Traffic and Congestion Control for ATM over Satellite to provide QoS

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Thesis submitted for the degree of Doctor of Philosophy

Centre for Communication Systems Research
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ABSTRACT

In broadband multimedia satellite networks it is necessary to multiplex bursty streams of traffic with differing Quality of Service (QoS) requirements to maximise the utilisation of the satellite link bandwidth. Providing the desired QoS of each service, in a multi-service environment is a major challenge for satellite networks. Asynchronous Transfer Mode (ATM) which provides hard QoS guarantees is suitable for a multi-service satellite environment. ATM has been developed as a vehicle for multimedia communications and is widely regarded as one of the most important and fastest-growing communications technology of this decade.

The design of suitable traffic and congestion control algorithms is one of the most important challenge for the success of an ATM-based satellite network. This thesis develops and optimises a traffic and congestion control mechanism which can provide users the required QoS for ATM over satellite networks.

In order to provide QoS differentiation for end-to-end communication it is proposed to use both loss and delay priorities, which are determined from the required Cell Loss Rate (CLR) and Cell Transfer Delay (CTD) parameters, for each service class. A multiple shared buffer scheduling (MSBS) policy considering both delay and loss priorities, is proposed and evaluated for scheduling and discarding of ATM cells. It is shown that both the CTD and CLR requirements of ATM services can be met by the MSBS scheme.

A combined preventive/reactive control scheme incorporating an adaptive Leaky Bucket (LB) is investigated for the satellite environment. It has been found that reactive control improves the cell loss due to congestion for time scales larger than the propagation delay.

As the satellite air interface bandwidth is currently one of the most expensive commodities in the service provision, an adaptive MAC protocol that can support the ATM service classes whilst maximising the bandwidth utilisation, is proposed and evaluated. The mapping of ATM service classes to MAC classes and the use of a prioritised request queue provides the QoS differentiation required by ATM networks. It is shown that a pure reservation system performs poorly for very bursty user traffic. The user population which can be supported using Random Access (RA) for very bursty users with short burst duration is higher. The system throughput can be maximised, by making this protocol adaptive to changing traffic characteristics. It is shown that the utilisation of the frame capacity and the total number of users served can be improved by using this protocol.
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<td>ISL</td>
<td>Inter Satellite Link</td>
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<td>NNI</td>
<td>Network Node Interface</td>
</tr>
<tr>
<td>NPC</td>
<td>Network Parameter Control</td>
</tr>
<tr>
<td>nrt-VBR</td>
<td>non-real-time Variable Bit Rate</td>
</tr>
<tr>
<td>OAM</td>
<td>Operation And Maintenance</td>
</tr>
<tr>
<td>OBP</td>
<td>On-Board Processing</td>
</tr>
<tr>
<td>OBRRM</td>
<td>On-Board Radio Resource Management</td>
</tr>
<tr>
<td>OBS</td>
<td>On-Board Switching</td>
</tr>
<tr>
<td>PCR</td>
<td>Peak Cell Rate</td>
</tr>
<tr>
<td>PDH</td>
<td>Plesichronous Digital Hierarchy</td>
</tr>
<tr>
<td>PDU</td>
<td>Protocol Data Unit</td>
</tr>
<tr>
<td>PM</td>
<td>Physical Medium</td>
</tr>
<tr>
<td>PRM</td>
<td>Protocol Reference Model</td>
</tr>
<tr>
<td>PRMA</td>
<td>Packet Reservation Multiple Access</td>
</tr>
<tr>
<td>PT</td>
<td>Payload Type</td>
</tr>
<tr>
<td>PTI</td>
<td>Payload Type Identifier</td>
</tr>
<tr>
<td>PHY</td>
<td>Physical Layer</td>
</tr>
<tr>
<td>QoS</td>
<td>Quality of Service</td>
</tr>
<tr>
<td>RA</td>
<td>Random Access</td>
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<td>RAL</td>
<td>Radio Access Layer</td>
</tr>
<tr>
<td>Rec.</td>
<td>Recommendation</td>
</tr>
<tr>
<td>RM</td>
<td>Resource Management</td>
</tr>
<tr>
<td>RPL</td>
<td>Radio Physical Layer</td>
</tr>
<tr>
<td>RS</td>
<td>Reed-Solomon</td>
</tr>
<tr>
<td>rt-VBR</td>
<td>real-time Variable Bit Rate</td>
</tr>
<tr>
<td>S/N</td>
<td>Signal to Noise ratio</td>
</tr>
<tr>
<td>SAR</td>
<td>Segmentation And Reassembly</td>
</tr>
<tr>
<td>SATM</td>
<td>Satellite-ATM</td>
</tr>
<tr>
<td>SCR</td>
<td>Sustainable Cell Rate</td>
</tr>
<tr>
<td>SDH</td>
<td>Synchronous Digital Hierarchy</td>
</tr>
<tr>
<td>SECBR</td>
<td>Severely Errored Cell Block Ratio</td>
</tr>
<tr>
<td>SES</td>
<td>Severely Errored Seconds</td>
</tr>
<tr>
<td>SP</td>
<td>Static Priority</td>
</tr>
<tr>
<td>STM</td>
<td>Synchronous Transfer Mode</td>
</tr>
<tr>
<td>TAT</td>
<td>Theoretical Arrival Time</td>
</tr>
<tr>
<td>Acronym</td>
<td>Description</td>
</tr>
<tr>
<td>---------</td>
<td>---------------------------------</td>
</tr>
<tr>
<td>TC</td>
<td>Transmission Convergence</td>
</tr>
<tr>
<td>TCP</td>
<td>Transmission Control Protocol</td>
</tr>
<tr>
<td>TDM</td>
<td>Time Division Multiplex</td>
</tr>
<tr>
<td>TDMA</td>
<td>Time Division Multiple Access</td>
</tr>
<tr>
<td>UBR</td>
<td>Unspecified Bit Rate</td>
</tr>
<tr>
<td>UNI</td>
<td>User Network Interface</td>
</tr>
<tr>
<td>UPC</td>
<td>Usage Parameter Control</td>
</tr>
<tr>
<td>USAT</td>
<td>Ultra-Small Aperture Terminal</td>
</tr>
<tr>
<td>VBR</td>
<td>Variable Bit Rate</td>
</tr>
<tr>
<td>VCC</td>
<td>Virtual Channel Connection</td>
</tr>
<tr>
<td>VCI</td>
<td>Virtual Channel Identifier</td>
</tr>
<tr>
<td>VD</td>
<td>Virtual Destination</td>
</tr>
<tr>
<td>VPC</td>
<td>Virtual Path Connection</td>
</tr>
<tr>
<td>VPI</td>
<td>Virtual Path Identifier</td>
</tr>
<tr>
<td>VS</td>
<td>Virtual Source</td>
</tr>
<tr>
<td>VSAT</td>
<td>Very-Small Aperture Terminal</td>
</tr>
<tr>
<td>WAN</td>
<td>Wide Area Network</td>
</tr>
<tr>
<td>WCT</td>
<td>Worst Case Traffic</td>
</tr>
</tbody>
</table>
1. INTRODUCTION
Recent developments in multimedia applications, which possess diverse traffic characteristics and Quality of Service (QoS) requirements, have led to the evolution towards an Asynchronous Transfer Mode (ATM)-based Broadband Integrated Services Digital Network (B-ISDN). The ultimate aim of this evolution is to ensure that everyone can have access to information, anytime, anywhere. To achieve this goal, satellites play an important role in the broadband infrastructure. Geostationary satellites offer several unique benefits to the overall telecommunications network: ubiquitous wide-area coverage, network flexibility, broadcast and multipoint-to-multipoint capabilities, as well as rapid network set-up and deployment.

Several GEO multimedia satellite communication systems such as Cyberstar, Astrolink, GE*Star and EuroSkyWay have been proposed [ELIZ96, FERN97, BULL97, LOSQ97]. They plan to use Ka-band, due to the larger bandwidth available at those frequencies and to facilitate multimedia services to fixed and portable terminals, which are delivered over ‘ATM like’ transport in the satellite network.

In a broadband multimedia satellite network, it is necessary to multiplex bursty traffic with different Quality of Service (QoS) requirements, to maximise the utilisation of the satellite link. Providing the desired QoS of each service, in a multi-service environment is not an easy task. The design of a suitable traffic and congestion control is one of the most important challenge for the success of an ATM-based satellite network. There has been extensive research over recent years on traffic and congestion control schemes for terrestrial networks. It is important to adapt the schemes proposed for terrestrial networks to the satellite environment. Due to the different properties of the satellite link from terrestrial links, some different mechanisms and protocols are needed for the satellite environment which will be investigated in this thesis.

Various control mechanisms have been proposed for ATM networks. Preventive control techniques, which attempt to prevent congestion by taking appropriate action before congestion occurs. A preventive control mechanism consists of the connection admission control (call level) and usage parameter control (cell level). Although preventive control tries to prevent congestion before it actually occurs the satellite
system may experience congestion due to multiplexing buffer or switch output buffer overflow. In this case, where the network relies only on the UPC and no feedback information is exchanged between the satellite and the source, no action can be taken once congestion has occurred.

Explicit Rate (ER) indication, in which the network notifies the sources, which can reduce their activity, of the exact bandwidth share it should be using in order to avoid congestion seems to be more effective considering the long satellite propagation delay. ER control is an end-to-end closed loop control scheme however it has been proposed to segment this control loop with the satellite acting as Virtual Source (VS) and Virtual Destination (VD) to halve the control loop delay.

It is widely accepted that a combination of three levels (cell, burst and call level) of traffic and congestion control is usually the most effective. However most work has focused on one of these mechanisms (in isolation), which are effective at different time scales. It will be investigated how these techniques can work in an integrated manner and which functionality’s are kept on the ground terminal.

In ATM cells there is one cell loss priority bit in the header which is considered by the network when discarding cells during congestion. However no provision for delay priorities is made in the ATM standards, since the delay experienced in the terrestrial network is small and at high bit rates (155 Mbit/s) the buffering delay is negligible. In contrast the satellite propagation delay is long and the bit rates of the satellite links are currently around 2-34 Mbit/s. Thus, due to the long propagation delay of the satellite link it is a much more difficult task to guarantee delay requirements (Cell Transfer Delay and Cell Delay Variation Tolerance). Therefore the use of delay priorities seems to be very promising for scheduling purposes in order to provide delay requirements. The use of delay priorities associated with the CTD requirement of each connection will be investigated in this thesis. Although there is no explicit provision in the standards for distinguishing different levels of delay priority, it is possible to use VPI/VCI values in the header. On entry to a switch, the VPI/VCI values are used to determine the outgoing port required, so it is a relative simple extension to use these values to associate delay priority and choose one of maybe a number of priority buffers at the output port.
Scheduling and buffer management algorithms using both loss and delay priorities on the ground terminal and on the output buffer of the on-board satellite switch for end-to-end QoS differentiation will be investigated for GEO satellite constellations.

Considering that satellite communication users need to access a shared medium, a mechanism to control the access to the transmission medium is needed once the connection has been set-up. The statistical multiplexing capability of ATM has to be extended to the air interface, since maximisation of resource utilisation is very important for the satellite environment.

Several Multiple Access Control (MAC) protocols have been developed to control the access to a shared communication channel capacity. The MAC is responsible for bandwidth scheduling and most of the MAC work so far does not consider any priorities. We thus extend the work to implement prioritised bandwidth scheduling taking into account the traffic contract and the ATM traffic classes.

1.1 Objectives

In order to design an integrated traffic control scheme, to minimise congestion and maximise resource utilisation whilst providing the different QoS requirements, the objectives of the thesis are:

- To accurately control the declared traffic parameters with a short reaction time, at the access point to the satellite network.

- Scheduling and discarding of traffic on-board the satellite according to loss and delay priorities.

- Extension of the statistical multiplexing capabilities of ATM to the satellite air interface, while providing the QoS of each service class.

- Investigating if supporting the ABR service class (which requires feedback) for maximising resource utilisation is efficient for satellite ATM networks.

1.2 Contributions of this Thesis

This thesis proposes a traffic and congestion-control mechanism to provide the required QoS of a range of service classes. The main idea is to associate loss and
delay priorities with the CLR and CTD requirements of each service class to meet individual delay and loss requirements. It is shown that the number of users sharing the same resources (such as buffer and bandwidth) can be maximised whilst still meeting the requested QoS. The major contributions of this thesis are given below.

- A buffer scheduling and discarding strategy called Multiple Shared Buffer Scheduling (MSBS) was proposed and evaluated. This scheme considers both delay and loss priorities. It has been shown that both delay and loss requirements can be met by the MSBS scheme. The complexity of MSBS is less than that of partial buffer sharing with nested thresholds, where lower cell loss rates for services sharing the same buffer might be obtained.

- The dimensioning of the Generic Cell Rate Algorithm (or equivalent leaky bucket) to police the declared parameters accurately and with a short reaction time to violations has been carried out for bounded and unbounded burst size. In order to reduce the reaction time of the policing mechanism, the LB with smoothing buffer has been analysed for unbounded burst size. Closed-form expressions were derived for the mean reaction time and the mean queuing delay. The accuracy of both expressions has been verified using simulations.

- It has been shown that considerable improvements in delay performance and satellite bandwidth utilisation are possible if on-board processing satellite technology and an adaptive MAC protocol is used. The traditional demand-assignment scheme using a ground terminal as control station has two important drawbacks: long set-up and reservation time and limited channel utility. Both are due to the long propagation delay of the satellite link. Both disadvantages can be removed by processing channel requests in the satellite to allocate frame slots.

- An adaptive reservation/random access MAC scheme, which supports all ATM service classes, is proposed and analysed. The mapping of ATM service classes to MAC classes and the use of a prioritised request queue provides the QoS differentiation required by ATM networks. It is shown that the utilisation of the frame capacity and the total number of users served is greatly increased by using this protocol to statistically multiplex traffic over the air interface. The
problem of variable channel-access delay of Packet Reservation Multiple Access (PRMA) and its variants is solved by using polling for Variable Bit Rate (VBR) terminals.

- It has been shown that, for a reactive control scheme, the number of cells transmitted on the satellite link before any feedback can be received is dependent on the terminal bit rate and on the satellite link propagation delay. Reactive control algorithms perform better for lower propagation delays and lower terminal bit rates. The performance of reactive control schemes can be improved by making them proactive. This means that the onset of congestion must be identified before congestion actually occurs. It has been found that the Explicit Rate (ER) control is successful in achieving the specified target utilisation value. ER has been found to be most effective if the congestion duration is longer than the propagation delay.

Some part of this work was carried out as part of the EPSRC-PCP-Link Global Integrated Personal Satellite multimedia Environment (GIPSE) and Mobile VCE projects. Appendix E shows a list of publications as a result of this research.

1.3 Organisation of this Thesis

Following the Introduction in Chapter 1, which summarises the motivation, objectives and achievements of this research, Chapter 2 presents an overview to ATM over satellite. First the advantages of ATM as the transfer mode for B-ISDN, and a summary of the ATM standards recommended by the ITU-T are given. This is followed by the reasons for using satellites. An introduction to link budgets which are used to determine the Bit Error Rate (BER) of the satellite channel is given. Then satellite system design issues such as choice of orbit, use of On-Board Processing (OBP), On-Board Switching (OBS) and Inter-Satellite Links (ISLs) are discussed. Furthermore, satellite access techniques are described. After providing information on ATM and Satellites the motivation for ATM over satellite transmission is explained. The role of satellites to seamlessly extend the B-ISDN and the proposed enhancement techniques to make transmission of ATM over satellite more robust are reviewed.

In order to design and develop network functions, such as traffic control, it is necessary to comprehend the characteristics and requirements of the traffic to be
carried. Chapter 3 therefore focuses on source characterisation. After summarising various proposed source parameters, which can be negotiated between user and network, service categories which may be supported by the network are identified. Each service category can support various services, depending on source parameters declared and QoS guarantees required by the user. The statistical behaviour of generic services such as voice, video, data and multimedia are provided and, based on this information, the criteria in selecting source models for services are explained. The Chapter concludes with long and short-range dependent traffic source models to represent individual services.

Chapter 4 explains the proposed traffic and congestion control framework, which is analysed throughout this thesis. Traffic controls for two network architectures with and without OBP are illustrated. The research carried out considers future satellite systems with on-board processing satellites which have cell switching capabilities. Both preventive and reactive control schemes are used, and the use of both delay and loss priorities for multi-service satellite ATM networks constitutes new work in this area. The performance of the proposed Multiple-Shared Buffer Scheduling (MSBS) scheme which considers delay and loss priorities is also evaluated in this Chapter.

In Chapter 5 the performance of the Usage Parameter Control (UPC) and Network Parameter Control (NPC), which are used to monitor and control the traffic in terms of conformity with the agreed traffic contract, are evaluated and optimised. First, the UPC performance parameters are described and the actions which can be taken by the UPC explained. Then the effects of Cell Delay Variation (CDV) on the cell stream and the necessity of introducing some tolerance to the policing algorithm is illustrated. It is shown how to dimension the Generic Cell Rate Algorithm (GCRA), chosen as UPC/NPC, to control the peak rate.

The emphasis of Chapter 5 is on minimisation of excess traffic entering the satellite network. Some tolerances are allowed when policing the negotiated traffic contract, in order not to discard conforming cells. Tight policing of sources is important as statistical multiplexing tolerances in the UPC might result in QoS degradation of other connections.
The control of the Sustainable Cell Rate (SCR), which is an upper bound on the mean rate for unbounded and bounded burst size, is investigated in order to assess the advantage of the Maximum Burst Size (MBS) parameter recommended by the ITU-T and ATM-Forum.

Chapter 6 analyses MAC schemes which coordinate access to the shared satellite radio link resources. First, the design objectives of a MAC protocol are summarised and then various proposed MAC schemes to support multiple services are described. Then the analysis of a proposed adaptive random/reservation access control scheme which can support all ATM service classes while providing the required QoS is presented.

In addition to preventive control, reactive controls can also be used for satellite ATM networks. The satellite may experience congestion due to buffer overflow at the multiplexing or on-board switch output buffer. Reactive control is a technique used to recover from a congested state and reactive schemes operate at time scales greater than the satellite propagation delay. Hence, the efficiency of feedback control schemes for different bit rates has to be investigated and the gain achieved by supporting the Available Bit Rate (ABR) service class, which uses rate-based control, has been evaluated.

Chapter 7 investigates the performance of a combined preventive/reactive congestion-control scheme using simulations, with emphasis on the utilisation improvement in supporting the ABR service class.

Finally, Chapter 8 will present the conclusions of this thesis and suggest directions for further work.
In Figure 1-1 the structure of the thesis is illustrated. Accordingly, Chapter 3 provides the traffic models. Then Chapter 4 provides the framework for the rest of the study and analyses the buffer scheduling and discarding strategy. This is followed in Chapter 5 by the optimisation of the preventive control function to limit the traffic entering the satellite network. Chapter 6 analyses the MAC which supports all QoS service classes by mapping of ATM service classes. Furthermore the statistical multiplexing capabilities of ATM are extended to the satellite air-interface. This is followed by Chapter 7 investigating the utilisation improvement achieved by supporting service classes which require feedback.
2. ATM OVER SATELLITE OVERVIEW

In this Chapter the reasons to choose ATM as transfer mode and satellite links as transmission medium are explained. Then the motivation to use ATM over satellite and related issues are discussed.

2.1 Asynchronous Transfer Mode (ATM)

Asynchronous Transfer Mode (ATM) has a feature which guarantees its success, namely the possibility to transport any service, irrespectively of its characteristics such as the bit rate, its quality requirements or its burstiness. This big advantage was one of the main motivations for the ITU-T to decide that ATM will be the transfer mode for the future Broadband Integrated Services Digital Network (B-ISDN).

ATM networks aim to combine the flexibility of packet-switched networks (Internet) with the Quality-of-Service (QoS) guarantees of circuit-switched networks (telephone network). Thus ATM networks have the potential to create a unified infrastructure that carries voice, video and data and will play a very important role in the near future.

Note that the phrase 'transfer mode' is used by ITU-T (formerly CCITT) to describe a technique which is used in a telecommunication network, covering aspects related to transmission, multiplexing and switching. In other words, the transfer mode defines how information supplied by network users is eventually mapped onto the physical network. The word asynchronous has been used for this transfer mode, since it allows asynchronous operation between the sender's clock and the receiver's clock. The difference between both clocks can easily be solved by inserting/removing empty/unassigned cells in the information stream (i.e. cells which do not contain useful information).

Asynchronous transfer of information was chosen in response to problems with the Synchronous Transfer Mode (STM) transmission used in telephone networks. In short, STM service is inflexible: circuits based on STM must have a bandwidth in multiples of a time slice, and a source wastes any portion of this bandwidth that it does not use. This problem is solved in a packet-switched network by introducing a header to describe, in particular, the data source and destination. The header allows intermediate switches to store packets, forwarding them when convenient. This allows
more complicated link sharing, since the receiver does not have to rely on a fixed frame format to decide the source of the data.

There are two ways to build a packet-switched network. The first is the datagram method, where each packet contains the full destination address. Addresses can be large, so if the average datagram length is small, this wastes bandwidth. In the second way to build a packet-switched network, packet headers carry identifiers instead of addresses, and each switch maintains a translation from the identifiers to a destination. This saves header space because identifiers are smaller than destination addresses. However, the mapping from an identifier to a destination must be set up at each switch along the path before data transmission begins. In other words a call set-up phase must precede a data transmission phase. This variant is called Virtual Circuit (VC) switching and is used by ATM.

**2.1.1 Advantages of ATM**

2.1.1.1 Flexible and future-safe

Advances in the state of the art of coding algorithms and VLSI technology may reduce the bandwidth requirements of existing services. New services may emerge with unknown characteristics. All these changes can be supported with success without modification of the ATM network and without loss of efficiency. The ATM systems (transmission, switching, multiplexing etc.) does not need to be modified.

2.1.1.2 Efficient in the use of its available resources

All available resources in the network can be used by all services, so that an optimal statistical sharing of the resources can be obtained. No resource specialisation exists in an ATM network, meaning that available resource can be used by any service.

2.1.1.3 One universal network

Since only one network needs to be designed, controlled, manufactured and maintained, the overall costs of the system may be smaller, due to economies of scale. These advantages will benefit all involved parties in the telecommunication world: customers, operators, and manufacturers.
The definition of ATM, is being finalised by ITU-T Study Group XVIII. This Chapter will describe the most important options taken by ITU-T.

2.1.2 Basic Principles of ATM

This section summarises the basic principles as put forward by ITU-T in Recommendation I.150 [ITUT95b].

ATM is considered as a specific packet oriented transfer mode based on asynchronous time division multiplexing and the use of fixed length cells. Each cell consists of an information field and a header. The header is primarily used to identify cells belonging to the same virtual channel within the asynchronous time division multiplex, and to perform the appropriate routing. Cell sequence integrity is preserved per virtual channel.

The information field of ATM cells is carried transparently through the network. No processing like error control is performed on it inside the network. All services (voice, video, data etc.) can be transported via ATM, including connectionless services. To accommodate various services, several types of ATM Adaptation Layers (AAL) have been defined, depending on the nature of the service, to fit information into ATM cells, and to provide service specific functions (e.g. clock recovery, cell loss recovery etc.). The AAL specific information is contained in the information field of the ATM cell.

2.1.3 ATM B-ISDN Protocol Reference Model

The Open Standard Interconnect (OSI) model of the International Standard Organisation (ISO) is very famous and used with success to model different communication systems. The same logical hierarchical architecture as used in OSI is used for the ATM B-ISDN network in Recommendation I.321 [ITUT91]. However, only the lower layers are explained. ITU-T has not yet determined the relation between ATM and OSI.

The B-ISDN Protocol Reference Model (PRM) for ATM is shown in Figure 2-1. As the N-ISDN PRM, it contains 3 planes: a user plane to transport user information, a control plane mainly composed of signalling information and a management plane, used to maintain the network and to perform operational functions. In addition, a third
dimension is added to the PRM, called the plane management, which is responsible for the management of the different planes.

![Diagram of the ATM/B-ISDN Protocol Reference Model](image)

**Figure 2-1 The ATM/B-ISDN Protocol Reference Model**

2.1.3.1 Physical Layer

The physical layer transports ATM cells between two ATM entities. It consists of two sublayers: the Physical Medium (PM) sublayer and the Transmission Convergence (TC) sublayer.

The physical medium sublayer includes only physical-medium dependent functions and provides bit transmission capability, including bit-transfer and bit-alignment. It includes line coding and electrical-optical transformation.

The transmission convergence sublayer performs all those functions necessary to transform a flow of cells into a flow of data units (e.g. bits) which can be transmitted and received over a physical medium.

Going from the physical layer to the ATM layer, the flow of data (in OSI terms, *service data units*) crossing this boundary is a flow of valid cells. Valid cells are those whose headers have no errors, error checking on the header having been performed in the transmission convergence sublayer.

Going in the opposite direction, from the ATM layer to the physical layer, the ATM cell flow is merged with the appropriate information for cell delineation and it also carries operation and maintenance information relating to this cell flow.
2.1.3.2 Asynchronous Transfer Mode (ATM) Layer

The ATM layer is fully independent of the physical layer used to transport the ATM cells. The following main functions are performed by this layer.

- The multiplexing and demultiplexing of cells of different connections into a single cell stream on a physical layer.
- Translation of the cell identifier, which is required in most cases when switching a cell from one physical link to another, in an ATM switch or cross connect. This translation can be performed either on a VCI or VPI separately, or on both simultaneously.
- Providing the user of a VCC or VPC with one QoS class, out of a number of classes supported by the network. Some services may require a certain QoS for one part of the cell flow of a connection and lower QoS for the remainder. The distinction between the connections is made by means of the CLP bit in the cell header.
- Management functions: the header of user information cells provides for a congestion indication and an ATM user to ATM user indication by using dedicated PTI codes.
- Extraction (addition) of the cell header before (after) the cell is delivered to (from) the adaptation layer.
- Implementation of a flow control mechanism on the user-network interface. This is supported by the GFC bits in the ATM cell header.

Recommendation I.361 [ITUT95c] describes the coding of ATM cells in detail. The cell structure finally selected by ITU-T contains a 48 octet information field and a 5 octet header. The octets are sent in an increasing order, starting with octet 1 of the header. Within an octet, the bits are sent in a decreasing order, starting with bit 8. For all fields of an ATM cell, the first bit sent is also the Most Significant Bit (MSB).

The ATM cell header consists of the following fields: Generic Flow Control (GFC), Virtual Path Identifier (VPI), Virtual Channel Identifier (VCI), Payload Type (PT), Cell Loss Priority (CLP), and Header Error Control (HEC).
GFC is a four-bit field providing flow control at the UNI for the traffic originated at user equipment and directed to the network, and does not control the traffic in the other direction (i.e. network-to-user traffic flow). The GFC field has no use within the network and is meant to be used by access mechanisms that implement different access levels and priorities. Accordingly, this field is used as a part of the VPI at NNIs, providing enhanced path-identification capabilities. Two modes of operation are defined for the GFC field: uncontrolled access and controlled access. The former is expected to be used in early ATM deployments and has no impact on the traffic users send to the network. In the case of controlled access, the flow rate of cells generated by the users is controlled at the UNI.

ATM is a connection-oriented technique, and virtual circuits are required to be established between the end nodes before transmission can start. As with any other packet-switching network, routing of cells is performed at every node for each arriving cell. VPI, an 8 or 12-bit field, together with VCI, a 16-bit field, contain the routing information of a cell. The header values are assigned to each section of a connection for the complete duration of the connection, and translated when switched from one section to another. If no sufficient resources are available the connection is refused to the requesting terminal.

Two sorts of connections are possible: Virtual Channel Connections (VCC) and Virtual Path Connections (VPC). A VPC can be considered as an aggregate of VCCs. When switching/multiplexing on cells is to be performed, it must first be done based
on the VPC, then on the VCC. This is shown in Figure 2-3. There we see an entity which only performs VP switching, and another entity which performs both VP and VC switching. However, the VP switching part may be idle, resulting in a pure VC switch.

**Figure 2-3 VC and VP Connections in ATM**

There are three bits in the ATM header to define the payload type. The seven values defined so far are given in Table 2-2. The other value of the PTI coding is reserved for a future function.

<table>
<thead>
<tr>
<th>PTI</th>
<th>Meaning</th>
</tr>
</thead>
<tbody>
<tr>
<td>000</td>
<td>User data cell, congestion not experienced, SDU type=0</td>
</tr>
<tr>
<td>001</td>
<td>User data cell, congestion not experienced, SDU type=1</td>
</tr>
<tr>
<td>010</td>
<td>User data cell, congestion experienced, SDU type=0</td>
</tr>
<tr>
<td>011</td>
<td>User data cell, congestion experienced, SDU type=1</td>
</tr>
<tr>
<td>100</td>
<td>Segment OAM flow-related cell</td>
</tr>
<tr>
<td>101</td>
<td>End-to-end OAM flow-related</td>
</tr>
<tr>
<td>110</td>
<td>Resource management cell</td>
</tr>
<tr>
<td>111</td>
<td>Reserved</td>
</tr>
</tbody>
</table>

**Table 2-1 Payload Type Indicators**

The CLP field of the ATM cell header is a 1-bit field used for cell-loss priority. Due to the statistical multiplexing of connections, it is unavoidable that cell losses will occur in B-ISDN. A cell with the CLP bit set may be discarded by the network during congestion, whereas cells with the CLP bit *not* set have higher priority and shall not be discarded if at all possible. Traffic control mechanisms therefore have to be designed to take account of the CLP.

The HEC field is used mainly for two purposes: to discard cells with corrupted headers and cell delineation. The 8-bit field, when used for the error check, provides single-bit error correction and a low-probability corrupted-cell delivery capabilities.
The HEC value is equal to the remainder of the division of the product \( x^8 \) and the polynomial of order 31 with coefficients being equal to the bits of the first four bytes of the header by the polynomial \( x^8 + x^2 + x + 1 \).

2.1.3.3 ATM Adaptation Layer (AAL)

The ATM Adaptation Layer (AAL) enhances the services provided by the ATM layer to a level required by the next higher layer.

AAL supports higher layer functions for the user, management and control planes. It also supports connections between ATM and non-ATM interfaces. Information received by the AAL from higher layers is segmented or collected to be inserted into ATM cells. Cells received by the AAL from the ATM layer are reassembled to form the information or read out.

The services which will be transported over the ATM layer are classified in 4 classes, each of which has its own specific requirements towards the AAL.

To obtain the 4 classes, the services are classified according to 3 basic parameters:

i) **Time relation between source and destination**: Some services have a time relation between source and destination, for some there is no such time relation. For instance, in 64 kbit/s PCM voice, there is a clear time relation between source and destination. On the other hand, information transfer between computers has no time relation. Sometimes services with a time relation are called real time services.

ii) **Bit Rate**: Some services have constant bit rate, others have a variable bit rate.

iii) **Connection Mode**: Services can be either connectionless or connection oriented.

Only 4 types out of theoretically 8 combinations of those 3 parameters result in valid existing services. Therefore ITU-T, has defined 4 classes, according to these basic parameters as described in Figure 2-4.

- **In Class A**, a time relation exists between source and destination. The bit rate is constant and the service is connection oriented. A typical example is voice of 64 kbit/s as in N-ISDN to be transported over ATM. The offering of this service over
an ATM network is also sometimes called circuit emulation. Another example is fixed bit rate video.

- In Class B, again a time relation exists between source and destination, for a connection-oriented service. However, the difference with Class A is that Class B sources have variable bit rate. Typical examples are variable bit rate video and audio.

- In Class C, there is no time relation between source and destination and the bit rate is variable. The service is connection oriented. Examples are connection-oriented data transfer and signalling.

- Finally, Class D differs from Class C in being connectionless. An example of such a service is connectionless data transport (e.g. switched multimegabit data services).

![Figure 2-4 Service Classes for Adaption](image)

Four types of AAL protocols have been recommended up to now by ITU-T, named AAL 1, AAL 2, AAL 3/4 and AAL 5. Recommendation I.362 [ITUT93b] states that CBR services will utilise AAL Type 1, but other AAL protocols for CBR are for further study. Connectionless data services will use the Type 3/4 AAL. Frame Relay services will use AAL 5. The specific association of the other services with an AAL type is still for further study.

**AAL Type 1**

Constant Bit Rate (CBR) services require information to be transferred between source and destination at a constant bit rate, after a virtual connection has been set up. The layer services provided by the AAL type 1 to the user are:

- Transfer of Service Data Units (SDU) with a constant source bit rate, and their delivery with the same bit rate.
ATM over Satellite Overview

- Transfer of timing information between source and destination.
- Transfer of data structure information.
- Indication of lost or corrupt information which is not recovered by the AAL itself, if needed.

A number of error indications, such as corrupted user information, loss of timing, buffer overflow and underflow, may be passed from the user plane to the management plane.

The SAR and CS sublayer functions are not in scope of this project, but can be found in detail in [PRYC93] for all AAL types.

AAL Type 2

The type 2 AAL offers a transfer of information with a variable bit rate. In addition, timing information is transferred between source and destination. Since the source is generating a variable bit rate, it is possible that cells are not completely filled, and that the filling level varies from cell to cell. Therefore, more functions are required in the SAR, which has not yet been defined by the ITU-R (only a possible way of the SAR has been mentioned).

AAL Type 3/4

ITU-T recommends the use of AAL 3/4 for transfer of data which is sensitive to loss, but not delay. The AAL may be used for connection oriented as well as for connectionless data communication. The AAL itself does not perform all functions required by a connectionless service, since functions like routing and network addressing are performed on the network layer. Two modes of AAL 3/4 are defined:

i) Message Mode

The AAL-SDU is passed across the AAL interface in exactly one AAL Interface Data Unit (IDU). This service is provided for transport of fixed or variable length AAL-SDUs.

ii) Streaming Mode

The AAL-SDU is passed in one or more AAL-IDUs. Transfer of these AAL-IDUs may occur separate in time. The service provides transport of long variable length...
AAL-SDUs. It also includes an abort service by which the discarding of a partially transferred AAL-SDU can be requested.

Both modes of service may offer the following peer-to-peer operational procedures:

i) **Assured Operation**
   Every SDU is delivered without any content modification caused by errors. Any corrupted or lost CS-PDU is retransmitted. In addition, flow control is supported between the endpoints. The use of this procedure may be restricted to point-to-point AAL connections.

ii) **Non-Assured Operation**
   In this case, an SDU may be delivered incorrectly or not at all. So, lost or corrupted CS-PDU are not retransmitted. The provision of flow control is optional.

**AAL Type 5**

According to the end-user equipment manufacturers and high speed, connection-oriented data service users, the AAL 3/4 as recommended by ITU-T is not really suited to their needs. The AAL 3/4 has a high overhead of 4 bytes per SAR-PDU of 48 bytes. Also the 10 bit Cyclic Redundancy Check (CRC) for detecting corrupted segments, and the 4 bit sequence number for detecting lost and misinserted segments, may not offer enough protection for conveying very long blocks of data.

Therefore the ATM Forum has specified a new type of AAL, called AAL 5, which also been adapted by the ITU-T. The objective is to offer a service with less overhead and better error detection below the CPCS layer. At this layer, the service of AAL type 5 shall be identical to the service provided by the CPCS of AAL type 3/4, except that no multiplexing is supported. If multiplexing is required at the AAL layer, it will occur in the SSCS layer. AAL type 5 can also be used for signalling across the UNI and NNI in the B-ISDN.
2.2 Satellite Networks

The principle advantages of satellite networks are their wide coverage and broadcasting capabilities. They enable fast extension of links to rural and remote areas. Satellite links are quick and easy to install irrespective of geographical constraints and are also not affected by natural disasters. Satellites have to be seen as complimentary to the terrestrial network and can also be used as back-up network.

2.2.1 Link Budget

The major consideration in planning an overall satellite link is the quality in the baseband. This is measured in terms of S/N for an analogue system and in terms of Bit Error Rate (BER) for a digital system. In both cases the quality of the link is proportional to the carrier to total noise (C/Nr) at the input of the receiver demodulator. The link budget is a calculation of the (C/N) power ratio at the receiving side of a transmission link, taking into consideration the transmission medium and the transmitter/receiver characteristics.

The received power $P_r$ is given by:

$$ P_r = G_t P_i G_r / L_{rs} \text{ where } L_{rs} = \left( \frac{4\pi R}{\lambda} \right)^2 $$

where $\lambda$ is the wavelength, $G_t$ the gain of the transmitting antenna in the receiver direction, $G_r$ the gain of the receiving antenna, and $R$ the distance between the satellite and the ground terminal. This means that the received power is inversely proportional to $R^2$.

The principal role of the receiver is to amplify the received signal to a usable level. Receiver noise, external background noise and interference from others transmitters corrupt this amplified signal. The received noise power spectral density is given by:

$$ N_0 = kT $$

where $T$ is the equivalent noise temperature of the receiving equipment (satellite)

$k$ the Boltzmann constant = $1.397 \cdot 10^{-23}$ W/Hz/K

Denoting the received carrier power level by $C$, the uplink carrier power-to-noise power spectral density ratio is expressed at:
ATM over Satellite Overview

\[(C / N_0)_u = EIRP_u \left( \frac{1}{L} \right)_u \left( \frac{G_r}{T_r} \right) \frac{1}{k} \]

where

\[(EIRP)_u = P_i G_i \] is the effective isotropically radiated power on the uplink

\[T_r \] the equivalent noise temperature of the receiving equipment (satellite)

\[L = L_{fs} + L_a \] the total loss: \( L_{fs} \) the free space loss, and \( L_a \) the additional losses, where \( L_a \) depends on \( L_{at} \), \( L_{pol} \), and \( L_{pol} \).

\( L_{at} \): attenuation by the atmosphere (which reach almost the geosynchronous altitude, but is really part of the first 1 000 km) and the ionosphere (from an altitude of 80 km to more than 180 km). As a consequence, LEO satellites have a slight advantage on both MEO and GEO satellites, which have the same \( L_{at} \).

\( L_{pol} \): losses caused by polarisation mismatch between the transmitting and receiving antennae; this mismatch is clearly more important for GEO than for LEO satellites, due to the distance.

\( L_{pol} \): represents losses caused by antenna depointing (ground terminal near the coverage boundary, errors in pointing, misalignment of the radio-electrical axis with the geometrical axis, imperfect satellite stabilisation, etc.). For a terrestrial observer, a geostationary satellite is immobile in the sky, so a GEO satellite has fewer \( L_{pol} \) losses than a LEO/MEO satellite.

Similarly, the downlink carrier power-to-noise power spectral density ratio is expressed at:

\[(C / N_0)_D = EIRP_D \left( \frac{1}{L} \right)_D \left( \frac{G_r}{T_r} \right) \frac{1}{k} \]

For the total link without interference from other systems the carrier power-to-noise power spectral density ratio is expressed at:

\[\left( \frac{C}{N} \right)_T^{-1} = \left( \frac{C}{N} \right)_u^{-1} + \left( \frac{C}{N} \right)_D^{-1} \]
2.2.2 Orbit Choice

At the highest level, the applicability of a satellite constellation to any system can be judged by considering the performance of the constellation in the following areas; path delay, complexity for networking and the number of satellites required.

A LEO system requires many satellites, especially for a line-of-sight system and also would require inter-satellite links to reduce the size of the ground segment. The major advantages of LEO systems are the very low path delay and attenuation of the signal, although inter-satellite handover will occur frequently and techniques to cope with this must be developed. A MEO system offers higher delays but a smaller and simpler constellation where individual satellite payloads will be much larger and more complex than in equivalent LEO systems. The problem of handover still exists, albeit at a lower rate. Finally, GEO systems offer a simplified user terminal due to the stationary satellite position with respect to the surface of the earth. However, problems of critical path delay and large path attenuation must be addressed, especially for real-time services.

Geostationary Orbit

- Geostationary systems require low satellite numbers for global coverage but introduce high delay and attenuation;
- no handover or tracking is required for personal terminals;
- coverage can be tailored to desired area/land mass by use of orbit positioning and spotbeams;
- low elevation angles at high altitudes makes communication difficult with small mobile terminals;
- Phased and fast deployment is possible. A single satellite can give good regional coverage.

Medium Earth Orbit

- satellite numbers in excess of 10 are required for global or near-global coverage, depending on elevation angles required;
- round-trip delay satisfies ITU recommendations for real-time interactive services (e.g. voice) once processing & network topology are included;
- non-geostationary satellites introduce the need for user terminal tracking of satellite and handover capability;
- a relatively low number of earth stations can be used for full connectivity;
- it is difficult to tailor coverage to the land masses by orbit specification alone. Results highlight minimum satellite elevation angle, coverage gaps and frequency co-ordination issues;
- phased deployment for inclined systems is difficult. At best this would be unattractive, and at worst temporal or regional coverage gaps would exist;

Low Earth Orbit
- the shortest latency and lowest signal attenuation is experienced with LEO;
- a large global capacity is possible but requires many satellites;
- a large number of satellites is needed to offer channel conditions comparative to a MEO constellation;
- polar LEO increases the number of satellites required compared with inclined LEO and redundant coverage would be effectively unused and wasted at extreme latitudes;
- very large handover overhead is involved with LEO systems.

Significantly more spotbeams are required on a satellite at MEO altitude than at LEO to obtain a given spotbeam size on the earth. This results in an increase in payload complexity and therefore cost, therefore the choice between a LEO or a MEO orbit could be based on the trade-off between the complexity of a MEO satellite payload and the number of satellites required for the LEO orbit.

Hybrid/Elliptical
- Hybrid systems enable different areas to be covered by separate satellite types and constellations (e.g. Matra Marconi Space’s WEST proposal, Hughes Spaceway GEO and MEO proposal, etc.)
- Elliptical eccentricity is limited by delay constraints, so a major advantage gained in elliptical broadcast systems is lost. Extensive geographic coverage and near-stationary properties are realised at apogee;
- for global coverage only a small improvement on MEO numbers of satellites is gained;
• satellite payload and control network will have to compensate for the variation in altitude, elevation angles, attenuation, delay.

2.2.3 On-Board Processing (OBP)

OBP is in itself a vast domain that is the subject of much activity in the USA, Japan, and Europe. All commercial civil satellites to date have used transparent transponders which consist of nothing more than amplifiers, frequency changers and filters. These satellites adapt to changing demands, but at what cost? The cost has been high space segment tariffs and high-cost, complex earth terminals. OBP aims to put the complexity in the satellite and to reduce the cost of the use of the space segment and the cost of the earth terminals. This is not without problems (details in [EVAN91]) and varying degrees of processing can be applied:

• regenerative transponder (modulation and coding)
• on-board switching
• access format conversion (e.g. FDMA-TDM)
• flexible routing.

They may not all be present in one payload and the exact mix will be a function of the application.

The advantages rendered by the use of OBP are summarised [EVAN91]:

• Regenerative transponders

The advantage of the regenerative scheme is that the up-link and down-links are now separated and can be designed independently of each other. With conventional satellites \((C/N)_U\) and \((C/N)_D\) is additive, with regenerative transponders they are separated. This can be translated into an improved BER performance as reduced degradation are now present. Regenerative transponders can withstand much higher levels of interference for the same overall \((C/N)_R\).

• Multirate communications

With OBP it is possible to convert on the satellite between low- and high-rate terminals. This allows ground terminals operating at various rates to communicate with each other via a single hop. Transparent transponders would require rate
conversion terrestrially and hence necessitate two hops. Multirate communications implies both multicarrier demodulators and baseband switches.

- **Reduced complexity earth-stations**

The effects of employing OBP on ground terminals can be summarised as follows:

- Lower earth-station transmit power/gain due to the reduced \( (C/N)_U \)

- Reduced complexity receivers as the TDM downlink means no burst mode demodulators.

These add up to much reduced complexity and cheaper ground terminals.

**2.2.4 On-Board Switching (OBS)**

On-board processing (OBP) satellites with high-gain multiple spot-beams and on-board switching (OBS) capabilities have been considered as key elements of new-generation satellite communications systems. These satellites support small, cost-effective terminals and provide the required flexibility and increased utilisation of resources in a bursty multimedia traffic environment.

Although employing an on-board switch function results in more complexity on-board the satellite, the following are the advantages of on-board switches.

- Lowering the ground station costs.
- Providing bandwidth on demand with half the delay.
- Improving interconnectivity.
- Offering added flexibility and improvement in ground link performance, i.e., this allows earth stations in any uplink beam to communicate with earth stations in any downlink beam while transmitting and receiving only a single carrier.
- In-band signalling for combined traffic and TT+C operation

One of the most critical design issues for on-board processing satellites is the selection of an on-board baseband switching architecture. Four types of on-board switches are proposed:

- Circuit switch
- Fast Packet switch
ATM over Satellite Overview

- Hybrid switch
- Cell switch (ATM switch)

These have some advantages and disadvantages, depending on the services to be carried which are summarised in Table 2-2.

<table>
<thead>
<tr>
<th>Switching Architecture</th>
<th>Circuit switching</th>
<th>Fast packet switching</th>
<th>Hybrid switching</th>
<th>Cell switching (ATM switching)</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Advantages</strong></td>
<td>• Efficient bandwidth utilisation for circuit switched traffic.</td>
<td>• Self-routing</td>
<td>• Handles a much more diverse range of traffic</td>
<td>• Self-Routing with a small VC/VP</td>
</tr>
<tr>
<td></td>
<td>• Efficient if network does not require frequent traffic reconfiguration</td>
<td>• Does not require control memory for routing</td>
<td>• Optimisation between circuit switching and packet switching</td>
<td>• Does not require control memory for routing</td>
</tr>
<tr>
<td></td>
<td>• Easy to control congestion by limiting access into the network</td>
<td>• Transmission without reconfiguring of the on-board switch connection</td>
<td>• Lower complexity than fast packet switch</td>
<td>• Transmission without reconfiguring on-board switch connection</td>
</tr>
<tr>
<td></td>
<td>• Provides flexibility and efficient bandwidth utilisation for packet switched traffic</td>
<td>• Easy to implement autonomous private networks</td>
<td>• Can provide dedicated hardware for each traffic type</td>
<td>• Easy to Implement Autonomous Private Networks</td>
</tr>
<tr>
<td></td>
<td>• Can accommodate circuit-switched traffic</td>
<td>• Provides flexibility and efficient bandwidth utilisation for all traffic sources</td>
<td>• Can accommodate circuit-switched traffic</td>
<td>• Provides flexibility and efficient bandwidth utilisation for all traffic sources</td>
</tr>
<tr>
<td><strong>Disadvantages</strong></td>
<td>• Reconfiguration of earth station time/frequency plans for each circuit set-up</td>
<td>• For circuit switched traffic higher overheads than circuit switching due to packet headers.</td>
<td>• Can not maintain maximum flexibility for future services because the future distribution of satellite circuit and packet traffic is unknown</td>
<td>• Can accommodate circuit-switched traffic somewhat higher overheads than packet switching due to 5 byte ATM header.</td>
</tr>
<tr>
<td></td>
<td>• Fixed bandwidth assignment (not flexible)</td>
<td>• Contention and congestion may occur</td>
<td>• Waste of satellite resources in order to be designed to handle the full capacity of satellite traffic</td>
<td>• Contention and congestion may occur</td>
</tr>
<tr>
<td></td>
<td>• Very inefficient bandwidth utilisation when supporting packet-switched traffic</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Difficult to implement autonomous private networks</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Table 2-2 Comparison of various switching techniques

From an efficiency-of-bandwidth point of view, circuit-switching is advantageous under the condition that the major portion of the network traffic is circuit-switched. However, for bursty traffic, circuit-switching results in a lot of wasted capacity.
Fast packet switching may be an attractive option for a satellite network carrying both packet-switched traffic and circuit-switched traffic. The bandwidth efficiency for circuit-switched traffic will be slightly less due to packet overheads.

In some situations, a mixed-switch configuration, called hybrid switching and consisting of both circuit and packet switches, may provide an optimal on-board processor architecture. However, the distribution of circuit- and packet-switched traffic is unknown, which makes the implementation of such a switch a risk.

Finally, fixed-size fast-packet-switching, called cell-switching, is currently an attractive solution for both circuit- and packet-switched traffic. Using statistical multiplexing of cells, it could achieve the highest bandwidth efficiency despite a relatively large header overhead (5 bytes) per cell (53 bytes).

In addition, due to on-board mass and power-consumption limitations, the ATM switch is especially well suited to satellite switching because of the sole use of digital communications, and VLSI digital circuits limit mass and power consumption. Since ATM is also the recommended transfer mode for B-ISDN, it is apparent that the use of ATM switching for satellite systems is attractive, making seamless integration of terrestrial/satellite networks possible.

OBS introduces some issues for the design and analysis of the satellite architecture. Many considerations previously the concern only of the ground segments now shift to the space segment. The on-board processor allocates bandwidth on demand and performs statistical multiplexing. This essentially changes the nature of the satellite from a deterministic system to a stochastic system. In a stochastic system, the arriving traffic is random and statistical fluctuations may cause congestion, where cell loss due to buffer overflow might occur. Thus, it is necessary to incorporate traffic and control mechanisms to regulate the input traffic.

2.2.5 Inter-Satellite Links (ISLs)

The use of inter-satellite links (ISL) for traffic routing has to be considered. It has to be justified that this technology will bring a benefit which would make its inclusion worthwhile or to what extent on-board switching, or some other form of packet switching, can be incorporated into their use.
The issues which need to be discussed when deciding on the use of ISLs include:

- networking considerations (coverage, delay, handover)
- the feasibility of the physical link (inter-satellite dynamics)
- the mass, power & cost restrictions (link budget)

The mass and power consumption of ISL payloads are factors in the choice of whether to include them in the system, in addition to the possible benefits and drawbacks. Also, the choice between RF and optical payloads is now possible as optical payloads have become more realisable and offer higher link capacity. The tracking capability of the payloads must also be considered, especially if the inter-satellite dynamics are high. This may be an advantage for RF ISL payloads.

**Advantages of ISLs**

- calls may be grounded at the optimal ground station through another satellite for call termination, reducing the length of the terrestrial 'tail' required.
- a reduction in ground-based control may be achieved with on-board baseband switching - reducing delay (autonomous operation).
- increased global coverage - oceans & areas without ground stations.
- single network control centre and earth station.

**Disadvantages of ISLs**

- complexity and cost of the satellites will be increased
- power available for the satellite/user link may be reduced
- handover between satellites due to inter-satellite dynamics will have to be incorporated
- replenishment strategy
- frequency co-ordination
- cross-link dimensioning

### 2.2.6 Satellite Access Techniques

This section examines the techniques which are available for allowing a number of users to communicate via a common satellite transponder. This is broadly termed 'multiple access', where the transponder's available power and bandwidth are shared between a number of different channels and earth stations, which may themselves
have quite different transmit powers and signal characteristics. Efficient use of these resources is important, while meeting the need of the user's traffic demands.

There are three principal forms of fixed multiple access:

- Frequency Division Multiple Access (FDMA)
- Time Division Multiple Access (TDMA)
- Code Division Multiple Access (CDMA)

Each has its advantages and disadvantages, and all of the above are in current use, together with hybrid schemes.

2.2.6.1 Frequency Division Multiple Access (FDMA)

FDMA is a traditional and popular technique, whereby several earth stations transmit simultaneously, but on different preassigned frequencies into a transponder.

Preassigned FDMA was attractive because of its simplicity and cheapness. Single Channel Per Carrier (SCPC) FDMA was commonly used for very thin-route telephony, VSAT systems and mobile services, but now it is too inflexible for bursty applications which have varying bandwidth requirements. In order to use bandwidth efficiently 'demand assignment' is commonly used (demand-assigned FDMA).

Access by multiple carriers, as in FDMA, can give significant problems with Inter-Modulation-Products (IMPs), and hence a few dBs of back-off from saturation is required. The resultant reduction in downlink EIRP may represent a penalty, especially when working to small terminals.

2.2.6.2 Time Division Multiple Access (TDMA)

In TDMA, each user is allocated a time slot in which information can be transmitted. The information burst of each user, together with reference bursts which provide synchronisation information, make up a TDMA frame. The simplest system has fixed assignment of slots (fixed-TDMA), although demand-assignment schemes can be very flexible, with allocation of burst length as required.

Only one TDMA carrier accesses the satellite transponder at a given time, and the full downlink power is available for that access. TDMA can achieve efficiencies in power utilisation of 90 percent or more and similar efficiencies in bandwidth utilisation.
because the guard time loss in efficiency can be kept small by accurate timing techniques. The high bandwidth utilisation of TDMA is the reason why it is widely used.

Clearly TDMA bursts transmitted by ground terminals must not interfere one with another. Therefore each earth station must be capable of first locating and then controlling its transmit burst time phase. Each burst must arrive at the satellite transponder at a prescribed time relative to the reference burst. This ensures that no two bursts overlap and that the guard time between any two bursts is small enough to guarantee a high transmission efficiency.

Synchronisation is the process of providing timing information at all stations and controlling the TDMA bursts so that they remain within their prescribed slots. All this must operate even though each Earth station is at a different distance from the satellite for instance, and considering the motion of the satellite with respect to the Earth.

GEO satellites are located at a nominal longitude and typically specified to remain within a “window” with sides of 0.1 degree as seen from the centre of the Earth. Moreover, the satellite altitude varies as a result of a residual orbit eccentricity of about 0.001. The satellite can thus be anywhere within a volume of space which is typically 75 km · 75 km · 85 km.

The tidal movement of the satellite causes an altitude variation of about 85 km, resulting in around trip delay variation of about 500 μs and a frequency change of signals due to the Doppler effect.

One can easily understand that the TDMA co-ordination is all the more complex as the satellite is not fixed in the sky (LEO or MEO satellites), which means that the propagation delay varies. As a consequence, TDMA is harder to implement for LEO satellites than for GEO satellites.

**TDMA Buffers and Timing Control**

Since the bit streams entering the ground terminal are continuous while the output of the TDMA modulator is bursts according to a time plan, the TDMA modem must contain a data buffer. This buffer stores the data bits received from one frame until the
next. The total storage required is $M$ bits for $N$ input bit streams of bit rate $f_{d_i}$ and frame period $\tau_f$, where

$$M = \sum_{i=1}^{N} f_{d_i} \tau_f$$

TDMA timing at a ground terminal can be slaved either to an actual clock on-board the satellite or to an earth terminal clock at a terminal designated as the master.

**TDMA Frame Rates and Formats**

The format of a TDMA frame can have many different formats which will be discussed in more detail in Chapter 7. A superframe of $N$ frames can be used to allow for some very low data-rate users desiring to transmit at a rate below the frame rate. The frame rate, for example might be 84 frames/sec, and a user terminal $i$ who desires to transmit at 32 kbit/s would transmit on the average on the average 384 bits (ATM cell payload) per frame. On the other hand if a user terminal wants to transmit at an average of 8 kbit/s it would transmit 384 bits per 4 frames. However, most users transmit one data burst per frame plus perhaps a timing burst for synchronisation.

Traditional TDMA bursts were subdivided into a preamble for receiver synchronisation, data bits addressed to various receive terminals and postambles that identify the end of the burst (to resolve carrier phase and frequency ambiguities). Currently preambleless modems are available that do not require pre- and postambles, minimising the burst overhead [CELA97]. Guard times ($T_g$) are required at the beginning of each burst, to prevent overlap of adjacent bursts from different ground terminals. This guard time (typically from 30nsec-300nsec) must be sufficient to account for system timing inaccuracies and tails from adjacent bursts caused by finite filter-response times.

**TDMA System Efficiency**

The efficiency of the satellite transponder with TDMA inputs and hard limiting depends on the guard times between the transmissions $T_{gi}$ of each terminal, the preamble and postamble times and the addressing time required for each transmit/receive terminal pair $T_{adj}$, the time utilised for timing-ranging function $T_R$, and the frame duration $T_f$. The maximum efficiency for all terminals occupying the frame is:
where $i$ is summed over all $N$ terminals in the network and $j$ is summed over all $N-1$ which can be addressed by the terminal. If all guard times and address times are identical, then the maximum efficiency is

$$\eta_{\text{max}} = \frac{T_f - [T_R + \sum_i T_{Ri} + \sum_j T_{ij}]}{T_f}$$

where it is assumed that all terminals are communicating with all the other terminals and the frame is fully utilised. The inefficiencies of the data channels due to channel coding are not considered in this calculations.

### 2.2.6.3 Code Division Multiple Access (CDMA)

CDMA is an access technique employing spread spectrum modulation, where each channel is modulated with a unique spreading function. The resulting wide-band signals from all users may be overlaid in a common RF bandwidth, and employ the same carrier frequency simultaneously. It is also known as Spread Spectrum Multiple Access (SSMA). A feature of spread spectrum is that operation is possible in the presence of high levels of uncorrelated interference, and this property of spread spectrum has important antijamming applications in military communications.

Each CDMA user combines his data signal of a few kHz bandwidth with a very wide-band spreading function. This has a bandwidth of typically several MHz and is derived from a pseudo-random code sequence, and the resulting transmitted signal then occupies a similar wide bandwidth. At the receiver, the input signal is correlated with the same spreading function, suitably synchronised, to reproduce the originating data. At the receiver output, the small residual correlation products from unwanted user signals amount to additive noise, known as self-interference. As the number of users in the system increases, the total noise level will increase and degrade the bit-error-rate performance. This will give a limit to the maximum number of simultaneous channels which can be accommodated within the same overall frequency allocation, and it can be shown that, theoretically, CDMA is inferior to FDMA or TDMA in terms of capacity for a given power and bandwidth. In practice
the performance can be superior to FDMA allowing for the latter’s limitation of guard bands and TWTA back-off. There is no need for network timing references as in TDMA, and speech duty cycles may be readily exploited. The overall merits or otherwise of CDMA are very scenario dependant, and the subject of considerable debate.

More details about fixed access, reservation access, random access and hybrid access will be provided in Chapter 7.

2.3 ATM over Satellite

A broadband network (such as B-ISDN) is needed as a result of recent developments in multimedia services. These services will have diverse traffic characteristics and Quality of Service (QoS) requirements. ATM can support all these services in an integrated manner, while providing the required QoS. This was the reasons why it was chosen as means of transport for the B-ISDN.

Fibre optic links are the preferred carrier for terrestrial ATM, however satellite systems play an important role in extending the B-ISDN to remote/rural areas where the deployment of terrestrial infrastructure is not economical.

Providing ATM over satellite has a number of advantages: a satellite can cover a large area, is immune to terrestrial disasters, provides bandwidth-on-demand capabilities and can be accessed at a relatively low cost by a large number of users with small terminals. ATM over satellite should be seen as complimentary to the terrestrial ATM, providing global multimedia services to anybody, anywhere, anytime. Apart from extension of ATM services over satellite, ATM brings benefits to the wireless environment, providing statistical gain, thereby promoting effective use of the available spectrum.

The role of satellites in high-speed networking will evolve according to the evolution of the terrestrial ATM-based B-ISDN. However two main roles for satellites can be identified:

- In the introductory phase, satellites will compensate for the lack of sufficient terrestrial high-bit-rate links mainly by interconnecting a few regional or national distributed broadband networks, usually called 'Broadband Islands'.
• In the mature phase, after the terrestrial broadband infrastructure will have reached some degree of maturity, satellites are expected to provide broadcast service and also cost-effective links to rural areas complementing the terrestrial network. In this phase satellite networks will provide broadband links to a large number of end users through a User Network Interface (UNI) for accessing the ATM B-ISDN. Satellites are also ideal for interconnecting mobile sites and provide a back-up solution in case of failure of the terrestrial systems.

2.3.1 Enhancement Techniques for Satellite ATM

ATM was designed for transmission on a physical medium with excellent error characteristics, such as optical fibre. Therefore, many of the features included in older protocols that cope with an unreliable channel are no longer part of ATM. While this results in considerable advantages (less overhead, increased throughput) in an optical network, it also causes severe problems when ATM is transmitted over an error-prone channel, such as the satellite link.

A geostationary satellite channel is often modelled as an Additive White Gaussian Noise (AWGN) channel. While the AWGN, which produces *random single bit errors*, is only an approximation, it is widely used and fairly accurate in many situations.

Satellite systems are usually power or bandwidth limited and in order to achieve reliable transmission Forward Error Correction (FEC) codes are often used in satellite modems. With such codes (typically convolutional codes), the incoming data stream is no longer reconstructed on a symbol by symbol basis. Rather some redundancy in the data stream, which generally increases the bandwidth, is used.

On average, coding reduces the bit error rate or alternatively decreases the transmission power needed to achieve a certain QoS for a given S/N ratio, at the expense of coding overhead. However, when a decision is made for a wrong data sequence, in general a large number of bits is affected, resulting in *burst errors*. Because ATM was designed to be robust with respect to random single bit errors, burst errors considerably degrade the performance of ATM.

Hence some enhancement techniques have to be used to make the transmission of ATM cells over the satellite link more robust. Different methods have been proposed
which are applicable for two scenarios. The performance of these scheme is directly related to the code rate (bandwidth efficiency) and/or the coding gain (power efficiency), provided the delay involved is acceptable to any ATM-based application.

2.3.1.1 Enhancement Techniques for Broadband Satellite ATM

Most recent research had focused on Scenario 1 for large earth stations operating at data rates higher than 2 Mbit/s. The underlying poor performance of ATM on satellite links is that an undesired concatenation of the convolutional channel code (FEC, inner code) and the ATM HEC code (outer code) takes place. Since the outer HEC code is only capable of correcting single bit errors, the errors at the output of the inner code should be dispersed by using an interleaver.

By interleaving the ATM cell headers (not the payload) of several cells, so called inter-cell interleaving, the performance of ATM in a random single bit error channel (e.g., AWGN channel) can be achieved [CHIT94]. Note that interleaving merely re-shuffles the bits on the channel (to spread the bit errors among ATM cell headers) and does not produce additional overhead which might decrease the overall bit rate. However, interleaving requires memory at the transmitter and the receiver, and it introduces additional delay. Assuming an average number of 30 bit errors in an error burst [KALT95], interleaving over 100 cell headers seems to be sufficient. This requires a memory of only about 10 kbytes and introduces a delay of 840 µs at 50 Mbit/s and a delay of 21 ms at 2Mbit/s. The performance improvement achieved by interleaving has been confirmed in several studies [AGNE93, KALT95, FAIR97]. Since the above interleaving scheme requires a continuous data stream, there are problems using it for portable terminals/USATs where single ATM cells may be transmitted.

Another way of correcting the burst errors due to FEC techniques applied to satellite links are Reed-Solomon (RS) codes. This type of block codes, which are based on symbols, have been identified as performing particularly well in concatenation with convolutional FEC codes, because of their ability to correct bursts of errors [AGNE96a].
Moreover, error bursts longer than what the RS code can correct should be spread over several blocks to take advantage of the error correction capabilities of the block code. This can be done by interleaving between the two codes.

2.3.1.2 Enhancement Techniques for Wideband Satellite ATM

For users to whom economical, rapid deployment and relocation is an important requirement, it is only practical to use smaller earth stations such as portable terminals/USATs. There is no clear definition of wideband, but we will define it as bit rates of 2.048 Mbit/s and below.

Since inter-cell interleaving is not feasible because only few cells may be transmitted from the terminal, mechanisms which protect single cell have to be found. Interleaving within an entire ATM cell (not only the header), so-called intra-cell interleaving, leads to a performance gain which is too small to be effective.

A better improvement can be achieved by using additional coding to protect the ATM cells. Note that this introduces additional overheads and therefore reduces the useful data bit rate. There are several reasons why FEC or concatenated FEC may not be suitable for enhancing ATM performance over wideband satellite links. First, if only FEC coding is used, than symbol interleaving is usually used to spread the burst errors over several ATM cell headers. The resulting interleaving delay (which is inversely proportional to the data rate) may be too large at a low rate for certain applications. Second if RS codes are used to correct burst of errors in concatenation with FEC either additional bandwidth has to be provided or the data rate has to be reduced.

The latest proposal for improvement of ATM performance for wideband satellite links is the construction of enhancing equipment which optimises the ATM protocols over a satellite link. This allows the data link layer to be optimised using a combination of protocol conversions and error control techniques. This approach allows commercial off-the-shelf ATM equipment to be used. At the transmitter standard ATM cells are modified to suit the satellite link. At the receiver, error recovery techniques are performed and the modified ATM cells (S-ATM cells) are converted into standard ATM cells.
The main aim of modifying standard ATM cell is to minimise the rather large ATM header overhead which is 5 bytes per 48 byte payload. Of the ATM header information, the address field (which is divided into the VPI and VCI) occupies 24 bits. This allows up to 16 million VC to be set up. Considering that in particular CBR connection cells all carry the same address information in the header, there may be methods not to duplicate the same information. Furthermore, wideband satellite links cannot support 16 million virtual connections and the use of 24 bits for address space may be considered a waste of bandwidth for this scenario.

One method to protect the ATM cell header when interleaving is not possible is the compression of the 24 bits address space to 8 bits and to store the duplicate header information (except the HEC field) of the previous cell [FAIR97]. The HEC is still computed over the first 4 bytes of the header and inserted into the fifth byte of the header. Therefore if a cell header contains errors, the receiver can store the payload in a buffer and recover the header information from the next cell provided that its header does not also contain errors. This method is only effective if the cell containing errors in the cell header has no errors in the payload. Simulations [FAIR97] show that this method provides considerable improvements in CLR compared to standard ATM transmission and even compared to interleaving.

Another alternative is to use a 3 byte HEC instead of a 1 byte HEC, which is inadequate for the satellite environment. For the mentioned (56,32) error correction scheme, candidate codes include a shortened form of the four-bit-error correcting (63,39) binary BCH code and a shortened form of the three-symbol-error correcting (15,9) RS code (with 4-bit symbols). There is possibly some preference for the latter from both implementation and performance considerations [MATR98].

### 2.3.2 Availability Considerations

Rec. I.356 [ITUT96a] provides the QoS class definitions and end-to-end network performance objectives. These objectives are given, for each performance parameter, as 'upper bound' that need to be met on a VC or VP for the duration of the connection. I.356 [ITUT96a] makes no reference to the ATM availability requirements although some preliminary ideas are given in draft Rec. I.357 [ITUT96c].
The total availability of the satellite network \( A_{\text{sat}} \) is dependent on the availability of the satellite \( A_{\text{satellite}} \), the availability of the satellite link \( A_{\text{propagation}} \) and the availability of the satellite resources \( A_{\text{congestion}} \).

\[
A_{\text{total}} = A_{\text{satellite}} \cdot A_{\text{propagation}} \cdot A_{\text{congestion}}
\]

From a dependability point of view, a portion of a B-ISDN ATM semi-permanent connection should have the following properties:

- The fraction of time during which it is in a down state (i.e. unable to support a connection) should be as low as possible,
- Once a connection has been established, it should have a low probability of being either terminated (because of insufficient data transfer performance) or prematurely released (due to the failure of a network component).

Availability of a B-ISDN ATM semi-permanent connection portion is defined as the fraction of time during which the connection portion is able to support a connection. Conversely, unavailability of a portion is the fraction of time during which the connection portion is unable to support a connection (i.e. it is in the down state) [ITUT96c]. A common availability model which is also used in [ITUT96c] is depicted in Figure 2-5.

The model uses four states corresponding to the combination of the ability of the network to sustain a connection in the available state and the actual use of the connection.

Two independent perspectives are evident from the model:

1. The Service perspective, where availability performance is directly associated with the performance perceived by the user. This is represented in Figure 2-5 by states 1 and 2, even in the case of an on/off source since the user is only concerned with the connection availability performance whilst attempting to transmit packets.
2. The Network perspective, where availability performance is characterised independently of user behaviour. All four states in Figure 2-5 are applicable.

Both service and network perspectives will be considered.
The criteria for entry into the unavailable state are for further study. In [ITUT96c] the onset of unavailability begins after 10 severely errored seconds (SES_{ATM}). These 10 seconds are part of unavailable time. A period of unavailability shall end with the occurrence of 10 consecutive seconds, none of which are SES_{ATM}. These 10 seconds are part of the available time. The 10-seconds criteria are supported using a sliding window with one-second granularity. A portion of a bi-directional B-ISDN connection is only available if both directions are available.

Two availability parameters are defined[ITUT96c]:

**Availability Ratio (AR):**

The service AR is defined as the portion of time that the connection portion is in the available state over an observation period, where the connection is in use. This is characterised in Figure 2-5 by the proportion of time in State 1 compared to the overall time in States 1 and 2.

The network AR is defined as the proportion of time that the connection is in the available state over an observation period, where the connection may not be in use. This is characterised in Figure 2-5 by the proportion of time in States 1 and 3 compared to the overall time in States 1 to 4.
Mean Time Between Outages (MTBO):

The service MTBO is defined as the average duration of a time interval during which the connection is available from the service perspective. Consecutive intervals of available time during which the user attempts to transmit cells are concatenated.

The network MTBO is defined as the average duration of a continuous time interval, during which the connection is available from the network perspective.

More information about availability and can be found in [ITUT96c]. As mentioned before, the QoS limits must be guaranteed to the end-user for the duration of the connection. This means that the 'upper bounds' should not be exceeded for the 'available time'. The availability for satellite links should also comply with any other network availability objective. It is therefore proposed [ITUT96c] that ATM satellite links that are designed to carry all QoS classes should offer the highest availability possible. In this thesis it is assumed that the QoS objectives are met for 99.8% of time (any month), or equivalently 99.96% of the year (in accordance to Rec. S.614 [ITUR96a] and S.1062 [ITUR96b]).

2.4 Discussions and Summary

Satellite communications (satcom) as a commercial enterprise is just over twenty-five years old. Until recently all commercial communications satellite used the Geostationary (GEO) orbit. The main reason for this is that a small number of satellites are sufficient for global coverage. Furthermore phased and fast deployment is possible. A single GEO satellite with a global beam will cover an arc of 18.080 km along the equator and it will radiate up to 81.35' of latitude on either side of the equator.

The only drawback of GEO satellites is the inherent long propagation delay which makes providing interactive real-time services difficult. Hence the recent non-GEO satellite constellation filings such as Teledesic and SkyBridge [FERN97, BULL97].

In this thesis GEO satellites are used because of their fast deployment and large coverage. The problems encountered under high propagation delays will be analysed and sometimes performance comparisons with MEO/LEO orbit provided.
OBP which aims to put complexity on the satellite to reduce the cost of the use of the space segment and the cost of the terminal will be used. Fixed-size fast-packet-switching, called cell-switching, is currently an attractive solution for both circuit- and packet-switched traffic. Using statistical multiplexing of cells, it could achieve the highest bandwidth efficiency despite a relatively large header overhead. Since ATM is also the recommended transfer mode for B-ISDN, ATM switching will be used for the satellite system in this thesis making seamless integration of terrestrial/satellite networks possible.

The multiple access scheme which will be used in this work will be TDMA because of the high throughput it provides compared to the other access schemes and because of its widespread use.

After providing the background on ATM and satellites the motivation for ATM over satellite was described. The role of satellites in the B-ISDN era and the physical impairments of the satellite link which have to be taken into account were explained. Proposed enhancement techniques to make the transmission of ATM cells over the satellite link more robust were reviewed. It was concluded that, for the direct access scenario, standard ATM cells have to be modified at the transmitter to suit the satellite link. At the receiver the modified ATM cells (S-ATM cells) are converted into standard ATM cells. The motivation for modifying standard ATM cells is to minimise the rather large ATM cell header overhead and maximise the error correction capability of the inner and/or outer (HEC) codes. Availability considerations were also dealt with in this Chapter. It was concluded that satellite systems must guarantee the QoS objectives during the satellite link availability. Specifically, since the ATM objectives need to be met during the duration of the connection, the satellite link must also be available during that period. Since the aim of this thesis is not to do research on the physical layer but on traffic control for the ATM layer, it is assumed that the QoS objectives are met for 99.8% of time (any month), or equivalently 99.96% of the year (in accordance to Rec. S.614 [ITUR96a] and S.1062 [ITUR96b]).
3. SOURCE MODELLING IN ATM NETWORKS

The aim of this research is to optimise traffic and congestion control functions for satellite ATM networks. In order to achieve this challenging task it is necessary to analyse and/or simulate such a network before actually implementing it. Modelling services using appropriate source models is therefore very important to enable simulations and analysis to be performed with the selected network architecture.

Source modelling is used to mimic the behaviour of a source. Traffic modelling on the other hand focuses on aggregated traffic patterns. Multiplexed models will capture the effects of statistically multiplexing bursty sources and will predict to what extent the superposition of bursty stream is smoothed. Hence traffic models will be used for designing connection admission control algorithms and for traffic engineering.

The most important application of source models is in predicting the QoS that a particular application might experience during different levels of congestion. Since this thesis is focusing on the QoS provided to individual services, source models will be used extensively. In some cases an aggregation of sources is also used with the assumption that homogeneous sources require the same QoS. Finally, some applications might want to use feedback control to ensure that minimal traffic is lost during periods of network congestion. Source models can be used to test various rate control algorithms.

3.1 Introduction

ATM networks are expected to support a diverse set of applications with a wide range of characteristics. Unfortunately, for the time being, there are no comprehensive measurements to satisfactorily address the characteristics of various types of B-ISDN applications in a realistically accurately.

Source characterisation at the macro level is defining the source traffic characteristics and its QoS requirements. The traffic characteristics of an application are the minimum set of parameters that a user can be expected to declare while providing the network with as much information as possible to effectively control network traffic and achieve high source utilisation.
This Chapter first provides information about traffic parameters used to describe the traffic characteristics of a source. Then QoS parameters which the user can negotiate with the network are explained.

After defining various parameters which can be negotiated between user and network, service categories supported by ATM networks are identified. Each service category can support various services depending on which traffic parameters can be declared and which QoS guarantees are required by the user. The statistical behaviour of generic traffic services such as voice, video, data and multimedia are provided and based on this information the criteria in selecting source models for traffic sources are explained. Finally a review of widely used source models, and their advantages in modelling particular traffic characteristics is provided.

3.2 Source Traffic Parameters and Descriptors

Source traffic parameters are used to describe traffic characteristics of a source. They may be quantitative or qualitative (e.g. telephone, videophone). For an ATM connection, traffic parameters are grouped into a source traffic descriptor, which in turn is a component of a connection traffic descriptor.

A source traffic descriptor is the set of traffic parameters of the ATM source. It is used during the connection set-up to capture the intrinsic traffic characteristics of the connection requested by a particular source. The set of traffic parameters in a source traffic descriptor can vary from connection to connection. A connection traffic descriptor characterises a connection at the User Network Interface (UNI). It consists of:

- Source traffic descriptor
- Cell Delay Variation Tolerance (CDVT)
- Conformance definition

The connection traffic descriptor is used by the network during connection set-up to allocate network resources and derive parameters for UPC. The conformance definition is used by the UPC to distinguish conforming and nonconforming cells without ambiguity.
An important issue is the set of traffic parameters to include in the source traffic descriptor. All parameters should be simple to be determinable by the user, interpretable for billing, useful to CAC for resource allocation, and enforceable by UPC. The set should be small but sufficient for the diverse types of traffic in B-ISDN.

Some proposed source traffic parameters which will be explained in detail are:

- Peak Cell Rate (PCR, p) and Cell Delay Variation Tolerance (CDVT)
- Sustainable Cell Rate (mean cell rate, SCR, m) and Maximum Burst Size (MBS)
- Intrinsic Burst Tolerance (IBT)
- Mean duration of the burst (t_m)
- Maximum Frame Size (MFS)

### 3.2.1 Peak Cell Rate and Cell Delay Variation Tolerance

The Peak Cell Rate (PCR) of the ATM connection is the inverse of the minimum interarrival time $T$ between two cells on a transmission link. It specifies an upper bound on the traffic that can be submitted on an ATM connection [ITUT96b]. The ATM Forum and ITU-T define the PCR and CDV tolerance using the Generic Cell Rate Algorithm (GCRA) and equivalent terminal model [ATMF94] [ITUT96b]. The reason for variation in the cell delay is that ATM Layer functions (e.g. cell multiplexing) may alter the traffic characteristics of ATM connections by introducing Cell Delay Variation (CDV). When cells from two or more ATM connections are multiplexed, cells of a given ATM connection may be delayed while cells of another ATM connection are being inserted at the output of the multiplexer. Similarly, some cells may be delayed while physical layer overhead or OAM cells are inserted. Consequently with reference to the peak emission interval $T$ (i.e. the inverse of the contracted peak rate), some randomness may affect the inter-arrival time between consecutive cells of a connection. The upper bound on the ‘clumping’ measure is the CDV Tolerance (CDVT). The CDVT at the public UNI, is defined in relation to the PCR according to the GCRA ($T, \tau_{UNI}$) where $\tau_{UNI}$ is the tolerance at the User Network Interface (UNI).

For the time being two extreme cases of characterising the CDVT [ITUT96b] have been identified:
Loose Requirements on CDV Tolerance
A large amount of CDV can be tolerated. In this case, only the specification of the maximum value of CDV tolerance $\tau_{\text{max}}$ that can be allocated to a connection is envisaged. $\tau_{\text{max}}$ is intended as the maximum amount of CDV that can be tolerated by the user data cell stream.

Stringent Requirement on CDV Tolerance
A connection should not be denied because of the required CDV tolerance if this CDV tolerance requirement is less than or equal to $\tau_{\text{PCR}}$ which is given by:

$$\frac{\tau_{\text{PCR}}}{\Delta} = \max \left( \frac{T_{\text{PCR}}}{\Delta}, \alpha \left( 1 - \frac{\Delta}{T_{\text{PCR}}} \right) \right)$$

where:
- $T_{\text{PCR}}$ is the peak emission interval of the connection (in seconds),
- $\Delta$ is the cell transmission time (in seconds) at the interface link speed,
- $\alpha$ is a dimensionless coefficient (suggested value is 80 [ITUT96b]).

3.2.2 Sustainable Cell Rate and Intrinsic Burst Tolerance
The Sustainable Cell Rate (SCR) is an upper bound on the average rate of the conforming cells of an ATM connection, over time scales which are long relative to those for which the PCR is defined. The Intrinsic Burst Tolerance (IBT) [ITUT96b] specifies the maximum burst size at the PCR or in other words the maximum deviation from the average rate. These parameters are intended to describe VBR sources and allow for statistical multiplexing of traffic flows from such sources.

The SCR and IBT traffic parameters are optional traffic parameters a user may choose to declare jointly, if the user can upper bound the average cell rate of the ATM connection. To be useful to the network provider and the customer, the value of the SCR must be less than the PCR. The SCR and the IBT (denoted as $\tau_{\text{IBT}}$) are defined by the GCRA ($T_{\text{SCR}}, \tau_{\text{IBT}}$). SCR and IBT belong to the ATM traffic descriptor [ITUT96b]. Translation from the Maximum Burst Size (MBS) to $\tau_{\text{IBT}}$ will use the following rule:

$$\tau_{\text{IBT}} = \lceil (\text{MBS} - 1)(T_{\text{SCR}} - T_{\text{PCR}}) \rceil \text{ seconds}$$

where $\lceil x \rceil$ stands for the first value above $x$ out of the generic list of values defined in [ITUT96b].
If the user has the knowledge of $\tau_{\text{int}}$ rather than of the maximum burst size, than the following rule applies:

$$\text{MBS} = 1 + \lceil \frac{\tau_{\text{int}}}{T_{\text{scr}} \cdot (T_{\text{PC}})} \rceil \text{ cells}$$

where $\lfloor x \rfloor$ stands for rounding down to the nearest integer value.

### 3.2.3 Mean Burst Period

The mean burst period ($T_{\text{mb}}$) is defined as the average time the source is transmitting cells at the peak rate. This parameter is widely used for bursty sources.

### 3.2.4 Maximum Frame Size

The Maximum Frame Size (MFS) is an upper bound on the number of cells send in a frame or in other words an upper bound on the frame size that a user can send. The MFS has to be smaller than the MBS.

### 3.3 Quality of Service Parameters

Quality of Service (QoS) is measured by a set of parameters characterising the performance of an ATM layer connection. These QoS parameters (referred to as network performance parameters by ITU-T) quantify end-to-end network performance at the ATM layer.

Six QoS parameters are identified by the ITU-T [ITUT96a] and ATM-Forum [ATMF96a] which correspond to a network performance objective. Three of these may be negotiated between the end-systems and the networks. One or more values of the QoS parameters may be offered on a per connection basis, corresponding to the number of related performance objectives supported by the network. Support of different performance objectives can be done by routing the connection to meet different objectives, or by implementation-specific mechanisms within individual network elements. The following QoS parameters are negotiated:

#### 3.3.1 Cell Loss Ratio (CLR)

Cell Loss Ratio (CLR) is the ratio of total lost cells to total transmitted cells in a population of interest. There are three different CLR definitions depending on the priority of the traffic. These are $\text{CLR}_p$, $\text{CLR}_{o1}$, $\text{CLR}_1$. The requested CLR will be an upper bound on the cell loss probability.
**CLR_{\text{h}}**
This is the ratio of total lost cell with high priority and the number of corresponding tagged cells to the number of CLP=0 transmitted cells.

**CLR_{\text{g=1}}**
The ratio of lost cells to the number of the total number of generated cells. This definition of CLR is used in this thesis.

**CLR_{\text{f}}**
The ratio of lost cells with CLP=1 to the number of transmitted cells with CLP=1.

### 3.3.2 Cell Transfer Delay (CTD)

This is defined as the elapsed time between a cell exit event at the measurement point 1 and the corresponding cell entry event at measurement point 2 for a particular connection. The mean CTD is the arithmetic average of a specified number of cell transfer delays. The requested CTD is an upper bound on the mean CTD.

### 3.3.3 Cell Delay Variation (CDV)

Two performance parameters associated with CDV are the **One-Point CDV** and the **Two-Point CDV** which are defined below.

#### 1-point CDV at an Measurement Point

The 1-point CDV (y\textsubscript{k}) for cell k at an MP is the difference between the cell’s reference arrival time (c\textsubscript{k}) and actual arrival time (a\textsubscript{k}) at the MP:

\[ y_k = c_k - a_k \]

The reference arrival time pattern (c\textsubscript{k}) is defined as follows:

\[ c_0 = a_0 = 0 \]

\[ c_{k+1} = \begin{cases} 
    c_k + T & \text{when } c_k \geq a_k \text{ or when cell } k \text{ does not arrive} \\
    a_k + T & \text{otherwise.}
\end{cases} \]

Positive values of 1-point CDV (early cell arrivals) correspond to cell clumping; negative values of 1-point CDV (late cell arrivals) correspond to gaps in the cell stream.
Cell Delay Variation between two MPs (2-point CDV)

The 2-point CDV \( (v_k) \) for cell \( k \) between MP\(_1\) and MP\(_2\) is the difference between the absolute cell transfer delay \( (x_k) \) of cell \( k \) between the two MPs and a defined reference cell transfer delay \( (d_{1,2}) \) between the same two MPs:

\[
v_k = x_k - d_{1,2}
\]

The absolute cell transfer delay \( (x_k) \) of cell \( k \) between MP\(_1\) and MP\(_2\) is the difference between the cell's actual arrival time at MP\(_2\) \( (a_{2k}) \) and the cell's actual arrival time at MP\(_1\) \( (a_{1k}) \):

\[
x_k = a_{2k} - a_{1k} \quad .
\]

The reference cell transfer delay \( (d_{1,2}) \) between MP\(_1\) and MP\(_2\) is the absolute cell transfer delay experienced by cell 0 between the two MPs.

Figure 3-1 Cell Delay Variation 2-point definition

where \( d_{1,2} \) Absolute cell transfer delay between MP\(_1\) and MP\(_2\)  
\( x_k \) Absolute cell k transfer time between MP\(_1\) and MP\(_2\)  
\( v_k \) 2-point CDV value between MP\(_1\) and MP\(_2\)

Figure 3-2 Definition of CDV length

Figure 3-2 illustrates the quality specification for the width of the CDV distribution. The delay time contains the fixed part composed of the propagation delay and the

\[1\] Variables \( a_{2k} \) and \( a_{1k} \) are measured with reference to the same reference clock
processing delay as well as the varying part due to the fluctuations in various delays (such as waiting time). The width of the CDV distribution is specified using the pair \((n, \Delta d)\) where \(\Delta d\) is the width of the value between the point where the tail of the varying part is \(10^{-n}\). The 2-point CDV is defined in [ITUT96a] as the upper bound on the difference between upper and lower \(10^{-n}\) quantiles of CTD\(^2\). Hence \(n\) should be taken as 8 and \(\Delta d\) for real-time services has been recommended as 3ms [ITUT96a]. For non-real-time services this value is expected to be between 600ms and several seconds [OHTA95].

The magnitude of the delay variation produced in the satellite link is larger than that of the terrestrial network hence CDV in the satellite link is a serious problem. The most widely used method to compensate for CDV is the use of a shaper buffer at the receiver terminal.

The maximum amount of CDV to be introduced by the terrestrial network is controlled by the Usage Parameter Control (UPC) which will be described in Chapter 5. However the main component of the delay variation introduced to the cell stream is due to the satellite access technique. Since TDMA (which is the access mechanism used in this thesis) is a kind of synchronous transfer mode retaining the asynchronous property of ATM cannot directly be realised. Sources can only send a certain number of cells each frame time. Hence the transfer interval of the user information cells at the input of the satellite network is not maintained and CDV is produced. The CDV introduced by the satellite network is usually equal to the access scheme frame time plus additional varying waiting delays in the buffers which can be bounded using the scheduling algorithm described in Chapter 4. More details on CDV measurements are provided in [ATMF96a, ITUT96a].

The following QoS parameters are not negotiated:

- **Cell Error Ratio (CER)**

  Cell Error Ratio (CER) is the ratio of total errored cells to the total of successfully transferred cells in a connection.

\(^2\) This is the difference between the \(10^{-n}\) and the \((1-10^{-n})\) quantiles of the CTD distribution during the life of the connection. \(10^{-n}\) was chosen because it allows for the proper engineering of delay buildout buffers when the overall CLR objective is \(10^{-n}\). The use of other quantiles for 2-point CDV specification is for further study.
• **Severely Errored Cell Block Ratio (SECBR)**
Severely Errored Cell Block Ratio (SECBR) is the ratio of total severely errored cell blocks to total cell blocks in a connection.

• **Cell Misinsertion Rate (CMR)**
Cell Misinsertion Rate (CMR) is the total number of misinserted cells observed during a specified time interval divided by the time interval duration.

Further information on ATM layer QoS may be found in ITU-T Rec. I.356 [ITUT96a].

3.4 ATM Service Categories

Services provided at the ATM layer, consists of different service categories which will be explained in this section.

3.4.1 **Constant Bit Rate (CBR) Service Category**
The CBR service category is used by connections that request a static amount of bandwidth that is continuously available during the connection. This amount of bandwidth is characterised by the Peak Cell Rate (PCR). CBR service is intended to support real-time (rt) applications requiring tightly constraint delay variation but is not restricted to these applications. Typical examples of CBR services include voice, video, and audio.

In the classical Synchronous Transfer Mode (STM) networks, fluctuating information rate must be converted into a CBR, namely the rate at which this STM network is operating. For instance, 64 kbit/s or 2 Mbit/s in N-ISDN.

CBR traffic is easy to manage in the network, since constant bandwidth is reserved for each CBR connection throughout its duration, independent of whether the source is actively transmitting or in a silent state. This is, however, an inefficient use of the transmission bandwidth, as illustrated in Figure 3-3. In particular, since the amount of information generated by most applications varies over time it is possible to reserve less bandwidth in the network than the application's peak bit rate, thereby allowing more connections to be multiplexed and increasing the resource utilisation. In initial deployments, a large portion of traffic in ATM networks is expected to be CBR voice,
video and audio. As time evolves, designers will have a better understanding of the dynamics of VBR traffic and be able to design efficient techniques to manage VBR traffic in the network, thereby achieving high resource utilisation.

### 3.4.2 Variable Bit Rate (VBR) Service Category

The traffic generated by a typical source, in general, either alternates between the active and silent periods and/or has a varying bit rate generated continuously. Furthermore, the peak-to-average bit rate (burstiness) of a VBR source is often much greater than one. Presenting VBR traffic to the network as CBR traffic means buffering, or rather artificially controlling its bit generation rate, which has the drawback of underutilization of network resources and QoS degradation. Although doing so simplifies the network management task, it is more natural to provide VBR service to VBR sources and thereby provide a better service and a framework to achieve higher resource utilisation. ATM networks offer this opportunity, thus the limitations of working at CBR disappears. VBR connections are characterised in terms of PCR, Sustainable Cell Rate (SCR) and Intrinsic Burst Tolerance (IBT).

![Figure 3-3 Bandwidth usage for VBR and CBR Services](image-url)

The VBR service category is usually divided into two categories namely: **real-time Variable Bit Rate** (rt-VBR) service category and **non-real-time Variable Bit Rate** (nrt-VBR) Service Category.

The rt-VBR service category is intended for rt-applications which require tight constrained delay and delay variation. The nrt-VBR Service Category on the other hand is intended for nrt-applications and no delay bounds are associated with this service category.
3.4.3 Available Bit Rate (ABR) Service Category

Many applications, mainly handling data transfer, have the ability to reduce their sending rate if the network requires them to do so. Likewise, they may wish to increase their sending rate if there is extra bandwidth available within the network. This kind of applications, which do not require bounds on delay and delay variation are supported by the ABR service category.

A rate-based flow control was specified [ITUT96b] which supports several types of feedback to control source rate in response to changing ATM layer transfer characteristics. This feedback is conveyed to the source through specific control cells called Resource Management (RM) cells. It is expected that an end-system that adapts its traffic in accordance with the feedback will experience a low cell loss ratio and obtain a fair share of the available bandwidth according to a network specific allocation policy.

On the establishment of an ABR connection, the end system specifies both a maximum and minimum required bandwidth. These are called Peak Cell Rate (PCR) and Minimum Cell Rate (MCR), respectively. The bandwidth available from the network may vary, but is guaranteed not to become less than the MCR.

3.4.4 Unspecified Bit Rate (UBR) Service Category

The UBR service category is intended for nrt-applications like traditional computer communication applications, such as file transfer and e-mail.

UBR service does not specify traffic related service guarantees. No numerical commitments are made with respect to the CLR experienced by a UBR connection, or as to the Cell Transfer Delay (CTD) experienced by cells on the connection. Congestion control for UBR may be performed at a higher layer on an end-to-end basis. The UBR service is indicated by use of the best effort indicator in the ATM user cell rate information element. Even if the PCR is not enforced it is still recommended to have the PCR negotiated, so that the source can discover the bandwidth limitation of the connection.
3.4.5 Guaranteed Frame Rate (GFR) Service Category

The Guaranteed Frame Rate (GFR) service is intended for users who are either not able to specify the range of traffic parameters to request most ATM services, or are not equipped to comply with the source behaviour rules required by existing ATM services. It is expected that many existing users fall in this category, as most applications are not currently equipped to select any of the traffic parameters required to establish ATM connections, or are not attached to the network through devices capable of effectively interacting with an ATM network (e.g. to comply with the ABR source behaviour).

The goal of the GFR service is to bring the benefits of ATM performance and service guarantees to users which require best-effort services with some minimum guarantees (MCR). It is UBR with some level of service guarantees. GFR requires minimal interactions between users and ATM networks, but the simplicity of the service specification for users does come at a cost in terms of requirements imposed on the network in order to efficiently support GFR. However the cost of these requirements is far outweighed by the potential benefits of making ATM technology more attractive to a broad range of users [ATMF98] (in particular Internet users).

<table>
<thead>
<tr>
<th>Traffic P:</th>
<th>CBR</th>
<th>rt-VBR</th>
<th>nrt-VBR</th>
<th>UBR</th>
<th>ABR</th>
<th>GFR</th>
</tr>
</thead>
<tbody>
<tr>
<td>PCR and CDVT</td>
<td>specified</td>
<td>specified</td>
<td>specified</td>
<td>specified</td>
<td>specified</td>
<td>specified</td>
</tr>
<tr>
<td>SCR, MBS, CDVT</td>
<td>n/a</td>
<td>specified</td>
<td>n/a</td>
<td>n/a</td>
<td>specified</td>
<td></td>
</tr>
<tr>
<td>MCR</td>
<td>n/a</td>
<td>n/a</td>
<td>specified</td>
<td>specified</td>
<td></td>
<td></td>
</tr>
<tr>
<td>MFS</td>
<td>unspecified</td>
<td>unspecified</td>
<td>unspecified</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>QoS P:</th>
<th>CDV</th>
<th>Maximum CTD</th>
<th>CLR</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>specified</td>
<td>unspecified</td>
<td>unspecified</td>
</tr>
<tr>
<td></td>
<td>unspecified</td>
<td>unspecified</td>
<td>unspecified</td>
</tr>
</tbody>
</table>

Table 3-1 List of ATM Service Traffic Categories Parameters

Notes:
1. CLR is low for sources that adjust cell flow in response to control information. Whether a quantitative value is specified is network specific
2. CLR is low for frames that are eligible for a service guarantee.
This Section has defined the ATM service categories. Table 3-1 provides a list of ATM QoS and traffic parameters and identifies whether and how these are supported for each service category. Then Table 3-2 provide the guaranteed network performance objectives of the network for a specific traffic class as recommended by the ITU-T Rec.I.356 [ITUT96a].

<table>
<thead>
<tr>
<th>Default Objectives</th>
<th>CTD</th>
<th>2-pt. CDV</th>
<th>CLR_{95}</th>
<th>CER</th>
<th>CMR</th>
<th>SECBR</th>
</tr>
</thead>
<tbody>
<tr>
<td>Class 1 (stringent)</td>
<td>400 msec</td>
<td>3 msec</td>
<td>3·10^{-7}</td>
<td>default</td>
<td>default</td>
<td>default</td>
</tr>
<tr>
<td>Class 2 (tolerant)</td>
<td>U</td>
<td>U</td>
<td>10^{4}</td>
<td>default</td>
<td>default</td>
<td>default</td>
</tr>
</tbody>
</table>

"U" means unbounded. When the objective of a parameter is specified as being "U" performance with respect to the parameter may, at times, be arbitrarily poor.

Table 3-2 Provisional QoS Network Performance Objectives.

### 3.5 Statistical Behaviour of Traffic Sources

#### 3.5.1 Voice

The statistics of a single voice source are composed of two phases and they normally depend on the technique of voice coding that is being used. The two periods are the active period and the silent period.

The POTS (Plain Old Telephony Service) has been using a fixed bandwidth digital channel at 64kbit/s. Modulation techniques such as adaptive differential pulse code modulation can be used to compress voice information to a constant bit rate with lower bandwidth requirements.

CBR voice in ATM networks is transmitted with AAL type 1 using the pulse code modulation technique. Recommendation G.711 [CCIT86] specifies 64 kbit/s CBR voice.

When voice signals are coded with a variable bit rate an active period of a voice source corresponds to a talk spurt, whereas a silent period corresponds to speech silence duration. The silent periods constitutes 60-65% of the transmission time of voice calls in each direction. More specifically, the average active and silent periods are measured to be respectively equal to 352 ms and 650 ms [SRIR86]. Furthermore,
in a normal conversation the active period fits the exponential distribution reasonably well while the duration of the silent periods is less well approximated by the exponential distribution [BRAD69]. Nevertheless, the most frequently used models of voice sources in the literature assume that the duration of both active and silent periods are exponentially distributed.

A single voice source can be modelled by an Interrupted Poisson Process (IPP) or by the on-off model. Multiplexed voice sources are best modelled by a Markov Modulated Poisson Process (MMPP).

### 3.5.2 Video

A promising service of ATM networks is video communication. It can be divided into still picture and motion picture video traffic. The investigation of video statistics started in the 1970s, but still little is known about the statistics for the arrival process of cells containing video information coded at high bit rates. Video is quite different than voice or data in that its bit streams exhibit various types of correlation's between consecutive frames. Video images have the following statistical components (which are dependent on the type of codec):

- **Line Correlation**: occurs when data at one part of the image is highly correlated with data on the same part of the next line (spatial correlation).
- **Frame Correlation**: data at one part of the image is highly correlated with data on the same part of the next image (temporal correlation).
- **Scene Correlation**: occurs because sequences of scenes may, to a greater or less extent, be coincidentally correlated with each other.
- **White Noise**: is a memoryless process and is uncorrelated.

Non-frame buffered video codecs have all four of the correlation's, whilst frame buffered video codecs (frames all always buffered before being sent) only have scene and white noise correlation [RACE93]. Scene correlation's can be reduced by multi-frame buffering.

Due to the various correlation's that video traffic exhibits it is inadequate just to measure the burstiness of video traffic. The following list summarises some desirable qualities for new measures:

- The measure should not yield just statistical values, but values that capture the characteristics of the rate variation over time.
The measures must be capable of evaluating the statistical multiplexing effect.

- The measures should allow easy modelling of video information sources.

The following measures have been proposed [OHTA94] to fulfil these kinds of conditions:

**Bit Rate Distribution**

The distribution and the probability density distribution of the encoded bit rate evaluated in single frame units. Along with the average bit rate and the variance, they are quite adequate for approximating the required capacity.

**Autocorrelation Function**

The autocorrelation function is a convenient measure for expressing the nature of temporal variations.

**Coefficient of Variation**

In order to express such phenomena as the signal delays that arise when a signal is buffered, the coefficient of variation is used, as a measure to investigate the multiplexing characteristics when variable-rate signals are statistically multiplexed.

**Distribution of Scene Duration**

The probability density distribution of intervals between scene changes.

Various models have been proposed to model video sources. In applications with uniform-activity-level scenes, the change in the information content of consecutive frames is not significant. A typical application of this form is a video telephone where the screen shows a person talking. In general, correlation's in video services with uniform activity levels last for short duration and decay exponentially with time. A first-order Autoregressive (AR) model is proposed in [NOMU89]. Another continuos-state AR model which is found to be quit accurate compared with actual measurements, is proposed in [MAGL88]. It has however to be noted that these models are not convenient for queuing analysis, but mostly used in simulation studies. In order to evaluate regions of extremely low probability (like cell loss), Markov models can be used.
The observation that intrascene bit-rate variations are smooth and that their sum should not exhibit sudden jumps was used by [MAGL88], to model the video source as a continues time, discrete state Markov model which is shown in Figure 3-6. This is a type of birth-death Markov model, and only transitions to adjacent states are possible.

In applications with nonuniform activity-level scenes like motion video, frames of high-activity scenes and scene changes contain large amounts of data followed by frames that contain less data. In addition to the short-term fast decaying correlation’s (temporal correlation’s) of uniform activity scenes, there is a long-term slow decaying correlation in the amount of information generated per frame, that occurs at times of scene changes. The Autoregressive Moving Average (ARMA) model is proposed [GRÜN91] to take into account the two types of correlation that occur in nonuniform activity-level scenes. The ARMA arrival processes are used in Monte Carlo simulations to estimate the probability-distribution function of the queuing delay and the mean and variance of the interdeparture time seen by the arriving cell. However, ARMA models cannot be used in the numerical and analytical analysis of queues.

A two-dimensional, continues time Markov model which is shown in Figure 3-7 [SEN90], is a generalisation of the model developed in [MAGL88] for uniform activity scenes. In two dimensions, it is now possible to model the bit-rate fluctuations in consecutive frames to include jumps to the higher or lower bit rates, thereby modelling the correlation at scene changes.

The Markov Modulated Poisson Process (MMPP) can be used to model the cell arrival process from video sources (see Figure 3-9). The interscene transition are given by a Markov chain. This model views bit-rate variations as changes in the number of packet arrivals. Furthermore, for ease of analysis, all distributions (the scene change interval and state persistence-time distribution) are assumed to be exponential distributions.

The MMPP can also be used when \( N \) independent video sources are multiplexed. Since the scene change interval distribution of the various sources are assumed to be exponential distributions, the scene change interval distribution of the multiplexed
model will be an exponential distribution with an average scene change interval of \( 1/N \) of that of a single video source.

### 3.5.3 Data

The term data is used for any application that uses coded text, that is, any application that is not voice, audio, video or still image. Despite the fact that data networks have been operational for a number of decades, traffic characteristics of some data sources are not well understood.

The main difficulty arises due to the fact that there is no typical data connection. Large amounts of data are transmitted in a file transfer on a rather continuous basis during the duration of the connection, whereas only a few hundred bytes are generated by an e-mail.

Furthermore, data connections are not generally established between two users, but between groups of users, as in the case of Local Area Network (LAN) interconnection. Although the data cell arrival process in ATM networks has not yet been identified, actual data packet arrival processes have been investigated.

It is well known that generation of data from a single data source is well represented by a Poisson arrival process (continuous time) or by a geometric inter-arrival process (discrete time). When information loss occurs, these kinds of services use retransmission as a way of recovering information. The retransmission of the complete data frame is executed every time there is cell loss.

**Interactive Data Transmission**

A single packet is generated at each time. This could be either a fixed length or a variable length packet. The length of the packet is represented by a certain distribution of fixed mean.

This traffic is of bursty nature, relatively short in length, and requires relatively small delay in transmission. Delay variance is not a major problem, but error free transmission is an important requirement.

Examples of such traffic are transaction/credit card verification, hotel/airline reservation, WWW access and various short message transmissions.
Bulk Data Transmission
The nature of the traffic is similar to the earlier case, but now messages consist of a number of packets. This is a batch arrivals case and the arrivals of the packets that make up the message are not independent. Since ATM networks have a fixed cell size, it may happen that a data packet of either variable or fixed size, is fragmented into several cells.

The performance requirements are similar to the previous case, but a slightly higher average delay might be acceptable. Examples for bulk data transmission are file transfer, database information acquisition etc.

Candidates to model data sources are the two state MMPP, also called Switched Poisson Process (SPP), including the Interrupted Poisson Process (IPP) and the Geometrically Modulated Deterministic Process (GMDP), including the on-off model.

3.5.4 Multimedia
The term multimedia is used to refer to the representation, storage, retrieval, transmission of multiple media, such as text, voice, graphics, image, audio and video.

Multimedia applications constitute a significant future market. Examples include teleconferencing, entertainment video, medical imaging, distance education, telemarketing and advertising. Each of these applications consists of two or more information types, which are listed above. It has to be noted that a strong correlation between successive cell arrivals is characteristic of many multimedia traffic sources.

Traffic models of information types (i.e. video, voice, data), which are put together in multimedia services are presented throughout this section. The extension of these models to characterise the integrated environment of a multimedia service is an important task and is currently under extensive study [ORS98g].

Despite this, the MMPP is widely used to model superposed traffic of different information types (as it will be explained in the next section), and could therefore be used to model multimedia traffic.
3.6 Criteria for the Selection of Source Models

There are, usually, several alternatives to represent a particular traffic source and different levels of complexity. The most important criteria upon which the selection was based is briefly discussed.

First of all, the chosen source model must be accurate with respect to our assumptions. It should be close to reality and the different parameters should not have only a statistical but also a physical meaning. Analytical and simulation results of the model should be compared, if possible, to measured performance of real traffic sources to verify the source models accuracy and validity.

From the analytical point of view, tractability (superposition/queuing) is an important feature. This means that the use of the source models in analysis should lead to solutions that lend themselves for numerical computation. In many cases, general methods such as iterations to solve systems of linear equations, aggregation methods to reduce the dimensionality, matrix analytical methods to solve structured Markov chains etc. could be applied. Often, the exploitation of the special structure of the processes involved, may make the model much more suitable for numerical solutions, without losing its probability interpretation.

Another important feature of source models is its generality and usability. A typical model should be able to represent a large class of sources with similar characteristics. Since most of the source models are also used in simulations, care should be taken so that it is possible to represent the model in a simulation environment. It is also important that the model is statistically stable, otherwise there may be significant problems in the overall network simulation model that might be difficult to detect. The statistical stability is measured over a period of time which is proportional to the highest level of resolution in time specified by the source model and its number of different states.

Finally the number of parameters of the model should be taken into account. This number is usually directly related to the complexity of the description of the model. The aim is to use a model that is adequate for our purpose, but uses a limited number of parameters. This makes the analysis easier and the computation faster.
3.7 Multi-State Traffic Source Models

In this Section traffic source models to characterise the traffic sources discussed in the previous section will be described.

3.7.1 General Modulated Deterministic Process (GMDP) Model

The GMDP is based on a finite state machine having \( n \) states. In each state, cells are generated with constant interarrival time \( T_i \) (therefore it is called a deterministic process), the index \( i \) being the state. The number of cells which are emitted in state \( i \) may have a general discrete distribution. Usually, the GMDP also includes silence states where no cells are generated and the duration of these states may also have a general discrete distribution. If the burst and silence duration’s have an exponential distribution then the model is called a Markov Modulated Deterministic Process (MMDP). The on-off model is a two state MMDP with one silence state.

The state transitions are governed by a transition matrix where each element denotes the probability of moving from state \( i \) to state \( j \) once the sojourn period expires.

Usually, voice traffic sources can be characterised when using this model with 2 states, whilst video traffic sources may need 3 states to be characterised.

3.7.2 The On-Off Model

One traffic model, which is widely used for the characterisation of ATM sources, is the on-off source model. This model has been successfully used to realistically model packetised speech, still picture and interactive data services. According to the on-off model the ATM cell stream from a single source is modelled as a sequence of alternating burst periods and silence periods. This model is a 2-state Markovian representation of an ATM source as shown in Figure 3-4. The duration of each burst is exponentially distributed with mean \( 1/a \) ms. During such a period ATM cells are emitted with constant interarrival time \( T \) ms, where \( T = 1/PCR \). After generation of the ATM cells an exponentially distributed silence period with mean value \( 1/b \) ms follows. This corresponds to a geometrically distributed number of packets per active period (i.e. burst), with mean value \( 1/(aT) \), followed by an exponentially distributed silence period, with mean value \( 1/b \).
Figure 3-4 The on-off source model

Note that this model is a special case of the General Modulated Deterministic Process (GMDP); it is equivalent to a two state Markov Modulated Deterministic Process (MMDP) with one silent state.

The on-off traffic source model as shown in Figure 3-4, can be described by the parameters \((\text{PCR}, m, \beta, t_m)\) as follows:

\[
\text{PCR} = 1/T, \quad t_m = a^i, \quad m = a^i/T(a^i + b^i) \quad \text{and} \quad \beta = (a^i + b^i)/a^i
\]

\(a\) and \(b\) are the transition rates, i.e. \(a\) is the inverse of the mean burst duration, \(b\) is the inverse of the mean silence duration, \(m\) is the mean cell rate, \(\beta\) is the burstiness and \(t_m\) is the average burst duration.

3.7.3 The Interrupted Poisson Process (IPP)

The IPP is a Poisson process that is alternatively turned on for an exponentially distributed period of time (active period where cells are emitted) and turned off for another exponentially distributed period of time (silent period), like the on-off model. The difference however is that during the active periods, the interarrival times of cells are exponentially distributed (i.e. in a Poisson manner).

The advantage of modelling the arrival process from a single voice source as IPP is that the aggregated arrival process from multiple sources can be modelled by an MMPP. This is due to the fact that the IPP is a special case of an MMPP and the superposition of MMPPs is an MMPP.
Let $r_1$, $r_2$, and $\lambda$ respectively denote the average duration of the active and silent periods, and the cell generation rate during the active period. The simplest way to determine these IPP parameters is to set the mean talkspurt length ($\mu' = t_{nn}$) to the mean sojourn time of the cell arrival process $r_1$, to set the mean silence period length ($\mu' = t_{nn}(1-\beta)$) to $r_2$ and to set the mean cell generation rate during a talkspurt $\lambda'$ to that of the cell arrival process $(T)$. But this parameter matching underestimates the performance, therefore the two-moments and peakedness method has been proposed [IDE88].

### 3.7.4 Interrupted Bernoulli Process (IBP)

The IBP is a discrete version of the IPP. Time is slotted, with a slot length being equal to the cell in the medium. A slot is either in an active state or in a silent state. A slot in an active state contains a cell with probability $a$ and no cell with probability $(1-a)$, while no cell arrive in a silent state. Given that the slot is in the active state (independent of whether the slot contains a cell or not), the next slot is also in the active state with probability $p$ and changes to the silent state with probability $(1-p)$. Similarly, given that the slot is in the silent state, the next slot is also in the silent state with probability $q$ and changes to an active state with probability $(1-q)$. Accordingly, both the active period, $Pr(X=x)$, and the silent period, $Pr(Y=y)$ are geometrically distributed. That is

$$Pr(X=x) = (1-p)^x p^x, \quad Pr(Y=y) = (1-q)^y q^y \quad x, y \geq 1$$

with respective average duration times $1/(1-p)$ and $1/(1-q)$.

### 3.7.5 The Birth-Death Process

A birth-death process is a Markov model where only transitions to adjacent states are generated. The continuous-time birth-death process shown in Figure 3-6 is used to model voice and video. This process can be viewed as the superposition of $N$ independent homogeneous on-off sources.

This continuous-time process is a fluid approximation model and bit rates can be seen as switching between states with discrete values, and the time spent in each state is given by a random Poisson time sequence.
For voice, instead of modelling the individual information sources, the total bit rate of $N$ independent active voice sources is modelled. To model the actual video source, bit rates can take on only discrete quantized values and are assumed to sampled at random Poisson times in the time domain. Figure 3-5 shows an example of sampling the bit rate at Poisson times. The quality of the approximation can be improved by reducing the amount of information in a quantised step and increasing the sampling rate. There are $N+1$ states, and the state quantization step is $A$ bits per frame. The probability of a transition to a higher bit-rate state is higher for lower bit rate states and decreases as the bit rate increases.

Figure 3-5 Poisson sampling and quantisation of the source rate.

If $p(i,j)$ is the transition rate from state $i$ to $j$, the birth and death rates are given [MAGL88] by:

\[
p(i,i+1) = (N-i) \cdot b \quad i<N
\]

\[
p(i,i-1) = i \cdot a \quad i>0
\]

where $a$ and $b$ are the transition probabilities.

The equilibrium probability of being in state $i$ is given by the binomial distribution:

\[
P(i) = \binom{N}{i} p^i (1-p)^{N-i}
\]
The two dimensional, continuous time birth-death process shown in Figure 3-7 can be used to model jumps to higher or lower bit rates in video scene changes. Each dimension of the model can be viewed as the one-dimensional birth-death process discussed above. \( c \) and \( d \) are the transition probabilities to low-activity and high-activity levels respectively. Note that the cell rate in state \( N \) is \( N\alpha \) and is \( A_s + NA_s \) in state \( (1,N) \).

![Figure 3-7 Two-dimensional birth-death process](image)

![Figure 3-8 State-transmission-rate diagram for aggregate source model](image)

If a single video source is modelled in this manner, the bit rate when multiple information sources are multiplexed can be modelled with the same structure. Thus the multiplexing of \( N \) video sources can be modelled with a state-transition-rate diagram like that shown in Figure 3-8.
3.7.6 The Markov Modulated Poisson Process (MMPP)

The MMPP has been extensively used to model various B-ISDN sources, such as voice, video, as well as characterising superposed traffic. It has the property of capturing both the time-varying arrival rates and correlation's between the interarrival times. Also, if individual traffic sources are modelled by an MMPP, the superposition of different sources can be described by an MMPP.

An MMPP is a doubly stochastic Poisson process. The arrivals occur in a Poisson manner with a rate that varies according to a n-state (phase) Markov chain, which is independent of the arrival process.

![Figure 3-9 Two-state MMPP](image)

As the simplest case, Figure 3-9 shows the 2-state MMPP (also called Switched Poisson Process (SPP)) having Poisson arrival rate $\lambda_j$ in phase j, j=1,2, which appears alternately exponentially distributed sojourn time with mean $r_j$. This is characterised by $(R, A)$ where $R$ is the infinitesimal generator of the underlying Markov chain and $A$ the arrival rate matrix, defined by:

$$R = \begin{bmatrix} -r_1 & r_1 \\ r_2 & -r_2 \end{bmatrix}, \quad A = \begin{bmatrix} \lambda_1 & 0 \\ 0 & \lambda_2 \end{bmatrix}$$

The n-state MMPP is similarly characterised by $(R, A)$ with each matrix of n x n size.

In special cases, the MMPP becomes a renewal process, which is characterised by statistically independent and identically distributed interarrival times.

If $\lambda_1=\lambda_2=\lambda$, the MMPP reduces to a poisson process with rate $\lambda$. If $\lambda_2=0$, it is called an Interrupted Poisson Process (IPP).

Superposition of MMPPs

The superposition of MMPPs is also an MMPP. Therefore it can be used to model superposed heterogeneous traffic. Consider $N$ MMPP models, each with parameters $R_i$,
and \( \Lambda \). Then, the transition rate matrix \( R \) and the arrival rate matrix \( \Lambda \) of the superposed process are:

\[
R = R_1 \oplus R_2 \oplus \ldots \oplus R_N \quad \Lambda = \Lambda_1 \oplus \Lambda_2 \oplus \ldots \oplus \Lambda_N
\]

where \( \oplus \) denotes the Kronecker sum defined below. We note that both \( R \) and \( \Lambda \) are \( k \times k \) matrices, where \( k = 1, \ldots, N \).

The Kronecker sum of two matrices \( R_1 \) and \( R_2 \) is defined as:

\[
R_1 \oplus R_2 = (R_1 \otimes I_{R_2}) + (I_{R_1} \otimes R_2)
\]

where \( I_{R_i}, i=1,2, \) is an identity matrix of the same order as matrix \( R_i \) and \( \otimes \) denotes the Kronecker product, which is defined for two \( C = \{ c_{ij} \} \) and \( D = \{ d_{ij} \} \) as:

\[
C \otimes D = \begin{bmatrix}
c_{11}D & c_{12}D & \ldots & c_{1n}D \\
\vdots & \vdots & \ddots & \vdots \\
c_{m1}D & c_{m2}D & \ldots & c_{mn}D
\end{bmatrix}
\]

As the number of the superposed processes increases, the number of states of the MMPP increases and it becomes very complex to solve queues with a large number of arrival streams. Due to the complexity of matrix analysis with a high number of states it is difficult to obtain the source model of multiplexed traffic sources. To reduce complexity, the superposed process may be approximated by a simpler process that captures important characteristics of the original process as closely as possible. The simplest model that has the potential to approximate an MMPP with a large number of states accurately, is the two state MMPP, which was described above.

### 3.8 Self-Similar Traffic Models

Several recent studies show that network traffic is self-similar, or exhibits long-range dependencies. Self-similar traffic is problematic for routing and congestion control algorithms because self-similar traffic is very different from conventionally considered traffic models such as Poisson or Markovian traffic.

It has been shown that the Ethernet LAN traffic in an R&D environment as seen aggregated on network cable segments which connect file-servers, minicomputers, workstations, personal computers, and printers is self-similar. Also, wide-area TCP traffic demonstrate that WAN packet arrival processes are better modelled by self-similar processes. In addition, it has been reported that VBR video traffic can be
modelled as self-similar processes based on the analysis of a VBR video traffic trace coded from a two-hour movie [WANG95].

Network traffic with self-similarity raises many issues:

- If much of network traffic is self-similar, are we then compelled to discard network studies based on other traffic models, in particular Poisson and two-state Markov traffic models?
- What is the relationship between a single source and aggregated network traffic?

A self-similar phenomenon exhibits structural similarity across a wide range of time scales. Self-similar behaviour is also called as fractal or scale-invariant. In the case of network traffic, self-similarity can be clearly observed from the plots of packet counts over different time scales. That is, if we depict a sequence of plots of the number of packets per time slot for different slot lengths, with self-similarity all plots looks very "similar" to one another and are distinctively different from white noise. In particular, with self-similarity the traffic does not have a natural burst length: at every time scale ranging from milliseconds to minutes or hours, traffic bursts look similar. It has been shown that self-similar traces are self-similar at all levels of aggregation [WANG95].

Work has been done and is still under progress to use a Markov chain to produce self-similarity on a finite timescale. A model which is quite easy to manipulate and depends only on three parameters has been proposed by [ROBE96]. This model can also be fitted to the measured data but only provides self-similarity over 'some' time scales. Recent work provides a simple physical explanation for the observed self-similarity of measured aggregate packet traffic in terms of the nature of the traffic generated by the individual source/destination pairs that contribute to the aggregate packet stream [WILL97]. Developing an approach originally suggested by Mandelbrot [MAND69] they showed that the superposition of strictly alternating ON/OFF sources ultimately gives the mathematical interpretation of the self similar traffic.

### 3.8.1 Aggregation of heavy-tailed on-off sources

The superposition of many (strictly alternating) independently and identically distributed (i.i.d.) ON/OFF sources, each of which exhibits the "Noah Effect", results in self similar aggregate traffic [LELA93]. The ON and OFF periods do not
necessarily need to be of the same distribution. By presenting the data traffic as ON/OFF source model (also known as "packet train model") it can be identified that the Noah Effect is the essential point of departure from traditional to self similar traffic modelling. Intuitively, the Noah effect for an individual ON/OFF source model results in ON and OFF periods, i.e. "train lengths" and "inter train distances" that can be very large with non-negligible probability. In other words, the Noah Effect guarantees that each ON/OFF source individually exhibits characteristics that cover a wide range of time scales. The Noah Effect is synonymous with the *infinite variance syndrome* - the empirical observation that many naturally occurring phenomena can be well described using distributions with infinite variance. In mathematical aspects the use of heavy tailed distributions with infinite variance (e.g. Pareto distribution) can accommodate the Noah Effect and the shape parameter $\alpha$ describing the "heaviness" of the tail of such a distribution gives a measure of the intensity of the Noah Effect. Therefore the mathematical vehicle for modelling such phenomena, the *heavy tailed* distributions with infinite variance (e.g., Pareto distribution, Weibull distribution) are used. Also a simple relation between the $\alpha$ (which describes the "heaviness of the tail of the corresponding infinite variance distribution, or equivalently the "intensity" of the Noah effect) and the Hurst parameter $H$ has been suggested as a measure of the degree of self similarity (or equivalently, of the "Joseph Effect") of the aggregate traffic stream. The aggregating of the traffic of many such source models produce a self similar superposition process [MAND69] with self similarity parameter $H = (3 - \alpha)/2$, where $\alpha$ is the parameter that characterises the "thickness" of the tail of the distribution [LELA93].

The processes which have strictly alternating ON and OFF periods agree with the ON/OFF source models commonly considered in the communications literature. The ON and OFF periods, moreover may have different distributions, either with infinite or finite variance. The main important result being that the superposition of many such packet trains exhibits, on large time scales, self similar behaviour.

### 3.8.2 Heavy-tailed distributions

The self similar processes show *long range dependence*. A process with long range dependence has an autocorrelation function,
r(k) \sim k^{-a} \text{ as } k \to \infty, \text{ where } 0<a<1

Thus the autocorrelation function of such a process decays hyperbolically as compared to the exponential decay exhibited by the traditional traffic models [Crov96]. Hyperbolic decay is much slower than the exponential decay, and since $a<1$, the sum of the autocorrelation values of such a series approaches infinity. This has the number of implications. First, the variance of $n$ samples from such a series does not decrease as a function of $n$ (as predicted by basic statistics for uncorrelated datasets) but rather by the value $n^{-a}$. Secondly, the power spectrum of such series is hyperbolic, rising to infinity at frequency zero, reflecting the "infinite" influence of long range dependence in the data.

The Pareto distribution used in the research have the property of being "heavy tailed". A distribution is heavy tailed if,

$$P[X>x] \sim x^{-a}, \text{ as } x \to \infty, \text{ } 0<a<2$$

That is, regardless of the behaviour of the distribution for small values of the random variable, if the asymptotic shape of the distribution is hyperbolic, it is heavy tailed. The simplest heavy tailed distribution is the Pareto distribution. The Pareto distribution is hyperbolic over its entire range; its probability density function (pdf) being,

$$p(x) = \alpha \beta^\alpha x^{-\alpha-1} \text{ for } \alpha, \beta > 0, x>\beta$$

and its cumulative distributive function (cdf) is given by,

$$P[X\leq x] = 1-(\beta/x)^\alpha \text{, } \alpha, \beta > 0, x>\beta$$

where $\beta, \alpha$ are the location parameter and the shape parameter of the distribution respectively. If $\alpha \leq 2$, then the distribution has infinite variance, and if $\alpha \leq 1$, then it has an infinite mean value. The mean ($\mu$) of the Pareto distribution is given by,

$$\mu = \alpha \beta / (\alpha - 1), \alpha > 1$$

The detailed choice of the parameters that appear in a distribution function is to some extent arbitrary. However three types of parameters are regarded as "basic" in the sense that they always have a certain physical or geometrical meaning. These are the location, scale and the shape parameters, the descriptions of which are as follows:
Location Parameter ($\beta$): The abscissa of a location point (usually the lower or mid point) of the range of the variate.

Scale Parameter ($\gamma$): A parameter that determines the scale of measurement of the quantile $x$.

Shape Parameter ($\alpha$): A parameter that determines the shape (in a sense distinct from location and scale) of the distribution function (and other functions) within a family of shapes associated with a specified type of a variate.

The symbols $\alpha$, $\beta$, $\gamma$ is used to denote shape, location and scale parameters in general, but other symbols are used in case where firm conventions are established. In Pareto distribution only the location parameter ($\beta$) and the shape parameter ($\alpha$) exists, whereas for the normal distribution the mean $\mu$, is a location parameter (the locating point is the midpoint of the range) and the standard deviation $\sigma$ is the scale parameter. The normal distribution does not have a shape parameter. In the heavy tailed Pareto distribution the location parameter $\beta$ gives the lowest value of the range, whereas the shape parameter $\alpha$ gives the heavy tailed property or the slowly decaying shape of the distribution. Some distributions such as Beta distribution have two shape parameters namely, $\alpha_1$ and $\alpha_2$. The Pareto distribution is also known as the double exponential distribution or the power law distribution has been used initially to model distributions of incomes exceeding a minimum value [ARNO83].

3.9 Summary

This section summarises the models that will be used to represent various service types. The exploration of self-similar traffic generator processes which are suitable for simulation and analysis is a topic currently drawing international interest. Within this thesis traditional models are used and Table 3-3 provides a summary of traffic models that are used to generate streams of cells appropriate for the corresponding service type.

The reason for using traditional Markov models for the research was the fact that a self-similar traffic model suitable for simulation was not available until the final stages of the thesis. The used models capture short-range dependent correlation effects and a long-range dependent model may have been more suitable for data traffic.
Table 3-3 Source Models user for Service Types

Although it was mentioned earlier that most traffic source parameter values are not known yet, some proposed traffic parameter values [BUTT91], [SYKA92], [AKIM93] of the future B-ISDN that will be used during the project are shown in Table 3-4. Note that some of the values will be varied in order to observe the affect of different burstiness values and peak rates on network performance. The used parameter values will be provided in the relevant section.

Table 3-4 Traffic Characteristics of some B-ISDN services

<table>
<thead>
<tr>
<th>Source</th>
<th>Peak Bit Rate</th>
<th>$t_{on}$</th>
<th>Burstiness $^*$</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Voice</td>
<td>32 kbit/s</td>
<td>352 ms</td>
<td>2.85</td>
</tr>
<tr>
<td>Still Picture</td>
<td>2 Mbit/s</td>
<td>87 kbit/s</td>
<td>23</td>
</tr>
<tr>
<td>Broadband Services</td>
<td>10 Mbit/s</td>
<td>2 Mbit/s</td>
<td>5</td>
</tr>
</tbody>
</table>

*Burstiness = Peak Bit Rate/Mean Bite Rate*
4. TRAFFIC AND CONGESTION CONTROL FRAMEWORK FOR ATM OVER SATELLITE

The design of a suitable traffic and congestion control scheme is the most important challenge for the success of an ATM based B-ISDN. Therefore it has been the subject of vigorous research over recent years. Considering the scarce satellite bandwidth available, traffic and congestion control for satellites is even more important than for the terrestrial network. In this thesis the framework developed by the ITU-T will be adopted to the satellite environment where possible. However due to different properties of the satellite link from terrestrial fibre-optic links, some different mechanisms and protocols are needed for the satellite environment.

According to ITU-T Recommendation I.371 [ITUT96b], the primary role of traffic control procedures is to protect the network so that it can achieve the required network performance objectives, e.g. in terms of cell loss probability or cell transfer delay. The same recommendation defines ATM layer traffic control as the set of actions taken by the network to avoid congestion. The latter is defined as a state of network elements (e.g. switches, concentrators, cross-connects and transmission links) in which the network is not able to meet the negotiated quality-of-service objectives for the connections already established or for any new connection request, because of traffic overload or control-resource overload. Congestion can be caused by unpredictable fluctuations of traffic flows or by fault conditions within the network and is to be distinguished from queue saturation, which may happen while still remaining within the negotiated quality-of-service.

The objectives of ATM layer traffic control for B-ISDN can be summarised as follows:

- **Flexibility**: It should support a set of ATM layer QoS classes sufficient for all existing and foreseeable services.
- **Simplicity**: The challenge is to design a simple ATM layer traffic control mechanism which minimises network equipment complexity while maximising network utilisation.
• **Robustness:** The requirement of achieving high resource efficiency under any traffic circumstance while maintaining simple control functions.

### 4.1 Reactive and Preventive Controls

Various control mechanisms have been proposed for ATM networks. These can be classified into two categories: reactive control and preventive controls.

Each class of control is applicable at different time scales. In particular, reactive schemes can necessarily operate at time scales greater than the propagation delay, whereas preventive techniques are designed to be effective at cell transmission times till the end of the connection duration.

#### 4.1.1 Reactive Control

Reactive control is a technique used to recover from a congested state. The traffic flow at the access points, based on the current traffic levels, are regulated within the network; access points therefore require indicators of congestion to be fed back from network nodes with enough lead time to react effectively. There are several potential problems with a reactive control approach in a broadband network (see Table 4-1 for reaction time). The feedback control loop would be sensitive to transient traffic behaviour and network topology (i.e. propagation delay), making it difficult to avoid overreactions while at the same time ensuring the guaranteed QoS is met. It would also be difficult to determine which terminals or end systems should be throttled back when each may have such diverse bandwidth requirements. Reactive Control schemes for satellite ATM networks will be discussed in more detail in Chapter 7.

The ATM cell length is 53 bytes=424 bits and the approximate data transmission speed is 150000 km/s on the fibre optic links [LEME93] and 300000 km/s on a satellite link.

Propagation time for fibre optic: \( T_{pf} = \frac{\text{Distance}}{\text{Speed of propagation on fibre}} \)

Propagation time for satellite link: \( T_{ps} = \frac{\text{Distance}}{\text{Speed of propagation in vacuum}} \)

Cells transmitted during the \( T_{pf} \) time: \( T_{pf} = T_{pf} \cdot \frac{\text{Bit Rate}}{424} \)

Cells transmitted during the \( T_{ps} \) time: \( T_{ps} = T_{ps} \cdot \frac{\text{Bit Rate}}{424} \)
Traffic and Congestion Control Framework for ATM over Satellite

Note that with a bit rate of 64 kbit/s, on a 38000 km distance (for GEO satellites), the time from the cell transmission to its acknowledgement (2·2·126.66 ms = 506.6 ms) is 76 times the cell transmission time. The sender can therefore emit 76 cells (=32kbit) before receiving any information about error or flow control from the other end. On the other hand for a bit rate of 2 Mbit/s the user can transmit 2·1194=2388 cells before receiving any feedback information. The number of emitted cells before receiving feedback can be halved by using a satellite which can act as virtual destination / virtual source. Table 4-1 shows the number of cells which can be emitted for a certain bit rate and propagation time.

<table>
<thead>
<tr>
<th>Throughput</th>
<th>Distance</th>
<th>$T_{pf}$ (μs)</th>
<th>$T_{pf}$ (μs)</th>
<th>$T_{tr}$ (cells)</th>
<th>$T_{ts}$ (cells)</th>
</tr>
</thead>
<tbody>
<tr>
<td>64 kbit/s</td>
<td>50 km</td>
<td>333</td>
<td>166</td>
<td>5E-2</td>
<td>2.5e-2</td>
</tr>
<tr>
<td></td>
<td>1000 km</td>
<td>6666</td>
<td>3333</td>
<td>1</td>
<td>0.5</td>
</tr>
<tr>
<td></td>
<td>10000 km</td>
<td>66666</td>
<td>33333</td>
<td>10</td>
<td>5</td>
</tr>
<tr>
<td></td>
<td>38000 km</td>
<td>253333</td>
<td>126666</td>
<td>38</td>
<td>19</td>
</tr>
<tr>
<td>2 Mbit/s</td>
<td>50 km</td>
<td>333</td>
<td>166</td>
<td>2</td>
<td>1</td>
</tr>
<tr>
<td></td>
<td>1000 km</td>
<td>6666</td>
<td>3333</td>
<td>32</td>
<td>16</td>
</tr>
<tr>
<td></td>
<td>10000 km</td>
<td>66666</td>
<td>33333</td>
<td>314</td>
<td>162</td>
</tr>
<tr>
<td></td>
<td>38000 km</td>
<td>253333</td>
<td>126666</td>
<td>1194</td>
<td>597</td>
</tr>
<tr>
<td>155 Mbit/s</td>
<td>50 km</td>
<td>333</td>
<td>166</td>
<td>122</td>
<td>61</td>
</tr>
<tr>
<td></td>
<td>1000 km</td>
<td>6666</td>
<td>3333</td>
<td>2434</td>
<td>1217</td>
</tr>
<tr>
<td></td>
<td>10000 km</td>
<td>66666</td>
<td>33333</td>
<td>24330</td>
<td>12165</td>
</tr>
<tr>
<td></td>
<td>38000 km</td>
<td>253333</td>
<td>126666</td>
<td>92456</td>
<td>46228</td>
</tr>
</tbody>
</table>

Table 4-1 Transmission and Propagation Times

4.1.2 Preventive Controls

In contrast to reactive control, preventive control techniques attempt to prevent congestion by taking appropriate actions before it actually occurs. It provides a fair allocation of bandwidth, by requiring at times of high network load levels, that each connection’s traffic flow remains within specified bounds as negotiated.
4.1.2.1 Connection Admission Control (CAC)

When using virtual circuits, data transfers are performed in three stages: virtual circuit (VC) establishment, data transfer, virtual circuit closure. At the admission control stage, the sender negotiates the quality of service parameters with the network. The connection admission control decides whether or not a connection can be accepted. CAC is defined as the set of actions taken by the network at the call set-up phase (or during call re-negotiation phase) in order to establish whether a VC or VP connection can be accepted or not. A connection request is accepted only when sufficient resources are available to establish the call through the whole network at its required QoS and to maintain the agreed QoS of existing calls. This also applies to re-negotiation of connection parameters within a given call.

The CAC has to be provided with the following information:

- Source Traffic Characteristics
- Required Quality of Service (QoS)

The parameters to characterise traffic have yet not been completely determined by the various standards bodies, but several parameters have been proposed (see Chapter 4). Accurate and simple characterisation of traffic sources is very important. Bad characterisation of the source could cause wrong decisions by the CAC.
Traffic and Congestion Control Framework for ATM over Satellite

**Bandwidth Allocation**

Bandwidth allocation deals with determining the amount of bandwidth required by a connection for the network to provide the required QoS. There are two alternative approaches for bandwidth allocation: deterministic multiplexing and statistical multiplexing.

In deterministic multiplexing, each connection is allocated its peak bandwidth. Although this can eliminate cell level congestion almost totally, it goes against the philosophy of the ATM framework since it does not take advantage of the multiplexing capability of ATM and restricts the utilisation of network resources.

An alternative method is statistical multiplexing. In this scheme, the amount of bandwidth allocated in the network to a VBR source is less than its peak, but necessarily greater than its average bit rate. This, allocated bandwidth which is less than the peak, is called *virtual* or *effective* bandwidth. Then, the sum of peak rates of connections multiplexed onto a link can be greater than the link bandwidth as long as the sum of their effective bandwidths is less than or equal to the provisioned link bandwidth.

The main difficulties in calculating the value of the effective bandwidth of a connection are:

- Guaranteeing the QoS requirements of individual connections;
- Assuring that the QoS provided to existing connections does not degrade to unacceptable levels when multiplexed together with new connections.

**CAC Algorithms**

A variety of CAC algorithms have been proposed. The aim is to produce an algorithm that is simple (in terms of processing and storage requirements) and efficient (to allow statistical multiplexing gain).

Various CAC algorithms to allocate an *effective* bandwidth to *N* homogeneous sources on an ATM link have been investigated in [ORS94] concluding that the fluid-flow approximation provides the most accurate results. Therefore the fluid-flow
approximation will be used for admission control calculations. A program for calculations is provided in Appendix C.

The Cell Loss Rate (CLR) must be evaluated for connection admission control as stated before. If the traffic parameters are quantitative, and a user specifies their value, CAC must assign a bandwidth guaranteeing the CLR objective for any cell arrival process satisfying the specified traffic parameter values.

The accuracy of the CAC algorithms proposed so far depends on the burstiness and predictability of the traffic flows. Bursty and unpredictable heterogeneous traffic incurs a higher risk of congestion and compels a lower utilisation factor. The algorithms proposed for assigning bandwidth to a heterogeneous mix of traffic are therefore either too complex (requiring long calculations) or inaccurate. The design of a CAC that is simple and can allocate bandwidth to a heterogeneous mix of traffic is for further study, and not in scope of this thesis.

4.1.2.2 Usage/Network Parameter Control (UPC/NPC)

After accepting the connection, the network must ensure that the traffic characteristics are kept within the values of traffic parameters negotiated during the establishment period. This policing mechanism is called Usage Parameter Control (UPC) and Network Parameter Control (NPC). The UPC function is performed at the User-Network Interface (UNI), whereas the NPC function is performed at the internetwork Network-Node Interface (NNI). Because of the identical nature of the functions being performed, the following sections will consider only the UPC, but the same discussion is applicable to NPC.

**UPC Functions**

Usage parameter control (also called policing) is defined as the set of actions taken by the network to monitor and control traffic in terms of conformity with the agreed traffic contract at the user access. The main purpose is to protect network resources from misbehaviour that could affect the QoS of other established connections. It does this by detecting violations of negotiated parameters and taking appropriate actions.

The monitoring task for usage parameter control and network parameter control is performed by:
• Checking the validity of VPI and VCI value (i.e. whether or not valid VPI/VCI values have been assigned)

• Monitoring the traffic entering the network from each active VCC and VPC in order to ensure that parameters agreed upon are not violated. This monitoring action is performed at the termination of the first VC link for virtual channel connections and the first VP link for virtual path connections.

The requirement where the UPC function is to be situated again illustrates the advantages of the VP concept within ATM networks. If a customer has VP that is established as a user-to-user VP, there is no requirement to perform UPC on the individual VCs, only on the VP as the VC link is not terminated within the network.

**UPC Actions**

If UPC detects a violation of the negotiated traffic contract, it can either discard cells or tag them for discard when the network is congested. One recommended way for tagging is to use the cell loss priority (CLP) bit in the header [ITUT96b]. When a cell with high priority (CLP=0) is tagged, UPC sets the CLP of that cell to low priority (CLP=1).

It is worth noticing that the original aim of introducing the CLP bit as a field in the header of the ATM cells was purely to attribute priority levels to cells. This leads to the present problem of using the CLP bit for two different purposes, as the tagged cell becomes indistinguishable from the cells that originally had low priority. An alternative tagging method would use a payload type field in the cell header to express the tag. Generally, one bit is not sufficient, to distinguish the priority of so many service classes. Especially considering that some video coding schemes need at least three priority levels for the encoded traffic.

Excessive action of the UPC/NPC on any connection are part of the overall network performance degradation and should remain of a very low probability. Quantification of this probability is within the scope of ITU-T Rec.I.356 [ITUT96a].

The access control reacts at the cell level and its performance will be investigated in more detail in Chapter 6.
4.2 Priority Control

To efficiently utilise the link capacity, in particular for a satellite link, statistical multiplexing must realise different QoS requirements for different types of services. Such requirements are commonly measured by cell loss rate and end-to-end cell delay. The relative significance of these two measures varies from one application to another. Real-time services, which require timely and synchronous delivery of cells, have stringent delay requirements. Other non real-time services require stringent loss requirements.

To provide delay requirements, time (or delay) priority mechanisms can be implemented for buffers at satellite ATM nodes. Time priority deals with the order in which cells leave the waiting area. This means that cells with higher time priority will experience a shorter delay. Space (loss) priorities on the other hand are used to achieve different cell loss rates. Cells with higher space (loss) priority will experience lower cell loss. Due to their simplicity, space priorities are preferred over time priorities in high-speed switching. The ATM standards explicitly support space priority, by the provision of a cell loss priority bit in the ATM cell header. Time priorities are not explicitly supported in the standards.

Ideally each traffic class, distinguished by its QoS requirements, should have its own designated set of buffers at each switching node. If per-VC queuing is used delay and loss priorities can be met more easily by the network. However when shared buffers are used prioritising traffic is essential. This allows resource sharing to be controlled by algorithms which favour one traffic class over another when necessary, to the benefit of all.

Considering that satellite ATM networks support many services with different characteristics and the long GEO satellite link propagation delay, the use of a combination of both space and time priority control is proposed.

4.2.1 Time Priority

The delay priority can be used by a scheduling algorithm for a mixed traffic at a switching node in a satellite ATM network. This switching node may also be an OBS satellite. The effect of time priorities is to decrease the delay for the higher priority
services at the expense of increasing the delays for the lower priority services while still providing the required QoS to all services.

Although there is no explicit provision in the standards for distinguishing different levels of time priority, it is possible to use the VPI/VCI values in the header. On entry to a switch, the VPI/VCI values are used to determine the outgoing port required, so it is a relatively simple extension to use these values to associate a time priority and choose one of maybe a number of priority buffers at the output port.

Several scheduling algorithms have received wide attention [HYMA93, CHAO94, DAIL95, LING96]. Due to its simplicity, many networks use the First-In-First-Out (FIFO) algorithm which applies the same service discipline to all cells independent of their performance objectives. In FIFO, packets are served in the order in which they arrive.

To allow simple differentiation of cells the Static Priority (SP) scheduling algorithm is often considered for scheduling real-time traffic in high speed networks. With SP, cells are given a level of time priority before they enter the network and a FIFO queue is maintained for each priority. Cells of the highest priority class are always served before cells of any other class. Either a separate buffer can be used for each priority or a shared buffer can be used. The use of a shared buffer increases the utilisation of the buffer but also increases the complexity since classes have to be treated differently within the buffer.

Note that SP scheduling is class dependent. Thus this policy will not be efficient if a higher priority class is large compared to a low priority class. Therefore the strict priority policy could cause QoS violations for the other traffic classes while high priority class delays are far from the allowed limits. In these cases, overall performance can be improved by delaying high priority cell within their QoS bounds. It is also important to reduce the loss rate incurred by some cells over others. Since loss rate objectives may vary greatly from application to another the loss priorities have also to be considered in the priority control.
4.2.2 Space Priority

Loss priorities for buffer access are called space priorities, as they deal with priorities regarding the utilisation of the space in the buffer. One bit in the ATM cell header is reserved to implement space priorities. The policing function and buffer access mechanism may use this bit to mark cells based on a negotiated contract for bandwidth allocation. Cells which violate the traffic contract are marked as low priority and are rejected to enter the network when congestion occurs.

One bit is usually not sufficient to distinguish between the cell loss requirements of multiple services hence we propose that a loss priority is assigned by the network to each individual VC/VP. By implementing multiple loss priorities it is possible to achieve very low loss probabilities, just for those services that require it, and this leads to a significant improvement in the traffic load that can be admitted to the network.

Two space priority schemes have been proposed and studied extensively for ATM buffers: the push-out scheme; and partial buffer sharing. In the push-out mechanism, all cells enter one shared buffer up to the maximum buffer size. If a high priority cell arrives at a saturated buffer that contains low priority cells, the last low priority cell to enter the buffer is discarded and its place is given to the high priority cell. A low priority cell arriving to find a full buffer is always discarded. The partial buffer sharing scheme reserves a part of the buffer for high priority cells only. If the queue is below a threshold size, then both low and high priority cells are admitted onto the queue. Above the threshold only high priority cells are accepted.

The push-out scheme achieves only slightly better performance than partial buffer sharing. However the implementation and buffer management for the push-out mechanism is more complex. Thus the Partial Buffer Sharing (PBS) provides the flexibility to adapt the system to a certain load situation by adjusting the threshold value. Generalisation of this method for more than two levels of priority is called Nested Threshold Cell Discarding (NTCD) mechanism. Under NTCD (Figure 4-2), the buffer is logically partitioned by a number of threshold levels which is equal to the number traffic classes. Cells of priority class $i$ enter the buffer up to a threshold level $T_i$. If the buffer occupancy is above $T_i$, arriving cells of class $i$ are dropped. The choice of the threshold values is very important for maximising the admissible load of
each traffic class. A substantial increase is possible, particularly if the difference in cell loss probability requirements is large.

![Image of NTCD with a shared buffer and n thresholds]

Figure 4-2 NTCD with a shared buffer and n thresholds

### 4.3 Application of both Space and Time Priorities

Most work in the field of traffic and congestion control has assumed a single service type. However, there will be a number of service classes in ATM networks. Ideally, each service class, distinguished by its QoS requirements, should have its own designated set of buffers at every switching node in the network. This allows resource sharing to be controlled by algorithms operating in both space and time to favour one traffic class over another when necessary, to the benefit of all. The traffic classes are distinguished in the time and space dimensions by their cell-delay and cell-loss priorities.

The proposal to use of both time and space priorities for satellite ATM networks can be justified as the following:

- loss sensitive cells have stringent loss requirements which can not be met by the sole implementation of a time priority mechanism.

- space priorities are not sufficient because of the real-time constraint of services, particularly in a satellite environment. Some cells may be useless to transport after a certain queuing time.

- in order not to disrupt cell sequence integrity, time priorities should be assigned only on a call basis whereas space priority could be assigned on a cell basis.

An effective method is to assign each service, during call setup, a service class with a delay and a loss priority. Real-time (rt) services are assigned with higher delay priorities to have lower Cell Transfer Delay (CTD), and loss sensitive services are assigned with higher loss priorities to have a smaller Cell Loss Rate (CLR). Cells with higher delay
priority will be send first and cells with lower loss priority will be discarded first when the buffer is full.

The work by [CHA094] proposes the pushout scheme to distinguish between loss priorities and scheduling of cells according to their delay priorities. Four traffic classes, combining low and high delay and loss priorities are considered, with buffers assigned to each service class. Similar work was done by [KRUN94] by using partial buffer sharing with nested thresholds for loss priorities, instead of the pushout scheme. In this scenario all non real-time traffic was assigned the same loss priority and only real-time traffic was assigned low and high loss priorities. Two buffers for rt and nrt services were used.

4.4 Traffic Shaping

Despite of the perfect regulation and perhaps smoothing of arriving traffic at the UPC, a satellite network is inherently capable of creating long bursts of cells. The underlying cause of the altering cell pattern along a VC is the cell delay variation or jitter, which occurs mainly due to the buffering of cells. The satellite access schemes like TDMA also change the characteristics of the cell stream by buffering cells and then transmitting them at a higher rate. Shortening of cell interarrival times which creates the ‘clumping of cells’ can increase the burstiness of a cell stream.

Since the satellite (or terrestrial) network can destroy the original smoothness of a stream, shaping the cell stream is required at the network node terminating the satellite network, in order to make it conform with its original pattern description.

Traffic shaping ensures that the user required QoS (CDVT) is provided and is part of the congestion avoidance framework. A traffic shaping function which tailors traffic according to the particular characteristics of the satellite environment might be typically located at:

- the originating equipment to ensure that the transmitted cell stream complies with the traffic contract and will be accepted by the network’s UPC function. In this case the shaper may be implemented in the UPC but the traffic characteristics may again be varied by the satellite network.
• the input port of the network, to ensure that the cell stream entering the network has the expected traffic characteristics.

• the output of the network to ensure that the outgoing cell stream meets the performance expectations of the destination equipment or the next network.

The problem is that a shaper is needed for every VC which increases the complexity of the terminal or switching node where the shaper is located. Furthermore buffering in the shaper creates additional cell delay and perhaps cell loss.

4.5 Satellite Network Architecture

Two satellite network architectures for the support of ATM services will be described. The first architecture uses existing bent pipe repeater satellites and interconnects fixed networks. The second network architecture employs a satellite with cell switching capabilities in order for fixed/portable terminals to access the ATM B-ISDN directly. The novelty of this architecture is its ability to multiplex and switch all the traffic onboard the satellite reducing the terminal complexity and cost. There are two variants of the described architectures which are not in scope of this thesis. One for the interconnection of mobile ATM networks and the other for direct mobile terminal access to the B-ISDN. These architecture need a mobility enhanced NNI and/or UNI and are not in scope of this thesis.

4.5.1 Interconnection of Broadband Islands

In the first scenario, satellite links provide high bit rate links between broadband nodes or broadband islands. The interfaces with satellite links in this mode are of the Network-Node Interface (NNI) type. This scenario is characterised by a relatively small number of large earth stations which have a relatively large average bit rate. Since only a small number of larger earth stations is required, the cost of the earth station has not a big impact on the suitability of the satellite solution.

The RACE CATALYST project was a demonstrator for this scenario and showed the compatibility of satellite technology with ATM and the terrestrial B-ISDN. The equipment developed during the CATALYST project has been able to interconnect ATM testbeds as well as existing networks such as DQDB, FDDI and Ethernet.
Traffic and Congestion Control Framework for ATM over Satellite networks, all using ATM. A detailed explanation of the system design and performance is provided in [POLE94, LOUV94, HADJ94, SUN96, SUN97].

Figure 4-3 shows the reference configuration for resource management, traffic and congestion control for the scenario where a number of broadband nodes are interconnected by satellite. The network architecture is using a bent-pipe satellite in which the hub acts as a network management centre for channel allocation and policing, but VSATs that have established a connection talk directly to each other via the satellite. In this case the propagation delay is a single satellite hop.

Since the expected traffic at these nodes is high, statistical multiplexing can be performed at the ground stations, hence making maximum use of the available air interface bandwidth possible and not necessarily requiring complex access schemes.

![Diagram](image)

**Figure 4-3 Traffic and Congestion Control for Scenario 1**

### 4.5.2 Direct Access to ATM/B-ISDN

In the second scenario the satellite system is located at the border of the B-ISDN and provides access links to a large number of users. This scenario is characterised by a large number of terminals with relatively low average and peak bit rates (up to 2Mbit/s) and a few gateway earth stations. The traffic at the user terminals is expected to show large fluctuations. Therefore the multiple access scheme will considerably effect the performance of the system. Furthermore the cost and size of the terminal
Traffic and Congestion Control Framework for ATM over Satellite

have a large impact on the suitability of the satellite solution. The objective is to interconnect a large number of users and make B-ISDN access affordable by lowering terminal cost and providing bandwidth on demand.

The 155 and 622 Mbit/s transmission rates conventionally associated with ATM are well above the maximum rates possible with today’s portable terminal technology. However, in practice, most individual users will usually require significantly lower traffic rates, especially if there are only a few data or voice terminals located at a remote location. This large number of users with bursty traffic will need a cost-efficient way to communicate between each other and occasionally access the B-ISDN/ATM network. This architecture does not necessarily need a central reservation/control unit on ground (Hub). Figure 4-4 shows the reference configuration for traffic and congestion control for the scenario where a high number of terminals want to directly access the B-ISDN.

![Diagram of Traffic and Congestion Control for Scenario 2](image)

**Figure 4-4 Traffic and Congestion Control for Scenario 2**

Network requirements in this scenario are for a full meshed point-to-point and point-to-multipoint system. Suitable satellite architectures for meshed networks are expected to employ a spot beam coverage pattern to achieve the high uplink and downlink gain required for mesh connectivity between portable terminals. Although the advantages and potential of new-generation satellite systems are promising, there are undoubtedly many challenging technical issues, which require further studies. The satellite is known as being bandwidth-, power-, and mass-limited. In addition,
radiation hardness is another concern. New techniques to improve the transmission performance are needed to reduce the size and cost of terminals and the payload power and bandwidth requirements. Efficient on-board processing and switching schemes with low power consumption, low complexity, and high performance are the key to keep the payload power and mass within a practically affordable range. Hence the aim is to only have the necessary functions on-board the satellite. These functions can include switching, channel set-up and for added flexibility and improvement in link performance.

4.5.2.1 Ground Segment

The functions of a portable ultra-small aperture terminal (USAT) are shown in Figure 4-5. The arriving cells are optionally controlled by the Usage Parameter Control (UPC) for their conformance with the traffic contract. Only the peak rate is controlled for CBR and ABR traffic. However for VBR sources also the sustainable cell rate is controlled using a dual Leaky Bucket (LB) configuration. The UPC parameters for the ABR type traffic are adaptive (dynamic LB) and changed according to the explicit rate feedback signal from the satellite. Details about the dimensioning and performance of the Leaky Bucket are provided in Chapter 6. It is important to note that the UNI for portable terminals is located at the access to the satellite network. In this case the traffic is not necessarily controlled in the terminal but at the access to the satellite. This means that the UPC functionality may also be on-board the satellite, integrated with the MAC which controls the access to the satellite air-interface bandwidth.

![Terminal Block Diagram](image)

**Figure 4-5 Terminal Block Diagram**
The cell scheduling module is responsible for scheduling CBR, VBR, ABR, GFR and UBR traffic. CBR and rt-VBR connections have high delay priority and maximum burst utilisation is achieved by detecting the silence periods of VBR connections. ABR and GFR traffic have higher loss priority than UBR. The cell processor also performs a number of functions. First the processing function sorts the cells by VP and assigns new VPIs. Then VPs are mapped into satellite packet addresses for on-board routing and satellite ATM headers are assigned. The packets are then scrambled before modulation for transmission. The procedure for the destination terminal primarily consists of the procedure above in reverse order.

4.5.2.2 The Space Segment

At present, most current satellites act as transparent repeaters (scenario 1). They simply receive a RF signal, amplify it and then re-transmit it to the Earth. To improve satellite link capability using spot beam coverage, such a technique becomes extremely limited. In the near future, the satellites are expected to contribute effectively to the scope of B-ISDN networks and viewed not merely as a repeater, but rather as a network node in its own right in a hopefully integrated space/terrestrial network.

On-board processing (OBP) satellites with high-gain multiple spot-beams and switching capabilities have been considered as key elements of new-generation satellite communications systems. These satellites support small, cost-effective terminals and provide the required flexibility and increased utilisation of resources in a bursty multimedia traffic environment.

OBP introduces some issues for the design and analysis of the satellite architecture. Many considerations previously the concern only of the ground segments now shift to the space segment. The on-board processor allocates bandwidth on demand and performs statistical multiplexing. This essentially changes the nature of the satellite from a deterministic system to a stochastic system. In a stochastic system, the arriving traffic is random and statistical fluctuations may cause congestion, where cell loss due to buffer overflow might occur. A possible payload block diagram for an on-board cell switching satellite is shown in Figure 4-6.
The output processor is responsible for some very important task such as monitoring the buffers for indication of congestion, scheduling and discarding of cells. In order to provide the required QoS of each service class the scheduling and discarding of cells has to be carried out according to time and loss priorities. This will be investigated in the next section.

Figure 4-6 Payload Block Diagram.

4.6 Scheduling and discarding of cells on-board the satellite based on space and time priorities

The Multiple Shared Buffer Scheduling (MSBS) policy which uses a loss priority queue for each traffic class with similar delay requirements, sharing a single server is proposed for scheduling. Cells entering the buffer are ordered first by loss priority with higher loss priority cells going towards the head of the queue, then within each loss priority class the order is FIFO (similar to the SP for scheduling). For this buffer model the allocation of the server to each of the buffer classes is done by dividing time into periods called cycles consisting of up to $L$ cells. Each cycle is further divided into $N$ subcycles (the number of buffers). During each subcycle, the link is allocated to the corresponding traffic classes with similar delay requirements. The buffer manager can dynamically adjust the length (in cells) of each subcycle according to the traffic load and mix (as the MARS policy [HYMA93]). If the served buffer is empty during a subcycle then the other buffer is served, instead of remaining in idle state. Thus whereas traffic classes requiring different delay requirements are placed in different buffers, cells with similar delay but different cell loss requirements are placed in the same buffer with different loss priorities.
4.7 Performance Evaluation of the MSBS strategy

Different service classes are considered in order to evaluate the performance of the scheduling and discarding strategy considering both delay and loss priorities. These are representing the service classes as described by [ATMF96a], namely CBR, rt-VBR, nrt-VBR, ABR and UBR. Table 4-2 shows the delay and loss priorities assigned to each service class. Note that there are two services which belong to the rt-VBR service class, but have different cell loss requirements. The traffic of the user requiring less stringent cell loss is entitled LP rt-VBR. Voice, for example may tolerate CLR of up to $10^{-3}$. LP does not imply tagged traffic since non conforming traffic is discarded by the UPC. It is assumed that the emergency video transmission requires a lower cell loss then nrt-data transfer such as ftp. Furthermore CBR and rt-VBR are assumed to have the same loss and delay priorities. All traffic sources except the CBR source are modelled by the on-off source model. The traffic parameters are given in Table 3-4 and the mean burst duration was taken as a function of the mean load. For a mean load of 0.1 the mean burst duration for data services was taken 20 cells.

<table>
<thead>
<tr>
<th>Service Class</th>
<th>Loss Priority</th>
<th>Delay Priority</th>
</tr>
</thead>
<tbody>
<tr>
<td>Emergency Video (rt-VBR)</td>
<td>3</td>
<td>1</td>
</tr>
<tr>
<td>Data (nrt-VBR and ABR)</td>
<td>2</td>
<td>0</td>
</tr>
<tr>
<td>Voice (rt-VBR)</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>WWW access (UBR)</td>
<td>0</td>
<td>0</td>
</tr>
</tbody>
</table>

Table 4-2 Example Loss and Delay Priorities for Service Classes

Simulations will be used for the evaluation of the MSBS policy because of the complexity of analytical solutions when more than two service classes with different priorities are present. More details about the reason for using simulations and the used simulation tool BONeS Designer is provided in Appendix A.

In the following experiments, the performance of the MSBS policy is investigated in terms of CLR and average cell delay. Other parameters were obtained and are reported briefly (e.g. average queue length and delay variance). Simulation run times were selected so that the traffic with the most severe performance looses more than 5000
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cells, so that results could be accurate. The simulation model and details of the MSBS policy can be found in Appendix B.

The rt-buffer was fixed at 120 and the nrt-buffer size was varied to observe the impact of the buffer size on loss and delay of nrt-services. The total server cycle $L$ was fixed as 24. The simulation parameters for the first experiment are shown in Table 4-3.

In order to observe the affects of the subcycle periods on the cell loss and delay of each service class the LP rt-VBR load is initially fixed at 0.35. This results in a total load of 0.85 with simulation parameters shown in Table 4-3.

<table>
<thead>
<tr>
<th>Service Class</th>
<th>Load</th>
</tr>
</thead>
<tbody>
<tr>
<td>CBR</td>
<td>0.1</td>
</tr>
<tr>
<td>rt-VBR</td>
<td>0.1</td>
</tr>
<tr>
<td>UBR</td>
<td>0.1</td>
</tr>
<tr>
<td>LP rt-VBR</td>
<td>varied</td>
</tr>
<tr>
<td>nrt-VBR</td>
<td>0.1</td>
</tr>
<tr>
<td>ABR</td>
<td>0.1</td>
</tr>
</tbody>
</table>

Table 4-3 Simulation parameters for MSBS policy experiment 1

Figure 4-7 (a) CLR (b) Mean delay vs real time cycle for nrt-buffer=480

Figure 4-7 and Figure 4-8 show the CLR and mean cell delay for various rt-cycle values. It can be seen that both delay and loss priorities can be satisfied, when the rt-cycle is selected according to the load and buffer size. A larger buffers for nrt-traffic is required to achieve low cell loss at the expense of longer delays which might not be
critical for nrt-traffic. Increasing the rt-cycle decreases the cell loss of rt-services increasing the cell loss for nrt-traffic.

Figure 4-8 (a) CLR (b) Mean delay vs real time cycle for nrt-buffer=960

For a nrt-buffer size of 480, a rt-cycle of 21 and an offered load with simulation parameters from Table 4-3 the cell loss and mean delay as a function of the load are shown in Figure 4-9. The CLR of the ABR service class is lower than for LP rt-VBR. In contrast the delay experienced by ABR cells is higher than LP rt-VBR cells, as required.

Figure 4-9 (a) CLR (b) Mean delay of MSBS vs mean load

A similar experiment was repeated with the following parameters shown in Table 4-4. The reason for varying the low priority traffic classes for both scenarios, is to observe how they affect the higher priority traffic classes. The rt-buffer is again fixed at 120 cells and Figure 4-10 to Figure 4-12 show the CLR and mean delay for each service class, as a function of the rt-cycle for various nrt-buffer values. The ABR and nrt-VBR load was fixed at 0.35 for a total mean load of 0.85 for the rt-cycle simulations. As the ABR and nrt-VBR service class requires a lower CLR, the ideal rt-cycle value
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is a function of the rt and nrt-buffer size. For a nrt-buffer size of 960, a rt-cycle of 15
would satisfy both delay and loss requirements. Thus by changing the subcycles
according to the mix of traffic it is possible to achieve the required loss and delay
requirements of each service class.

<table>
<thead>
<tr>
<th>Traffic Class</th>
<th>Load</th>
</tr>
</thead>
<tbody>
<tr>
<td>CBR</td>
<td>0.2</td>
</tr>
<tr>
<td>rt-VBR</td>
<td>0.1</td>
</tr>
<tr>
<td>UBR</td>
<td>0.1</td>
</tr>
<tr>
<td>LP rt-VBR</td>
<td>0.1</td>
</tr>
<tr>
<td>ABR and nrt-VBR</td>
<td>varied</td>
</tr>
</tbody>
</table>

Table 4-4 Simulation parameters for MSBS policy experiment 2

Figure 4-10 (a) CLR (b) Mean delay vs real time cycle for nrt-buffer=300

Figure 4-11 (a) CLR (b) Mean delay vs real time cycle for nrt-buffer=480
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Figure 4-12 (a) CLR (b) Mean delay vs real time cycle for nrt-buffer=960

In order to show that the subcycles have to be changed according to the traffic load, the rt-cycle has been fixed at 16 and the ABR and nrt-VBR load varied as specified in Table 4-4. Figure 4-13 clearly shows that the rt-cycle has to be modified according to the load of each service class in order to provide a certain QoS. Otherwise the CLR of some sources may increase drastically as the load increases.

Figure 4-13 (a) CLR (b) Mean delay vs mean load for a fixed rt-cycle of 16

Figure 4-14 (a) Mean buffer occupancy (b) variance of delay for rt-cycle of 16
4.8 Summary

This Chapter has defined the different methods and algorithms which are part of the proposed traffic and congestion control mechanism for a scenario where many users with bursty traffic want to access the ATM/B-ISDN. Both preventive and reactive control schemes, which are applicable at different time scales will be used. Preventive control consists of CAC and UPC. The fluid-flow approximation will be used as CAC algorithm and is presented in Appendix C. The UPC which controls the traffic accessing the network will be investigated in Chapter 5.

The importance of traffic shaping to ensure that the user required QoS (CDVT) is met and the typical locations of shaping devices were also discussed.

As the emphasis is to maximise the utilisation of resources and minimise earth station cost an on-board switching satellite with Priority Control, Congestion Control and Radio Resource management functions will be used.

The scheduling and discarding of cells which belong to services with different QoS requirements necessitates the use of cell scheduling and discarding mechanisms considering both time and delay priorities.

In order to be able to provide the different delay and loss requirements to each service class, services are assigned time and space priorities to maximise utilisation of resources. The scheduling and discarding of cells at the terminal and on-board the satellite are done according to these priorities. A multiple shared buffer scheduling (MSBS) policy was proposed and evaluated in this Chapter, which considers both delay and loss priorities. It has been shown that both delay and loss requirements can be met by the MSBS scheme.

The novelty of the MSBS proposal was to use the same buffer for services with similar delay requirements and different loss requirements. Scheduling is performed according to the delay priority and push-out is used as discarding strategy. Although the push-out scheme does not require the dimensioning of multiple thresholds for different loss priorities, the performance of the service classes in the same buffer is determined by their traffic load. Hence the complexity of MSBS using push-out is less compared to partial buffer sharing with nested thresholds, where lower cell loss rates
for services sharing the same buffer might be obtained (assuming optimum dimensioning of the multiple thresholds).

The seamless extension of existing ATM protocols to wireless nodes is an important issue and the use of standard ATM protocols to support seamless wired and wireless networking is possible by incorporating a new radio specific protocol sublayer into the ATM protocol model. Considering that satellite communication uses multiple access on a shared medium, a MAC layer, which is not present in traditional ATM networks is needed. The MAC protocol plays a central role as means of accessing the Radio Physical Layer (RPL) from the ATM layer. It must extend the statistical multiplexing of ATM to the air interface considering the negotiated QoS. MAC protocols for ATM over satellite will be investigated in Chapter 6.

It has been shown that for a reactive control scheme the number of cells transmitted on the satellite link before any feedback can be received, is dependent on the terminal bit rate and the satellite link propagation delay. It is expected that reactive control algorithms will perform better for lower propagation delays and lower terminal bit rates. The performance of a combined preventive-reactive control scheme for a satellite environment will be investigated in Chapter 7.
5. Usage Parameter Control Performance Evaluation

Usage Parameter Control (UPC) and Network Parameter Control (NPC) perform similar functions at different interfaces. The main purpose is to protect network resources from misbehaviour that could affect the QoS of other established connections. The UPC function is performed at the User-Network Interface (UNI), whereas the NPC function is performed at the internetwork Network-Node Interface (NNI). Because of the identical nature of the functions being performed, the following sections will consider only the performance of the UPC, but the same discussion is applicable to NPC. The emphasis is on minimising excess traffic entering the satellite network and the accurate policing of the declared parameters.

5.1 UPC Performance Parameters

Two performance parameters have been identified, to be considered when assessing the performance of UPC/NPC mechanisms [ITUT96b].

5.1.1 Response Time

The time to detect a given situation that involves non-conforming cells on a VPC/VCC under given reference conditions. This time should ideally be small so that violations of the traffic contract can be detected in a relatively short time.

5.1.2 Transparency (Accuracy)

The accuracy with which the UPC/NPC initiates appropriate control actions on a cell stream in which some cells are non-conforming and the accuracy to avoid inappropriate control actions on a stream of conforming cells for the same set of reference conditions.

A method to determine the ratio of non-conforming cells to a negotiated cell rate at a given interface is defined in ITU-T Rec. I.356 [ITUT96a]. A 1-point measurement process computes the ratio $\gamma_m$ between the number of cells exceeding the traffic contract and the total number of submitted cells.

The transparency of a UPC/NPC mechanism can be defined by the accuracy with which this mechanism approaches the ideal mechanism, i.e. the difference between the reference policing ratio $\gamma_m$ and the actual policing ratio $\gamma_e$. A positive difference
means that the UPC/NPC is taking less policing action than the measurement process would do. A negative difference means that excessive policing action is taken by the UPC/NPC.

A UPC/NPC performance requirement relating to accuracy is as follows [ITUT96b]:

For cell rate control (either peak or sustainable cell rate), the UPC/NPC should have an accuracy of at most 1% larger cell rate admission into the network than the cell rate used in the conformance definition (for cell rates larger than 160 cells/s). For cell rates between 100 cells/s and 160 cells/s, the accuracy is specified as 1.6 cells/s (which is 1% of 160 cells). Although this is a requirement on the capability of the UPC, a network operator is not required to set the parameters of the UPC to be within the specified margin.

Cell delay and cell delay variation introduced by the UPC/NPC is also part of the delay and delay variation allocated to the network.

5.2 The Effects of Cell Delay Variation

Before cells arrive at the UPC, they may experience random delays due to physical functions in the terminal equipment, medium access controls in customer equipment and ATM multiplexing in the network infrastructure. Below both aspects of CDV (clumping and dispersion effect) are explained.

Some cells experience a shorter delay than the preceding ones. During some transient periods, cells are passing closer than expected from each other; therefore the observed PCR is larger than the source cell rate. This is the clumping effect. If the connection PCR is controlled, the policing mechanism has to take into account this clumping effect in order not to systematically discard cells which experience a shorter queuing delay than the preceding ones.

Some cells experience a larger delay than all the preceding ones so that interarrival times can be larger than \( T \). A “gap” occurs in the cell stream. This is the dispersion effect. In order to overcome this dispersion effect, when circuit emulation is needed, a smoothing equipment is implemented at the receiving side which buffers a sufficient number of cells in order to fill any gap in the cell stream [GRAV93].
Since CDV can change the traffic characteristics of sources, the UPC should take CDV into account when deciding whether cells are violating the traffic contract. The user can require stringent CDV on his applications. In [ITUT96a] the maximum CDV for the stringent traffic class, allowed to be introduced by the network, is 3ms.

However when the UPC takes CDV into account, it introduces tolerances in the amount of traffic from a source. It is thus possible that some cells that are violating the traffic contract are not discarded. One way to compensate for CDV is to use a buffer before the UPC. Although this process called shaping can reduce clumping, it does so at the expense of delaying the traffic generated by the source.

### 5.3 UPC Algorithms

A number of desirable features of the control algorithm can be identified as follows:

- Capability of detecting any illegal traffic situation.
- Selectivity over the range of checked parameters (i.e. the algorithm could determine whether the user behaviour is within an acceptance region)
- Rapid response time to parameter violations
- Simplicity of implementation

Several methods to control peak-rate, mean-rate and different load states within several time-scales have been studied. The Leaky Bucket (LB) is generally agreed to achieve the best performance compromise of the mechanisms studied for policing. It was first introduced in [TURN86]. Since then a number of variants have been proposed. Recently the ATM-Forum [ATMF94] and ITU-T [ITUT96b] adopted the Generic Cell Rate Algorithm (GCRA) to describe source parameters and also recommended it for UPC. The GCRA is a reference algorithm that is used to define the cell conformance of a cell stream to the negotiated value of a cell rate, assuming that a tolerance is allocated. There are two equivalent versions of this algorithm: the virtual scheduling algorithm and the continuous state leaky bucket algorithm.

#### 5.3.1 Virtual Scheduling Algorithm

The virtual scheduling algorithm is shown in Figure 5-1. The actual arrival time of the \( n^{th} \) cell, \( t(n) \), is compared with its Theoretical Arrival Time (TAT), which is the expected arrival time under the assumption that cells are spaced equally in time with
distance $T$. The algorithm is intended to ensure that the cell rate is not greater than $R = T^{-1}$ on average, with some tolerance dependent on $\tau$; that is, cells will not arrive too much earlier than their TAT. The cell is deemed to be conforming if $t(n) > \text{TAT} - \tau$; otherwise it is nonconforming (too early). The TAT for the next cell is calculated as a function of $t(n)$. If the $n^{th}$ cell is conforming and $t(n) < \text{TAT}$, then the next TAT = $\text{TAT} + T$. If the cell is conforming and $t(n) \geq \text{TAT}$, then the next TAT = $t(n) + T$. Nonconforming cells are not counted in the update of the TAT.

![Figure 5-1 Virtual scheduling algorithm: (a) cell is too early and nonconforming; (b) cell is early but conforming; (c) cell is conforming](image)

**5.3.2 Continuous-State Leaky Bucket Algorithm**

The continuous-state LB is an equivalent version of the GCRA algorithm and can be viewed as a finite capacity bucket whose real-valued content leaks at a continuous rate of 1 unit of content per time unit and whose content is increased by the increment $T$ for each conforming cell. If at a cell arrival the content of the bucket is less than or equal to the limit value $\tau$, then the cell is conforming, otherwise the cell is nonconforming. The bucket size is $(T + \tau)$. The implementation requires a simple real-valued up/down counter to reflect the contents of the token bucket.
5.3.3 Traditional Leaky Bucket Algorithm

This is a more widely used description of the LB with parameters leak rate $r$ and bucket size $M$. The basic idea behind this LB mechanism is that each incoming cell adds a token to the bucket. Tokens are leaking at constant rate $r$. The size of the bucket imposes an upper bound on the burst length and determines the number of cells that can be transmitted back to back, controlling the burst length. Provided that the burst is short, the bucket will not fill and no action will be taken against the cell stream. However, if a long burst of higher-rate cells arrives, the bucket will fill and the UPC function will take actions against cells in that burst. The tolerance allowed for the connection depends on the size of the token buffer ($M$) and the token leak rate ($r$). Conceptually, the tokens can be viewed as arrivals to a finite-capacity, single-server queue with deterministic service time. It is also obvious that the LB enforces the rate $r$ and allows temporary bursts above the rate $r$ depending on the bucket size $M$. The GCRA $(T, \tau)$ corresponds to the LB $(1/T, \tau T + 1)$ and the LB $(r, M)$ corresponds exactly to the GCRA $(1/r, (M-1)/r)$.

5.4 Leaky Bucket Variants

A number of variations of the basic leaky bucket are possible. Instead of discarding/tagging cells when the token bucket is full, arriving cells can be allowed to queue in a data buffer ($B_0$). The input or data buffer smoothes the burst by spacing the cells at the cost of introducing some delay.

The two types of enforcement action that can be taken within the LB scheme (cell discarding or marking), and whether or not a user buffer is used, gives rise to mainly four different versions of the LB. It is worth noting that the maximum burst size (or intrinsic burst size) parameter was not recommended by the ATM Forum until the end of 1994 [ATMF94]. Thus till then it was assumed that the user had to declare only the PCR and mean cell rate.

5.4.1 Unbuffered LB

Fluid flow analysis of the unbuffered LB for packet voice and still picture in [BUTT91] conclude that one LB can not police both peak and mean rate. An explicit formula for the CLR is derived. This was followed by performance comparisons of multiple unbuffered LBs and a single LB to police source parameters concluding that
multiple LBs are more effective in policing the source traffic [LIAO92]. Using
discrete-time analysis [FRAT92] derived equations for the cell-loss and the
approximate reaction time of the LB. The use of a dual LB configuration to police
statistical parameters such as the mean rate by [ORS94] concluded that the required
CLR can be achieved but that the reaction time still is sometimes not sufficient.

5.4.2 Buffered LB

The performance of the buffered LB assuming Poisson arrival (which might not be
the case in ATM networks) was investigated in [SIDI89]. Using discrete-time analysis
he derived equations for the cell loss and the mean waiting time in the user buffer.
The smoothing effect of the user buffer on the packet stream was also investigated by
deriving the interdeparture time (time between successive departures of cells from the
system) equations. This analysis was extended to continuous-time Markov Arrival
Processes (MAP) in [BERG91b].

Analysis, assuming a discrete-time environment and a batch arrival process was done
by [AHMA93]. His work is based on matrix analytical techniques. This work is
extended to a general finite state discrete MAP by [SOHR94]. Then, in order to model
correlated interarrival times the same analysis is carried out for a MMPP as input
traffic [KIM92].

For an on-off source model the work of [ELWA91], derives closed-form expressions
for cell loss using fluid-flow analysis. Later, assuming a truncated hyper-exponential
function (with squared coefficient of variation 50, 100) as traffic source model the
performance of the buffered LB was assessed in [LEME93]. This is repeated for an
IPP as traffic source by [SAKA93], where expressions (using discrete-time analysis)
for cell loss and cell delay are derived. This paper concludes that the LB parameters
(leaky rate) must be set at values in access of the user declared values (for statistical
parameters). The performance of this configuration is superior to the unbuffered LB
since the reaction time is much shorter at the expense of some queuing delay
introduced by the data buffer.
5.4.3 Cell Tagging

Analytical expressions for the statistical gain by using a unbuffered LB with and without cell tagging are derived in [HSIN93]. The paper concludes that tagging gain is observed when sources have very small average bursts and network elements small buffer sizes. However no comparison of analytical and simulation results is provided. In [CHA091a] all four LB schemes are compared, for a discrete-time IPP as traffic source. The paper concludes that the buffered LB using cell tagging performs best in terms of cell loss rate. An architecture (actually three) to implement this mechanism is also proposed. This work is repeated in [CHAN94] for a MMPP as traffic source, again concluding that the buffered LB with cell tagging is most effective. A comparative studies of variants of the buffered LB is done in [WU94]. An adaptive LB is investigated where the LB parameters are changed according to the network traffic load. These are compared to the adaptive LB with cell tagging and the adaptive LB with cell tagging where traffic sources have different priorities. The simulation results show that the adaptive LB with cell tagging and priority mechanism has the best cell loss performance.

5.5 Control of the Peak Rate

During the connection set-up-phase it is necessary to negotiate two parameters as part of the traffic contract for peak rate policing [ITUT96b]. These parameters are the Peak Cell Rate (PCR) \( R_{\text{PCR}} = \frac{1}{T_{\text{PCR}}} \) and the Cell Delay Variation Tolerance (CDVT) \( \tau_{\text{PCR}} \).

The aim of this Section is to evaluate the performance of the LB for different traffic arrival patterns and different CDVT values.

The LB will be used as UPC, equivalent to the GCRA \((T_{\text{PCR}}, \tau_{\text{PCR}})\). The parameters of the LB are the leak rate \( r = \frac{1}{T_{\text{PCR}}} \) and the token pool size \( M = \left( \tau_{\text{PCR}} / T_{\text{PCR}} + 1 \right) \). Thus the GCRA \((T_{\text{PCR}}, \tau_{\text{PCR}})\) corresponds to the LB \( (1/T_{\text{PCR}}, \tau_{\text{PCR}} / T_{\text{PCR}} + 1) \) and the LB \((r,M)\) corresponds exactly to the GCRA \((1/r, (M-1)/r)\). Note that the LB is incremented by 1 for each incoming cell.

If a stream is not conforming to GCRA \((T_{\text{PCR}}, \tau_{\text{PCR}})\) because the actual peak rate \( R = 1/T \) received by the UPC is higher than the contracted peak cell rate \( R_{\text{PCR}} = 1/T_{\text{PCR}} \), cells are discarded because the traffic contract is violated. The CDVT \( \tau_{\text{PCR}} \) is to be interpreted as a
measure of the amount of CDV which is allowed to be introduced by the access network.

The behavior of the LB can be investigated for violation of the negotiated peak rate. Let $Y$ be a factor of increase in the contracted peak rate. Then:

$$\text{Actual Peak Rate } (R) = Y \cdot \text{Contracted Peak Rate } (R_{PCR})$$  \hspace{1cm} (5.1)

In ITU-T Rec.I.356 [ITUT96a] the reference policing ratio ($\gamma_M$) is defined as:

$$\gamma_M = \frac{\text{number of cells exceeding contract entering network}}{\text{number of generated cells}}$$

This reference policing ratio ($\gamma_M$) must be smaller than 1% and is very much dependent on the call holding time.

The Cell Discard Ratio (CDR) is defined as:

$$\text{CDR} = \frac{\text{number of discarded cells}}{\text{number of generated cells}}$$

The "ideal" throughput behaviour is given by:

$$\text{CDR} = 1 - \frac{T_{PCR}}{T} = 1 - \frac{1}{Y} \text{ for } T < T_{PCR} \quad \text{CDR} = 0 \text{ for } T > T_{PCR}$$  \hspace{1cm} (5.2)

![Figure 5-2 Ideal throughput behaviour of a UPC function](image)

Two views of the ideal throughput behaviour of a UPC algorithm are shown in Figure 5-2. The left side is a graphical representation of equation (5.2), while the other Figure shows the cell rate entering the network, as a function of the cell rate received at the UPC input (normalised to the contracted PCR). If the cell rate arriving at the UPC is less than the contracted PCR than the submitted cell rate enters the network.
Otherwise if the submitted cell rate exceeds the contracted PCR, then the contracted PCR is admitted to the network.

The UPC algorithm does not behave exactly as in the ideal case due to the CDV tolerance. The larger the CDVT the more consecutive cells can enter the network at rates higher than the contracted PCR.

The reaction time of the LB to increases in the contracted PCR (or the time for excess traffic to fill the bucket) is:

\[
T_R = \frac{M}{Y_{R_{PCR}} - R_{PCR}} = \frac{\tau R_{PCR}}{(Y - 1)R_{PCR}} = \frac{\tau}{Y - 1}
\]  

(5.3)

The maximum burst size which can enter the network is found by multiplying the reaction time by the increase in the peak rate and adding one for the first cell which will fill the buffer:

\[
MBS = \left[ Y_{R_{PCR}} \frac{\tau R_{PCR}}{Y_{R_{PCR}} - R_{PCR}} \right] + 1
\]

\[
MBS = \left[ \frac{\tau}{T_{PCR} - \frac{1}{Y} T_{PCR}} \right] + 1
\]

or

\[
MBS = \left[ \frac{\tau Y}{T_{PCR} Y - 1} \right] + 1
\]  

(5.4)

The Minimum Silence Period (MSP) for the LB to empty completely in cells is:

\[
MSP = \frac{Y_{R_{PCR}} MBS}{R_{PCR}} - MBS = MBS(Y - 1)
\]  

(5.5)

The Worst Case Traffic (WCT) entering the network is defined as the cell stream which is conforming to the negotiated traffic contract, and which requires the greatest amount of resources.
An on-off type source with bursts of length MBS, generated at higher rates than the PCR and then with silence of length MSP is a typical worst case traffic source for the LB controlling the PCR.

The effects of the increase in the PCR and different values of $\tau_{PCR}$ on the MBS entering the network is illustrated below in Figure 5-3.

![Figure 5-3 MBS entering the Network vs Increase in the contracted PCR](image)

**Figure 5-3 MBS entering the Network vs Increase in the contracted PCR**

As it can be seen larger $\tau_{PCR}$ values lead to longer bursts entering the network at higher than the contracted PCR. If the increase in the PCR is small, the reaction time of the LB is slow and larger bursts can enter the network. Thus in order to prevent large bursts of traffic at higher rates than the contracted PCR, the CDVT has to be chosen very carefully. Resource allocation should be done according to the tolerances allowed at the traffic contract.

A method for dimensioning the CDVT of a LB for peak rate policing based on a remote quantile of the delay distribution of an arbitrary cell has been proposed by [NIES90].

The value of $\tau$ can be found for an expected delay variation (EDV) and connection link rate ($R_c$) as [NIES90]:

$$\tau_{PCR} = 2 + [\text{EDV} \cdot R_c / 384]$$  \hspace{1cm} (5.6)

For a maximum CDV of 3ms and a connection link rate of 128 kbit/s or 2Mbit/s, $\tau_{PCR}$ is found to be 3 or 18 respectively.
The policing and spacing functions can be merged by using a data buffer. The worst case delay for a bucket size of 1 and a data buffer size of \((3-1=2)\) and \((18-1=17)\) would be 5.6ms and 4ms respectively. Thus the cell stream arriving at the UPC can be spaced to conform to the contracted minimum cell interarrival time, at the expense of some buffering delay.

Since bounding the CDV is very important, the work of [NIES90] has been extended to take into account the correlation's between successive cell arrival epochs [GUIL91]. Furthermore analytical, simulation and measurement results of the performance of the LB for CBR streams and various values of \(\tau\) are provided in [HOEK95]. Dimensioning the CDV for more complex networks and multiplexed traffic is described in [ROBE96].

### 5.6 Control of the Mean Rate

The peak cell rate was controlled by setting the token generation rate to the peak rate, resulting in a relatively small buffer size to overcome the CDV problem. However there are several problems to be addressed in policing the Mean Cell Rate (MCR).

The bursty nature of some ATM traffic types makes it very difficult to detect violations effectively. The peak rate of bursty traffic may be many times higher than the average rate. A sudden arrival of a burst of cells at peak rate can easily overflow the LB causing excessive cell losses. This happens even when the traffic source has been adhering to the negotiated mean rate in the long run. This problem can be avoided by choosing a large buffer in order to minimise the probability of dropping/marking cells which conform to the traffic contract. However, the larger the bucket size, the longer it takes to detect a traffic violation.

In other words, for a general stochastic process, one needs measurement over a long time period in order to detect a violation. A long reaction time is therefore needed if we want to accurately control the mean rate. On the other hand, a policing mechanism has to ensure that users with short connection times don't exploit the network. Thus for fast reaction, short estimation intervals are required.

The problem has been tried to solve by using a leaky bucket. Then shaping the traffic before policing it, using a buffered Leaky Bucket was proposed. For sources which
are not time sensitive the total buffer $B = B_0 + M$ can be distributed between the token buffer ($M$) and the smoothing buffer ($B_0$). For time sensitive traffic sources however a smoothing (shaper) buffer alone is not sufficient requiring another solution to overcome the long reaction time.

A proposal by the ATM-Forum [ATMF94] to upper bound the source Mean Cell Rate (MCR) and limit the number of cells in a burst at the peak cell rate was accepted by the ITU-T [ITUT95d]. The upper bound on the source mean cell rate was called Sustainable Cell Rate (SCR) and the maximum number of cells allowed at peak rate was called Maximum Burst Size (MBS).

Currently three parameters have to be negotiated as part of the traffic contract for mean rate policing [ITUT96b]. These parameters are the Sustainable Cell Rate (SCR) $R_{SCR} = 1/T_{SCR}$, the intrinsic burst tolerance $\tau_{inr}$ characterising the Maximum Burst Size (MBS) at the peak cell rate and the CDVT $\tau'_{SCR}$.

The selection of the SCR and MBS is still an open issue since there are infinite number of couples of values to chose from, for a VBR source. The aim of this Section is to optimise the UPC so that the SCR can be tightly policed with a fast reaction time. The LB will be used as GCRA $(T_{SCR}, \tau_{SCR})$. Note that $\tau_{SCR} = \tau_{inr} + \tau'_{SCR}$. If the user has the knowledge of $\tau_{inr}$ rather than the MBS, than the following rule applies [ITUT96b]:

$$\text{MBS} = 1 + \lfloor \tau_{inr}/(T_{SCR} - T_{PCR}) \rfloor \text{ cells} \quad (5.7)$$

Before starting the analysis it is worth noting that policing of the SCR traffic descriptor can be done only for high priority (CLP=0) and depending if tagging is used or not there are three different conformance definitions for policing the SCR. Figure 5-4 shows the possible configurations.

For configuration 1, cells with CLP=0 and CLP=1 have to conform to both the PCR(0+1) and SCR(0+1) conformance tests. For configuration 2, there are two conformance definitions. CLP=0 cells are conforming if they are conforming to both PCR(0+1) and SCR(0) conformance tests. If tagging is used than the CLP=0 has only to conform to the PCR(0+1) test and is tagged instead of discarded if is does not
conform to the SCR(0). For a CLP=1 cell to be conforming it has only to pass the PCR(O+1) test.

![Flowchart showing SCR Policing](image)

**Figure 5-4 SCR Policing (a) Configuration 1 (b) Configuration 2**

### 5.6.1 Performance of the Unbuffered Leaky Bucket

Let us first investigate the performance of the widely used unbuffered leaky bucket. Note that this configuration has no real data buffer ($B_0=0$), introducing no queuing delay to the cells. The analysis is first done for an unbounded burst size, and then for the case where the maximum burst size is specified.

5.6.1.1 The unbuffered LB for sources with unbounded burst size

Calculations and simulation results show that the token generation rate $r$ must be at least slightly higher than the Mean Cell Rate (MCR) in order not to discard too many conforming cells for a reasonable reaction time.

Expressing the token generation rate in symbols yields:

$$\text{Token generation rate } (r) = E\cdot \text{MCR} = R_{scr}$$  \hspace{1cm} (5.8)
where tolerance factor $E > 1$. The question now becomes: which value of $E$ and $M$ (bucket size) should be used in order for a well behaved source to experience a cell discard ratio of the same order as its QoS? Due to the oversize factor $E$, it is expected that some sources may exceed the mean rate. However as we are policing the SCR which is an upper bound on the MCR, it is possible to allocate resources according to the oversize factor.

The fluid flow analysis by [ANIC82], [BUTT91] derives closed form expressions to determine the cell discard ratio of the LB $(r,M)$ for a on-off source with exponentially distributed burst and silence periods. The bit flow is considered as a continuous variable. $Y$ and $X$ are the continuous buffer states in the instants where two consecutive bursts arrive (Figure 5-5).

![Fluid Flow Approximation](image)

**Figure 5-5 Fluid Flow Approximation**

During the burst, whose duration is $Z$, the buffer state grows at rate:

$$b = p - r \text{ bits/s}$$

After the burst, during the inactivity period of duration $L$, the buffer state decreases at the rate of $r$ bits/s.

At the instant a burst begins, the buffer state is a Markov chain, described by the equation:

$$X = Y + b \cdot Z - r \cdot L$$

We assume that $r < p$, otherwise the buffer is always empty.
The calculations to obtain the formula for the cell discard ratio (5.9) can be found in [BUTT91]. A program written in BASIC to do the calculations is given in Appendix D.

\[
CDR = \frac{p \cdot r}{p \lambda_1 - \lambda_2} \frac{\lambda_1 - \lambda_2}{\lambda_1 e^{\lambda_2 M} - \lambda_2}
\]

where

\[
\lambda_1 = \frac{1}{t_{on} b}, \quad \lambda_2 = \frac{1}{t_{off} a}
\]

p: peak arrival rate of the ATM cells in bits/s when source is at the on-state
r: leak rate in bits/sec
M: token bucket size in bits
t_{on}: the average duration of the on-state (peak duration)
t_{off}: the average duration of the off-state (silence duration)

Simulations have been used to prove the accuracy of the analytical results. The simulation model created using BONeS Designer is shown in Appendix B. The reason for choosing a high CDR was to make comparison of analytical with simulation results possible. Having observed that the analytical results are accurate and that a large token buffer M is necessary for r=MCR, the token generation rate was increased.

Figure 5-6 (a) and (b) show the CDR of the LB for broadband services and packetised voice respectively. The figures show the CDR as a function of the increase in the MCR for various leak rates. In symbols:

\[
\text{Actual cell rate (R)} = Y \cdot \text{MCR}
\]

where Y is the factor of increase in the MCR.

![Figure 5-6 CDR vs Y for (a) Broadband Source (b) Packetised Voice](image-url)
It can be seen that the larger the LB size $M$ the better the control. However, this comes at a cost, the reaction time also increases. In other words, the lag time between the jump in average bit rate and its detection becomes large. Note that $M$ decreases as the token generation rate $r$ increases for the same CDR. This provides a fast reaction to big jumps in the mean rate. However, because of the increase in the token generation rate small increases in the average go undetected. These considerations show that there is a trade-off between fast reaction and tightness of control. The LB approaches the ideal behaviour as $M$ increases and $r$ decreases.

![Image](figure5.png)  

**Figure 5-7** LB Parameters of (a) Broadband Source (b) Packetised Voice

In order to be able to chose the necessary LB parameters according to the requirements of the service in terms of Cell Loss Rate (CLR), the CDR has to be dimensioned lower than the CLR. The design CDR indicates the ratio of cells discarded although they conform to the traffic contract over a long observation time. The analytical model was run for various design CDR’s to determine the required token buffer size as a function of the leak rate as shown in Figure 5-7 (a) and (b) for broadband sources and packet voice respectively.

An extreme example to demonstrate the large token buffer size needed to control the mean cell rate is the still picture source which has a burstiness of 23. Figure 5-8 shows the required token buffer size as a function of $E$, for various design CDR’s.
In order to calculate Worst Case Traffic (WCT) entering the network, expressions for the reaction time ($T_R$) and maximum burst size (MBS) are derived below:

\[ T_R = \frac{M}{Y R_{SCR} - E R_{SCR}} = \frac{M}{(Y - E) R_{SCR}} \]  \hspace{1cm} (5.11)

\[ T_R = \frac{d R_{SCR}}{(Y - E) R_{SCR}} \]  \hspace{1cm} (5.12)

Figure 5-8 LB Parameters of Still Picture Source

Figure 5-9 Reaction Time vs Y for (a) Broadband Source (b) Still Picture

Figure 5-9 shows the worst case reaction time of the LB as a function of increase in the mean ($Y$ is the increase factor) for a Broadband source and Still Picture source (for a CDR of $10^{-7}$). Here the arrival rate of the cells is assumed to be $Y$ times the SCR. The equations derived by the ITU-T [ITUT96b] assume the cell arrival rate to be equal to the PCR. In this case $Y$ is equal to the burstiness of the broadband and still picture source (5 and 23 respectively). The reaction times as a function of $E$ (oversize factor) are shown in Figure 5-10.
The Maximum Burst Size (MBS) entering the network can be found by multiplying the reaction time by the increase in the mean rate and adding one for the first cell which fill the buffer:

\[
MBS = \left[ \frac{M}{(Y - E)R_{SCR}} \right] YR_{SCR} + 1 = \left[ \frac{MY}{Y - E} \right] + 1
\]  
(5.13)

or for E=1 (no tolerance if the IBT is specified)

\[
MBS = \left[ \frac{YR_{SCR} \tau R_{SCR}}{YR_{SCR} - R_{SCR}} \right] + 1 = \left[ \frac{\tau}{T_{SCR} - \frac{1}{Y} T_{SCR}} \right] + 1
\]  
(5.14)

Note that this equation is different from the ITU-T equation [ITUT96b] which assumes that bursts are generated at the peak cell rate. For bursts at peak cell rate, substituting \( T_{SCR} \) instead of \( (1/Y)T_{SCR} \) will give the same equation. We are investigating the reaction time of the LB for the case when the source generates traffic at rates lower than the peak rate but higher than the contracted SCR. Equation (5.13) is used for the MBS calculations as the oversize factor \( E \) has always to be larger than 1, for the unbounded burst size analysis.

Figure 5-11 (a) shows the maximum reaction time of the LB for a Packet voice source and Figure 5-11 (b) shows the maximum burst size which can enter the network for various leak rates and a CDR of \( 10^{-7} \).
Figure 5-11 (a) Reaction Time (b) MBS which can enter the network vs Y for packetised voice source.

In the case that the burst arrives at the peak rate, Y will be equal to the burstiness of the packet voice source, which is 2.85. The reaction time and MBS entering the network as a function of E are shown in Figure 5-12 (a) and (b).

Figure 5-12 (a) Reaction Time (b) MBS which can enter the network vs E for packetised voice source.

The Minimum Silence Period for the LB to empty completely is:

\[ MSP = Y \left( \frac{Y}{E} - 1 \right) MBS \]  

(5.15)

Thus for a connection of short duration equal to the reaction time, no action can be taken against excess traffic because of the allowed tolerance. The worst case traffic which can pass a dual LB configuration policing PCR and SCR is a on-off source which emits bursts of size MBS and then waits for the MSP before sending a burst again.
The choice of the best compromise between tight control and prompt reaction has to be determined by the network manager because a slack in one of them might deteriorate the QoS requirement of other users. Therefore both properties are generally required. In this case two LB’s in parallel can be used with different token buffer (M) values. A cell is discarded whenever one of the buckets overflows. This results in a triple LB (shown in Figure 5-13) to police the peak and the mean cell rate. The values of M for the dual LB configuration can be found for the required reaction time and tightness of control from equation (5.11), which can be used for any service class. The main drawback of the dual LB is that it provides only fast reaction to large and sudden jumps in the mean. If for example we chose E=1.1 and E=1.6 then the reaction time for small increases in the mean (10%-60%) will be long. The same problem applies to multiple buckets in parallel. To overcome the problem of large token buffers and thus slow reaction either the buffered leaky bucket can be used or the burst size can be bounded.

![Figure 5-13 Triple LB to police both peak and mean cell rate](image)

5.6.1.2 The unbuffered LB for sources with bounded burst size

In the previous section we have shown that both tight policing of the SCR and fast reaction time is difficult to achieve for bursty sources with unbounded burst size. If however the intrinsic burst tolerance parameter is declared by the user than the MBS entering the network can easily calculated from equation (5.14). Hence if the tolerance to be allowed (the MBS) for policing the upper bound on the mean (the SCR) is known, than policing the SCR becomes as straightforward as policing the PCR.
5.6.2 The Performance of the Buffered Leaky Bucket

Source traffic which is not very time sensitive may be delayed by introducing a smoothing buffer which spaces the cell. In this case the maximum allowed delay to be introduced by the access control has to be known.

5.6.2.1 The buffered LB for sources with unbounded burst size

The fluid flow approach is used to analyse the LB mechanism. The mechanism is based on each cell to get one token from the token buffer ($M$) before entering the network. If no token is left the cell queues in the data buffer ($B_0$). The analysis also applies to the unbuffered LB by taking $B_0=0$. The source model differs from the on-off model only in the assumption of uniform generation and transmission of information. Here information is considered to have a continuous nature, as if it were a fluid. Thus the burst scale is considered neglecting the effects of the cell scale to simplify computation.

![Fluid-flow model of the LB mechanism](image)

**Figure 5-14 Fluid-flow model of the LB mechanism**

A detailed derivation of the formulas using the model in Figure 5-14, is given in Appendix D. In this section we will summarise the closed-form expressions to calculate the Cell Discard Ratio (CDR), mean queuing delay (if a smoothing buffer is used) and the mean reaction time.

The Cell Discard Ratio (CDR) obtained, using the approach of [ELWA91a] is:

$$CDR = (\frac{1}{\rho} - 1) \frac{a}{b} (\frac{p - 1}{r}) e^{(zr)}$$

(5.16)

where $z = \frac{a + b}{p - r} (1 - \rho)$ and $p = m/r$. 
It can be seen that the CDR decreases exponentially with $B=M+B_0$. Thus the cell loss performance of the leaky bucket algorithm depends only on the sum of the buffer sizes (Note, that the token buffer is only a model representation. The data buffer, on the other hand, is a real buffer). For the unbuffered leaky bucket, the cell loss probability can be obtained by setting the data buffer $B_0=0$.

The CDR formula (5.9), also derived using fluid analysis, gives the same results for dimensioning the LB. Another important performance parameter of the LB is the reaction time required to detect non-conforming cells. If the token buffer is dimensioned according to the burst tolerance the required CDR can in most cases not be achieved. Therefore some part of the total buffer $B$ has to be distributed to the data buffer. This results in faster reaction but introduces some delay. Thus in order to calculate the reaction time and queuing delay of the LB, the expressions derived in Appendix D are given below.

The mean reaction time of the leaky bucket is given by:

$$T_r = \frac{Q_M n_c}{(Y - E)m} \quad (5.17)$$

where $Y$ is the increase factor in the mean rate of the source, $n_c=424$ the number of bits per cell and $E\cdot m$ is the token generation rate.

The mean delay which is introduced when using a data buffer is given by:

$$\overline{D_{bn}} = \frac{Q_{bn} n_c}{(1-P_L)r} \quad (5.18)$$

$$\overline{Q_M} = \frac{\rho (1-e^{\Delta M})}{z \Delta (B)} + \frac{M}{\Delta (B)}$$

$$\overline{Q_{bn}} = \frac{\rho (e^{\Delta B} - e^{\Delta M})}{z \Delta (B)} + B_0 (1 - \frac{1}{\Delta (B)})$$

$\overline{Q_M}$ is the mean token buffer contents and $\overline{Q_{bn}}$ is the mean data buffer contents where

$$\Delta(x) = 1 - \frac{a (p-r)}{b} e^{(px)}$$

The accuracy of the fluid-flow model has already been investigated for some sources in [BUTT 91]. In order to verify the accuracy of this model for various other ATM sources, simulations were done using BONEs Designer
According to the required CDR and tightness of control, $B$ is determined from equation (5.16) which is derived using the fluid-flow approach. $B$ is the total buffer which can be distributed between the data buffer ($B_d$) and the token buffer ($M$). The mean queuing delay for a certain data buffer size can be determined from equation (5.18). Comparing numerical with simulation results it was observed that expression (5.18) is very accurate for conforming traffic. Figure 5-15 shows the mean delay introduced by the data buffer for different leak rates and a CDR of $10^{-7}$ for a broadband source.

![Figure 5-15 Mean Delay of the buffered LB for a Broadband source]

The above figure which shows the mean delay introduced by the data buffer has to be used with care, since this is the delay for conforming traffic. However when the source increases its activity above the agreed mean rate than the buffering delay experienced by all cells will increase.

For example if $E=1.6$ than the total buffer $B$ to achieve a CDR of $10^{-7}$ is 1713. If $B_d$ is chosen to be 1000 cells than the mean delay for conforming traffic is 0.1 ms and the maximum delay for nonconforming traffic is 62.3 ms.

The maximum queuing delay which can be introduced by using a data buffer is:

$$D_{B_d} = \frac{B_d \cdot 424}{(1 - CDR)^r}$$

(5.19)

Using this equation the maximum data buffer size which can be used for the worst case traffic can be determined according to the maximum delay to be introduced by
the UPC. For a leak rate of 3 Mbit/s a CDR of $10^7$ and maximum delay of 5 ms to be introduced by the UPC the maximum data buffer size can be found to be 82 cells.

Note that for sources which are not time sensitive and where longer delays can be introduced one LB with the token generation rate near the mean bit rate is sufficient to control the SCR. In this case the total buffer $B$ is distributed in such a way that the token buffer is small (in the range of the number of cells emitted during one burst) for fast reaction and the data buffer is large introducing a high delay. Thus a dual LB mechanism would be sufficient in this case to police both PCR and SCR. Alternatively a dual LB with buffer can be used to police the mean rate. This idea was also analysed in [GRIF96].

5.6.2.2 The buffered LB for source with bounded burst size

As for the unbuffered LB with bounded burst size the MBS tolerance which can enter the network can be calculated from equation (5.13) if not stated explicitly by the user. If the maximum delay to be introduced by the UPC is known than the size of the data buffer can be found from equation (5.19).

5.7 The Adaptive Leaky Bucket

This Section proposes the use of an adaptive LB to improve the performance of a preventive control scheme by using a feedback control loop.

Although preventive control tries to prevent congestion before it actually occurs there may be various reasons for the network to experience congestion (such as switch buffer overflow). In this case, where the network relies only on the UPC and no feedback information is exchanged between the network and the source, no action can be taken once congestion has occurred. Therefore, some form of feedback control must be incorporated in the congestion control mechanism.

It has to pointed out that the ABR service class was recommended in 1995 [ATMF95a, ITUT95d] and that at the time of this research the dynamic LB [ITUT96b] was not defined. When this research was carried out, it was assumed that it will be possible in ATM networks to throttle the activity of the sources. Hence this work is also applicable to the ABR service class.
After the definition of the ABR traffic class which incorporates a feedback control loop an adaptive LB was needed to control the change in source parameters. This LB was called dynamic LB and is recommended by the ITU-T and ATM-Forum. Again it has to be noted that our innovative work predates the recommendation.

Figure 5-16 Congestion Control Scheme using feedback to change the parameters of the LB

The congestion control scheme using the adaptive LB is shown in Figure 5-16. Sources are statistically multiplexed and if the multiplexer detects the onset of congestion, information is send to the sources, instructing them to limit their activity. Upon receipt of the congestion signal the source limits its mean bit rate by a prespecified rate called the 'throttling rate'. The same congestion information is also sent to the UPC but delayed, by two times the transmission delay, so that the source can change its parameters. The transmission delay is equal to the sum of propagation delay and switching delay from the point of congestion to the source. Thus the LB is dimensioned dynamically according to the state of congestion within the network resulting in an adaptive LB. After the source receives information that congestion is over it sends cells at a rate calculated according to prespecified increase mechanism. The leak rate and bucket size can be calculated from equation (5.9) and (5.11) respectively. Chapter 7 will evaluate the performance of a combined preventive/reactive control scheme using the adaptive LB.
5.8 Summary

This Chapter has discussed the parameters for assessing the performance of a UPC/NPC mechanism, namely the response time and accuracy of control. The reasons for CDV like clumping and dispersion were explained. Then the UPC algorithm recommended by the ITU-T and ATM-Forum was described. Variants of the GCRA (or equivalent LB) which use buffering, priorities and tagging were discussed. The rest of the Chapter describes the dimensioning of the GCRA (or equivalent LB) for policing the peak and mean rate for bounded and unbounded burst size. The aim is to police the declared parameters accurately and with a short reaction time to violations in order to minimise non-conforming traffic entering the satellite network.

The peak bit rate can be controlled relatively easy, setting the leak rate to the source peak rate with a small token buffer to account for the CDV. Analytical expressions for the reaction time, and the MBS entering the network for a certain CDVT are provided.

The mean bit rate, and thus the burstiness, can only be controlled by at least one more LB whose parameter setting strongly depends on the source characteristics to be controlled.

The performance the LB without smoothing buffer is carried out for unbounded and bounded burst size. The numerical results for the unbuffered LB for unbounded burst size, presented as figures in this Chapter, show that a very high LB size must be used in order to sensitively detect violations of the mean rate (by setting the token generation rate near the mean rate). This implies that a very long time is necessary to detect a violation of the mean bit rate. On the other hand, if a small LB size is chosen, for fast reaction, the token generation rate must be much higher than the mean bit rate, so that the source can increase its mean rate up to the token generation rate, without any action being taken.

Considering this trade-off, and the fact that both sensitivity to detect violations of the mean bit rate and fast reaction is necessary in most cases, a policing mechanism called the triple leaky bucket mechanism has been proposed. This mechanism consists of one LB policing the peak rate and two LBs policing the mean rate and thus the burstiness. The drawback of the triple LB configuration is that tight control and fast reaction are
provided, but not both together. One LB provides fast reaction to larger increases in
the mean whereas the other has slower reaction providing tight control.

The control of the SCR (upper bound on the MCR) when the burst size is bounded by
the MBS parameter (or IBT) is straightforward as policing the PCR. The problem is
whether the user will be able to declare these values.

In order to find the optimum solution for the trade-off between fast reaction and
queuing delay, for the unbounded burst size, we continued the fluid-flow analysis of
the LB with smoothing buffer for unbounded burst size. Closed-form expressions
were derived for the mean reaction time (5.17) and the mean queuing delay (5.18) of
the buffered LB. The accuracy of both expressions has been verified using
simulations. However, it has been found that the values calculated using these
expressions have to be used with care, because the required QoS parameters (such as
end-to-en delay and CDVT) are strict upper bounds.

When considering the maximum delay introduced by the data buffer, it was realised
that only small data buffers can be used for rt-services. However for nrt-services the
use of a smoothing buffer seems to be very attractive in reducing the reaction time.
The same conclusions for the buffered LB are also applicable to sources with bounded
burst size.

Finally we proposed to improve the performance of a preventive control scheme using
feedback control. Since preventive control tries to prevent congestion before it
actually occurs the network relies only on the UPC and no action can be taken once
congestion has occurred within the network. For this reason we proposed to
incorporate a congestion control technique called Backwards Explicit Congestion
Notification (BECN) to the control scheme. The parameters of the LB are changed
according to the state of congestion within the network resulting in an adaptive LB.
The comparison of traffic control schemes with and without feedback is presented in
Chapter 7.
6. A MAC SCHEME FOR ATM OVER SATELLITE TO PROVIDE QoS

Providing and guaranteeing Quality of Service (QoS) requested by a user is an important task and within ATM networks the negotiated service requirements have to be fulfilled using several methods of traffic management. Although the functionality and the QoS requests by users in a wireless environment is similar to the one needed in the wired ATM network environment, the more difficult and restrictive conditions of the satellite air interface require different methods in order to satisfy the user needs while achieving high utilisation of the satellite link bandwidth.

In order to achieve maximum utilisation of the available satellite bandwidth for bursty sources the protocol stack at the ATM satellite air interface has to behave like a usual ATM multiplexer. The multiplexing around the air interface has to be coordinated for access to the shared satellite resources in such a way that the QoS of all ATM services can be guaranteed for each established connection.

In this Chapter first an overview of various proposed MAC schemes for ATM over Satellite is given. Then the ATM service classes are mapped onto MAC service classes to simplify the design of an optimum MAC protocol. Finally the performance of an Adaptive Random-Reservation Medium Access Control (MAC) protocol which can support all ATM service classes while providing the required Quality of Service (QoS) is analysed.

Our study focuses on parameter optimisation of the multiple access schemes for ATM over a GEO satellite with on-board processing capabilities, considering various traffic mixes of CBR, rt-VBR, nrt-VBR and UBR.

The adaptive MAC protocol was designed to allow statistical multiplexing of ATM traffic over the satellite air interface, especially for the independent and spatially distributed terminals. It is shown that the potential user population which can be served is considerably increased by statistically multiplexing bursty traffic over the air interface.
The use of standard ATM protocols to support seamless wired and wireless networking is possible by incorporating a new radio specific protocol sublayer into the ATM protocol model [ATMF96b] as shown in Figure 6-1.

Considering that satellite communications uses multiple access on a shared medium, a MAC layer, which is not present in traditional ATM networks is needed. The MAC protocol plays a central role as means of accessing the RPL from the ATM layer. The access scheme refers to the physical layer multiplexing technique to a share a common channel among multiple users of possibly multi-services. The problem of statistical multiplexing at the satellite air interface is slightly different to that in the fixed network as illustrated in Figure 6-2. In the fixed network the problem is associated with control of bandwidth on an outgoing link from some multiplexing point after buffering has occurred. It is implicitly assumed that the access links from the source are dimensioned in such a way that they do not impose any constraints on the traffic (e.g. sources can transmit at their peak bit rate). In the air interface the constraint is on the bandwidth available in total to all sources before the buffering/multiplexing point.

The Satellite-UNI (S-UNI) interface has to contain support for mapping of user terminal ATM connections to the shared satellite access link. A key issue is the mapping of ATM service classes such as CBR, VBR, ABR, GFR and UBR to the satellite channel, so as to maintain the required QoS for each VC.

In this Chapter an efficient MAC protocol for the satellite environment is proposed which maps all ATM service classes to MAC classes. The proposed protocol is evaluated for a mix of different service classes.
6.1 Framework for Proposed MAC protocol

6.1.1 Design Objectives

The MAC protocol has to be designed to allow statistical multiplexing of ATM traffic over the satellite air interface, especially in the uplink for the independent and spatially distributed terminals. The following design objectives are taken into consideration:

- Maximise the slot utilisation, especially for bursty traffic.
- Guarantee the QoS requirements for all service classes.
- Maximise frame efficiency by minimising overheads.

The minimisation of overheads is not an easy task, especially for ATM which was designed for channels with very good error characteristics (BER around $10^{-10}$). To minimise cell loss over the satellite link, channel coding has to be used to make the
transmission more robust. A Logical Link Control (LLC) header to facilitate error recovery mechanisms is optional and not in scope of this study. Finally a satellite specific header with satellite routing and Radio Resource Management (RRM) fields is added to form a MAC packet as shown in Figure 6-3.

Figure 6-3 Encapsulation of ATM Cells to MAC Packets and mapping to TDMA Frame.

6.2 Access Schemes

MAC layer access schemes can be typically categorised into four classes: Fixed Access, Random Access, Demand Assignment Multiple Access (DAMA) and Adaptive Access. The first three techniques have evolved to meet the needs of constant high traffic with long duration's, sporadic traffic with short to medium duration's, and sporadic traffic with long duration's, respectively [BOHM93]. Finally adaptive access is used to meet the needs of multiple media which consists of traffic with all of the above characteristics.

With Fixed Assignment schemes a proportion of the total available bandwidth is exclusively allocated to each terminal. Although these provide the best QoS performance, they do not offer statistical multiplexing at the MAC layer (only at the terminal / gateway assuming it is operating as a concentrator). This basic scheme is the easiest to set up and control. Each station has a share of the total bandwidth independently of the other stations, even if it not transmitting information. Delay and loss requirements can be guaranteed but the satellite air interface bandwidth is poorly exploited when a station is not transmitting, since unused bandwidth cannot be assigned to another station.
Fixed assignment schemes have traditionally been used for satellite trunk systems and include FDMA and TDMA. Both suffer from low utilisation of the capacity when the traffic becomes bursty. In addition, they are not versatile in that the frame or the bandwidth has to be reconfigured each time users are added.

**Random Assignment** schemes (such as ALOHA) simply attempt to transmit data as required and typically use a stochastic back-off mechanism to resolve conflicts. Due to the continued possibility of conflicts they are unsuitable for stream-type information flows such as voice. This type of access is well suited to networks containing a large number of stations where each station transmits short bursts with low average traffic rates. The possibility of collision is always present and each retransmission represents an extra delay. This access scheme is ideal for non real-time connectionless interactive services. No guarantees on delay and cell loss can be provided using this access scheme.

**Demand Assignment** approaches use a reservation protocol to limit the possibility of collisions. Explicit reservation schemes introduce greater latency as they take longer to reserve the required bandwidth. With implicit schemes a successful initial transmission within a slot automatically reserves the same slot in succeeding frames. This access scheme is more flexible than the fixed access and also provides the required QoS. The utilisation of the satellite resources is increased by making use of a reservation procedure. DAMA also has the potential to serve an almost infinite population like random access.

There are two variants of DAMA namely, fixed-rate DAMA and variable-rate DAMA. Fixed-rate DAMA is ideal for connections where the amount of bandwidth allocated for that connection will not be changed until the connection is terminated. On the other hand using variable-rate demand assignment the bandwidth of a connection can be adjusted according to the change of the data transfer rate.

**Adaptive Assignment** mechanisms utilise various MAC protocol variants depending upon the instantaneous traffic conditions. All protocols have their strength and

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3 The Combined Free - DAMA protocol is a hybrid scheme[NGOC96] where slots are first allocated on a request basis as for "standard" DAMA approaches, and any remaining slots are then assigned to terminals according to some distribution strategy. At each terminal these "free" slots can be used for transmitting information without having to first request bandwidth thus reducing the overall latency.
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weaknesses; no access scheme outperforms all others under every condition. To accommodate a combination of traffic types, channels can be partitioned into several sections, each operating under its own protocol.

Adaptive protocols attempt to provide good performance over a large range of conditions; the access scheme itself changes, adapting to network traffic load fluctuations, yielding an access procedure appropriate for the actual traffic type mixture. Thus to meet the design objectives in a multi-service environment an Adaptive Access mechanism seems to be the best choice. In the following sections we will consider various proposals.

Most proposals for broadband ATM over satellite consider Time Division Multiple Access (TDMA) techniques. The background and motivation for using TDMA in this research were given in Chapter 2. Within the TDMA proposals a further distinction is made according to the number of carrier frequencies used on the uplink and downlink: Frequency-Division Duplex (FDD), which uses two frequencies, and Time-Division Duplex (TDD), which uses only one frequency carrier. A random access technique called slotted ALOHA (or a variant of this protocol) is used by most TDMA protocols for the initial reservation of slots from the terminal to the control station. This transmission is done on the uplink (UL) channel in contention among the terminals. Consequently, most proposals do not analyse the communication from the control station (satellite) to the terminals, called downlink (DL), which is realised by a TDM technique, since the satellite has total control of the channel resources through a scheduler (which was analysed in Chapter 4).

6.2.1 Packet Reservation Multiple Access (PRMA) and Variants

PRMA was proposed by Rutgers WINLAB [GOOD89] for packet-oriented voice transmission over a wireless channel. It cannot be used for wireless ATM networks because PRMA suffers from a serious problem of variable channel access delay which does not suit the stringent requirements of some of the ATM services.

Several other protocols have been proposed for enhancing the performance of PRMA (DTDMA/PR [RAYC94], DSA++ [PATR96], PRMA/DA [KIM96]). All these enhancement PRMA protocols dealt with improving channel efficiency more and
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providing some kind of fairness for data applications. However, the variable channel access delay problem has not been resolved.

To summarise MAC protocols for wireless ATM suffer from the problem of cell (or burst) access delay. To solve this problem, the defined MAC protocol must possess some sort of mechanisms to deal with admitted idle VBR, ABR and UBR users separately from newly requesting access users. Therefore, control and management mechanisms such as priority and polling need to be considered [KUBB97]. The following proposals are MAC schemes for satellites and consider polling and/or priorities.

6.2.2 FIFO Ordered Demand Assignment-TDMA (FODA-TDMA)

In FIFO Ordered Demand Assignment / Information Bit Energy Adapter (FODA/IBEA) the TDMA frame is divided into 3 subframes, which are the control subframe, the stream subframe and the datagram subframe (for bulk and interactive data) [CELA91], as shown in Figure 6-4.

The control subframe contains four small slots assigned cyclically to all the active stations. They are used by the stations to send the requests for stream and datagram capacity as well as to send control information. The boundary between the stream and datagram subframe is movable.

Fixed-rate Demand Assignment Multiple Access (DAMA) is used for stream traffic and variable-rate DAMA is used for datagram traffic. The datagram allocations generally change in each frame since each station requests slots according to the status of its queue. This system uses three queues with different priorities at each ground station. The stream traffic FIFO queue has highest priority, followed by interactive data traffic FIFO queue. Bulk data traffic has the lowest priority since it is not delay sensitive.

![FODA-TDMA Frame Structure](image)

Figure 6-4 FODA-TDMA Frame Structure
A new station can enter the network by using an access slot which has a fixed position at the end of the frame and its frequency is every 32 frames.

FODA/IBEA which has centralised control was developed in the framework of the Olympus project and its validity has been confirmed by simulation and experimental results [CELA91]. FODA/IBEA does not support VBR traffic, however FODA/IBEA Derived also supports VBR video traffic (FID/VBR) [CELA95].

6.2.3 Anticipated Reservation Protocol
The frame structure of the anticipated reservation protocol [ZEIN92] is similar to the FODA scheme. It consists of three parts: reservation subframe, bursty subframe and stream subframe with movable boundary between the three parts. Delay sensitive traffic is conveyed using fixed-rate DAMA and delay unsensitive bursty traffic uses an anticipated reservation protocol. Anticipated reservation means that the request for bursty traffic is sent when the terminal receives the first cell instead of the common store and forward approach. Another signalling packet has to be sent to the satellite when the complete burst has been received to reserve the necessary bandwidth.

The main drawback of this scheme is that if the burst duration is smaller than the reservation cycle, some capacity is lost.

6.2.4 Combined/Fixed Reservation Assignment (CFRA)
This access scheme is a combination of the anticipated reservation protocol [ZEIN92] and the buffer threshold method and has been called CFRA [ZEIN95]. It tries to improve the performance of the anticipated reservation protocol by distinguishing between short and long bursts. A fixed rate-DAMA assignment of $R_{min}$ is made at the beginning of a burst. If a burst is longer than a certain number of cells, extra capacity is requested. After a time-out interval where no cells are received by the station a capacity deallocation is requested. Fixed-rate DAMA is used for stream traffic as in previous schemes.

In a recent study [CELA97a, CELA97b] the performance of FODA/IBEA and CFRA is compared, concluding that CFRA is more suitable for interconnecting clusters of only a few stations where the traffic load is light. On the other hand FODA/IBEA is better suited to interconnect networks, with heavy traffic and many hosts.
6.2.5 Movable Boundary Random/DAMA Access

A scheme which can support video, voice and file transfer in a connection-oriented mode as well as connectionless interactive data is proposed in [NGUY90]. The frame is divided in three, as shown in Figure 6-5, by movable boundaries: A reservation subframe operated in random access mode, a Slotted Aloha channel for bursty data traffic, and a DAMA channel for all other traffic. DAMA channels have priority over random bursty traffic. This means that the DAMA allocation is done first and the remaining bandwidth is available for slotted Aloha access. UBR traffic could be send using random access since no QoS guarantees are provided for this service class. In [BOHM93] a performance evaluation of this proposal for voice, video and file transfer is carried out, concluding that this scheme substantially improves the performance of the interactive data users.

![Figure 6-5 Movable Boundary Random/DAMA Frame Format](image)

6.2.6 Combined Free/Demand Assignment Multiple Access (CFDMA)

The CFDAMA scheme [NGOC96] first allocates bandwidth to pending requests (fixed-rate and variable-rate DAMA). Reservation can be made using pre-assigned or random-access request slots, or piggy-backing on a data packet. Then it assigns the remaining capacity to terminals on a round-robin fashion. In this way it saves on reservation time for the bursty traffic.

The delay throughput performance is quite sensitive to population size. For low to medium sizes of terminal population pre-assigned reservation has better performance. For large populations and low throughput, random access reservation is more efficient because it reduces the long waiting time for a request slot.

6.3 Mapping of ATM Service Classes onto MAC Service Classes

To simplify the conceptual design of the MAC protocol, ATM service classes can be mapped onto MAC service classes.
6.3.1 Mapping of Constant Bit Rate (CBR) Service Category

Fixed-Rate DAMA is ideal for connections with a constant bit rate such as the CBR service class in ATM networks. Before a connection is set-up, the terminal and satellite negotiate the Quality of Service (QoS) parameters. These QoS parameters determine the characteristics of the connection. Since the parameters will not be modified during the connection, the amount of bandwidth allocated for that connection will not be changed until the connection is terminated. For ATM CBR connections the Peak Cell Rate is allocated to the terminal.

6.3.2 Mapping of real-time Variable Bit Rate (rt-VBR) Service Category

Real-time Variable Bit Rate (rt-VBR) services can also be supported with fixed-rate DAMA. For real-time services, the amount of bandwidth assigned to the connection should be close or equal to the Peak Cell Rate (PCR) to avoid cell delay. The major drawback of this scheme is that a major portion of the bandwidth is wasted when the cell transfer rate is lower than the assigned bandwidth. The major difficulty to employ variable-rate DAMA in an ATM over GEO satellite systems is the effect of the large propagation delay. The computing and negotiation process between the satellite and the terminal may be too long for real-time VBR services and result in unacceptable QoS. The use of variable-rate DAMA for rt-VBR is only possible if the arriving traffic can be predicted one hop delay in advance. Since this is not possible except in some special cases fixed-rate DAMA will be used for rt-VBR.

A scenario where fixed-rate DAMA is efficient for rt-VBR services is when the terminal can multiplex traffic from multiple services. In this case the aggregate traffic can be approximated as a constant cell flow by using a small amount of shaping.

6.3.3 Mapping of non-real-time Variable Bit Rate (nrt-VBR) Service Category

VBR services which are not time sensitive can be assigned an effective bandwidth which is between the mean cell rate and PCR. Since the required bandwidth of VBR sources changes with time, there may be instants when the cell transfer rate is higher than the amount of bandwidth (effective bandwidth) assigned to that connection. In this case cells can be buffered in the terminal and in case the queue exceeds a certain threshold more bandwidth can be requested. Thus using variable-rate DAMA the
bandwidth of a connection can be adjusted according to the change of the data transfer rate.

Alternatively burst reservation can be used for nrt-VBR services. The terminal requests bandwidth equal to the PCR when a burst arrives. A common example of variable-rate demand assignment with central control is polling: each user terminal is addressed sequentially by a central station, for transmission requests. This way contention is avoided after the initial call set-up. It is important that the terminal can buffer the arriving burst till the reservation has been made. The proper operation of a centrally controlled system, however, depends on the reliability of the controller.

6.3.4 Mapping of the Available Bit Rate (ABR) Service

ABR is an ATM-layer service where the network bandwidth offered to an end-user application may alter dynamically during the lifetime of the connection. The user can specify a Minimum Cell Rate (MCR) and PCR. The MCR is the minimum bandwidth guaranteed by the network and using Resource Management (RM) cells the network informs the source about the allowed cell sending rate which is a value between MCR and PCR. The MCR can be zero and in this case the ABR service class is a best effort service class. Except for the case when the MCR is zero fixed rate DAMA can be used to guarantee the MCR and additional bandwidth can be requested using variable-rate DAMA. However if the MCR is zero and the bursts are very small then Random Access becomes an attractive alternative.

6.3.5 Mapping of Unspecified Bit Rate (UBR) Service Category

No numerical commitments are made for the UBR service class and this service category is intended for non-real time applications. UBR services could be supported by variable-rate DAMA. However the fact that this service class has the lowest priority (because no commitments to CLR are made) has to be considered. We propose that UBR could transmit data directly to the unoccupied data slots without reservation. The unreserved slots are broadcasted on the downlink to be accessed by random access. This is particularly appealing for bursty interactive services with short duration, for which the long slot reservation delay is unacceptable.
6.3.6 Mapping of the Guaranteed Frame Rate (GFR) Service Category

The GFR service class [ATMF98] is similar to ABR in that it provides minimum service guarantees, but without the requirement of feedback control. Accordingly traffic not exceeding a maximum frame size will obtain low cell loss up to the MCR. Any traffic above this rate will be treated as best effort. This service category is particularly interesting for Internet users, who want to have some minimum throughput (as opposed to UBR which was purely best-effort). Fixed rate DAMA can be used to guarantee the MCR and additional bandwidth can be requested using variable-rate DAMA.

6.4 The Random-Reservation Adaptive Assignment Protocol

The TDMA frame of the adaptive assignment protocol is divided into Reservation slots, Control slots, Data slots and Random Access slots, as shown in Figure 6-6. The protocol is based on the proposals by [BOHM93, CELA91, ZEIN91] with modifications to achieve the design objectives for multi-service networks. The most important contribution of this work is prioritised queuing of requests on-board the satellite and the use of random access for the UBR service class.

![Figure 6-6 Frame Structure](image)

A Reservation slot is that period of time in which terminals report their initial bandwidth requests to the On-Board Radio Resource Management (OBRRM) module. There are only a few Reservation slots available and a terminal selects one at random without knowing whether another station is using the same slot. The flow diagram for the initial reservation procedure, without queuing of request, is shown in Figure 6-7. It shows how the simulation model was broken into two parts. If more than one terminal selects the same reservation slot, a collision occurs and terminals have to retransmit
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after waiting for a *mean retransmit waiting time* determined by the collision resolution algorithm. The MAC protocol ensures that the collision probability stays low. The reason for using reservation slots is because ATM networks support different services which have different loss and delay requirements.

![Flow Diagram for Initial Reservation of TDMA simulation model](image)

Figure 6-7 Flow Diagram for Initial Reservation of TDMA simulation model

If a single request was received for a reservation slot (successful request), the OBRRM module tries to allocate the necessary Data slots. If no Data slots are available, the request can either be blocked (called blocking probability) or queued. We propose to queue successful requests in a prioritised queue so that the terminal does not need to compete with other terminals for a reservation slot again. By queuing successful reservations, requests can be allocated data slots according to their priorities.

Once the Data slots are reserved (successful reservation), an acknowledgement is transmitted to the terminal in Time Division Multiplex (TDM) mode on the downlink frame. Data slots represent the part of the frame in which a terminal can transmit its message after a successful reservation. In every frame there are many data slots and the OBRRM module will assign data slots to a particular successful request. A data slot is assigned to at most one terminal and therefore there is no possibility of collision.
On the other hand Random Access (RA) slots represent the part of the frame in which terminals can transmit without the need of making a reservation. The slots available for random access are broadcasted in the downlink frame. This part is for services which don’t want to wait for the lengthy reservation procedure. In random access mode it is not possible to guarantee a certain QoS to users although the protocol will try to minimise the number of collisions to maximise throughput by using an adaptive collision resolution algorithm. Random Access should only be used by UBR sources with relatively small burst length since RA terminals are not allowed to reserve slots and have to content for each MAC packet.

Unless the number of reservation slots per frame is carefully adjusted the result would be either low capacity utilisation and long delays (too many reservation slots, less capacity available for information transmission) or network backlog (too few reservation slots resulting in successive collisions and high delay). The number of reservation slots should be fixed for system behaviour where the number of collisions can be controlled by broadcasting a message in the downlink that services with lower priority should not send/resend requests till the collisions have been resolved. Our analysis has shown that two (one) reservation slots provide adequate performance for an average call holding time of longer than six (fifty) seconds. However when the number of collisions can’t be controlled (very short average call holding time) new reservation slots can be added by reducing the number of control slots.

The requests for dynamic slot allocation are done using the Control Slots which are assigned, on a round-robin basis (polling) to all inactive terminals which request the variable-rate DAMA MAC class. The number of control slots is set to eight to minimise the frame overhead.

State diagrams illustrating the various states of the CBR and VBR sources are shown in Figure 6-8 and Figure 6-9 respectively. The registration of the terminal is not shown on the diagram and a terminal which wants to transmit its initial request scans the corresponding broadcast signalling channel. Then it contends with the other terminals for the reservation slots. If the contention is successful then the OBRRM module will try to assign the required capacity. Once the capacity is granted the CBR terminal obtains the capacity for the connection duration. On the other hand, the VBR
terminal transmits the initial burst and then starts waiting in the polling mode for the next control slot to reserve bandwidth.

![Diagram](Image)

**Figure 6-8 State-diagram of CBR source**

The satellite frame introduces a constant delay equal to the frame length, on the cells of a stream connection. Therefore the selection of the frame size should be small enough to satisfy the delay limit of real-time services (400ms) [ITUT96a] taking into account the satellite propagation and processing delays and the delay introduced by the terrestrial B-ISDN.

### 6.5 Analysis of an Adaptive Random-Reservation MAC Protocol

We assume that the underlying satellite channel is slotted in time where the slot length is equal to the encoded MAC layer packet. This is because of the fixed size of ATM cells, and to limit the segmentation and re-assembly.

The considered time division frame has a frame length equal to a CBR packet generation time ($T_{cBR}$). This is to emphasise the importance of the MAC's ability to provide uninterrupted service to a CBR user. The simulated TDMA reservation system is designed to support up to 2.048 Mbit/s on the uplink.
One of the main evaluating factors of various access protocols is the normalised average throughput defined as:

\[
\text{throughput} = \frac{\text{the number of occupied timeslots in a frame}}{\text{the number of time slots in a frame}}
\]

The parameter which is typically varied to obtain performance curves is the (average) load of the system.

The offered traffic load \( (\rho) \) of the system can be calculated by adding the offered load of each service class:

\[
\rho_s = \sum_s \lambda_s h_s \frac{\text{PCR}_s}{\beta_s} \quad \text{[kbit/s]}
\]

where \( \lambda_s \) = average number of calls originating per minute for service type \( s \)

\( h_s \) = average call holding time of calls of service type \( s \)

\( \text{PCR}_s \) = PCR of service type \( s \) [kbit/s]

\( \beta_s \) = Burstiness of service type \( s \) = Peak rate/mean rate

The offered load as a ratio of the bit rate of an elementary channel with capacity \( C_e \) is:

\[
\rho = \sum_s \lambda_s h_s \frac{\text{PCR}_s}{\beta_s C_e} \quad \text{[Erlang]}
\] (6.1)

The normalised average load load (\( \rho_n \)) is then:

\[
\rho_n = \rho / \text{Number of time slots}
\] (6.2)

and the average number of active user terminals per service \( s \) is:

\[
N_{u,s} = \beta_s \rho C_e / \text{PCR}_s
\] (6.3)

Given a certain normalised average load, this number can be calculated backwards for known load shares between the different service classes and the corresponding PCR and burstiness values.

The MAC packet slot period \( C_e \) has been chosen to support a 32 kbit/s CBR stream and corresponds to one frame unit of 384 un-coded information bits every uplink frame. This results in a frame period of 11.9 ms to transmit 84 ATM cells per second.
using AAL5. There are 64 encoded MAC packet slots to support 2.048 Mbit/s of traffic per spot-beam on the uplink. The actual uplink transmission rate is higher due to ATM and MAC layer overheads. The call holding time was assumed 1 minute if not stated otherwise.

**6.5.1 CBR Simulation Results**

The MAC simulations were carried out using BOnes DESIGNER and all simulation models are described in Appendix B. The CBR simulations were carried out to validate simulation results with well known mathematical equations. For the mixed service environment it was not possible to use analytical methods hence simulations had to be used.

In the simulations we assume a Poisson arrival of calls and negative exponential distribution of call holding times, thus we can use the Erlang-B equation for a loss system with zero waiting room on-board the satellite and assuming that a blocked customer won’t try to call again.

Under these assumptions the Erlang-B table can be used to determine how many calls can be supported for a certain number of data slots and a required blocking probability. Note that each CBR connection was assumed 32 kbit/s.

![Figure 6-10 Comparison of Theoretical and Simulation Results of Blocking Probability vs Traffic Load](image)

**Figure 6-10 Comparison of Theoretical and Simulation Results of Blocking Probability vs Traffic Load**

The loss system simulation model was compared to the theoretical Erlang-B results to validate the simulation accuracy. As shown in Figure 6-10 the simulation results are within the 90% confidence interval with maximum 10% difference due to collisions.
The next step was to modify the simulation model so that blocked customers could retransmit their requests after a certain \emph{mean retransmit waiting time}. Again zero waiting room was assumed on-board the satellite, which means that if slot allocation is not possible the call is rejected and not queued till a slot becomes available.

The \emph{mean retransmit waiting time} is very important since the total number of calls attempting to request a reservation is equal to the number of newly generated plus retransmitted ones. If the blocked request retransmits too early they might be blocked again, because no data slot may be free resulting in a drastically increased blocking probability.

![Figure 6-11](image)

**Figure 6-11** (a) blocking probability (b) mean call reservation delay (Scenario 1).

Simulations were run for various mean retransmit waiting periods for a normalised traffic load of 0.86 at a mean call holding time of 1 minute (Scenario 1). The number of reservation slots was fixed at two, as it was observed (for Scenario 2) that one reservation slot could result in too many collisions. More reservation slots could have been used but this would increase the frame overhead and decrease the frame efficiency. It can be seen from Figure 6-11 (a) that the longer the mean retransmit waiting time the lower the blocking probability.

As the mean retransmit waiting time approaches infinity, the system behaviour approaches the pure loss system where the Erlang-B equation applies. However the longer the retransmit waiting time, the longer becomes the mean reservation delay as shown in Figure 6-11 (b).
Another important observation from Figure 6-11 (a) is that queuing requests on-board the satellite greatly reduces the blocking probability. Furthermore, the size of the on-board queue does not have a large impact on the mean reservation time for the chosen load as seen from Figure 6-11 (b). The impacts of increasing load on the reservation time will also be investigated.

Repeating the above simulations for a normalised load of 0.86 and for a mean call holding time of 6 seconds (Scenario 2) it can be observed from Figure 6-12 (b) that there is a point where the mean reservation delay is minimum. Any retransmit waiting time below this value means that the terminal retransmits too early and that it’s request most probably won’t be successful again (either due to collision or blocking).

Figure 6-12 (a) blocking probability (b) mean reservation delay for Scenario 2.

The optimum mean retransmit waiting time has to be calculated by the collision resolution algorithm and broadcasted on the downlink. Note that this value will increase as the load increases and it may not be the same for each service class. Thus services with lower priority might have to wait longer before retransmitting.

Figure 6-13 (a) Blocking Probability (b) Mean Reservation Delay vs Load using the Static Scheme.
First the mean retransmit waiting time was fixed at 1.5 seconds for a call holding time of 1 minute. The blocking probability and mean reservation delay of the system is shown in Figure 6-13 (a) and (b) respectively.

It was observed that an adaptive mean retransmit waiting time improves the system performance. Figure 6-14 shows the blocking probability and mean reservation delay as a function of the normalised load for a static and adaptive mean retransmit waiting time. It can be clearly seen that an adaptive mean retransmit waiting time reduces the blocking probability. The mean call reservation delay is not reduced significantly for this case.

![Figure 6-14 (a) Blocking Probability (b) Mean Reservation Delay vs Load using the Adaptive Scheme.](image)

As the offered traffic load is increased so does the number of collision since there are only a limited number of reservation slots available. Figure 6-15 shows the probability that a reservation request collides with the request of another terminal as a function of the increase in the offered load for a static and adaptive mean retransmit waiting time. The collision probability for the adaptive mean retransmit waiting time (exponential distribution proportional to the number of collisions) is lower, in particular when no or a small on-board request queue is used. Note the sharp increase in the collision probability when the normalised load exceeds 0.83 for the static mean retransmit waiting time. The reason for this, is the increase in the blocking probability. Since blocked calls are retransmitted after a certain time an increase in blocked calls results in more reservation requests and thus more requests collide. Collisions can be minimised by queuing calls if no empty slots are available. Thus calls which
successfully access a reservation slot do not have to compete with other stations when no data slot is available at the time of the request.

![Collision Probability vs Normalised Traffic Load](image)

**Figure 6-15 Collision Probability vs Normalised Traffic Load**

Figure 6-16 shows the normalised throughput of the data slots, where one would be 2.048 Mbit/s throughput. As it can be seen a throughput of up to 95% of the system data capacity can be achieved by using an on-board request queue. This does not take into account the overheads which may be up to 45%. Hence the total system bandwidth utilisation may only be 50%. However statistical multiplexing will result in increased efficiency for the use of the available bandwidth.

![Normalised Throughput as a function of the Traffic Load](image)

**Figure 6-16 Normalised Throughput as a function of the Traffic Load**

While queuing of request on-board the satellite improves the utilisation and system performance in terms of blocking and collision probability, it does not contribute significantly to the mean reservation delay for stable system behaviour.
6.5.2 VBR Simulation Results

As for CBR, rt-VBR traffic will also use fixed-rate DAMA because of the CTD constraints of rt-traffic. Nrt-VBR traffic on the other hand can be supported by both fixed-rate and variable-rate DAMA.

Each nrt-VBR source represented by an on-off source model described in Chapter 3, where $a^{-1}$ is the mean burst period and $b^{-1}$ is the mean silence period which are both exponentially distributed. The burstiness is defined as:

$$\beta = \frac{PCR}{SCR} = \frac{a^{-1} + b^{-1}}{a^{-1}}$$  \hspace{1cm} (6.4)

6.5.2.1 Use of fixed-rate DAMA and effective bandwidth assignment for nrt-VBR traffic

For a bursty nrt-source with a certain PCR and SCR an effective bandwidth can be reserved. Bursts arriving at the PCR, have to be buffered since the service rate is equal to the assigned effective bandwidth. Hence the larger the assigned effective bandwidth the smaller the mean queuing delay but the larger the unused bandwidth, as shown in Figure 6-17. The PCR of the bursty source is assumed 64 kbps and the burstiness is 5. As it can be seen from Figure 6-17 there is a trade-off between queuing delay and bandwidth utilisation. If the user has no real-time constraints than a lower effective bandwidth assignment can be made and more users supported.

![Figure 6-17](image)

**Figure 6-17** (a) Mean Queuing Delay (b) Maximum number of supported users

The reason why effective bandwidth assignment is not recommended for rt-services is apparent from Figure 6-18 where the maximum queuing buffer contents and the maximum queuing delay is shown.
6.5.2.2 Use of Variable-rate DAMA and burst reservation for nrt-VBR traffic

Next the use of variable rate-DAMA for nrt-VBR is investigated. Larger buffers can be used on-board the satellite since there are no real-time constraints for nrt services. If a control station is used to assign satellite resources then each burst reservation request has to be first send to a ground terminal. This request will be processed and then transmitted to the satellite resulting in a round-trip delay of approximately 270ms for a GEO satellite. This delay can be prevented by using an OBP satellite with OBRRM functions.

Furthermore, by queuing requests and by using an adaptive collision resolution algorithm the blocking and collision probabilities can be minimised. Thus this section will mainly focus on the performance of the system in terms of utilisation and reservation delay.

Nrt-VBR connections using variable rate DAMA first have to reserve the initial capacity by using a Reservation slot and can then use a Control Slot assigned to them in round-robin fashion for any other requests. Thus there are two different reservation delay values. The delay for the initial capacity request using the reservation slot is called ‘Call Reservation Delay’ (or only ‘Reservation Delay’) and the delay for the reservations using the control slot is called ‘Burst Reservation Delay’. The slot reservation is relinquished during the silence periods. In this way the air interface bandwidth is shared between multiple sources achieving statistical multiplexing on the satellite air interface.
The load can be calculated from Equation (6.1). Simulations (parameters are shown in Table 6-1) are carried out for traffic with burstiness ($\beta$) of 5, 10 and 20. Note that for all the simulations the number of active terminals is equal to the call arrival rate. The confidence interval for all simulations is 95% and not shown on the graphs for the neatness of the results.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Brief Description</th>
<th>Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>Average number of CBR and rt-VBR calls (per minute)</td>
<td>The number of calls generated are assumed of Poisson arrival with a certain mean which determines the arrival rate</td>
<td>20-60</td>
</tr>
<tr>
<td>CBR, nrt-VBR and UBR Call holding time</td>
<td>The average call holding time per CBR, VBR and UBR terminal (in minutes)</td>
<td>1 minute</td>
</tr>
<tr>
<td>Average number of nrt-VBR calls (per minute)</td>
<td>The number of calls generated are assumed of Poisson arrival with a certain mean which determines the arrival rate</td>
<td>0-32\beta</td>
</tr>
<tr>
<td>Required Bandwidth for CBR</td>
<td>The requested bandwidth per terminal</td>
<td>32 kbit/s</td>
</tr>
<tr>
<td>Required Bandwidth for nrt-VBR source</td>
<td>The requested bandwidth per terminal</td>
<td>64 kbit/s</td>
</tr>
<tr>
<td>Reservation Slots/Frame</td>
<td>The number of reservation slots per Frame</td>
<td>2</td>
</tr>
<tr>
<td>Data Slots/Frame</td>
<td>The number of data slots per Frame Period.</td>
<td>64</td>
</tr>
<tr>
<td>On-board Request Buffer Size</td>
<td>This buffer queues requests when no data slots are available reducing the blocking probability</td>
<td>100</td>
</tr>
<tr>
<td>Header Period</td>
<td>The length of the header (in ms)</td>
<td>1</td>
</tr>
<tr>
<td>Frame Period</td>
<td>The length of the frame (in ms)</td>
<td>11.9</td>
</tr>
<tr>
<td>Mean VBR Burst Period</td>
<td>The mean duration of the burst for VBR traffic</td>
<td>100 ms</td>
</tr>
<tr>
<td>Mean VBR Silence Period</td>
<td>The mean duration of the inactivity period of VBR traffic following the burst</td>
<td>$100*(\beta-1)$ ms</td>
</tr>
<tr>
<td>Control Slots</td>
<td>The number of signalling slots used by nrt-sources to request or relinquish bandwidth</td>
<td>8</td>
</tr>
<tr>
<td>Propagation Delay</td>
<td>The delay introduced through the satellite link</td>
<td>135 ms</td>
</tr>
</tbody>
</table>

Table 6-1 Simulation Parameters

The simulation results show (Figure 6-19) that a normalised information throughput of up to 0.95 can be achieved. The reservation delay remains reasonably low, for low burstiness values. As the burstiness is increased so does the mean call reservation
delay as more sources need to be multiplexed on the satellite air interface to achieve high utilisation.

![Diagram](https://via.placeholder.com/150)

**Figure 6-19** (a) Utilisation (b) Reservation Delay for nrt-VBR services with various burstiness values.

Choosing a low number of control slots to minimise frame overheads results in increased burst reservation delay as can be seen in Figure 6-20. The increase in burst reservation delay become more visible as the number of active terminals increases. If each source is assigned an individual control slot then there will be large frame overhead for $N$ sources while achieving very low burst reservation delays. Note that $N$ (number of terminals) can be calculated from Equations (6.3). Since the terminals are expected to buffer bursts at the PCR for the burst reservation delay period, a reduced burst reservation delay results in smaller buffer requirement for the terminal, to avoid cell loss due to buffer overflow.

As it can be seen from Figure 6-20 (b) the maximum reservation delay for VBR is too long to support rt-applications. Even for lower orbits such as Low Earth Orbit (LEO) the maximum burst reservation delay may be too high under high loads [BOST98].

The mean burst duration has been assumed 100ms for all the simulations. If the mean burst duration is shorter than the frame duration than the efficiency of the reservation procedure is dependent on the burstiness of the source. The throughput for very bursty traffic with short burst duration is not very high as will be shown for UBR traffic. However it is anticipated that most traffic using the VBR service class won’t be very bursty.
Figure 6-20 (a) Mean (b) Maximum Burst Reservation Delay

6.5.3 CBR and nrt-VBR Analysis

Next the effects of CBR traffic on nrt-VBR traffic is investigated by fixing the mean nrt-VBR load at 0.4 and varying the CBR load. The increase in the call and burst reservation delay as a function of CBR load is shown in Figure 6-21. The increased delay is due to the fact that as the total offered load is increased, it exceeds the system capacity and hence request have to be buffered which results in longer delays till bandwidth allocation can be made. Again the burst reservation delay is minimised by allocating a control slot for each VBR source as shown in Figure 6-21 (b). For eight control slots the burst reservation delay increases since control slots are assigned in round-robin fashion to terminals which can only request bandwidth when the control slot is assigned to them.

Figure 6-21 (a) Mean Reservation Delay (b) Mean Burst Reservation Delay
The achieved throughput is similar for various burstiness values as shown in Figure 6-22, due to the low number of collisions and blockings.

![Figure 6-22 Total System Throughput vs Normalised CBR Load](image)

**6.5.4 CBR, VBR and UBR Analysis**

The novelty of the adaptive MAC protocol is its ability to support lower priority UBR traffic in both reservation and random access mode according to the terminals burst duration.

The UBR load is calculated in the same way as for VBR traffic (equation (6.1)) and the throughput of UBR for a pure-reservation system, (simulation parameters shown in Table 6-1) in a scenario with a high number of UBR sources with very bursty traffic ($\beta=200-5000$) is shown in Figure 6-23(a). The number of arrivals can be calculated from equation (6.3). The PCRUBR is assumed 32 kbit/s and the UBR burst period is varied from 13 ms (1 cell) to 416 ms (depending on the burstiness).

As it can be seen the amount of carried traffic remains unacceptably low, even for low to medium load due to the high number of collisions. The pure-reservation MAC throughput increases for traffic with longer burst duration (or lower burstiness). Only for burstiness values lower than 200 can a throughput higher than 25% of the frame capacity be achieved.

The offered UBR traffic load is dependent on the burstiness of the traffic and Figure 6-23 (b) shows the delay as a function of the UBR terminal numbers, (which is also the number of arrivals per minute) for $\beta_{UBR}=2307$. As it can be seen, UBR traffic is increasing the call reservation delay of higher priority CBR and VBR traffic. Since all
the terminals content for the same reservation slots, UBR traffic increases the collision probability and the access delay.

![Graph 1: Throughput vs normalised UBR load](image1)

![Graph 2: Reservation Delay vs Arrival Rate per minute](image2)

**Figure 6-23 (a) Throughput vs normalised UBR load (b) Reservation Delay vs Arrival Rate per minute**

To improve the UBR throughput and the access delay we propose that UBR sources access the MAC slots remaining after the reservation procedure by Random Access (RA). This way the lengthy reservation procedure is avoided and the number of collisions reduced. The reservation delay of terminals using reservation (CBR and VBR terminals) is unaffected by RA terminals.

The UBR throughput using RA is limited by 36.8% of the available RA capacity (theoretical Slotted Aloha limit). This throughput is higher than the pure-reservation throughput for very bursty traffic ($\beta=500-5000$). Our simulations showed that in order to minimise RA collisions and access delays, the UBR load has to be kept around 25% of the available RA capacity.

**6.6 Summary**

As user demands become more complex, satellite networks are expected to support a much wider range of services. As satellites will play an important role in the deployment of ATM networks, we addressed the optimisation of the capacity allocation scheme, using performance results for an adaptive MAC scheme.

Considerable improvements in delay performance and satellite bandwidth utilisation are possible if next-generation satellite technology (OBP) and an adaptive MAC protocol is used. The traditional demand-assignment scheme using a ground terminal
as control station has two important drawbacks: long set-up and reservation time and limited channel utility. Both are due to the long propagation delay of the GEO satellite link and can be removed by processing channel requests in the satellite to allocate frame slots. Furthermore, the problem of variable channel access delay of PRMA and its variants is solved by using control slots which are assigned to the VBR terminals.

The mapping of ATM service classes to MAC classes and the use of a prioritised request queue provides the QoS differentiation required by ATM networks. It was shown that a pure reservation system performs poor for very bursty user traffic. The user population which can be supported using RA for very bursty users with short burst duration is much higher.

It was also shown that by using both variable-rate DAMA and RA, the utilisation of the frame capacity and the total number of users served is greatly increased by statistically multiplexing traffic over the air interface.
7. Performance Evaluation of Congestion Control Schemes for OBP Satellites

Maintaining high utilisation of satellite link resources while satisfying the users' traffic contract is essential for satellite ATM networks where satellite air-interface bandwidth is the most expensive commodity. However, striving for high utilisation of a resource (satellite link bandwidth) without proper allocation may lead to long queuing delays and losses resulting in a low throughput.

Although preventive traffic control techniques reduce the probability of buffer overflow, it is not possible to eliminate temporary periods of congestion due to multiplexing buffer or switch output buffer overflow on-board the satellite. Furthermore the utilisation of resources is usually not very high due to the admission control mechanism (part of preventive control), which has to calculate how many sources can be statistically multiplexed without too many of them (exceeding the link capacity) being active at the same time. Hence some capacity is always unused in order to prevent congestion. Since the aim is to maximise the utilisation of the scarce satellite bandwidth, best-effort services are supported to utilise the bandwidth remaining from the guaranteed services.

In periods of congestion, the satellite network throughput can be considerably reduced by retransmission of cells by the sources, increasing the network traffic and turning the momentary buffer overflow to sustainable periods of cell loss. Therefore, in addition to preventive counterparts, reactive control mechanisms are necessary to monitor and avoid prolonged congestion on the satellite and notify sources when congestion is detected.

Thus congestion control mechanisms which are used to avoid sustainable periods of cell loss and come into play after the network is overloaded are expected not to be very effective for temporary congestion periods. There may be short time-intervals where the demand for capacity is higher than the available capacity and till the congestion control reacts, the reason for congestion may already be over (due to the high GEO satellite link propagation delay feedback can only be received after 250ms). Therefore 'congestion avoidance' mechanisms are necessary for reaction before the network is overloaded, i.e. congestion is predicted (also called pro-active congestion
control). Congestion avoidance is part of the congestion control and aims at minimising temporary congestion but can result in underutilisation of the link capacity.

The use of feedback control enables ATM networks to support best-effort service classes such as ABR. The efficiency of supporting these kind of best-effort service classes over satellite under high propagation delay will be investigated in this Chapter.

7.1 Reactive Control Framework

Reactive control mechanisms have been satisfactorily used in low-speed packet-switched networks. However the performance and effectiveness of feedback-based schemes in the presence of non-negligible propagation delay in high speed network is an important issue. This results in large amount of cells generated, depending on the bit rate of the connection, before any feedback information is received by the source. Accordingly, reactive control mechanism are not as effective in satellite ATM networks as they are in terrestrial ATM networks. The effectiveness of a reactive mechanism in a satellite ATM environment mainly depends on the connection duration, the congestion duration (the burst length) and the round-trip propagation delay.

Ensuring both high utilisation of the available bandwidth and a minimum CLR requires efficient as well as fair congestion control schemes. These schemes should achieve three goals [YIN94]:

- The QoS negotiated during the connection establishment has to be satisfied for each source.
- Unused bandwidth should be fairly distributed among the active connections.
- During congestion the connections that are using more bandwidth than was negotiated at the connection establishment should be given the opportunity to reduce their rate before the network starts discarding traffic in excess of the negotiated QoS.

Various reactive control mechanisms have been proposed which can be classified according to:

- Congestion Detection Methods,
- Congestion Notification Techniques,
- Action to control the source.
These have been studied to find out the most effective combination and the relevant implementation complexity.

### 7.1.1 Congestion Detection Methods

The congestion detection module acquires information about the state of the network and indicates congestion and normal state according to certain thresholds. Mainly two methods are used for congestion detection which are described below.

#### 7.1.1.1 Monitoring the Buffer Occupancy

This method of congestion detection is widely used because of its simplicity in implementation. If the queue occupancy of a network element, such as switch buffer, exceeds a certain threshold a congested state is detected. As soon as the monitored buffer occupancy drops below the threshold a normal state is indicated. Thus employing only one threshold for congestion detection provides no robustness against instantaneous and rapid fluctuations of the buffer contents. Incorporating two buffer thresholds avoids this rapid fluctuation between congested and normal states.

However, the proper dimensioning of both, the threshold indicating congestion ($T_u$) and the threshold indicating normal state ($T_L$) is very important. The value of $T_L$ should be relatively smaller than that of $T_u$ to avoid oscillations of the transitions between congested and normal state.

#### 7.1.1.2 Monitoring the number of active sources

Monitoring the arrival rate of the traffic to estimate the number of active sources is an alternative congestion detection method. If the number of active sources exceeds a certain threshold congestion is detected. Then if the number of active sources drops below a second lower threshold a normal state is indicated. Again the setting of both thresholds is very important for the performance of the reactive control.

The advantage of using the number of bursts in progress as means of congestion detection is that it may detect the onset of serious congestion sooner. Thus making it possible to take the control action earlier (congestion avoidance), compared to a buffer occupancy based scheme, to account for the propagation delay.

To illustrate this with an example assume a link capacity of 2.048 Mbit/s and 75 multiplexed bursty sources each with a 64 kbit/s peak rate. In this case the
multiplexing buffer will start filling if more than 32 sources are active and buffer occupancy based monitoring can detect congestion only if the 33rd source gets active. On the other hand the scheme monitoring the number of active sources can already indicate the onset congestion when 29 or 30 sources are active (if desired). This however results in lower utilisation of the network since congestion is indicated before the actual capacity of the link is fully used.

7.1.2 Congestion Notification Techniques

According to the information provided by the congestion detection module, this module applies control action to the source. There are mainly three techniques to control the sources by informing them about congestion within the network.

The main objective of dynamic notification schemes for bursty sources is to assign rates such that even though the sum of all allocated rates is greater than the system capacity, with very high probability the received aggregate rate of guaranteed services will be below system capacity. The probability of congestion has to be kept low and in periods of congestion best-effort ABR, GFR and UBR traffic is discarded first. As mentioned earlier conforming ABR and GFR traffic will experience a low CLR.

7.1.2.1 Explicit Backward Congestion Notification (EBCN)

In an EBCN scheme once congestion is detected a special Resource Management (RM) cell is prepared and sent to the source nodes of all connections that pass through the congested node. This method minimises the time it takes to notify a source that there is a congested node along its path. Since a special cell is used for this purpose, the 48-byte payload (minus overhead, used to specify that the cell is carrying a congestion indication) of the cell can be used to include a variety of information about the congested node. Accordingly, sources can react to congestion along their path. Despite these advantages, EBCN is not accepted by the standardisation committees as a viable approach for congestion indication in terrestrial ATM networks. This is mainly due to the use of special cells that impose a considerable processing burden on intermediate nodes.
Although this congestion indication technique is not attractive for terrestrial networks with a high number of intermediate nodes, it is feasible to implement EBCN for OBP satellites to minimise the feedback delay from the source of congestion.

If this congestion notification technique is applied, the tuning of the thresholds used in the congestion detection mechanism is very important to have a high throughput. Thus when evaluating the performance of the EBCN scheme the amount of EBCN cells generated has to be investigated. By using filters it is possible to minimise the number of EBCN cells at the expense of a little delayed reaction. The use of the EBCN scheme on satellites with cell switching capabilities is very attractive, due to the inherent broadcast capabilities of the satellite.

Considering that feedback information may be lost either due to the satellite link conditions or due to overload a positive rate control scheme is more suitable for the satellite environment. Proportional Rate Control Algorithm (PRCA) is a proposal for terrestrial network which could also be applied to satellite networks. In this scheme the rate can only be increased if a positive indication, in this case a RM cell, was received. Otherwise, the rate is decreased after each sent RM cell. Rate reductions and increases are done on proportion to the current sending rate.

The performance of this scheme has been evaluated [SISA96] with the result that high utilisation can be achieved when setting a high congestion threshold. However the main drawback of this scheme was the unfair distribution of bandwidth to the sources and the long time it takes to reach a high throughput.

7.1.2.2 Explicit Forward Congestion Notification (EFCN)

In the EFCN scheme once congestion is detected all cells passing through that trunk are marked until the congestion period ends (when the normal state is indicated). Once the EFCN bit of a cell is set at a node, it cannot be modified by any other node along the path to the receiver. A reserved bit in the cell header that is defined by the ITU-T may be used for this purpose. A cell received at a destination node with the EFCN bit set indicates that there is a congested node along the path of this connection. In this scheme, receivers have no information on which nodes in the network are congested and do not indicate the particular connection the marked cell belongs to that is not conforming to the negotiated connection parameters.
The performance of this scheme is highly dependent on the distance between source and destination. If the propagation delay is low (like in ATM LANs) than the EFCN may be efficient but as the propagation delay increases by the time a marked cell arrives at a receiver it is possible that the congested nodes which marked the cell are no longer congested. Therefore for high propagation delays, receivers do not react to congestion indication very quickly. Instead, statistics are collected to accurately determine whether there is momentary or sustained congestion along the path. If it is decided that the latter is the case, then the receiver sends a notification back to the source.

In a satellite environment where the propagation delay is high the EFCI would not be effective. Depending upon the satellite delays, this could require an implementation to employ Virtual Source / Virtual Destination (VS/VD) behaviour at the satellite in order to halve the effective control loop delays. A satellite acting as VS/VD will divide the end-to-end control loop into separately controlled segments by acting like a (virtual) destination on one segment, and like a (virtual) source on the other. The VS/VD option is specified in the ABR specifications [ATMF96a].

7.1.2.3 Explicit Rate Indication

Explicit Rate (ER) indication notifies the source of the exact bandwidth share it should be using in order to avoid congestion. This method seems to be very effective considering the long satellite propagation delay. Various algorithms have been proposed for the calculation of the fair bandwidth share per connection [KALY97].

ER indication can be used for ABR which is an ATM-layer service where the network bandwidth offered to an end-user application may alter dynamically during the lifetime of the connection, by the use of feedback control.

ABR requires the use of Resource Management (RM) cells, shown in Figure 7-1, to be inserted periodically into the same Virtual Channel Connection cell stream as the associated data. With the Explicit Rate (ER) variant of ABR, the RM cells include information stating a source’s current bandwidth allocation, the Current Cell Rate (CCR) and a requested bandwidth, held in the ER field. The source sets the CCR field to its current cell transmission rate which cannot exceed its assigned Allowed Cell Rate (ACR). It also sets the ER field to the maximum bandwidth it would wish to use.
The RM cells are then transmitted periodically within the data cell stream. At each network component the CCR value is monitored and the ER field lowered to a revised level if the component cannot accept the current requested setting. This may happen in both the forward and return ABR flow path. The RM cells are identified by PTI field in the ATM cell header as well as the RM protocol identifier.

<table>
<thead>
<tr>
<th>ATM Header</th>
<th>RM Protocol Identifier</th>
<th>Function specific fields</th>
<th>Reserved</th>
<th>EDC (CRC-10)</th>
</tr>
</thead>
<tbody>
<tr>
<td>5 octets</td>
<td>8 bits</td>
<td>45 octets</td>
<td>6 bits</td>
<td>10 bits</td>
</tr>
</tbody>
</table>

EDC: Error Detection Code

**Figure 7-1 Resource Management Cell Format**

Once the RM cells are received back at the ABR source, the ACR is immediately reduced to the returned ER value if this is less than the current ACR. However, if the returned ER value is greater than the current ACR, the source is permitted to increase its ACR setting to the lower of the revised ER value or the current ACR plus its Additive Increase Rate (AIR). Additional rules define how the source should behave in the absence of data to transmit or if backward RM cells have been lost or delayed. ER control which employs Virtual Source/Virtual Destination behaviour at the satellite can halve the control loop delay.

**7.1.3 Actions to control the source**

7.1.3.1 Adaptive Rate Control

In adaptive rate control, the rate of the source is controlled by feedback information. The variants of the adaptive rate control depend on the source parameters to be controlled and on how these parameters are changed. The Available Bit Rate (ABR) service class aims to adjust the allocated bandwidth according to the state of the network to achieve maximum resource utilisation. The source activity can either be reduced by reducing the peak rate, increasing the burstiness or by adaptive source coding. The effects of these control action are explained below.

- Reduction of peak rate
The impact of reducing the peak rate is that it will increase the amount of delay that it takes to emit a burst. For example the transmission time of a large image file can be
increased from 0.5 sec to 1 sec by reducing the peak rate to half. Thus decreasing the peak rate results in increased delay. Despite this fact this is the most widely used control action in many papers [MATR93][YIN94][ATAI94]. The main advantage of peak rate reduction is that it provides fast control compared to other adaptive source control methods.

- Increment of burstiness

The burstiness can be increased by:

  i) Decreasing the mean burst length. This can be achieved by data compression techniques which may reduce the QoS somewhat.
  
  ii) Increasing the mean off (or silent) period. This is equivalent to momentarily choking the traffic sources and thereby reducing service.

- Dynamic Source Coding

When a congestion indication is received by a source, the rate at which traffic is submitted to the network may be reduced temporarily. For delay-insensitive traffic, cells may be delayed in the source buffers. For real-time delay-sensitive traffic, buffering is not a desired alternative. Either the source traffic submission rate can be reduced or not-so-essential cells that may be reconstructed from other cells at the receiver can be marked with a low cell-loss priority before they are transmitted. For example, voice frames can be transmitted with the most significant four bits in one cell and the least significant four bit in another. The CLP bit of the latter is set before they are transmitted to provide the option to discard these cells before the others during congestion. In the case of video services, the quantisation step may be increased to reduce the amount of information generated per frame. Similar to voice frames, video frames with different levels of significance may be transmitted with different cell-loss priorities.

7.1.3.2 Adaptive Credit Control

In the window, or credit based procedure a source is allocated a maximum number of cells it can transmit into the network before receiving an acknowledgement of correct reception at the destination. Control messages from either the destination or from a congested point within the network are then used to reduce the window size or credit of outstanding packets. Since the ITU-T and ATM Forum chose to adopt a rate based scheme [ATMF96a, ITUT96b] the credit-based control won't be discussed in further detail.
7.1.4 Application of Reactive Congestion Control Schemes in WANs

The performance investigation of reactive control in wide-area high-speed networks is also of interest. Although the delay for satellite ATM networks will be higher, the techniques used for wide area networks (WANs) could be applied to the satellite environment.

The performance of an EFCI scheme for the specific case of file transfer where the transmission time of a data burst is much longer than the round-trip propagation delay time is investigated in [YIN94]. The occupancy of the queue was used to control the peak rate showing that for this particular case with delay insensitive traffic, improvements in cell loss due to congestion can be achieved. Again restricted to data traffic, a combined reactive/preventive control approach is proposed [MATR93]. A queue length threshold is again used to detect congestion and BECN is used to notify the source and policing mechanism (Leaky Bucket). When a certain threshold is exceeded the source peak rate is reduced and the LB parameters are set to a predefined congestion setting, reducing the number of generated tokens. The results show that improvements in system performance can only be achieved for propagation delays up to 300 cell times (due to the small buffer sizes chosen). A very good paper on congestion detection by monitoring the arrival rate of the traffic employing two thresholds [ATAI94] shows how significant improvements in cell loss can be achieved for propagation delays up to 50ms assuming a peak bit rate of 10Mbit/s. In order to achieve fast reaction the congestion state is indicated before the full link capacity is reached (this means while 10 sources can be in the on state without exceeding the link capacity, congestion is indicated if only seven or eight of the sources are in the on state) thus reducing utilisation. The presented results are generally for a fixed buffer size of 1 Mbit which is a slightly large buffer for a typical switch or multiplexer. In cases where high link utilisation is not required or where buffer threshold detection is not sufficient to react fast enough, the proposed method is a good solution. Finally dynamic source coding using BECN for video sources is investigated in [DAGI95a, DAGI95b]. The results show that using two thresholds, cell loss improvements due to congestion can be achieved for delays up to 4 ms.
7.2 Performance Evaluation of Reactive Control for OBP Satellites

This section investigates how the performance of a preventive control scheme using a Leaky Bucket (LB) can be improved by using a feedback control loop. The effects of the transmission delay and the system parameters on the system performance are therefore examined.

First a combined preventive/reactive control scheme for satellites is described where the rate of the sources is reduced by using Binary Congestion Notification at the onset of congestion. Then a more advanced feedback control loop is evaluated where Explicit Rate Indication is used to control the activity of the ABR sources.

The combined preventive/reactive control scheme for satellites using the adaptive LB is shown in Figure 7-2. The congestion control scheme has two modules which are the congestion detector and the traffic controller. The main function of the congestion detector is to detect the onset of congestion. Then the ABR sources will be notified to reduce their activity (currently only the ABR class supports feedback control).

Whatever assumptions are made on the traffic entering the ground terminal, the multiplexing and multiple access functions in the up-links alter its statistical characteristics. In order to evaluate the performance of the OBP satellite we therefore chose to use simulations.
The bandwidth delay product (BW*delay) is a very important parameter to assess the
effectiveness of reactive control mechanisms. It is defined as:

\[ BW\text{-}delay = \text{Propagation delay} \cdot \text{Peak Rate} / 424 \text{ (cells)} \]

This equation shows the number of cells which can be transmitted by a source during
the propagation delay time. Twice this value of cells can be transmitted before any
feedback can be received by a source.

7.2.1 Reactive Control using Binary Congestion Notification

Monitoring the arrival rate to detect the onset of congestion will be used for
congestion avoidance. Once the congestion onset threshold is exceeded, binary
congestion notification (BCN) is used for reduction of the ABR source rate by a
prespecified throttling rate. ABR source rate reductions and increases are done in
proportion to the current sending rate thereby enhancing the fairness of the rate
control scheme. The same congestion information is also sent to the UPC (adaptive
LB) but delayed by two times the transmission delay so that the source can change its
parameters. The transmission delay is equal to the sum of propagation delay and
switching delay from the point of congestion to the source.

Once the measured arrival rate drops below a second lower threshold, notification is
sent to the sources that congestion is over. After the source receives information that
congestion is over it increases its activity by a prespecified rate.

The choice of the higher onset threshold (\(T_\text{h}\)) and the congestion over threshold (\(T_\text{L}\)),
has to be made such that the probability of changing states from \(T_\text{h}\) to \(T_\text{L}\) in less than
two times the propagation delay time is negligible. This can usually be accomplished
by allowing sufficient distance between \(T_\text{h}\) and \(T_\text{L}\). For the GEO satellite propagation
delay however it may be more difficult to detect fluctuations in the aggregate traffic.

There are only four distinct periods realisable for this control scheme as shown in
Figure 7-3. \(t_1\) is denotes the time when the arriving traffic exceeds the onset of
congestion and \(t_2\) denotes the time when the arriving traffic drops below lower the
threshold which indicates that congestion is over. The cone-shaped regions at time \(t_1\)
and \(t_2\) are meant to show boundaries of the most likely trajectories of the arrival
stream over periods \([t_1, t_1+d_1+d_2]\) and \([t_2, t_2+d_1+d_2]\). Note that the ECN indicating the
rates should be reduced, sent at time $t_1$ will not arrive at the source until time $t_1 + d_1$. It is assumed that when the ECN is received, all sources reduce their sending rate immediately. Thus, it would take another $d_2$ seconds for the reduced arrival rate to reach the destination. Therefore during period I the effective rate is still the initial rate. During the entire period II, the ABR sources are transmitting with a reduced rate and thus congestion is reduced. Due to new burst arrivals before this reduction, there is a possibility that overflow may occur in period I. The chosen threshold $T_H$ should reduce this probability to tolerable limits.

![Diagram](image)

**Figure 7-3 Realisation of aggregate arrival rate**

In order to evaluate the performance of BCN for the satellite environment the method based on measuring the arrival rate is chosen to detect the onset of congestion as early as possible. This is done by estimating the number of active sources and comparing the arrival rate with the onset threshold. When the measured arrival rate reaches the onset threshold, information is send to the sources (in this case ABR sources) to reduce their activity. A similar approach was used in [CHU94a, CHU94b].

For simplicity we assume that all $N$ multiplexed VBR sources are identical and that all ABR sources are modelled by the persistent source which always sending with the maximum permitted rate. Different simulations [BARN95] have shown that this model imposes the heaviest constraints on the network and is therefore very appropriate for testing the throughput and cell loss of the ABR service. CBR sources are not considered since they do not affect congestion temporarily except for a long term reduction of transmission capacity. The feedback propagation delay to each of
the sources is assumed the same. Alternatively two times the largest transmission delay to the source could be used before changing the parameters of the LB.

The number of multiplexed VBR sources \( N \) depends on the Connection Admission Control (CAC) which decides if a new call can (or cannot) be accepted. The algorithm which was used for CAC was derived using fluid-flow analysis (see appendix C). The accuracy of this algorithm was verified in [SYKA93] and [ORS94] for the superposition of \( N \) identical ATM sources on an ATM link.

A VBR source modelled by an on-off source is used in the simulations. The source has the following traffic parameters: burstiness=2.85, average burst period=100 ms and the peak rate is varied. The bandwidth allocated to each source is called Effective BW (EBW). If the GEO satellite link propagation delay is 19 cell times for a source peak rate of 64 kbit/s (from Table 4-1)

In order to observe the performance improvement of feedback using simulation, an EBW of 27.3 kbit/s for \( N=75 \) sources with 64 kbit/s peak rate and an on-board switch output buffer of 200 cells was chosen. This gives us a link bandwidth of 2.048 Mbit/s (total link capacity 4Mbit/s) for a CLR of \( 1.95 \times 10^{-3} \). In other words congestion will occur when more than 32 VBR sources are active. The reason for choosing a rather high CLR is to make comparison with simulation results possible. If the remaining capacity is assigned to ABR sources than for the static case (without feedback) the CLR has been found to be \( 10^{-3} \) (using simulations). The aim is to select appropriate \( T_H \) and \( T_L \) values in order to achieve lower cell loss due to congestion.

Figure 7-4 shows the CLR of the on-board buffer as a function of the buffer size for various upper threshold \( (T_H) \) values. For a uplink propagation delay of 135 ms the bandwidth delay product was fixed at 38 cells. The additive increase factor was 2 and the multiplicative decrease factor was 4 to enable fast reduction of the source rate at the onset of congestion. Higher multiplicative decrease factors (which result in less aggressive source rate reduction) could not reduce the CLR sufficiently.
Figure 7-4 CLR vs Buffer Size for various $T_{II}$ values.

The simulation results show that the feedback policy results in a lower CLR than the static policy. As expected the CLR decreases with the buffer size. For this scenario it was observed that aggressive reduction in the source rate resulted in a fast recovery from a congested state, however leading to underutilisation of the link as shown in Figure 7-5. The shown utilisation values when using feedback for congestion avoidance are unacceptable for the satellite environment. Having observed that the achieved mean utilisation becomes even worse with increased delay it was decided not to further study the BCN scheme.

Figure 7-5 Utilisation of the link capacity vs $T_L$ for various upper threshold values

Of final interest for the BCN scheme was the transient response when the load on the network suddenly changes. As observed by other researchers [KOLA96, SISA96] and shown in Figure 7-6 the source rates oscillate between a minimum rate (MCR) and a peak value that progressively decreases as the number of active sources increases. The shown ABR capacity values are multiples of the VBR peak rate. Hence for a VBR
Performance Evaluation of Congestion Control Schemes for OBP Satellites

peak rate of 64 kbit/s, an ABR capacity of 32 (half the total capacity) means 2 Mbit/s is shared between all ABR sources.

To summarise Figure 7-6, each congestion period is followed by an underload period which causes the control to increase the load again. The large satellite link propagation delays directly contribute to the amplitude of the oscillations in the source rate and queue fill. From this it can be conclude that a simpler EFCI based method will be adequate in a LAN environment where the propagation delays are small but they are not suitable for a GEO satellite environment since it may take several round-trip delays for the sources to reach the optimum transmission rate to maximise the throughput.

![Figure 7-6 Change in ABR capacity as a function of time (in seconds)](image)

**7.2.2 Reactive Control using Explicit Rate Notification**

BCN uses only a single bit to inform sources to reduce or increase their traffic. Although this scheme is particularly appealing for a satellite-based network because of its inherent broadcasting capability, it may take several round trips before the source will adjust to the right rate. This is unacceptable for satellite links with long propagation delays. A better strategy is for the on-board switch to send RM information to the stations which send RM cells to the source containing the rate it should change to. Various algorithms have been proposed for the calculation of the fair bandwidth share per connection [KALY97]. It has been shown that using the ER mechanism on terrestrial networks the control system performs well with respect to low buffer occupancy, fast convergence, fairness and relatively high throughput.
Few papers have looked at the performance improvements achievable for satellite networks [FAHM96].

In order to evaluate the improvement in performance in terms of CLR we use the simulation parameters in Table 7-1. The satellite is assumed to act as VS/VD to halve the feedback delay. Note that the propagation delay values in cell times can be found from Table 4-1 as a function of the service peak rate. According to the ERICA scheme [JAIN96] the utilisation of the resources is monitored for a certain 'averaging interval' and then the average utilisation is compared to the 'target utilisation' (u). The sources are send feedback in RM cells about the exact rate they should be sending at to obtain minimum cell loss. We also included a third parameter called 'utilisation margin' which was the allowed variation from the utilisation target before sources were informed to change their rate. The propagation delay to all earth stations was assumed identical. The persistent source was used as ABR source model and the on-off model for VBR sources.

<table>
<thead>
<tr>
<th>Simulation Parameters</th>
<th>Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>Target utilisation</td>
<td>0.85-0.9</td>
</tr>
<tr>
<td>Mean burst period</td>
<td>54, 216 cells</td>
</tr>
<tr>
<td>ABR&lt;sub&gt;PCC&lt;/sub&gt;, VBR&lt;sub&gt;PCC&lt;/sub&gt;</td>
<td>64-256 kbit/s</td>
</tr>
<tr>
<td>Link Capacity</td>
<td>2-16 Mbit/s</td>
</tr>
<tr>
<td>Number of VBR sources</td>
<td>75</td>
</tr>
<tr>
<td>Burstiness</td>
<td>2.85</td>
</tr>
<tr>
<td>Averaging Interval</td>
<td>2-20 cells</td>
</tr>
<tr>
<td>Buffer Size</td>
<td>1-200 cells</td>
</tr>
<tr>
<td>Satellite Propagation Delay</td>
<td>135ms</td>
</tr>
<tr>
<td>Utilisation Margin</td>
<td>0.01-0.05</td>
</tr>
</tbody>
</table>

Table 7-1 Simulation parameters

First for a VBR sources with a mean burst period of 54 cells the buffer size was fixed at 200 and the target utilisation at 0.9 to determine the optimum 'averaging interval' and 'utilisation margin'. Figure 7-7 (a) shows how the required backwards RM cells can be minimised by increasing the utilisation margin. However as the utilisation margin increases so does the CLR as shown in Figure 7-7 (b). The reason for the knee in the CLR for a utilisation margin of 0.035 is the fact that this value minimises feedback information but still is effective enough for congestion avoidance. Larger utilisation margin values reduce the congestion avoidance property for high loads as the system capacity can be reached before the feedback is received by the sources.
The averaging interval and utilisation margin which minimised the feedback information and CLR for different propagation delay values was found to be 5 and 0.035 respectively.

![Figure 7-7](image)

**Figure 7-7** (a) Feedback information (b) CLR vs utilisation margin for different averaging interval values.

Figure 7-8 shows the CLR of the on-board buffer as a function of BW*delay product (in cells). The simulation results show that the CLR increases with the BW*delay reaching the value of the static policy as the BW*delay exceeds the congestion duration (mean burst period). Hence reactive controls will be effective to control congestion at time scales of the bandwidth delay product and sustainable periods of congestion.

The measured mean utilisation were 0.83±0.12 and 0.87±0.13 for target utilisation values of 0.85 and 0.9 respectively. Higher $u$ means more link bandwidth assigned to each ABR connection as shown in Figure 7-9 (for a PCR of 128 kbit/s).

![Figure 7-8](image)

**Figure 7-8** CLR vs BW*Delay for a target utilisation of 0.85 and 0.9
Figure 7-9 ABR capacity as a function of time for (a) $u = 0.85$ and (b) $u = 0.9$

Figure 7-10 shows the CLR as a function of the buffer size for various target utilisation values ($u$). The bandwidth delay product was fixed at 38 cell times. As it can be seen the cell loss increases very sharply as the target utilisation increases.

Figure 7-10 CLR vs On-Board Buffer size for various values of $u$

The effect of the bandwidth delay product on the performance of the ER control is shown in Figure 7-11. The target utilisation is fixed at 0.9 and as the exceeds the mean burst period (and hence the congestion duration) the feedback control is not effective anymore.
Figure 7-11 CLR vs On-Board Buffer Size for various BW*delay values

To show the effects of the VBR mean burst period and hence the time of congestion on the performance of the ER control scheme, the VBR mean burst period was taken to be 216 cells.

Figure 7-12 CLR vs BW*Delay for a target utilisation of 0.9

Comparing Figure 7-8 and Figure 7-12 it can be clearly seen that the ER scheme is effective in reducing cell loss for much higher BW*delay values for the second scenario where the mean burst period is 216 cells. This is due to the fact that reactive feedback control schemes can only be effective at the propagation delay time scale. Hence if the temporary congestion duration is shorter than the propagation delay than reactive controls won't be effective. Therefore buffers to absorb burst scale congestion will be needed on-board the satellite, in particular for nrt-services, since the rt-services can always be scheduled on-time by using scheduling scheme such as the MSBS strategy (which was described in Chapter 4).
7.3 Summary

For the reactive control using BCN, it was observed that each congestion period is followed by an underload period which causes the control to increase the load again. The large satellite link propagation delays directly contribute to the amplitude of the oscillations in the source rate and queue fill. Furthermore the measured utilisation was lower than 70% for low cell loss which is unacceptable for the satellite environment. BCN based methods could be adequate in a LAN environment where the propagation delays are small and underutilisation of resources is not a problem but they are found to be not suitable for a GEO satellite environment.

In contrary to the BCN scheme the ER control is successful in achieving the specified ‘target utilisation’ value. The limiting factor on the target utilisation is the required buffer size to achieve low cell loss. In practice a target utilisation between 0.85 and 0.9 is achievable.

The ER scheme has been found to be most effective in reducing cell loss if:

- the propagation delay is low.
- the number of cells send before feedback is received is small (small terminal PCR-propagation delay product).
- the congestion duration is longer than the propagation delay from the point of congestion. If the congestion duration is shorter than the propagation delay than reactive controls won’t be effective.
8. CONCLUSIONS AND FURTHER WORK

For a satellite ATM network to efficiently support services with different QoS requirements, there must be loss and delay priorities associated with the Cell Loss Rate (CLR) and Cell Transfer Delay (CTD) parameters of the individual connections.

In order to provide the different delay and loss requirements for each service class and to maximise utilisation of resources, services were assigned delay and loss priorities according to their CLR and CTD requirements. The scheduling and discarding of cells at the ground terminal and on-board the satellite are done according to these priorities. A Multiple-Shared-Buffer-Scheduling (MSBS) policy, which considers both delay and loss priorities, was proposed and evaluated. It has been shown that both delay and loss requirements can be met by the MSBS scheme. On the other hand, scheduling schemes like Static Priority (SP) can only distinguish one class of priority. The complexity of MSBS using push-out is less than that of partial buffer sharing with nested thresholds, where lower cell loss rates for services sharing the same buffer might be obtained.

The performance of the Generic Cell Rate Algorithm (GCRA), or equivalent leaky bucket for policing the traffic entering the satellite network was evaluated. The aim was to police the declared traffic parameters accurately and with a short reaction time to violations of the parameters.

It was shown that the peak bit rate can be controlled relatively easily, setting the leak rate to the source peak rate with a small token buffer to account for the Cell Delay variation (CDV). Analytical expressions were derived for the reaction time, and the Maximum Burst Size (MBS) entering the network.

The performance of the Leaky Bucket (LB) without a smoothing buffer was evaluated for bounded and unbounded burst sizes. The numerical results for the LB without buffer and unbounded burst size show that a very high LB size must be used in order to sensitively detect violations of the mean rate (by setting the token generation rate near the mean rate). This implies that a very long time is necessary to detect a violation of the mean bit rate. On the other hand, if a small LB size is chosen for fast reaction, the token generation rate must be much higher than the mean bit rate, so that...
the source can increase its mean rate up to the token generation rate without any action being taken.

Considering this trade-off, and the fact that both sensitivity to detect violations of the mean bit rate, and fast reaction are necessary in most cases, a policing mechanism called the *triple leaky bucket mechanism* has been proposed. This mechanism consists of one LB policing the peak rate and two LBs policing the mean rate and thus the burstiness. The drawback of the triple-LB configuration is that tight control and fast reaction are provided, but not together. One LB provides fast reaction to larger increases in the mean whereas the other has slower reaction providing tight control.

The control of the Sustainable cell rate (the upper bound on the mean cell rate) when the burst size is bounded by the Maximum Burst Size (MBS) parameter is straightforward as policing the PCR. The problem is whether the user will be able to declare the MBS parameter.

In order to find an optimal solution for the trade-off between fast reaction and queuing delay, for the unbounded burst size, we continued the fluid-flow analysis of the LB with smoothing buffer. Closed-form expressions were derived for the mean reaction time and for the mean queuing delay of the buffered LB. The accuracy of both expressions has been verified using simulations. However, it has been found that the values calculated using these expressions have to be used with care, because the required QoS parameters (such as end-to-end delay) are strict upper bounds.

When considering the maximum delay introduced by the data buffer, it was realised that only small data buffers can be used for rt-services. However for nrt-services the use of a larger smoothing buffer seems to be very attractive in reducing the reaction time. The conclusions for the buffered LB are also applicable to sources with bounded burst size.

In order to control access to the satellite bandwidth resources and to extend the statistical multiplexing capabilities of ATM to the satellite air-interface a MAC scheme which can distinguish between service classes with different QoS is needed.
It has been shown that considerable improvements in delay performance and satellite bandwidth utilisation are possible if next-generation satellite technology (OBP) and an adaptive MAC protocol is used. The traditional demand-assignment scheme using a ground terminal as control station has two important drawbacks: reservation time and limited channel utility. Both are due to the long propagation delay of the satellite link. Both disadvantages can be removed by processing channel requests in the satellite to allocate frame slots. Furthermore, the problem of variable channel access delay of PRMA and its variants is solved by using control slots which are assigned to the VBR terminals.

The mapping of ATM service classes to MAC classes and the use of a prioritised request queue provides the QoS differentiation required by satellite ATM networks. It has been shown that a pure reservation system performs poorly for very bursty user traffic. The user population which can be supported using Random Access (RA) for very bursty users with short burst durations is much larger.

It was also shown that by using both variable-rate DAMA and RA, the utilisation of the frame capacity and the total number of users served is dramatically increased by statistically multiplexing traffic over the air interface. Therefore an adaptive protocol seems to be the best solution in a multi-service satellite ATM network, where different type of services with different characteristics have to be supported.

Finally, we proposed to improve the performance of the preventive traffic control scheme using feedback control. Since preventive control relies only on the UPC no action can be taken once congestion has occurred within the network. For this reason incorporation of a congestion control technique using feedback is proposed. The parameters of the LB are changed according to the state of congestion within the network, resulting in an adaptive LB. Two reactive control schemes, one using a single bit (BCN) and one using the exact rate the source should send at (ER) were compared with the static preventive control policy.

For the reactive control using BCN, the capacity utilisation obtained by using simulations was lower than 70% for low cell loss, which is unacceptable for the satellite environment. Thus BCN based methods were found to be unsuitable for a GEO satellite environment.
In contrast to the BCN scheme the ER control is successful in achieving the specified target utilisation value. The limiting factor on the target utilisation is the required buffer size to achieve low cell loss. In practice a target utilisation between 0.85 and 0.9 is achievable according the obtained simulation results. The ER scheme has been found to be most effective in reducing cell loss if the propagation delay is low, the number of cells send before feedback is received is small (small terminal PCR·propagation delay product) and the congestion duration is longer than the propagation delay from the point of congestion. If the congestion duration is shorter than the propagation delay than reactive controls won’t be effective.

The results of this thesis have been disseminated in various workshops, conferences, books and journals. Appendix E shows a list of publications as a result of this research. Contributions were also made to the PCP-LINK GIPSE and EU-ACTS WISDOM projects as well as to COST actions 253 and 257.

**Further Work**

Although the objectives of this research have been achieved, it would be interesting to see some of the aspects discussed in this thesis implemented in an experimental set-up. Involvement with the ACTS WISDOM project may provide an opportunity for this. Any experimental results can then be fed back to the simulation model for further performance optimisation and validation.

The inherent assumption in the analysis is the negative exponential distribution of burst sizes. Current research indicates that some traffic sources have self-similar behaviour, which is different from the Poisson distribution. Hence, the results must be validated for self-similar traffic. A simplifying assumption for most protocol studies was that channels are error-free. Also, the effect of fading on satellite channels has to be included in a more detailed analysis.

The Explicit Rate (ER) algorithm for reactive control which performs best under long propagation delay could be determined by comparing schemes with the best performance and smallest computational complexity, as all ER calculations must be performed in real-time.
Another interesting extension of this work is the evaluation of TCP/IP traffic throughput using GFR in a mixed service environment. The TCP/IP flow control and the ATM control mechanisms would have to be optimised to achieve a high throughput.
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A. SIMULATION MODELLING

The central issues in the design, implementation, and operation of communication networks are performance evaluation and trade-off analysis. The performance measures used to evaluate communication networks vary, depending on the type of network being analysed and the application. In this thesis the performance of a satellite ATM network to provide the required QoS to services is investigated.

The performance of a given network can be evaluated using direct measurements, analysis of mathematical models, and/or simulation.

A.1 Why use Simulation?

Measurements, though the most direct method of network performance evaluation, are very expensive and time consuming and can only be used on existing networks. Because an on-board processing satellite with cell switching capabilities is currently not available, there was no opportunity to make measurements on a satellite ATM network.

Mathematical analysis is the second approach that can be used for evaluating the network performance. The main advantages of analyses based on mathematical models are that they can produce results quickly and that they can provide considerable insight on the relationship between network performance and design parameters. Unfortunately, the development of tractable mathematical models requires many restrictive assumptions. It is generally impossible to develop closed-form solutions, except in idealised and often oversimplified cases. Nevertheless, analytical models can be effective and their validity can be checked by using simulation.

In many applications however, explicit performance evaluations defies closed-form mathematical analysis. When this fact is considered together with the expensive and time-consuming nature of physical measurements, simulations are often the best answer. Although simulation procedures can also be time-consuming, they are more accurate, since they are not based upon as many assumptions as analytical procedures. Furthermore, it is easier to incorporate empirical models and measured data into simulations.
A.2 Discrete-Event Simulation of Communication Networks

The simulation technique typically used to evaluate network performance is discrete-event simulation. In discrete-event simulation, various components of the actual network under study are represented in a computer program. The events that would occur during the actual operation of the network are mimicked during the execution of the program. The functions of the simulation program are to generate events and then simulate the network's response. The simulation also gathers data during the simulation and computes performance measures.

A.2.1 Time-Advance Mechanisms

Because of the dynamic nature of the discrete-event simulation models, the current value of the simulation time is recorded as the simulation proceeds and there is also a mechanism to advance simulation time from one value to another. The variable in a simulation model that gives the current value of simulation time is called simulation clock. The unit of time for the simulation clock is never stated explicitly when a model is written in a general-purpose language such as Fortran, Pascal or C and it is assumed to be in the same units as the input parameters.

Historically, two principal approaches have been suggested for advancing the simulation clock: next-event time advance and fixed-increment time advance. Note that the second is a special case of the first. The first approach is used by all major simulation languages and by most people coding their model in a general-purpose language. BONeS Designer also uses the first approach.

With the next-event time-advance approach, the simulation clock is initialised to zero and the times of occurrences of future events are determined. The simulation clock is then advanced to the time of occurrence of the first (most imminent) of these future events, at which point the state of the system is updated to account for the fact that an event has occurred, and the times of occurrences of future events is also updated. Then the simulation clock is advanced to the time of the new (most imminent) event, the state of the system is updated and future events are determined, etc. This process of advancing the simulation clock from one event time to another is continued until eventually some prespecified stopping condition is satisfied. Since all state changes
occur only at event times for a discrete-event simulation model, periods of inactivity are skipped over by jumping the clock from event time to event time. It should be noted that the successive jumps of the simulation clock are generally variable (or unequal) in size.

A.2.2 Components of a Discrete-Event Simulation Model

Although simulation has been applied to a great diversity of real-world systems, discrete-event simulation models all share a number of common components and there is a logical organisation for these components that promotes the coding, debugging and future changing of a simulation model’s computer program. In particular, the following components will be found in most discrete-event simulation models using the next-event time-advance approach:

*System state*: The collection of state variables necessary to describe the system state at a particular time.

*Simulation clock*: A variable giving the current value of simulated time.

*Event list*: A list containing the next time when each type of event will occur.

*Statistical counters*: Variables used for storing statistical information about system performance.

*Initialisation routine*: A subprogram to initialise the simulation model at time zero.

*Timing routine*: A subprogram that determines the next event from the event list and then advances the simulation clock to the time when that event is to occur.

*Event routine*: A subprogram that updates the system state when a particular type of event occurs (there is one event routine for each event type).

*Library routines*: A set of subprograms used to generate random observations from probability distributions that were determined as part of the simulation model.

*Report Generator*: A subprogram that computes estimates (from the statistical counters) of the desired measures of performance and produces a report when the simulation ends.

*Main program*: A subprogram that invokes the timing routine to determine the next event and then transfer control to the corresponding event routine to update the system state appropriately. The main program may also check for termination and invoke the report generator when the simulation is over.
A.3 Introduction to BONeS Designer

BONeS Designer provides a motif graphical environment for capturing the design (architecture) of communication networks and simulating the performance of the captured designs. The user specifies a network design by drawing a hierarchical block diagram with building blocks from the user-extendible BONeS Designer model library. The possibility of using a block diagram for simulation makes this simulation package a very attractive choice. The main advantage of BONeS compared to other simulation packages is the possibility of debugging all simulations in the interactive mode, where all variable and data structure values can be observed. This was the main reason to choose BONeS for the required simulations.

Modelling elements at the lowest level are called primitives; written in C, these accept data structures as inputs and performs simple operations such as modifying fields within the data structures and return data structures as outputs.

A typical BONeS Designer session consists of four steps:

1. Creating Data Structures
2. Constructing Block Diagrams
3. Running Simulations
4. Evaluating the Results

A.3.1 Data Structure Editor (DSE)

The data structures necessary to complete the specification of a BONeS Designer model are created using a graphical data structure editor (DSE). Data structures are defined hierarchically and can have an arbitrary number of fields. Fields can contain simple entities, such as the sequence number and time stamps. These data structures are defined to meet the needs of the simulation and do not necessarily duplicate the actual packet structure in the network being simulated.

The first step for a model design is, as explained, to create a data structure. Figure 3-1 shows the defined data structure to represent an ATM cell. Every simulation model in this section uses this DS but not all data fields have been used during the simulation.
Appendix A Simulation Modelling

<table>
<thead>
<tr>
<th>Name</th>
<th>Type</th>
<th>Subrange</th>
<th>Default Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sequence Number</td>
<td>INTEGER</td>
<td>[0, +Infinity)</td>
<td>0</td>
</tr>
<tr>
<td>Time Created</td>
<td>REAL</td>
<td>[0, +Infinity)</td>
<td>0.0</td>
</tr>
<tr>
<td>Traffic Type Identification</td>
<td>INTEGER</td>
<td>[0, +Infinity)</td>
<td>0</td>
</tr>
<tr>
<td>Traffic Source Identification</td>
<td>INTEGER</td>
<td>[0, +Infinity)</td>
<td>0</td>
</tr>
<tr>
<td>Priority</td>
<td>INTEGER</td>
<td>[0, +Infinity)</td>
<td>0</td>
</tr>
<tr>
<td>CDV</td>
<td>REAL</td>
<td>(0, +Infinity)</td>
<td>...</td>
</tr>
<tr>
<td>Delay Priority</td>
<td>INTEGER</td>
<td>[0, +Infinity)</td>
<td>...</td>
</tr>
</tbody>
</table>

Figure A-1 ATM cell data structure

A.3.2 Block Diagram Editor (BDE)

The network model is also constructed graphically, using the BONeS Designer *block diagram editor* (BDE). Primitives and other blocks are placed on the workstation screen and connected to form protocol functions. These functions are in turn grouped to form nodes, and nodes are then connected by communication links to form a topological network model. Hierarchy and some form of model aggregation are used to manage the complexity of models of large networks. The user specifies parameters of the individual blocks at some point prior to simulation. Measured characteristics of traffic and communication links can easily be incorporated into BONeS models.

A.3.3 Simulation Manager

Once the network model definition is complete, BONeS Designer performs a variety of error and consistency checking and automatically translates the graphical model of the network into a C program. An event-driven simulation of the network model is then executed, with user-specified values for model parameters. During the simulation, data structures created by source models flow along connection lines to various processing modules, which may alter the content of data structures and/or modify their path through the block diagram. Once the simulation has been started, the *simulation manager* provides status information about the simulation in a monitor window, detailing the current simulation clock time, stop time and whether error and warning messages have occurred. Eventually, data structures arrive at sink modules, where they are taken out of the system. A BONeS simulation continues until there are no more data structures in the block diagram or until the simulation clock reaches a user-specified stop time.
The simulation manager also provides the capability to record the sequence of execution of block diagrams during a simulation run. The recorded data can later be played back to show the execution of a model or to aid in debugging block diagrams.

A.3.4 Post Processor (PP)

During the simulation, BONeS Designer collects data at various points in the network using a variety of probes. The user specifies the location and type of probes. The BONeS Designer library manager takes care of the storage and retrieval of simulation models, programs, parameter values, and simulation data. The data collected during the simulation are analysed and displayed graphically using the post processor. The PP has built-in analytic functions for performing statistical operations on the data.

To display results using the PP, the user typically goes through the following steps:

- selecting simulation
- selecting probe(s)
- applying a conditional or filter to the data probe (not necessary)
- specifying the X- and Y-axis expressions

A.4 Verification of Simulation Results

The verification of the simulation results was the most time consuming task before starting the simulations. If the simulation model seemed to work right, after debugging using the animator of BONeS, a simple model available in the literature with certain source parameters was implemented to compare the results.

Obviously most of the simulations could not be verified since they were new contributions. Hence throughout this thesis it is mentioned where it was possible to compare the results with existing work.

A.5 Statistical Significance of the Results

The performance measures found using BONeS (or any other simulator) are random variables. It is necessary to make sure the results are statistically significant before meaningful conclusions can be drawn from simulation results. The confidence interval width is traditionally used to measure the significance of a simulation result.
There is an x percent chance that the true value of a parameter lies within the x percent confidence interval for the parameter. For example if a 95% confidence interval for cell loss rate is 0.0010±0.0005, then based on the data available, there is a 95% chance the mean cell loss rate lies between 0.005 and 0.0015. The width of the confidence interval reflects the statistical certainty of the estimator.

**A.5.1 Calculating Confidence Interval Widths**

A confidence interval is usually constructed from a set of $n$ unbiased estimates ($X_i$) of the parameter of interest ($\mu$). A confidence interval is usually constructed on the assumptions that the $n$ (where $n \geq 2$) samples are independent and come from the same normal distribution. The confidence interval can be found because $(\mu - X_i)/\sigma_{x_i}$ has a distribution with $n-1$ degrees of freedom. If $n$ is fairly large ($n \geq 10$) then $(\mu - X_i)/\sigma_{x_i}$ can be assumed to be normal with little loss of accuracy.

The requirement for the samples ($X_i$) to be normal and from the same distribution is satisfied if the $X_i$s are each means calculated during a BONeS simulation. The use of the same system and parameters (except for random number seeds) insures the samples come from the same distribution. The Central Limit Theorem suggests that $X_i$s will be normal if they are calculated by adding many random variables (i.e. taking a mean).

Successive samples of network performance measures are rarely independent. There are three commonly applied techniques for obtaining confidence intervals of network performance:

- Independent replication of the simulation
- Batch means
- Regenerative Sampling

**Independent Replication**

The simplest way to obtain independent $X_i$s to iterate the simulation over the **global seed** parameter to provide several completely independent simulation runs. If we are interested in the steady state value of the parameters, it is necessary to let the simulation run until steady state is achieved before gathering statistics. A common choice (which
Appendix A Simulation Modelling

has no theoretical foundation) is to spend 10% of the simulation time in set-up. More set-up time is required for a system with heavy traffic and long queues.

The simplest way to use independent replications in BONEs is to use the batch mean probe with the number of batches parameter set to 1. This produces the mean value for a statistics for the entire simulation iteration (except for the setup time). The simulation is iterated over the global seed parameter to produce independent samples. Ten iterations over global seed is a rule of thumb requirement for normal distribution of samples to hold in practice.

**Batch Means**

The method of batch means breaks a single long simulation run into batches, which are non-overlapping segments of time during the simulation. Separate averages, \( X_i \), are calculated for each of the \( n \) batches. In a stable network, the covariance between samples decreases as the time between samples increases. If the time between samples is large enough, the samples should have almost no covariance. The batches must be long enough that the means from successive batches have little covariance. Batches which are too short lead to positively correlated \( X_i \)s which cause the confidence interval width to be underestimated.

It is necessary to wait until the system has reached steady state before beginning the sampling. The batch mean technique is only applicable to steady state performance.

**Regenerative Sampling**

Any stable, non-degenerate queuing model will visit a given state (usually the empty state) repeatedly. The visits to this state are called regeneration points and the time between visits are called regenerative cycles. Statistics for successive regenerative cycles are independent.

The regenerative method has a firmer theoretical basis than batch means, but is much more difficult to apply in practice. The regeneration points in a complex network can be so far apart that it can be impossible to simulate an entire cycle. Applying the regenerative method using BONEs requires that the model have a mechanism to detect the visit to the regeneration state. This method can only be applied to specific models in
BONeS and no general support for this method is available in the Post Processor or model library.

A.5.2 Results and Discussion

For the simulation results obtained, both the independent replication and the batch mean method have been used to determine the confidence interval. First the batch mean method has been applied but the results were not always satisfactory since the difference between batches was too large. The problem was to determine the length of each batch in order to have no covariance between batches and thus independent samples. The procedure of obtaining the time necessary between batches to obtain independent samples was time consuming and therefore this method was not appropriate.

![Graph showing confidence intervals](image)

**Figure A-2 90% Confidence Interval for Iterations with different Global Seeds for (a) Blocking Probability (b) Reservation Delay**

The results obtained by the independent replication method, which was used next, were satisfactory. The global seed was iterated 5 times to obtain 5 independent measurements. The simulation time was simply adjusted to 10000 bursts of the source to reach the steady state with 10% of the simulation time in set-up. Figure 3-2 shows the 90% confidence interval plots of the blocking probability and reservation delay for the CBR reservation MAC system. Because of the vast number of measurements done it is impossible to show or indicate the confidence interval of each measurement. However the measurements of various replicas were generally within the 90-95% confidence interval.
Appendix B Simulation Models

B. Simulation Models

B.1 Traffic Source Simulation Models

B.1.1 The On-Off Model

As explained in chapter 3 the on-off model has burst and silence periods which are exponential distributed. The simulation model shown in Figure B-1 uses the ‘TNow>=Burst Length ?new’ block to produce an exponential distributed burst with a mean duration specified by the user by the parameter \textit{mean burst period}. The simulation model works in a way that at simulation time 0 the ‘Init’ block generates a pulse. The following block ‘TNow >=Param?’ checks if the simulation has reached its end, by comparing \textit{TNow} (the current simulation time) to the stop time when the simulation ends. If the simulation has not reached its end the False (F) output of this module triggers the ‘execute in order (EIO)’ module. This module caused the most problems during the simulation model design because of its time-advance mechanism for paths, therefore a full description of it is given below:

The ‘execute in order’ module is used to order the execution of modules in a block diagram. The paths are executed as far as possible in order of their output numbers. The outputs are simply copies of the input. Note that there is no simulation time delay between the time that this module receives its input and the time that its output paths begin execution. This module simply orders the execution of the output paths as follows:

As many of the path branches as possible along the first output path are executed before any of those an the next path are executed. There are a number of conditions along the output path that would cause the execution of the next output path to begin.

The simplest case is when the entire output path and all branches along it have been executed. Typically, the path and its branches will end with a ‘Sink’ module or some other terminator module. At this point, the next output path begins execution.

Another case is when the path leads to a delay module. Execution continues only up to the delay module. If there are no other branches that can be executed in this output path, then the next output path begins execution. A third case is when a path leads to a
module that requires more than one input, and all of those inputs are not currently available. If there are no other branches that can be executed in this output path, then the next output path begins execution.

The 'execute in order' module can be useful when more than one operation is to be performed at the same simulation time, and the order of the operations is important.

Figure B-1 The On-Off Simulation Model

The first output of the 'EIO' module triggers the reset input of the 'TNow>=BurstLength ? new' module shown in Figure B-2. Then first a exponential number with a mean specified by the argument Mean Burst Period is generated and fed into an accumulator. Second the current simulation timer TNow is also fed to the accumulator where it is added to the exponential number and stored. Now the second output of the 'EIO' module triggers the Start input of the 'TNow>=Burst Length ? new' module. This input is connected to three modules which are activated at the same time. The accumulator output is enabled and the stored content is fed into the 'R>=' comparator module. At the same time the other input of the comparator is fed with the current simulation time. If the current simulation time has not reached the time (burst started+burst duration) then a logic zero is fed into the switch which outputs a trigger to the false output. If the burst duration end is reached a logic 1 is fed to the switch which triggers the true output.

If the burst duration has not ended the false output of the 'TNow>=BurstLength ? new' module sends a trigger to the second 'EIO' module which generates a trigger to generate an ATM cell and then executes the 'fixed abs. Delay' module. This delay module produces a gap between ATM cells with a user specified parameter interarrival time. Again the Start input of the 'TNow>=Burst Length ? new' module
is triggered and the current simulation time is compared with the time (burst started + exp. burst duration). This loop is executed till the burst duration has ended. Then the True (T) output of this module enables the ‘abs delay’ module whose delay time is specified by the ‘exponential range parameter’ module with a mean determined by the user as mean silence period. After this exponential silence duration, TNow is compared with the simulation stop time. This loop is executed till TStop has been reached.

**Figure B-2 ‘TNow>=Burst Length ? new’ Module**

**B.1.2 IPP Simulation Model**

The only difference of the IPP model from the on-off model is the interarrival time of cells, which is exponentially distributed with a parameter mean interarrival time specified by the user, instead of being fixed (Figure B-3).

**Figure B-3 IPP Simulation Model**
B.1.3 MMPP Simulation Model

As explained in chapter 3, the IPP is a special case of the MMPP with one silence period. Now the MMPP with two states has no silence duration, but an exponential distributed $2^{nd}$ state duration specified by the user with parameter mean burst period $2$, which has an exponential distributed cell interarrival time specified also by the user with parameter mean interarrival time $2$ (Figure B-4).

![MMPP Simulation Model](image)

Figure B-4 MMPP Simulation Model

B.1.4 Birth-Death Simulation Model

The birth-death process can be used to model multiplexed homogeneous on-off sources and was described in chapter 3. The transition rate probabilities from state $i$ to a higher state is $(N-i)/t_{\text{off}}$ and the probability to a lower state is $i \cdot t_{\text{on}}$, where $N$ is the number of sources. The simulation model first executes a loop to calculate the smallest burst duration and silence duration for the $N$ sources. Then the probabilities for transition to a lower or higher state are calculated. According to the result the current state is updated. The accuracy of this model has also been compared with analytical results.
Figure B-5  N-state birth-death process simulation model
B.2 Static Loss Priority Scheduling Simulation Model

Static loss priority scheduling can only distinguish one class of priority. Here the cell loss priority is used to distinguish the scheduling priority between service classes. First the loss priority field of the arriving cell is compared to a constant C. If the arriving cell has priority class 4 (HP rt-VBR) then this cell is only admitted into the buffer if the number of HP rt-cells did not exceed the threshold for this service class. This is done to limit the number of cells of this HP service class. The number of class 4 cells entering and exiting the queue is determined by means of a up/down counter module.

If the arriving cell has loss priority class 2 (LP rt-VBR) than again the cell is only admitted if the buffer contents did not exceed the HP-rt-VBR threshold in order to upper bound the number of rt-cells in the queue. The number of class 2 cells entering and exiting the buffer are not taken into account using the HP rt-cell counter. The reason for this is the fact that class 2 cells get always served after class 3 cells. Hence if class 2 cells are counted, then after some time class 4 cells won’t be admitted because the rt cell limit has been reached. The scheduling of cells according to the CLP and the threshold are illustrated below in Figure B-7. If no class 4 cells are
admitted then class 3 gets always served first and the CLR of class 4 will be very high.

<table>
<thead>
<tr>
<th>LP-nrt</th>
<th>LP-rt</th>
<th>HP-nrt</th>
<th>HP-rt</th>
</tr>
</thead>
<tbody>
<tr>
<td>CLP=1</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>CLP=2</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>CLP=3</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>CLP=4</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Figure B-7** Scheduling according to loss priorities with push-out

The 'stats module pri2' collects statistics such as the number of generated and lost cells of each service class. These are used for plotting the CLR graphs. The delay probes are placed at the output of the FIFO queue with priority (shown in Figure B-6). A generic probe was placed inside the queue module to collect the final queue statistics.

**Figure B-8** Statistics module collecting the number of generated and lost cells
B.3 Multiple Shared Buffer Scheduling (MSBS) Simulation Model

According to the MSBS policy, service classes with similar delay requirements and different loss requirements are queued in the same buffer. Cells are discarded according to their loss priorities and served according to their delay priorities.

In this simulation model two delay priorities (real-time and non-real-time) are considered, hence two buffers are used. Real-time services are queued in the rt-buffer and nrt-services in the nrt-buffer. The rt-buffer is served according to the selected rt-cycle.

Figure B-9 MSBS simulation model

Figure B-10 FIFO with priority with reject and check on queue contents
Appendix B Simulation Models

If there is no cell in the rt-queue then the nrt-buffer is served. In order to check whether the queue is empty during a serving cycle the FIFO with priority with reject module had to be modified. The modified module is shown in Figure B-10. The modification of the FIFO queue module consists of an addition of a ‘get queue length’ module which verifies if the buffer is empty, by comparing the queue length with zero (I==C? module). If the buffer is empty than the other buffer gets served. The MSBS policy can be summarised with the flow-chart shown in Figure B-11. After the relevant buffer has been served, the current subcycle status has to be compared with the rt and nrt-subcycle in order to determine which buffer has to be served next.

Figure B-11  Flow-chart describing the MSBS policy

Again the number of generated and lost cell per service class statistics were collected using the ‘stats module pri’ which is shown in Figure B-12. The delay statistics of each service class were collected using the delay probes placed at the output port of the relevant buffer (as shown in Figure B-9). Furthermore a generic probe has been placed in each of the FIFO queues to collect the final queue statistics (shown in Figure B-10).
Figure B-12  Statistics module collecting the number of generated and lost cells
B.4 Leaky Bucket Simulation Model

The leaky bucket, shown in Figure B-13 is simply a counter which increases with the number of cell arriving (each arriving cell adds one token to the bucket). The upper bound of the threshold is the LB size. The counter (bucket) is decremented with the leaky rate (user specified parameter *token generation interarrival time*). If cells arrive and the counter didn’t reach the threshold, they can enter the network. If the counter has reached its limit, cells arriving are non-conforming.

Figure B-13  Leaky Bucket Simulation Model

The arriving cell first comes to the 'Execute in order' block. This block controls the sequence of processes which happen at the same time. First the LB token buffer size (threshold) is fed to a comparator. Then the counter is incremented by one. Next the output trigger is enabled to also feed the counter contents to the comparator. If the threshold is greater than the counter content, the cell can pass to the output. Otherwise no permission (trigger) will be given to the cell. Tagging, dropping or buffering are possible actions to be taken for non-conforming cells. It has to be pointed out that the counter can’t exceed a user specified threshold with parameter *Leaky Bucket Size* and can not be less than zero. This has been done using the second comparator which compares the counter with zero. If it is less than zero the counter will be reset to its reset value 0. In a similar way if the counter exceeds the LB threshold it will simply be decreased to the threshold value by feedback.

Preventive Control Simulation Model using the Leaky Bucket

Figure B-14 shows a complete preventive control mechanism consisting of a source a policing device (the leaky bucket) and a queue from which the cells access the
network. The principle of this mechanism is that cells can access the network only if the leaky bucket sends a trigger (control signal) to the queue. If the LB is full the cell has to wait in the queue till the LB counter drops below the threshold. In the case when the queue is full, the arriving cell is simply rejected (dropped). This model was used to determine the performance of the LB as policing device. The results for violations of the negotiated parameters have been shown in chapter 5. For the unbuffered LB performance the FIFO size was chosen zero.

![Figure B-14 Preventive control simulation model using LB](image-url)

Figure B-14 Preventive control simulation model using LB
B.5 MAC Simulation Models

B.5.1 CBR Simulation Model

The TDMA MAC system build for the study is broken down into two portions:
- the action of the terminals
- the action of the on-board radio resource management (OBRRM) block

These two portions are presented by two modules called ‘req. allocation’ and ‘Terminal’, shown in Figure B-18 and Figure B-20 respectively. The two probes on the Resource memory collect statistics about the mean occupancy of the resources and the mean buffer occupancy if requests are queued.

Two different CBR terminal modules have been designed for the simulations. The terminal module without retransmission is shown in Figure B-17. Arrivals are modelled by a Poisson pulse train, where the mean interarrival time of call arrivals is specified by the Mean call arrival (per minute) parameter. Each trigger of the ‘Poisson pulse train’ module generates a TDMA Data Structure (DS) shown in Figure B-16. The fields in this data structure are used to collect various statistics such as delay and requested PCR. Note that the requested PCR of the service was stored in the message length field.
The time of the request, the requested PCR and the chosen RA row slot are stored in the relevant fields of the TDMA DS. The time of reservation field of the DS is assigned the value zero, for reasons which will be explained later. All DS are sent to the OBRRM module via the propagation delay module.

TDMA DS which return from the OBRRM module are first checked for their time of reservation. If the time of reservation field in the DS is still zero than no reservation has been made for this request and this means that either no capacity was available or that a collision has occurred. In the terminal shown in Figure B-17 unsuccessful requests are simply dropped. However in the terminal module shown in Figure B-18, unsuccessful requests are retransmitted after a exponentially distributed random time proportional to the mean retransmit waiting time parameter. Furthermore a new random access row slot is selected.

Successful request should be cleared after the mean call holding time which is assumed to have negative exponential distribution. This is done in the ‘Terminal’ module taking into account the propagation delay. The freeing of resources should have been done in the OBRRM module, however to simplify the model and reduce traffic between the modules this approach has been used (clearing the resources within the Terminal module). Two probes in this module store the mean reservation delay and the number of generated calls.

**Name**: TDMA DS [Example: TDMA]  
**Date**: Thursday, 5/21/98 07:59:41 pm GMT

<table>
<thead>
<tr>
<th>Name</th>
<th>Type</th>
<th>Subrange</th>
<th>Default Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Terminal ID</td>
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<td>0</td>
</tr>
<tr>
<td>RowSlot Requested</td>
<td>INTEGER</td>
<td>[0, +Infinity)</td>
<td>0</td>
</tr>
<tr>
<td>Message Length</td>
<td>INTEGER</td>
<td>[0, +Infinity)</td>
<td>0</td>
</tr>
<tr>
<td>Time of Creation</td>
<td>REAL</td>
<td>[0, +Infinity)</td>
<td>0.0</td>
</tr>
<tr>
<td>Time of Reservation</td>
<td>REAL</td>
<td>[0, +Infinity)</td>
<td>0.0</td>
</tr>
<tr>
<td>Time of Completion</td>
<td>REAL</td>
<td>[0, +Infinity)</td>
<td>0.0</td>
</tr>
</tbody>
</table>

**Figure B-16  TDMA Data Structure**
Figure B-17 Simulation model for ground terminals without retransmission

Figure B-18 Simulation model for ground terminals with retransmission

The satellite on-board radio resource management (OBRRM) module is shown in Figure B-20. Each requests for capacity is first stored in a queue while a vector counter, counts the number of requests per reservation slot. At the end of each frame period, all requests are released (triggered by the 'uniform pulse train' module with interarrival parameter frame period). If the number of requests per reservation slot is more than one, then a collision is assumed and the request is send back to the
‘Terminal’ module. The number of collisions counter is incremented by one for each colliding request.

If there was no collision then the required PCR of the request is determined. If the capacity is available it is assigned to the terminal and the time of reservation field is stamped. If no capacity is available the request can either be queued or rejected without reservation. In both cases the DS is sent back to the ‘Terminal’ module. The number of blockings and total requests (including retransmission) are stored in counters. At the end of the simulation time the TDMA Stats new DS, shown in Figure B-19 is created. This statistic DS will be used for most MAC simulations. The number of total requests, collisions and blocking are written in the relevant fields.

Name: TDMA stats new [TDMAnew]
Date: Thursday, 5/21/98 09:04:33 pm GMT

<table>
<thead>
<tr>
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<th>Type</th>
<th>Subrange</th>
<th>Default Value</th>
</tr>
</thead>
<tbody>
<tr>
<td># of total requests</td>
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<td>(0, +Infinity)</td>
<td>...</td>
</tr>
<tr>
<td># of Blockings</td>
<td>INTEGER</td>
<td>(0, +Infinity)</td>
<td>...</td>
</tr>
<tr>
<td># of Collisions</td>
<td>INTEGER</td>
<td>(0, +Infinity)</td>
<td>...</td>
</tr>
<tr>
<td># of VBR requests</td>
<td>INTEGER</td>
<td>(0, +Infinity)</td>
<td>...</td>
</tr>
<tr>
<td># of UBR Requests</td>
<td>INTEGER</td>
<td>[0, +Infinity)</td>
<td>...</td>
</tr>
</tbody>
</table>

Figure B-19  TDMA Stats Data Structure

Figure B-20  OBRRM module simulation module for CBR analysis
B.5.2 VBR Simulation Model

Initially the simulation model shown in Figure B-21 has been used to evaluate the performance of a MAC scheme where an effective bandwidth is allocated to a VBR source. The source model outputs cells which are stored in a buffer. The buffer is triggered according to the bandwidth assigned to the source. At the output of the queue statistics such as mean and maximum reservation delay are collected.

![VBR simulation model with effective bandwidth allocation](image)

**Figure B-21**  VBR simulation model with effective bandwidth allocation

The VBR MAC simulation model using burst reservation is broken down into two main portions (as the CBR model). The actions of the VBR terminals and the actions of the OBRRM module. There are two modules to represent the activity of the VBR sources. Figure B-23 shows the module which generates the initial VBR requests,

![MAC simulation model for VBR analysis](image)

**Figure B-22**  MAC simulation model for VBR analysis
which are send to the Reservation slots using random access (RA). This module is nearly identical to the ‘Terminal’ module of the CBR system. The only difference is that the call holding time has been set to the burst period and the addition of a ‘Insert Terminal ID’ module which is used in the ‘OBRRM’ module to distinguish initial and burst reservation requests.

**Figure B-23 Initial terminal request to access Reservation slots using RA**

The module shown in Figure B-24 is generating the burst reservations of the active terminals. The beginning and end of burst is generated using an ‘N-state birth-death process’ module which is shown in Figure B-25. The burst reservation request for more bandwidth is delayed by the maximum time it takes for a control slot to poll the terminal. This may be more than one frame time depending on the number of inactive stations.
Appendix B Simulation Models

Figure B-24  Burst reservation of VBR terminals using Control slots

Figure B-25  Birth-death process generating beginning and end of burst

The birth-death process was already described and hence the description of Figure B-25 will not be repeated here. It has to be noted that when a source becomes active a burst reservation request is triggered and when a source becomes inactive a resource release request is triggered.
The OBRRM module shown in Figure B-26 is used to allocate bandwidth to both CBR and VBR requests. First the comparison module checks whether the incoming DS is an initial request using Reservation slots or a Burst reservation using Control slots. For initial requests the OBRRM module behaves as described for the CBR case. If the arriving DS is a burst reservation than the request is put into the prioritised queue for bandwidth allocation. A threshold is also implemented for requests which cannot be queued for a long time. All the request statistics are collected in the statistics module shown in Figure B-27. The collected statistics are stored in the TDMA Stats DS which is shown in Figure B-19.
Appendix B Simulation Models

Figure B-27 Statistics module to collect VBR and CBR statistics

B.5.3 CBR+VBR Simulation Model

Figure B-28 Simulation model for CBR+VBR analysis

The simulation model shown in Figure B-28 consists of modules which were described so far for the CBR and VBR simulations. They are put together where CBR
and VBR requests can compete for resources. The OBRRM module (Figure B-26) does all the bandwidth allocation and statistics are collected in the statistics module (Figure B-27).

### B.5.4 CBR+VBR+UBR Simulation Model

#### Figure B-29 Simulation Model for CBR+VBR+UBR analysis

The simulation model shown in Figure B-29 consist of the terminal modules for the different service classes and a modified version of the ‘OBRRM’ module. The ‘UBR terminal request’ and ‘UBR Terminals’ modules are nearly identical to the ‘VBR terminal request’ and ‘VBR Terminals’ modules. The difference are the different parameter names and terminal ID’s which are inserted into the TDMA DS. The modified ‘OBRRM’ module for CBR+VBR+UBR reservations is shown in Figure B-30. The only additions are modules to distinguish the different Terminal ID fields used for different service classes. This way the CBR, VBR and UBR requests can be distinguished.
Figure B-30 OBRRM module for CBR+VBR+UBR analysis

B.5.5 UBR Random Access simulation model

For the scenario where CBR and VBR use reservation and UBR uses RA a clever modification has been made to the simulation model. As stated before, reservation slots are used for the initial access in random access mode and the ‘OBRRM’ module checks if more than one terminal selected the same slot. This part of the ‘OBRRM’ has been reused to determine whether the ‘UBR terminals’ module generating RA request had a successful reservation or collision. The ‘UBR terminals’ module is nearly identical to the ‘VBR terminals’ module with different parameters and a different terminal ID.


B.6 Reactive Control Simulation Models

B.6.1 Reactive Control Simulation Model using BCN

For reactive control using BCN, congestion is detected by monitoring the number of active sources. If the active sources exceed a certain threshold the ABR rate is reduced by the multiplicative decrease factor. On the other hand if the arrival drops below a second lower threshold the ABR rate is increased by the additive increase factor. The simulation model is shown in Figure B-31. The ‘Uniform Pulse Train’ determines the observation interval. The number of cells generated and cells lost are stored in counters which are triggered at the end of the simulation time to calculate the CLR. Two probes are placed on the shared ‘Resource’ to obtain buffer occupancy and resource utilisation statistics.

![Figure B-31 Congestion Control Model using the number of active sources to detect onset of congestion and BCN to notify sources.](image)

B.6.2 Reactive Control Simulation Model using ER control

The simulation model using ER control is very similar to the model using BCN. There are two main differences. First, congestion is detected by monitoring the utilisation for a certain time interval. If the average utilisation exceeds the target utilisation and the utilisation margin then congestion onset is detected. The sources are not simply notified to reduce their activity, but they are told the exact rate they should be sending at. The simulation model is shown in Figure B-32. The ‘QR Fractional Utilisation rt’
module calculates the mean utilisation each averaging interval. The mean utilisation is then compared to the target utilisation. If the difference is more than the utilisation margin then the new rate of each ABR source is determined. In order to observe the change in the ABR rate and mean utilisation, probes were placed so that samples were taken after each averaging interval. Furthermore CLE and buffer occupancy statistics were collected.

Figure B-32 Congestion Control Model monitoring average utilisation to detect onset of congestion and ER to notify sources.
Appendix C Connection Admission Control Algorithms

C. Connection Admission Control Algorithms

C.1 The Fluid Flow Model

In the physical model of [ANIC82] a buffer receives cells from a finite number of statistically independent and identical traffic sources that asynchronously alternate between exponentially distributed periods in the 'on' and 'off' states. In the on state the source transmits at a uniform rate. The buffer depletes uniformly with the allocated bandwidth rate of $BW$ bits/s.

The source model differs from the on-off source model presented in chapter 3 only in the assumption of uniform generation and transmission of information. Here information is considered to have a continuous nature, as if it were a fluid; there is no discretization of information in cells. Similarly, service is carried out in a uniform manner, not on a per cell basis.

Without loss of generality, the unit of time is selected to be the average on period $(t_{on})$. With this unit of time, the average off period is denoted by $1/\lambda$. That means

$$t_{off} = t_{on} / \lambda$$

$$\lambda = \alpha / (1 - \alpha)$$

where $\alpha = 1/\beta = \text{mean rate/peak rate} = t_{on} / (t_{on} + t_{off})$ denotes the on-state time fraction.

Again, without loss of generality, the unit of information is chosen to be the amount generated by a source in an average on period. Thus, at the on state, the source transmits at the uniform rate of 1 unit of information per unit of time. The output link capacity is $C = BW / p$ units of information per unit of time where $p$ is the peak rate of the source. The buffer size of $B$ cells is converted to a buffer size of $x = B \cdot 424 / (p \cdot t_{on})$ units of information.

First let us find the overflow probability of the buffer, in order to calculate the cell loss probability.

Now, if at time $t$ the number of on sources equals $i$, two elementary events can take place during the next interval $\Delta t$, a new source can start or a source can turn off. Since the on and off periods are exponentially distributed, the respective probabilities are $(N-i)\lambda\Delta t$ and $i\Delta t$. 

C-1
Appendix C Connection Admission Control Algorithms

The differential equations governing the equilibrium buffer distribution can be derived as follows: If \( P_i(t,x), \ 0 \leq i \leq N, \ t \geq 0, \ x \geq 0 \) is the probability that, at a time \( t \), \( i \) sources are on and the buffer contents does not exceed \( x \), then

\[
P_i(t + \Delta t, x) = (N - (i - 1)) \lambda \Delta t P_{i-1}(t, x) + (i + 1) \Delta t P_{i+1}(t, x) + [1 - ((N - i) \lambda + i) \Delta t] P_i(t, x - (i - C) \Delta t) + O(\Delta t^2)
\]  

(C.1)

These equations merely reflect the fact that only transitions between adjacent states are allowed. Passing to the limit de \( t \to 0 \) we obtain:

\[
\frac{\partial P_i}{\partial t} + (i - C) \frac{\partial P_i}{\partial x} = (N - i + 1) \lambda P_{i-1} - ((N - i) \lambda + i) P_i + (i + 1) P_{i+1}
\]  

(C.2)

Our interest is only in time-independent, equilibrium probabilities. Therefore we define \( F_i(x) \) as the equilibrium probability that \( i \) sources are on and the buffer content does not exceed \( x \), and set \( \partial P_i/\partial t = 0 \) to obtain the following set of equations:

\[
(i - C) \frac{dF_i}{dx} = (N - i + 1) \lambda F_{i-1} - ((N - i) \lambda + i) F_i + (i + 1) F_{i+1}, \ 0 \leq i \leq N
\]  

(C.3)

In matrix notation,

\[
D \frac{d}{dx} F(x) = M F(x), \ x \geq 0
\]  

(C.4)

where \( D = \text{diag}(-C, 1-C, 2-C, \ldots, N-C) \)

Let

\[
G(x) = \text{Pr}(\text{buffer content} > x) = 1 - \sum_{j=0}^{N} F_j(x), \ x \geq 0
\]  

(C.5)

\( G(x) \) is the probability of overflow beyond \( x \).

The procedure of [ANIC82] to solve the differential equations in (C.4) to obtain the equilibrium probabilities \( F(x) \) and the probability of overflow \( G(x) \) is summarized below.

The procedure is based on using the expression:

\[
F_j(x) = F_j(\infty) + \sum_{i=0}^{N-C-1} e^{zi \Delta t} a_i \phi_i
\]  

(C.6)

\( F_j(\infty) \) is the probability that \( j \) out of \( N \) sources are on simultaneously.
Appendix C Connection Admission Control Algorithms

Obviously,

\[ \sum_{j=0}^{N} F_j(\infty) = 1 \quad (C.7) \]

\[ \sum_{j=0}^{N} F_j(x) = \sum_{j=0}^{N} F_j(\infty) + \sum_{j=0}^{N} \sum_{i=0}^{N-C-1} e^{xi} a_i(\Phi_i)_j \quad (C.8) \]

Substituting (C.7) in (C.8) yields:

\[ G(x) = - \sum_{j=0}^{N} \sum_{i=0}^{N-C-1} e^{xi} a_i(\Phi_i)_j \quad (C.9) \]

\( z_i \) are the negative eigenvalues of \( D^1M \) and \( \Phi_i \) the associated eigenvectors. \( (\Phi_i)_j \) is the \( j^{th} \) component of the \( i^{th} \) eigenvector. Now, in [ANIC82] there is a explicit formula for the calculation of stable or negative eigenvalues:

\[ z_i = \frac{-B(i) + \sqrt{B^2(i) - 4A(i)C(i)}}{2A(i)} \quad (C.10) \]

where

\[ A(i) = (N/2 - i)^2 - (N/2 - C)^2 \]
\[ B(i) = 2(1 - \lambda)(N/2 - i)^2 - N(1 + \lambda)(N/2 - C) \]
\[ C(i) = -(1 + \lambda)^2((N/2)^2 - (N/2 - i)^2) \]

The \( j^{th} \) component of the \( i^{th} \) eigenvector is calculated as follows:

\[ (\Phi_i)_j = (-1)^N \sum_{m=0}^{N-j} \binom{N}{j} \binom{N-i}{j-m} r_{ji}^{j-i+m} r_{2i}^{N+i+j+m} \quad (C.11) \]

where

\[ r_{ji} = -\frac{(z_i + 1 - \lambda) + \sqrt{(z_i + 1 - \lambda)^2 + 4\lambda}}{2\lambda} \quad (C.12a) \]
\[ r_{2i} = -\frac{(z_i + 1 - \lambda) - \sqrt{(z_i + 1 - \lambda)^2 + 4\lambda}}{2\lambda} \quad (C.12b) \]

Finally, the coefficients \( a_i \) in the solution expression are obtained from:

\[ a_i = -\left(\frac{\lambda}{1 + \lambda}\right)^N \prod_{n=0}^{N-C-1} \frac{z_n}{z_n - z_i}, \quad n \neq i, \quad 0 \leq i \leq N - C - 1 \quad (C.13) \]

Having developed the analytical formula for buffer overflow, the ATM cell loss probability can easily be obtained. Considering that information is lost whenever the
incoming flow from the $i$ sources being at the on state exceeds the service rate $C$ and the buffer is filled beyond $x$, the actual buffer size. Hence

$$CLR = \sum_{j=C+1}^{N} (j - C)(F_j(\infty) - F_j(x))$$  \hspace{1cm} (C.14)

$$CLR = \sum_{j=C+1}^{N} (j - C)(- \sum_{i=0}^{N-C-1} e^{ix} a_i(\phi_i)_j)$$  \hspace{1cm} (C.15)

This formula can be used to calculate the cell loss rate (CLR) resulting from the superposition of $N$ identical ATM sources on an ATM link. A BASIC program used to calculate the bandwidth per source is given below. The formula has the advantage of taking into consideration all of the system parameters, including the buffer size of the multiplexer queue as well as the burst length of the multiplexed sources. For these reasons it performs better than the approximate flow model.

Program to calculate the CLR for $N$ homogeneous multiplexed on-off sources

```
DIM Z(200),A(200),RA(200),RB(200),PHI(200)
SO=0
INPUT "NUMBER OF SOURCES";N
INPUT "LAMBDA";LA
INPUT "ton";ton
INPUT "PEAK BIT RATE OF SOURCE";P
INPUT "BUFFER SIZE (IN CELLS)";NO
INPUT "EFFECTIVE BANDWIDTH ALLOCATED TO SOURCES";BW
C = BW / P; C = INT(C)
X = NO * 424 / (P*ton)
V = N - C - 1
FOR K = 0 TO V
  A = (N/2-K) ^ 2 - (N/2-C) ^ 2
  B = 2 * (1-LA) * (N/2-K) ^ 2 - N * (1+LA) * (N/2-C)
  CK = -(1+LA) ^ 2 * ( (N/2) ^ 2 - (N/2-K) ^ 2 )
  Z(K) = (- B - SQR (B ^ 2-4*A*C) ) / (2*A)
  RA(K) = ( ( -(Z(K)+1-LA) + SQR( (Z+1-LA) ^ 2 + 4*LA) ) ) / (2*LA)
  RB(K) = ( ( -(Z(K)+1-LA) - SQR( (Z+1-LA) ^ 2 + 4*LA) ) ) / (2*LA)
```
Next K
For U = 0 To V
    YU = 1
Next F = 0 To V
If F = U Then Goto 10
    YU = YU * Z(F) / (Z(F) - Z(U))
10 Next F
A(U) = -YU * (LA / (1 + LA))^N
Next U
K = 0
For I = C + 1 To N
    FO = 0
    For K = 0 To V
        AD = 0
        For J = 0 To K
            NK = N - K : IJ = I - J
            If IJ > NK Then Goto 20
            Q = NCR(K, J) * NCR(NK, IJ) * RA(K)^(K-J) * RB(K)^(N-K-I+J)
            AD = AD + Q
        20 Next J
    PHI(K) = AD * (-1)^(N-I)
    FO = FO + A(K) * PHI(K) * EXP(Z(K) * X)
Next K
SO = SO + (I-C) * (-FO)
Next I
Print SO
End
C.2 The Approximate Flow Model

This approximation considers only the dominant eigenvalue $z_0$, hoping that the remaining negative eigenvalues do not contribute significantly to the final result. The cell loss rate of (C.15) becomes

$$CLR = \sum_{j=C+1}^{N} (j - C)(-e^{z_0 \alpha} a_0(\phi_0)_j)$$  \hspace{1cm} (C.16)

The constant $a_0$ is:

$$a_0 = -\left(\frac{\lambda}{1 + \lambda}\right)^N \prod_{n=1}^{N-C-1} \frac{z_n}{z_n - z_0}$$  \hspace{1cm} (C.17)

and $(\Phi_0)_j$ is as follows:

$$(\phi_0)_j = \binom{N}{j} \left(\frac{N}{C} - 1\right)^{N-j} \quad 0 \leq j \leq N$$  \hspace{1cm} (C.18)

Computer program to calculate the CLR for $N$ homogeneous on-off sources using the approximate fluid-flow model

```
DIM Z(200), PHI(200)
INPUT "NUMBER OF SOURCES"; N
INPUT "LAMBDA"; LA
INPUT "ton"; ton
INPUT "PEAK BIT RATE OF SOURCE"; P
INPUT "BUFFER SIZE (IN CELLS)"; NO
INPUT "EFFECTIVE BANDWIDTH ALLOCATED TO SOURCES"; BW
C = BW / P: C = INT(C)
X = NO * 424 / (P*ton)
FOR K = 0 TO N-C-1
  A = (N/2-K) ^ 2 - (N/2-C) ^ 2
  B = 2 * (1-LA) * (N/2-K) ^ 2 - N * (1+LA) * (N/2-C)
  CK = -(1+LA) ^ 2 * ( (N/2) ^ 2 - (N/2-K) ^ 2 )
  Z(K) = (-B - SQR(B ^ 2 - 4*A*C) ) / (2*A)
NEXT K
```
Appendix C Connection Admission Control Algorithms

\[ \text{YU} = 1 \]

\[ \text{FOR } F = 1 \text{ TO } N-C-1 \]

\[ \text{YU} = \text{YU} \times \frac{Z(F)}{Z(F)-Z(0)} \]

\[ \text{NEXT } F \]

\[ A = -\text{YU} \times \left( \frac{LA}{1+LA} \right)^N \]

\[ \text{SO} = 0 \]

\[ \text{FOR } I = C+1 \text{ TO } N \]

\[ \Phi(I) = NCR(N,I) \times (N/C-1)^N \]

\[ \text{SO} = \text{SO} + \left( I-C \right) \times A \times \Phi(I) \times -\exp\left( Z(0) \right) \]

\[ \text{NEXT } I \]

PRINT SO

END
D. Fluid Flow Analysis of the Buffered LB

In this section an analysis of the buffered LB, shown in Figure 5-14 is carried out. The fluid-flow approach is used for this purpose following the work of [ELWA91]. The data buffer serves to buffer and smooth out bursts of cells arriving so that a smaller token buffer is required and control can be exerted more quickly. The price paid is an introduction of cell queuing delay.

The traffic source is, most generally, represented by a an $N$-state Markov-modulated fluid source which is the fluid equivalent of the MMPP. Transition between states are governed by an underlying Markov chain. This chain is represented by an $N \times N$ generating matrix $M$.

The diagonal rate parameters $\mu_i$ is given as:

$$\mu_i = -\sum_{j=1}^{N} \mu_{ij} \quad 1 \leq i \leq N$$

(D.1)

Row sums are thus all zero. The steady-state probability $\pi_i$ that the Markov chain is in state $i$ is given by solving the matrix equation

$$\pi M = 0$$

(D.2)

with

$$\pi = [ \pi_1, \pi_2, \ldots, \pi_i, \ldots, \pi_N ]$$

(D.3)

the row vectors of the $N$ steady-state probabilities.

This model differs from the MMPP model described in chapter 3 in the input traffic rates. These are not state-dependent Poisson rates, as in the case of the MMPP, but constant flows at these rates as appropriate to a stochastic fluid source.

Returning to Figure 5.14, the fluid flow model of the buffered LB, the objective is now to find the steady-state occupancy of the two buffers, $X$ and $Y$ (the data and token buffers respectively). From this analysis performance measures such as CLR, mean waiting time and mean reaction time can be found.

Note two facts from the LB technique:
Appendix D Fluid Flow Analysis of the Buffered LB

1. The data buffer can be occupied by cells only if the token buffer is empty. Otherwise the cell would be transmitted. Hence

\[ Y = 0, \quad X > 0 \]

2. Conversely, the token buffer can contain tokens only if the data buffer is empty. If cells were queued, they would capture a token and be transmitted. Hence

\[ X = 0, \quad Y > 0 \]

Both conditions are encapsulated in the single condition:

\[ XY = 0 \]

Following the approach of [ELWA91] we define the following "Virtual buffer" random variable:

\[ W = X - Y + M \]  \hspace{1cm} (D.4)

Note that because of the conditions on the random variables \( X \) and \( Y \),

\[ 0 \leq X \leq B_p, \quad 0 \leq Y \leq M \quad XY = 0 \]

\( W \) has the following properties:

1. \[ 0 \leq W \leq B = M + B_p \]  \hspace{1cm} (D.5)

2. In the range \( 0 \leq W \leq M \),

\[ X = 0, \quad W = M - Y \]  \hspace{1cm} (D.6)

3. In the range \( M \leq W \leq B = M + B_p \),

\[ Y = 0, \quad W = X + M \]  \hspace{1cm} (D.7)

Since \( X \) and \( Y \) appear at disjoint ranges of \( W \), one can find the statistics of \( X \) and \( Y \) from the statistics of \( W \) by using (D.7) and (D.6) respectively.

Because of the traffic source is assumed as a stochastic fluid process, \( X, Y \) and \( W \) are all continuous random variables (r.v.'s). In order to find the statistics of \( W \), and from it those of \( X \) and \( Y \) we proceed to the steady-state analysis. We assume that stationary conditions hold, i.e., as time goes on the system settles down to statistical equilibrium and define the joint probability.
Appendix D Fluid Flow Analysis of the Buffered LB

\[ F_i(x) = \text{Prob}[W \leq x, S=i] \] (D.8)

as the probability that the buffer occupancy is less than or equal to \( x \) with \( S=i \) the state of the input source Markov chain, \( 1 \leq i \leq N \). Note from the definition of \( W \) that this implies

\[ F_i(0) = \pi_i \] (D.9)

\( F_i(x) \) is given by the solution of differential equations derived by the incremental time analysis of the fluid-flow approach in [ANIC82].

The system's stationary-state equation in vector form is:

\[ \frac{dF(x)}{dx} D = F(x)M \] (D.10)

with

\[ F(x) = [F_1(x), F_2(x), ..., F_N(x)] \]

and

\[ D = \text{diag} [\lambda_i - r] \] (D.11)

\( r \) is the token arrival rate and \( M \) the \( N \times N \) matrix. Clearly we must have \( \lambda_i - r \) for all \( i \).

The solution to equation (D.10) is the weighted sum of exponentials in the eigenvalues of the matrix \( MD^{-1} \) given by:

\[ F(x) = \sum_{j=1}^{N} a_j \phi_j e^{z_j x} \] (D.12)

with \( z_j \) the \( j \)-th eigenvalue, \( \phi_j \) the corresponding eigenvector given as the solution to the eigenvector equation

\[ z_j \phi_j D = \phi_j M \quad 1 \leq j \leq N \] (D.13)

Now, in [ANIC82] there is an explicit formula for the calculation of stable or negative eigenvalues:

\[ z_i = \frac{-B(i) \pm \sqrt{B^2(i) - 4A(i)C(i)}}{2A(i)} \] (D.14)

where

\[ A(i) = (N/2-i)^2 - (N/2-r)^2 \]

\[ B(i) = 2(1-\lambda_i)(N/2-i)^2 - N(1+\lambda_i)(N/2-r) \]
Appendix D Fluid Flow Analysis of the Buffered LB

\[ C(i) = (1+\lambda)^2((N/2)^2-(N/2-i)^2) \]

The \( a_j \) constants are to be determined by invoking \( N \) boundary conditions. Since \( \pi M = 0 \), one eigenvalue of (D.13) must be zero. Calling this eigenvalue \( z_0 \), its associated eigenvector is \( \phi_i = \pi \). Hence (D.12) can be simplified to

\[ F(x) = a_1 \pi + \sum_{j=2}^{N} a_j \Phi_j e^{\xi_j x} \]  

(D.15)

To find the unknown constants \( a_i \) we now need to establish \( N \) boundary conditions. To establish these we note that for some states of the Markov chain the arrival rate (\( \lambda \)) must be higher than the token generation rate (\( r \)) resulting in filling states. Otherwise if \( \lambda \) is less than \( r \) the system is undeload resulting in emptying states.

If \( r > \lambda_i \) all \( i \); then the token buffer would never empty and each cell could enter the network without any possibility of control. On the other hand if \( \lambda_i > r \), all \( i \) the token buffer would be empty most of the time and the data buffer would always fill-up resulting in high cell loss because all states result in filling of the buffer with no emptying state. In other words there must exist some filling and emptying states. Hence all \( N \) states of the Markov chain modulating the source arrival rate can be divided into two disjoint sets:

\[ S_e = (\lambda_i - r < 0) \] (emptying states)  

(D.16)

and

\[ S_f = (\lambda_i - r > 0) \] (filling states).  

(D.17)

Without loss of generality, assume that the Markov chain states are such that the \( \lambda_i \)'s monotonically increase with the state number. Hence we have

\[ \lambda_1 < \lambda_2 < \ldots < \lambda_a < \lambda_{a+1} < \ldots < \lambda_N \]

Let \( \lambda_a \) correspond to the largest source rate such that the system is in the emptying range. Then we must have \((\lambda_a - r) < 0\), \((\lambda_{a+1} - r) > 0\) and \( \lambda_a < r < \lambda_{a+1} \).

First we investigate the boundary conditions for the \( L \) emptying states. Since the data arrival rate is less than the token arrival rate in these states, the data queue of Fig. 5-17
Appendix D: Fluid Flow Analysis of the Buffered LB

is tending to empty \((X \to 0)\) and the token buffer (or bucket) is tending to fill \((Y \to B_x)\). Hence

\[ W = X - Y + M \to 0 \]

For these states then, the probability that the virtual buffer or data buffer is full tends to zero, or we have

\[
\Pr [W = B, S = i] = \Pr [X = B_v, S = i] = 0 \quad i \in S_e \quad (D.18)
\]

Note that

\[
\Pr [W = B, S = i] = \Pr [W \leq B, S = i] - \Pr [W \leq B', s = i]
\]

Thus we have

\[
\Pr [W \leq B, S = i] = \Pr [W \leq B', s = i] = \pi_i
\]

\[
F_i(B^-) = \pi_i \quad i \in S_e \quad (D.19)
\]

This provides \(L\) of \(N\) boundary conditions. For the remaining \((N-L)\) filling states the token buffer of Fig. 5-17 tends to empty \((Y \to 0)\) and the data buffer tends to fill \((X \to B_p)\). Hence,

\[ W = X - Y + M \to B_p + M = B \]

For these states the probability that the virtual queue is empty must be zero. Thus

\[
\Pr [W \leq 0^+, S = i] = F_i(0^+) = 0 \quad i \in S_p \quad (D.20)
\]

(D.19) and (D.20) provide the \(N\) boundary conditions from which to find the \(N\) unknown constants \(a_i\), \(1 \leq i \leq N\). The \(N\) equations to be solved to find the unknown \(a_i\)’s can be written in scalar form as follows:

\[
F_i(B^-) = \pi_i = a_1 \pi_1 + \sum_{j=2}^{N} a_j \Phi_{ji} e^{\lambda_j \mu^-} \quad i \in S_F \quad (D.21)
\]

and

\[
F_i(0^+) = 0 = a_1 \pi_1 + \sum_{j=2}^{N} a_j \Phi_{ji} \quad i \in S_F \quad (D.22)
\]

\(\phi_{ji}\) is the \(i^{th}\) component of the \(j^{th}\) eigenvector.
We now apply this analysis to the two-state on-off source model. For this case,

\[ M = \begin{bmatrix} -a & a \\ b & -b \end{bmatrix} \]  \hspace{1cm} (D.23) \\

\[ D = \text{diag}(-r, \lambda - r) \]  \hspace{1cm} (D.24) \\

and

\[ \pi = \begin{bmatrix} b \\ \frac{a}{a+b} \end{bmatrix} \]  \hspace{1cm} (D.25) \\

The arrival rates for the on-off model are \( \lambda_1 = 0 \) and \( \lambda_2 = \text{peak rate} (p) \).

For this example there is only one eigenvalue \( z \) to be found. For this case we have from (D.14):

\[ z = \frac{a+b}{p-r}(1-p) \]  \hspace{1cm} (D.26) \\

where now

\[ \rho = \frac{p}{r} \left( \frac{a}{a+b} \right) = \frac{m}{r} \]  \hspace{1cm} (D.27) \\

Note that \( p \cdot \frac{a}{a+b} \) is the mean load (m) and the parameter \( \rho = \frac{m}{r} \) corresponds to the normalized load.

Since there is only one negative eigenvalue for the on-off model the solution of \( F(x) \) (D.22) is given by:

\[ F(x) = a_1 \pi + a_2 \phi e^{ax} \]  \hspace{1cm} (D.28) \\

The single eigenvector \( \Phi = [\Phi_1, \Phi_2] \) appearing in (D.28) can be found using the eigenvector equation:

\[ z \phi D = \phi M \]  \hspace{1cm} (D.29) \\

where we have dropped the subscripts of \( z \) and \( \Phi_j \) in (D.13) since there is only one term. Substituting \( z \) given by (D.26), \( D \) by (D.24) and \( M \) by (D.23), we have:

\[ \frac{\phi_1}{\phi_2} = \frac{p-r}{r} \]  \hspace{1cm} (D.30) \\

Since the eigenvectors can only be found to within a constant we chose \( \Phi_2 = 1 \) and \( \Phi_1 = (p/r - 1) \). We thus have:
The two unknown constants $a_1$ and $a_2$ are found using the boundary conditions (D.21) and (D.22). For the on-off source with $N=2$ state 1 with $\lambda_1=0$ is the emptying state and state 2 with $\lambda_2=p$ is the filling state. We therefore must have $0<r<p$ for the fluid flow analysis to provide a stationary solution. This implies that the single eigenvector $z$ of (D.26) is negative if the parameter $p$ defined by (D.27) is less then 1 (which is always the case).

From (D.27) we have using (D.31):

$$F_1(B^-) = \pi_1 = a_1 \pi_1 + a_2 \left( \frac{p}{r} - 1 \right) e^{dr}$$

with $\pi_1=b/(a+b)$ the probability that the source is in the off-state.

Similarly from (D.22) and (D.31) we have:

$$F_2(0^+) = 0 = a_1 \pi_2 + a_2$$

with $\pi_2=a/(a+b)$ the probability that the source is in the on-state.

Solving (D.32) and (D.33) simultaneously for $a_1$ and $a_2$ we get

$$a_1 = \frac{1}{I - \frac{a}{b} \left( \frac{p}{r} - 1 \right) e^{dr}}$$

and

$$a_2 = -\pi_2 = a_1 \cdot a/(a+b)$$

Note that we have dropped the - from $B$ since it is no longer needed here. Using (D.34) and (D.35) in (D.31), and following the notation of [ELWA91], we finally get the probability distribution functions,

$$F_1(x) = \pi_1 \frac{\Delta(x)}{\Delta(B)}$$

$$F_2(x) = \pi_2 \frac{1 e^{(r)x}}{\Delta(B)}$$

where
Appendix D Fluid Flow Analysis of the Buffered LB

\[ \Delta(x) = \frac{a (p-r)}{b r} e^{\alpha x} \]  
(D.38)

Since \( F(x) = F_1(x) + F_2(x) \) we obtain

\[ F(x) = Pr[W \leq x] = \frac{1 - \rho e^{\alpha x}}{\Delta(B)} \quad (0 \leq x \leq B) \]  
(D.39)

All performance parameters of interest can be obtained from (D.36) and (D.37). Note that the probability distributions are a function of the parameter \( B = M + B_n \).

First let us determine the mean throughput \( \gamma \) for a general \( N \)-state Markov chain model in order find the CLR. Since every cell requires a token for transmission, the average token throughput must equal the average cell throughput. We thus have:

\[ \gamma = r - \sum_{i=1}^{N} (r - \lambda_i) F_i(0) \]  
(D.40)

The second term here represents the mean token loss due to a full token buffer.

For the on-off model we have:

\[ \gamma = r - (r-0) F_i(0) = r[1 - F_i(0)] \]

and knowing that

\[ F_i(0) = a_i \pi_i - a_i \pi_i (p/r - 1), \quad a_i = 1/\Delta(B) \quad \text{and} \quad \rho = p/r \cdot a/(a+b) \]

the normalised throughput is found as:

\[ \frac{\gamma}{r} = 1 - \frac{\rho}{\Delta(B)} \]  
(D.41)

The cell loss rate is easily obtained, using

\[ CLR = 1 - \sum_{i=1}^{N} \frac{\gamma}{\lambda_i \pi_i} = 1 - \frac{\gamma / r}{\rho} \]  
(D.42)

As already noted \( m = a/(a+b) \cdot p \) in the on-off case and more generally

\[ \sum_{i=1}^{N} \frac{\lambda_i \pi_i}{i} \]

for the \( N \)-state Markov chain model corresponds to the mean Poisson arrival rate.
For the on-off case we consider the range $\rho=p/r < 1$ ensuring that $p>r$ (that there exists a filling state). This is the range of operation in which we would like the CLR to be very small. From (D.41) and (D.42), using (D.38) to evaluate $\Delta(B)$, we then get:

$$CLR = \left(\frac{1}{\rho} - 1\right)l - \Delta(B) = \left(\frac{1}{\rho} - 1\right)\frac{a}{b}(\frac{p}{r} - 1)e^{it} << 1$$

(D.43)

Since $z<0$ in this range, the loss probability decreases exponentially with $B=M+B_0$. Thus the cell loss performance of the leaky bucket algorithm depends only on the sum of the buffer sizes (Note that the token buffer is only a model representation. The data buffer, on the other hand, is a real buffer). For the unbuffered leaky bucket, the cell loss probability can be simply obtained by setting the data buffer $B_0=0$.

Since we are not only interested in the cell loss rate (CLR) but also in the delay introduced by the data buffer and the reaction time we continue the analysis of [ELWA91].

Having found the distribution of $W$, we can now find from it the distribution of the token buffer content $Y$ and the data buffer content $X$. Specifically,

$$\Pr(\text{data buffer full})=1-F(B)$$

$$\Pr(\text{token buffer full})=F(0)$$

$$\Pr(X \leq x, S=i)=F_i(x+M) \quad (0 \leq x \leq M)$$

$$\Pr(Y \leq y, S=i)=\pi_i F_i(M-y) \quad (0 \leq y \leq M)$$

The mean length of the token buffer (in cells) is:

$$\overline{Q}_M = \int_0^M y \, dz + MF(0) \quad z = [1 - F(M - y)]$$

(D.44)

By using partial integration we find:

$$\overline{Q}_M = \frac{\rho(1 - e^{2M})}{z\Delta(B)} + \frac{M}{\Delta(B)}$$

(D.45)

The mean reaction time of the leaky bucket is given by:
where $Y$ is the increase factor in the mean rate of the source, $n_c=424$ the number of bits per cell and $E \cdot m$ is the token generation rate.

The mean length of the data buffer is:

$$Q_{B_0} = \int_0^{B_0} x \, dx + B_D \cdot F(B_D) \quad z = F(x + M)$$

Applying partial integration we obtain:

$$Q_{B_0} = \frac{\rho(e^{zB} - e^{zM})}{z \Delta(B)} + B_D(1 - \frac{1}{\Delta(B)})$$

Thus the mean delay which is introduced when using a data buffer is given by:

$$D_{B_0} = \frac{Q_{B_0} n_c}{(1 - CLR)r}$$

**LB program using Fluid Flow Model to determine CLR**

```
INPUT "peak burst rate"; b
INPUT "leaky rate"; a
INPUT "mean burst period"; mb
INPUT "mean silence period"; ns
INPUT "memory"; mg
m = mg * 424
lam = 1 / mb
mam = 1 / ns
bur = lam / (b - a)
sil = mam / a
far = bur - sil
x = (b - a) / b
y = EXP(far * m)
clr = (x * far) / (bur * y - sil)
PRINT clr
```
E. Publications

The following papers have been published on work related to the thesis.


