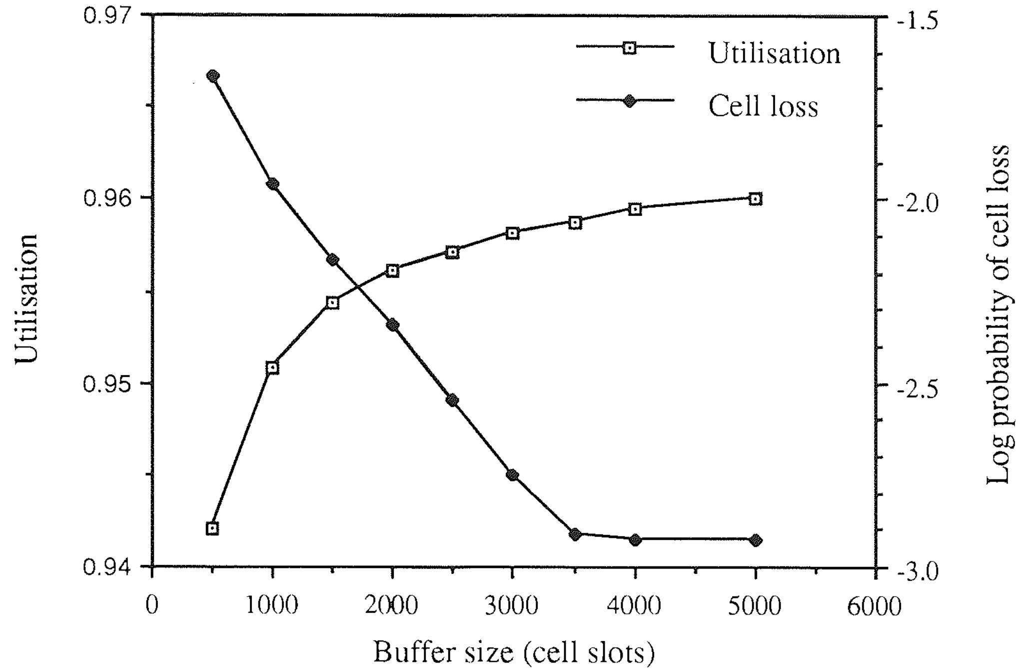



Figure 6.14 Variation of network node utilisation and probability of cell loss against buffer size for $W_a$ = 20 cells (Tq = 400 cell slots and R = 0.8).

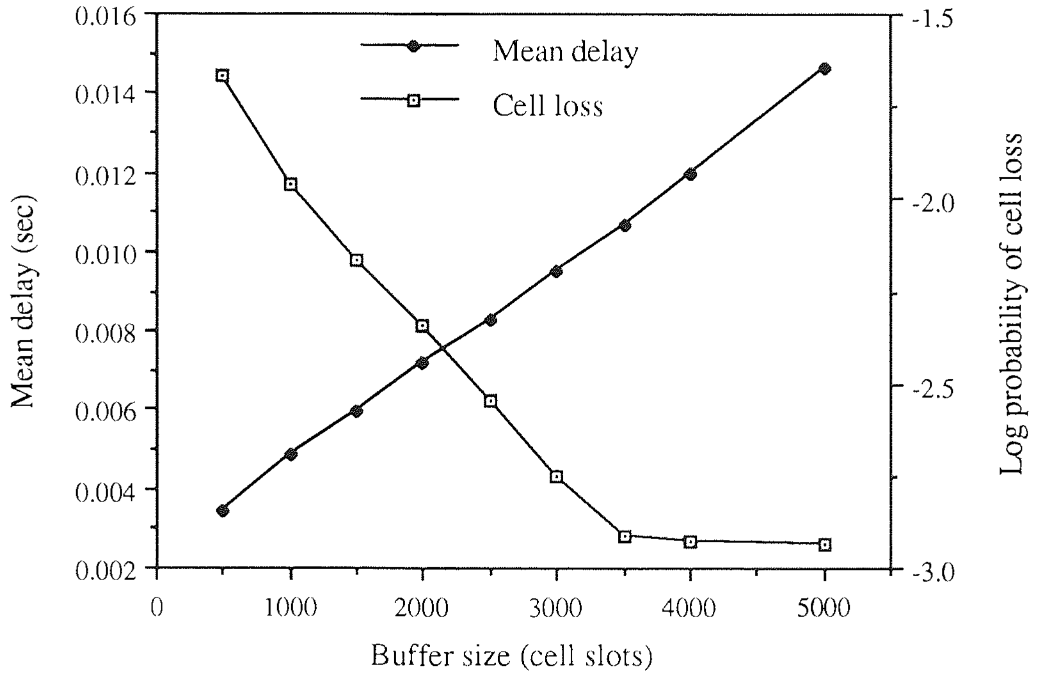



Figure 6.15 Variation of mean delay and probability of cell loss against buffer size for $W_a$ = 20 cells (Tq = 400 cell slots and R = 0.8)

## 6.6 Summary

In this chapter, a congestion avoidance scheme has been proposed for data services which requires stringent loss limit but can tolerate delay. The congestion avoidance scheme is based on a flow control window mechanism. The window size can be dimensioned according to the available buffer size or the required delay limit while keeping the losses due to buffer overflow almost negligible.

A dynamic window mechanism with explicit feedback notification allows the users to increase and decrease their load based on the network congestion status. Only a minimal amount of feedback from the network is used, one bit in the cell header. The feedback is a form of signalling to the users that congestion is about to occur. The users then have to monitor the amount of traffic that is placed in the network.

This scheme is distributed and can adapt well to the dynamic state of the network and converges to an efficient operating point. It is quite simple to implement with a low operational overhead. The scheme also maintains fairness in service provided to multiple users.

# CHAPTER 7

## SIMULATION MODELLING

### 7.1 Introduction

Modelling is a necessary step before going to the expense of implementing a full working system. A model is a representation of a system is some form, other than the entity itself. It aids a designer in explaining, understanding and improving a system. A model of a system may be an exact replica of the system (although executed in different material and to a different scale), or it may be an abstraction of the system's salient properties. A system design process is therefore assisted by modelling since experiments can be performed from which inferences can be drawn about the system without actually building it.

The degree of abstraction in modelling varies depending the types of model used. The most general model, and hence the most widely used model, is the mathematical model. It is the most abstract idealisation of a problem or a system which involves many simplifying assumptions in order to be analytically solvable. A model that is far too theoretical could deviate from a practical implementation of the problem.

The performance of packet switched oriented networks can be analysed mathematically through queueing theory [131]. In such networks, queueing arises very naturally since packets that arrive at a node must wait in a buffer before being transmitted on the appropriate outgoing links. A packet arrival process at a node queue is commonly assumed to follow a Poisson arrival process with an average rate. The packets in the queue are served according to some specified service discipline, normally an exponential service discipline. Such a simple single server queue with FIFO discipline, can be modelled as an M/M/1 queue.

In more recent developments of packet switched oriented networks such as ATM networks, the application of the traditional queueing theory may not accurately describe the network performance. The general distribution arrival process and deterministic service time due to a mixture of integrated teleservices and fixed cell length requires a more sophisticated queueing model analysis and more simplifying assumptions to be made.

Furthermore, the communication systems considered are often so complex that it can be extremely difficult to build analytically tractable performance models. However, due to the versatility, a simulation model can be employed in the design process to synthesize and evaluate such a system and provide a better insight into the model performance.

## 7.2 Modelling by simulation

Simulation modelling involves building a mathematical-logical model of a real system and experimenting with the model objectives. Some degree of abstraction is often necessary and considerable effort may be required to develop a performance model which accurately reflects the real system under study. Reference to the model objectives should always be made when deciding if an element of a system is significant and, hence, should be modelled. The model should be easily understood, yet sufficiently complex to realistically reflect the important characteristics of the real system without including unnecessary details.

Once a model has been developed and input data has been established, the model is translated into an acceptable computer form. A simulation model can be programmed using any general purpose language. However, there are several advantages in using a simulation language package such as SLAM, SIMSCRIPT, GPSS etc, including assistance in model formulation by providing a set of concepts for articulating the system description, which obviously saves programming time. A comprehensive review of such packages can be found in [132]. The choice of a particular simulation language however, depends on the type of system to be simulated and the efficiency of the program execution.

The performance of the simulation model must be evaluated through verification and validation. The verification task is to ensure that the translated model executes on the computer as intended while validation checks that the simulation model is a reasonable representation of the system.

### 7.2.1 Classification of simulation models

Simulation modelling assumes that a system can be translated in terms of system state descriptions. A system can be characterised by a set of variables, with each combination of variable values representing a unique state or condition of the system. The simulation is a portrayal of the dynamic behaviour of a model from state to state as a result of manipulation of the variable values in accordance with well defined operating rules. These rules in communication networks such as packet switched-based networks are largely based on the network protocols.

A system can be simulated in two different ways depending on the nature of the change of the system state;

Continuous simulation - The state of the system represented by dependent variables changes continuously over simulation time. Continuous simulation involves the characterisation of the behaviour of a system by a set of equations, such as difference, or differential equations and can consist of stochastic components. The dynamic behaviour of the state variables simulates the real system.

Discrete simulation - A change of the system state occurs discretely at specified points in simulated time, referred to as event times and the associated logic for processing the changes in state is called an event. The state of the system remains constant between event times. The dynamic portrayal of the system obtained by advancing simulated time from one event to the next is called the event-oriented approach.

In this work, SLAM II (Simulation Language for Alternative Modelling) has been used to simulate a high speed packet switched oriented network. It is a simulation oriented language which can be used for both discrete and continuous simulation. SLAM simulation statements can also be combined with standard FORTRAN routines. Furthermore, SLAM possesses an event trace facility which aids in the model verification by chronologically stepping through the system's events.

## 7.3 Simulation modelling using SLAM II

Throughout the research work, discrete event simulation has been used to model an ATM-based network. Basically, the simulation model can be divided into three parts. The progress of the modelling work first begins with a simple model which concentrates on the superposition of various services at the access node. The second part of the model continues with the embellishment of enforcement function and the last part involves a more refined communication between nodes.

In SLAM discrete event simulation, the state of the system is similar to the network model and is represented by variables and entities which have attributes and may belong to a file. These entities represent the cells within the network. Associated with each entity are a set of attributes such as cell arrival time, source address, cell number in a talkspurt etc. Events are used to model the start and completion of an activity.

During the execution of the simulation, the events are invoked and the model moves from state to state as the entities engage in activity. An example of system state that can exist on a network node is either the busy or idle state of the network server. The dynamic behaviour is then obtained by sequentially processing events and recording the status value at event times.

The organisation of a SLAM II program for discrete event modelling is illustrated in figure 7.1. The Slam II processor begins by reading the SLAM II input statements and initialising the SLAM II variables. A call is then made to subroutine INTLC which specifies additional initialising condition for the simulation. The processor then begins execution of the simulation by sequentially removing events according to the lowest event times from the event calendar.

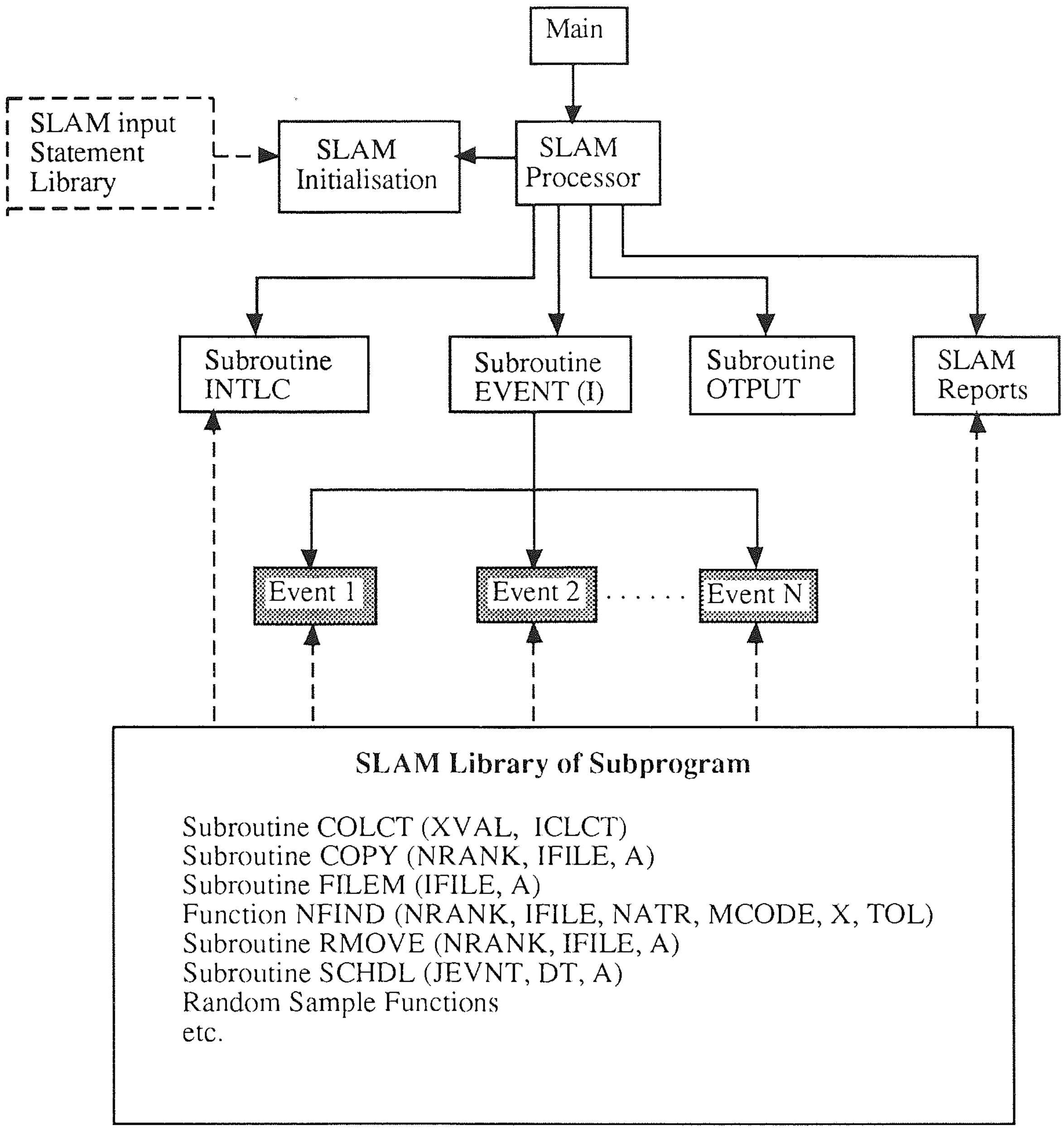



Figure 7.1 SLAM II organisation for discrete event modelling

### 7.3.1 Event definitions

The behaviour of the simulated network in the previous chapters utilised five basic events, which dictate the activity and hence the state of the network. These events which characterised the dynamic behaviour of the network are briefly described below.

***Arrival event*** - cells from different service classes are generated in this event according to their respective arrival processes as described in section 7.4. Various statistics on cell type, source address, cell arrival time, cell number, etc. are also collected and kept in the associated attributes.

***Update event*** - is used to update the server every cell slot time. A cell slot time is equivalent to the time to transmit a cell on the transmission link. In this case, an ATM cell of 53 bytes on 150 Mb/s link requires 2.83 us. The queue occupancies in the access and network nodes are also updated and the neccessary cell transfer is carried out depending on the priority mechanism and the state of the server. Cells are lost if the buffer space in the network node is exceeded. If the explicit feedback mechanism is employed, the network node average queue length is computed and updated every one round trip delay.

***Window event*** - data cells that have been generated by the arrival events are immediately scheduled to undergo the window event. This event enforces data cells to be transferred to the network node in windows. Cells that arrive when a window is full are buffered at the access node.

***Transmit event*** - collects statistics of the various attributes, e.g. cell delay and cell loss, on successful cell transmission. The server is reset and made available for service.

***Feedback event*** - receives the explicit feedback notification from the network node and makes a decision on the window size to alter the load already placed in the network. The decision is based on the state of the average queue length acknowledged by the update event.

The flow of events for data cells using the previous description is illustrated in figure 7.2

```
WHILE execuition is incomplete
    BEGIN
        arrival event
        REPEAT
            BEGIN
                window event
                REPEAT
                    BEGIN
                        update event
                        IF transmission successful  THEN
                            transmit event
                        IF threshold queue exceeded  THEN
                            feedback event
                    END
                UNTIL time constraint exceeded
            END
        UNTIL time constraint exceeded
    END
```

Figure 7.2 Flow of events for data cell

### 7.4 Traffic models

In this work the on/off model has been taken as the basis of source modelling in an integrated network. This model has been used successfully to describe voice traffic with silence removal [46, 47]. It can also represent an accurate model for file transfer. Even a video source can be modelled as a superposition of a number of independent on/off virtual mini-sources [48]. The corresponding discrete on/off model has also been suggested by [41, 57] for source modelling in an integrated environment.

### 7.4.1 Continuous bit rate source model

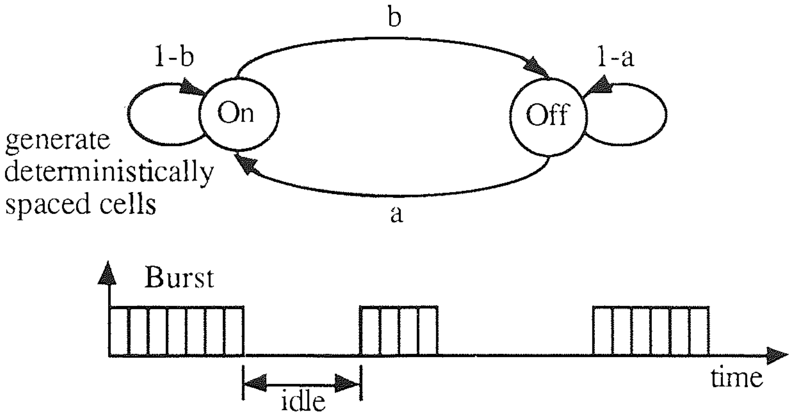
A specific source model can be defined with respect to the cell generation process. For example, a Continuous Bit Rate (CBR) voice source of 64 kb/s and a file transfer of 2 Mb/s are characterised by the continuous transmission of cells at their respective peak bit rates throughout the connection time. Thus, CBR teleservices traffic is generated in a deterministic way. CBR traffic is an extreme case of on/off traffic in which the burst length tends to infinity and the inactive period tends to zero.

### 7.4.2 Variable Bit rate source models

Variable Bit Rate (VBR) teleservices can be categorised into two types [64], VBR/Start-Stop and VBR/Continuous. The VBR/Start-Stop teleservices are defined by the alternation of activity and dormant silence periods. VBR/Continuous teleservices involve a more complex stochastic process and may be characterised by a bit rate variation during a virtual call, e.g. coded video telephony. This type of VBR teleservice is not considered in the research.

In the VBR/Start-Stop model, a source can be in two states, an on-state in which the source is transmitting cells with a fixed peak bit rate Bp and an off state in which the source is silent. It is assumed that the duration of the on and off state are exponentialy distributed. The model can be represented by a two state Markovian chain as depicted in figure 7.3 and it requires only three parameters to fully describe the traffic source, the peak bit rate Bp, the mean bit rate Bm and the mean burst length L. The value of Bp/Bm specifies the burstiness of the traffic source. Different hypotheses, which take into account different source behaviours (eg. alternation of very short and very long idle periods), require the definition of an appropriate set of traffic descriptor parameters and a specific model.

In this case a VBR 64 kb/s voice source traffic is assumed to alternate between talkspurts of mean length 1.36 sec and silent periods of mean length 1.76 sec. The values of these durations may vary depending on the threshold level of the speech activity detector used [134]. The voice source can be approximately characterised by a set of parameters, peak bit rate Bp = 64 kb/s, mean bit rate Bm = 28 kb/s and mean burst length of L= 227 cells.




Mean burst length $L = 1/b$

$$\text{Mean bit rate } Bm = \frac{a\,Bp}{a+b}$$

Figure 7.3 Two-state Markov traffic flow model

Bursty VBR data source traffic such as LAN-to-LAN interconnection, files transfer, etc. may be characterised by the same assumptions leading to a different set of traffic descriptor parameters. To facilitate the simulation work, data traffic is assumed to be characterised by Bp = 10 Mb/s, Bm = 2 Mb/s and L = 100 cells. The burstiness of the data source is varied by changing the mean bit rate. A high peak bit rate data source is considered since future data services are moving towards high speed systems.

The data cell format differs slightly from voice cell and video cell . A data cell is made up of 44 bytes of information field instead of 48 bytes. Four bytes are used by the ATM adaptation layer (AAL) as shown in figure 7.4 for interface purposes to the service user layer in the user, control and management planes. It is assumed that voice and video cell formats do not require the AAL facility and hence the whole 48 bytes are used for carrying voice and video information.

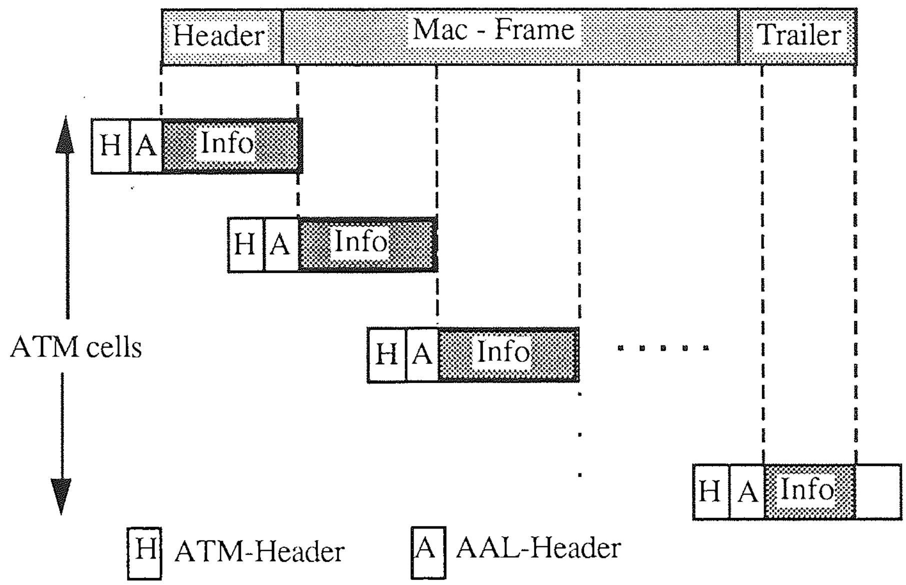



Figure 7.4 Eveloping and segmentation of a Mac-Frame

The video source used for the simulation experiments is modelled according the model proposed by [48]. It is assumed that the codec from a videophone scene generates 30 frames/s and had a mean output bitrate 3.9 Mb/s. The peak bit rate average over a whole frame is 10.575 Mb/s. In this model, the coder rate is modelled as a continuous-state, discrete-time stochastic process. Suppose $\lambda(n)$ represents the bit rate of a single source during the nth frame. A first order auto-regressive Markov process $\lambda(n)$ is generated by the recursive relation. Thus, the actual bit rate of the model during the $n^{th}$ frame is

$$\lambda(n) = 0.8781\ \lambda(n-1) + (0.1108\ w(n)\ 7.5\ \text{Mb/s})$$

where $w(n)$ is a sequence of Gaussian random variables with mean $\eta$ equal to 0.572 and variance equal to 1.

Since the model does not take into account the variation of the bit rate during a single frame, two extreme cell generation processes are possible:

1) the cells are sent equidistantly according to the actual bit rate of the frame. This is the smoothest traffic that can be generated and has a coefficient of variation for cell interarrival of about 2

2) the cells are sent equidistantly according to the maximum bit rate of the codec. This implies a pattern with one burst and one silence period during each frame. This

variant leads to a highly variable output stream with a coefficient of variation of about 10.6 for the cell interarrival time.

The latter generation process is used in the simulation since it produces the largest cell variation and thus produces a more conservative result.

The various types of input sources are modelled according to their statistical characteristics from a sample distribution and are varied using some chosen random number. Thus the input trafic models at the cell level are mixed stochastic and deterministic processes. The outputs from the simulation model are also probabilistic and statistical in nature.

## 7.5 Network model

Since ATM is basically designed for connection oriented services, a connection must be established before cells are transferred along the negotiated route. A simplified ATM network model is shown below in figure 7.5. N traffic sources access the network through a network interface and are multiplexed onto a high speed link to reach a network node. The different network nodes are connected according to the network topology through high speed links or trunks. In order to describe adequately the network behaviour the model must focus on the main resources, the link bandwidth and the bufffer capacity that are shared by the users

At the access node, the input traffic is the aggregate of the source traffic flows of each connection routed through that node. However, as a connection is followed along its route through several network nodes, its flow characteristics could change depending on the degree of interference due to buffering at each node and the network topology through which it is routed. If the interference has a negative effect on the traffic flow it would be necessary to account for it. Since the traffic flow characteristics of a connection along its route are a complex function of the traffic at each node, which can change during the connection, it is highly undesirable to formulate new resource reservation rules due to new changes in traffic flow characteristics as the network topology and routing evolves.

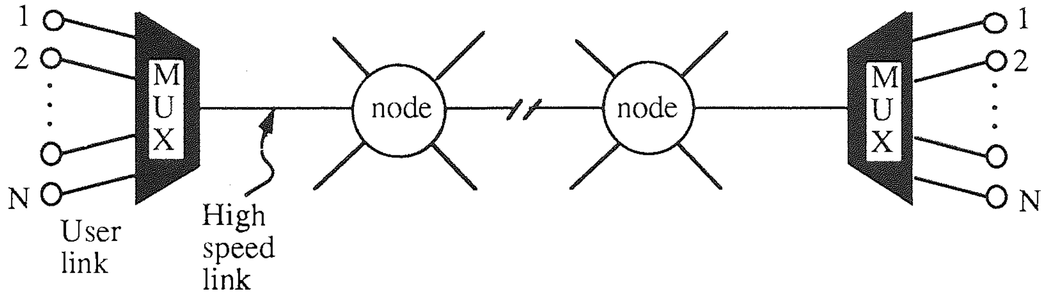



Figure 7.5 Network model

In the modelling design principle, it is assumed that an individual connection's traffic flow is not altered appreciably by any network node. This assumption is justified since it has been discovered by [122] that bursty traffic which causes large interference at the node buffer tends to become less bursty after passing through several nodes and thus experiences a positive smoothing effect at the interior of the network. Therefore, the input of any intermediate multiplexing or switching node can be modelled as the aggregate of the source traffic flows of all connections routed through that connection. This is a form of component decoupling and simplifies the simulation work by concentrating on an individual network node.

The simulation model is used for cell level modelling. Below are several assumptions made on the output buffer of a switch in a node;

- time is slotted with a cell slot duration of one cell service time (2.83 μs). Cell service time can start only at the beginning of a slot , with one cell being transmitted at a time.
- cell arivals occur between cell slot boundaries and cell departures occur at cell slot boundaries.
- service discipline is FIFO
- buffer has limited capacity.

### 7.6 Design of the simulation experiments

Since the simulation model is stochastic in nature, random elements of the model will produce outputs that are probabilistic. A stochastic process is said to be stationary if the underlying joint distribution of the random variables in the process remains invariant as time progresses. Throughout the research, stationary results have been used in the performance evaluation of the system. Various simulation design strategies must also be adopted in the simulation experiments as they dictate the inference one can make about an actual system. In particular, the design strategies should include the experimental conditions which will affect the simulation response.

In order to obtain a steady state result of a simulation, the simulation run must be long enough to suppress the initial state of the simulation. Furthermore, the steady state results should meet the performance measures of interest. The simulation results in this work were obtained from runs having a total simulation time up to $1.75\text{x}10^6$ time units each. For the network model concerned, one time unit is equivalent to one cell transfer time.2.83 us. Therefore, each simulation run lasted for about 5 seconds of real time. This corresponds to a maximum of almost $1.0\text{x}10^6$ cells being successfully transmitted through the transmission link excluding those produced in the initial state. Thus, the simulation results will provide a performance measure in terms of probability of cell loss down to $1.0\text{x}10^{-6}$.

The start-up policy used for settling the initial condition assumed that each voice and video source begins generating cells in bursts at a time taken from a uniform distribution in the range 0 to 200000 cell time units. More confident results were obtained by discarding the output during the transient portion of the simulation run which mainly depends on the initial conditions. The performance statistics were collected only after the simulation had reached a steady-state. The difficulty is how to determine exactly the length of the transient phase. This can be solved by having longer simulation run times such that the effect of the initial configuration becomes insignificant. The deletion of the initial statistics during the truncation period tends to reduce the bias on the output results from the initial condition.

A truncation period of 750000 cell time units had been assumed during which no statistics were measured. This is equivalent to about two seconds in real time. With simulation length of $1.75\text{x}10^6$ cell time units, longer truncation periods were found not to significantly affect the measured variables. Experiments using the same truncation periods on a much longer simulation run time of $3.5\text{x}10^6$ showed a very small and insignificant improvement in the output results. A simulation run of $1.75\text{x}10^6$ is preferable as it requires less CPU execution time.

An attempt was also made to validate the simulation model. A similar simple network model based on early work done by [74] was studied and its parameters were used in the basic simulation model for comparison. The simulation results obtained showed a very good performance agreement in terms of probability of cell loss. The simulation results were also used to validate an approximate analysis in chapter 3. They showed a far better performance and can be used to validate the analysis.

## 7.7 Summary

The network and its communication protocols have been modelled using simulation techniques due to the complexity associated with exact mathematical analysis. The simulation model was carried out using an event oriented discrete simulation technique, a common method used for modelling queueing networks.

A component decoupling principle is assumed without loss of generality. The whole simulation model was developed through the process of evolution, first the superposition of homogeneous and heterogeneous sources onto an access node, then the inclusion of the enforcement function, moving on to the introduction of the smoothing effect and finally the implementation of the feedback mechanism.

The simulation program can easily be modified to include more facilities for the network under consideration due its modularity. This can be done by adding more variables and subroutines in the program. The model can also be used as a 'test-bed' to allow different types of source model to be randomly generated and various network assumptions to be made and tested. Model verification has been carried out by inspection using event trace in SLAM. The accuracy of the behaviour of the models has been validated through comparison with other models of similar work.

# CHAPTER 8

## CONCLUSIONS

### 8.1 Introduction

The concept of B-ISDN has undergone considerable discussion and evolution over the past few years. ATM has been accepted as the most promising transfer technique for future B-ISDN. It is conceived as being a simple, flexible means of supporting a continuum of transport rates and it offers potential efficiency improvement over synchronous transfer through statistical sharing of network resources (bandwidth, buffers, processing, etc.) by multiple ATM connections.

With the B-ISDN goals of supporting diverse service and traffic mixes, and of efficient network resource engineering, the design of traffic and congestion flow control becomes an important challenge. Since ATM is a packet oriented transfer technique, it shares much in common with the conventional packet switched network for which successfully functioning congestion control methods exist. However, most of the conventional approaches to congestion control cannot operate at the extremely high broadband speeds. Many new issues arise and must be dealt with in designing the traffic and congestion flow controls including the large bandwidth-delay product, control implementability at broadband speeds, diverse quality of service requirements and the problem of traffic characterisation.

Public B-ISDN is envisaged to mature by the year 1995. However, this is subject to the progress of standardisation activity carried out by CCITT, mainly by Study Group XIII. While the transition towards the realisation of B-ISDN may take some time, it will be useful if ATM technology is adopted in advance for private networks. Private broadband ATM-based networks are not totally subject to standardisation and can be optimised in terms of performance and economy. They are also more flexible and easier to implement since they may have limited applications and less implementation constraints.

In this research, the study of traffic control has been carried out considering a private broadband wide area network. In such an environment, wideband leased lines combined with ATM technology can offer advanced and sophisticated features such as LAN interconnections and multimedia communications for business applications prior to the public B-ISDN network.

## 8.2 Review of the proposed traffic control for private ATM based network

In this work, traffic controls encompass other controls including the CAC, traffic flow enforcement, congestion, flow and priority controls. These traffic control capabilities are similar to those identified by CCITT Recommendations I.1311 [3] but are not bound by standard procedures. The traffic control procedures are carried out complementarily by the end terminals, the network access points and the internal elements of the network. The control functions operate at call level and cell level. This allows traffic management and congestion control to function on different time scales with distributed control capabilities. The various traffic control functions studied in this work for a private ATM-based network are briefly described below.

### 8.2.1 Call level - connection set-up phase

At the call level, during the connection set-up procedures, call acceptance and denial based on the specification provided by the user is carried out by the CAC. The CAC makes a decision according to the category of the call. Among the four categories identified, there are basically two types of bandwidth allocation; the peak bandwidth allocation and statistical bandwidth allocation. Those service catogeries under statistical mode can be further divided into two; bandwidth allocation based on the bufferless fluid flow mechanism and average bandwidth allocation. These services are accepted at the risk of a certain probability of cell loss. The amount of link bandwidth that can be consumed by the deteministic mode is fixed to a maximum limit of $C_d$, leaving the rest for statistical mode services. However, this division is only logical in the sense that the limit can be adaptively changed.

### 8.2.2 Cell level - data transfer phase

The traffic flow enforcement function considered in this work incorporates a smoothing function. It is suitable for data traffic that has a stringent cell loss requirement but can tolerate delay. The mechanism based on the window scheme restricts the source negotiated parameter such as the average bit rate by allowing a maximum number of cells to be transferred within a window interval. A smaller window interval admits a smaller number of cells and thus smooths out the cells arriving in bursts.

Connection requests that have been accepted into the network and have undergone the traffic enforcement function will be treated equally in the network unless they are specified by some priority mechanism to be carried out within the internal elements of the network. In this work, it is assumed that two ATM bearer services exist. The two levels of priority are specified by the priority bit or the reserved bits in the header. The two bearer services differentiate between loss sensitive traffic and delay sensitive traffic. The required QOS for each class is guaranteed through a partial buffer sharing with threshold.

There are two types of buffer in the network node; a short buffer for class 1 delay sensitive traffic and a long one for class 2 loss sensitive traffic. Class 2 traffic which consists of data traffic (e.g. LAN and file transfer), can be throttled in the case of congestion in the internal elements of the network. This is made possible by having some means of conveying congestion status information across the network. At the congested node an explicit congestion notification is set in the header of a cell and forwarded along the path to the destination terminal. The destination terminal will signal back to the source to react to the acknowledgement by throttling its end-to-end throughput based on the window mechanism. The performance of this feedback mechanism in providing a congestion free network for data services depends very much on the bandwidth propagation delay product. For a private environment, in this work, the round trip delay is restricted to less than 25 ms.

## 8.3 Evaluation of the various traffic controls capabilities

As listed out in chapter 2, the goals of a traffic and congestion control architecture should be to achieve simplicity, robustness, flexibility and controllability. In this work, these factors are taken into account with the addition of optimality in the network utilisation. To be more precise, optimality here is related to bandwidth and buffer utilisation in the network. In the broadband environment, bandwidth may be cheap and can be traded off with network controllability. However, in private networks using leased lines users would strive for optimum network utilisation with regard to bandwidth and buffer memory for economy purposes. The various traffic control functionalities are qualitatively assessed against the above goals and quantitatively evaluated based on the QOS measure. The results are summarised in table 8.1.

Table 8.1 Evaluation of traffic controls capabilities

| | CAC (overall) | Traffic enforcement | Priority | Explicit Feedback |
|---|---|---|---|---|
| Simplicity | ✓ | ✓ | ✓ | ✓ |
| Flexibility | ✓ | ✗ | ✓ | ✗ |
| Robustness | ✓ | ✗ | ✓ | ✓ |
| Optimality | ✓ | ✓ | ✗ | ✗ |
| Conservative measure QOS | ✓ | ✓ | ✓ | ✓ |

On the whole, the CAC, which includes the bandwidth allocation algorithms, satisfies the various goals. The optimum network utilisation may not be achieved using the bufferless fluid flow mechanism since its measurement does not take into account the buffer provided at the network access point. However, this is optimised by the acceptance of category 2 data traffic based on its average bit rate, at the expense of delay. The allocation of peak bandwidth to the traffic with large interference and CBR traffic may unavoidably reduce the network utilisation but is limited by the maximum virtual link capacity $C_d$. This strategy is considered fair since it prevents the network from being monopolised by certain types of traffic.

Deterministic bandwidth allocation for limited virtual link capacity with sufficient buffer memory will always guarantee almost negligible cell loss. On the other hand, although statistical bandwidth allocation risks some probability of cell loss, the QOS measure computed by the bufferless fluid flow model is still a conservative measure due to the presence of a buffer in a real system. QOS for average bandwidth allocation to data traffic is also a conservative measure since a sufficiently long, finite buffer size is reserved. The maximum size of the buffer depends on the end-to-end window interval. In general, the proposed CAC is simple, flexible and robust and can be carried out on-line since it does not involve complex mathematical computation.

On the contrary, the traffic enforcement function which is based on the window mechanism does not achieve all the required goals. Nevertheless, it is simple to implement and can achieve maximum network utilisation by allowing average bandwidth to be enforced without discarding cells and hence more calls can be accepted. However, this is done at the expense of delay due to cells being buffered and is thus a conservative QOS measure, assuming that the delay is within the limit. The traffic enforcement function may be used to enforce traffic to a bit rate higher than the actual mean bit rate if agreed by the CAC, by introducing an overdimensioning factor. This will reduce the delay experienced by cells due to buffering. However, this function is still not as flexible and robust compared to leaky bucket described in [92]. The main advantage of this method is that the window mechanism can be extended for flow control using an explicit feedback mechanism.

The partial buffer sharing with threshold priority mechanism achieves all the goals except optimality. The network performance at cell level may not achieve the maximum buffer utilisation if the threshold queue is not properly dimensioned according to the class 1 bearer service load. This is mainly due to the procedure of accepting class 2 cells into the main buffer. Despite this, the dimensioning of the two buffers provides a conservative QOS measure. The mechanism is also robust since the threshold can be changed adaptively depending on the load. Being flexible, fair and simple, this mechanism offers the best compromise between performance and implementation.

Finally, the congestion and flow control strategy using an explicit feedback notification aims at congestion avoidance rather than reacting on an already congested network. This strategy is only appopriate for data services that can be flow controlled and hence tolerate delay. Therefore, it is not flexible. The mechanism is quite robust to any changes in the ECN threshold queue and the reduction factor, at the cost of the network utilisation and cell loss. However, the cell loss can be maintained small at reduced network utilisation with an appropriate ECN threshold queue, reduction factor and buffer size dimensioning. Fairness is also preserved through distributed reactions shared by multiple users. It is also simple to implement since it involves only a minimal amount of operational overhead ( one bit in the cell header) and a small amount of processing time.

In general, the various traffic control functions discussed above provide reasonable function capabilities within the stated limited applications. Although the optimality of the network utilisation is not achieved in all the functions, the other more important merits such as the conservative QOS measure and simplicity of the various functions outweigh this disadvantage. However, the optimum network utilisation in the CAC is already an excellent indicator of resource usage consumed by the various service catogeries. The other functions are service dependent and, therefore, their inefficiency in certain aspects may affect only the related services.

The overall achievement of the various traffic control capabilities summarised in table 8.1 suggests that they are appropriate for a private ATM-based network. Such a network could be easily coupled with the future public B-ISDN since it is based on ATM technology. However, more research work and experiment has to be carried out before such a network can be realised. In particular, the effectiveness of traffic control including congestion and flow control must be further improved by adding more facilities. Further recommendations for future work are listed in the following section.

## 8.4 Recommendations for further work

* The priority scheme in this work differentiates between two bearer service classes, class 1 delay and class 2 non-delay sensitive traffic. It is necessary to provide loss priority among the different service categories in class 1. For example, CBR traffic should be given higher priority than statistical traffic. It is suggested that the push-out scheme could be used to push out the lower priority traffic when the buffer is full. Further work is required to examine the effect of this scheme on individual traffic cell loss and the effect on the cell sequence intregrity.

* Investigation of the possibility of within-call parameter renegotiations which will allow the network to satisfy efficiently the dynamically changing service needs of a call. For example, consider a large file transfer consisting of a long train of cells during a burst followed by a long inactive period. It would be a waste of resources to reserve the whole buffer for the whole duration of the call, which may last for several seconds or minutes. Instead, buffer reservation can be negotiated for the duration of the average burst length and released during the inactive period.

* Investigation of the possibilities of selective cell discarding under congestion conditions in the internal elements of the network . This is appropriate for services of category 1 and 3, for example voice and video sources, where the embedded coding scheme can produce essential cells and enhancement cells. Essential cells are given higher priority than the enhancement cells by marking in the cell header. Discarding the low priority cells containing less important imformation may cause the least effect on the quality of the service. Another possibility is marking or tagging violating cells in the traffic enforcement function at the edge of the network and discarding these cells when they arrive at a congested node.

* Throughout the work, the principle of component decoupling has been assumed for the design of each node along the path to the destination. However, the multiplexing

of bursty traffic and different mixes of class 1 traffic at diferent nodes along the path may alter individual connections. It is therefore necessary to examine the behaviour of individual connections at the various nodes in order to further optimise resource utilisation within the network.

* The network node in the simulation model is assumed to be error free. The window flow control with ECN feedback is meant to prevent congestion. However, cells may be lost due to a network fault and therefore, an error control is required to counter it. This subject is for further study.

Only a few recommendations for future work are listed above. Practical work is required to verify that the various control functions are operable and give the required performance. It is also desirable for tractable mathematical models to be derived for the analysis of network performance.

## 8.5 Summary

In this chapter, a review of traffic control procedures that has been studied during this research has been briefly made.The performance of each component of the traffic controls has been evaluated against the high level goals. On the whole they are simple, highly practical and can provide an increase in network performance in terms of bandwidth usage.

The results are promising and suggest that the various congestion and flow contol functions proposed are adequate for advanced packet oriented private broadband wide area networks. However, more work is still required to improve the related traffic control functions. Several recommendations for further work have been given while other valuable related work is very much to be encouraged.

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Flexible multi-service acceptance control for private broadband ATM-based Network
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4. N. Fisal and R. L. Brewster
Congestion/Flow control using explicit feedback mechanism in private ATM-based network
To be presented in IEEE Malaysian International Communication Conference, Kuala Lumpur, Nov.1993